April 1998 $7.20 £3.75

Studio sound

THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION & BROADCAST

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LOUDSPEAKER TESTS

Introducing the definitive review

The ELLIOT SCHEINER Interview

MANIC STREET PREACHERS IN SESSION
POSTING LOST IN SPACE
EXPLAINING PLUG-INS
DVD AUDIO DEFINED

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Warner Bros, Burbank – Stage 3

Warner Bros, Burbank – Stage 4

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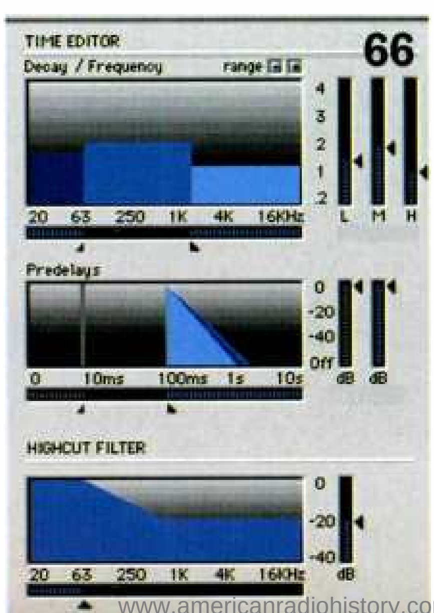
From the small screen to the big silver screen—*Lost in Space*

Below: Taking a shine to the Fleetwoods; plug in to processing; and two flavours of American outboard from Avalon and Aphex



Editorial

Skipping a generation

ONE OF THE MOST FASCINATING aspects of watching a previously technologically under-developed market upgrading its technology is the reasoning behind the chosen route as this illustrates a purity of thought that is uncluttered by the mind-set enforced by a continuous programme of updating. Frequently these involve skipping a generation of apparently intermediate technology but there are other instances where generations are skipped by manufacturers and not the users. The leap to affordable digital mixing, for example, has glossed over and seemingly missed the very real requirement that still exists for affordable but large and fader and mute automated analogue consoles with enough mic pres and pots under the hand to record and mix live music well. The chances are we may never see such desks even though the high-end continues to make its living on more expensive variants of the same theme.

The benefit of intermediate technology is that it allows us to adapt and prepare for the next stage but most importantly it serves as a thought-structuring discipline. With the current state of play with DVD Audio I would suggest that the leap has been so great that no one can actually see or agree where it ought to land. We are presented with a throbbing spreading mass of potential that is difficult to tie down with the main players resorting to contrived solutions in order to contain it. The purity of thought was lost in the excitement of the jump and it's already showing the beginnings of an inelegant lash-up. The degree of interest and discussion on the potential of DVD Audio eclipses anything that has ever come before, yet there is still relatively little to show for it. Knowing the forces that are at work in this issue you've got to ask how long it can stay up in the air and will it not so much land as crash.

Zenon Schoepe, executive editor

Loud and clear

IT IS COMFORTABLY OVER 10 years since *Studio Sound* last ran a loudspeaker review. It's not that there haven't been any new speakers in the intervening period, it's just that neither of the magazine's editors have felt it appropriate to sully the title with the kind of subjective review many other magazines have favoured. And in the absence of a suitably qualified and independent reviewer, that would have been the only option.

Certainly, setting up a pair of speakers in a working environment and running a variety of programme material through them gives a reviewer something of an indication of how well they perform—within a specific acoustic environment—but the printed word is no way of relaying that impression to a reader. The result, as we all know, is reams of references to 'solid' and 'smooth' bass, 'clean' and 'glassy' treble, 'sucked-out' mid ranges and 'mid-bass boom', 'punch', 'tension', 'edginess', 'airiness', 'in-yer-face' factors, and so on. It may help a magazine garner advertising dollars, but it doesn't help its readers to determine whether a pair of speakers suits your requirements.

The intention, then, of beginning a programme of objective reviews of close-field monitors is to eliminate the intangible elements of listening tests and provide measurements that are both useful in themselves and also allow valid comparisons to be made between speakers, safe in the knowledge that they were derived under identical test conditions—something rarely possible with manufacturers' own specifications. (The way in which the tests relate to real-world performance will be fully explained.) None of this is to say that there is no value in listening, quite the opposite, but there is no value in relaying inconsistent subjective impressions and suggesting that meaningful judgements can be made from them.

That loudspeaker manufacturers themselves recognise this is without doubt, as you discover when the certainty of a review loan is tempered or withdrawn upon mention of anechoic and reverberant chambers. And to be fair, manufacturers should be cautious when submitting to a measurement regime—they need to know that the tests will represent relevant aspects of a loudspeaker's performance in a meaningful way. They are also right to be concerned over the absence of any listening aspect to a test, after all, a speaker is meant to be listened to. But we've already been over that. Ultimately, I would have to feel cautious about a speaker whose manufacturer declines to submit it for evaluation.

Welcome, then, to *Studio Sound*'s speaker tests—they're your best indication of performance outside of your ears.

Tim Goodyer, editor

Studio Sound

Incorporating Broadcast Engineering

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Fax: +44 171 401 8036.
Email: mfpag001@mfpag001.demon.co.uk
Net: www.prostudio.com/studiosound

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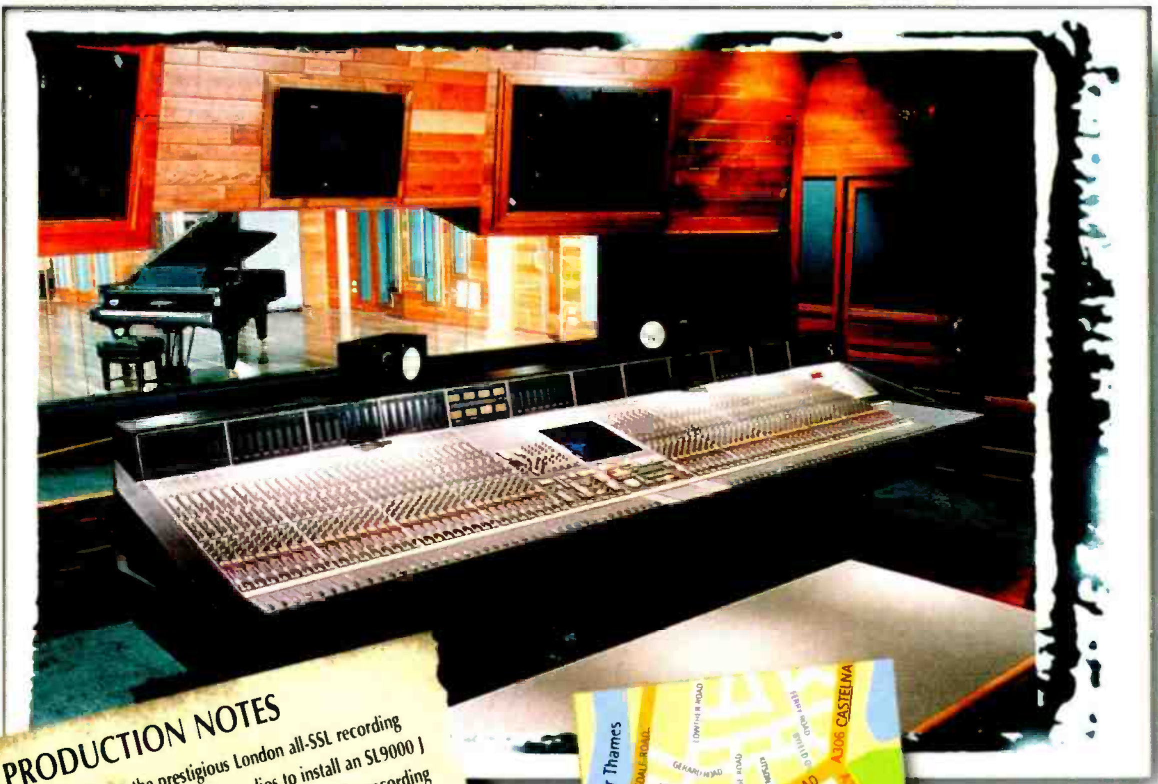

PRODUCTION NOTES

Olympic Studios, the prestigious London all-SSL recording facility, was one of the first UK studios to install an SL9000 J Series console, and was recently the venue for the recording and mixing of the new Eric Clapton album 'Pilgrim'. Engineering the sessions was Alan Douglas, who obviously enjoyed working with the Solid State Logic console. "The 9000 has remarkable capabilities. The bottom end is fantastic, the top end is really open and the automation is brilliant. The J is the best sounding console that I have ever worked on."

OLYMPIC STVDIOS

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music week awards 98
BEST STUDIO

Solid State Logic

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Tel: +44 (0)1865 842300
Fax: +44 (0)1865 842118
E-mail: sales@solid-state-logic.com
http://www.solid-state-logic.com

SL 9000 J SERIES

New York	Los Angeles	Tokyo	Paris	Milan	Toronto	Singapore
+1 (1)212 315 1111	+1 (1)213 463 4444	+81 (0)3 5474 1144	+33 (0)1 3460 4666	+39 (0)2 2622 4956	+1 (1)416 431 9131	+65 (0)285 9300

Another fine Messe...

Germany: The Frankfurt Musikmesse is perhaps the only event that strikes fear in the hearts of regular show-goers. There's something about the sheer size (Messe myth has it that one individual marked his progress via a pedometer and clocked up a frightening 11.3km in one day), the expense of staying in Frankfurt (now largely in line with what it costs to get over to NAMM in the US and the sunshine), and the rather disappointing city combine to create a show that has to be endured rather than enjoyed.

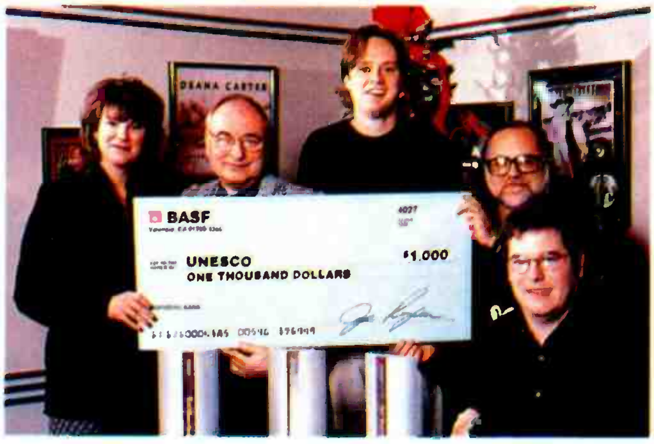
It may have been more comfortably removed from its proximity to the European AES Convention than it was last year but it moved closer to NAMM which, while smaller, still pipped it for the majority of announcements with the exception of Steinberg and Emagic's agreement to co-operate on their MIDI + Audio futures (see later in this issue).

It is the largest audio related gathering on earth, yet the truth about Frankfurt is that it is now too big to be covered by any individual, be they a distributor, dealer or punter. You have to contend with the reality that most established manufacturers have already sorted out their distribution and are there predominantly to show willing, new companies are hard to find in the throng, and few can be pressed to admit that they really enjoy the invasion of the bag-snatchers on the public days. Frankfurt remains a phenomenon.

Zenon Schoepe

UK: Cotswold-based Chipping Norton Studios has installed an Otari Radar hard-disk recorder-editor system. Set up and managed by Richard Vernon, the facility was designed by Neil Grant and houses a Trident 90 console. Headlining clients include Radiohead and Portishead. Chipping Norton Studios. Tel: +44 1608 643636.

UK: This year's Distinguished Engineers' Audio Federation Awards continued the 25-year tradition of celebrating pro-audio and raising cash for deaf children's charity. Being the 25th event, this year's proceedings included a gold disc—sponsored by *Studio Sound* and sister title *Pro Sound News Europe*—for each of the 26 founder members. Pictured with his gold disc is founder Stephen Court; not pictured are the remaining 25, who includes such luminaries as Ken Townsend, Adrian Kerridge, Keith Grant, Malcolm Toft and John Bauch. The event was held in the luxury of the Compleat Angler Hotel in deepest Buckinghamshire. DEAF: www.interstudio.co.uk/isl/deaf.htm



US: producer Peter Collins, along with engineer Joe Baldridge and Nashville's Emerald Sound Studios, are the latest recipients of a BASF Master Award for their work on Jewell's single 'You Were Meant for Me'. A reel of half-inch SM468 gained the production its eligibility and the climb to a No. 1 chart position secured the Award which also carries a $1,000 donation to Unesco in the name of the recipient.

Reference and information

Recent publications of note include an instructional video, *Think like a Producer*, from American Guy Marshall who teaches at the Musician's Institute and Santa Monica College. Unlike other videos that address purely technical aspects of music and production, Marshall aims to explain the producer's mind-set to those either seeking to become one or needing to work with one. The 'dynamics and language' of the producer are highlighted and applied to a wide range of studio activities including vocals, drums and multitracking. *Think like a Producer* is available from Tutte & Babe Music in both NTSC and VHS versions costing $24.

Philip Newell's new book, *Recording Spaces* (Heinemann, ISBN 0240515027), deals with the acoustics of recording rooms from the perspective of an author active in the music and recording field for some 30 years. Room acoustics, isolation, variable acoustics, lighting and ventilation are just a few of the topics addressed in 182 pages—although Newell's dedication alone would warrant a recommendation.

Coming in at 168 pages, meanwhile, is HHB's 1998 catalogue featuring the gamut of products from post to electronic instruments from a wide range of manufacturers—including HHB itself.

Tutte & Babe Music.
Tel: +1 310 395 4835.
Heinemann Publishers Oxford, Halley Court, Jordan Hill, Oxford OX2 8EJ, UK.
HHB Communications, UK.
Tel: +44 181 962 5000.

Parsons Leaves Abbey Road

UK: After a remarkable brief period, Alan Parsons is leaving his job as EMI Studios group vice president to become creative consultant and associate producer.

Quoting pressure from forthcoming commitments outside the studio, Parsons reckons to be unable 'to continue to do justice to my executive role at EMI'. His new position will see Parsons remain involved with DVD and surround projects, merchandising and PR, as well as conducting his career as a producer and artist.

No announcement about Parsons' successor had been made at the time of going to press.

UK: Raymond Gubbay's £2m production of Puccini's *Madame Butterfly* continues its run in-the-round at London's Royal Albert Hall to critical acclaim. The production uses a new translation from the Italian by Amanda Holden and with sound design by Bobby Aitken and a sound system from Autograph Recording including a 360° cluster of 11 Meyer MSL-4s, two tower arrays of 22 UM-1s and concealed UPM-1s to lower the sound image. Off-stage clusters also use Meyer PSW-1s and MSL-4s with mics being predominantly Sennheiser SK50s and MKE2s.

Australia: Bathed in controversy, the 8th FINA World Swimming Championships were notable for illegal drug use and a stack of ARX equipment. Provided by Perth AudioVisual, the 922 Concert Series and PowerMax 2 speakers were driven by SX1500 amps while the broadcast audio split was achieved using DI-6S line splitters. ARX. Tel: +61 3 9555 7859.

Colin Sanders Memorial

UK: The Oxford University Church of St Mary the Virgin was almost full when it opened its doors for SSL-founder Colin Sanders memorial service last month.

The earlier funeral had been a moving affair attended by just some of Sanders' many friends and, for the most part, the memorial shared the sense of loss held by those present—despite the humour most notably injected in the address made by Gyles Brandreth's, one-time business partner and self-proclaimed nemesis of Sanders' otherwise remarkable business success.

The congregation contained a broad representation of industry faces including Sir George Martin, who also spoke, and was treated to music performed by the choir of Magdalen College under Bill Ives, the London Chamber Orchester lead by Christopher Warren-Green with a selection of soloists. The performances included a playback of Procol Harum's 'Whiter Shade of Pale' in which Sanders had been involved, and a piece entitled 'For Colin' composed by Nick Bicât.

From the Church, many of the congregation accepted the wider invitation to continue at Harris Manchester College and finally at a local club, a procession designed to see Sanders' absence accepted in a manner more in keeping with his personality and philosophy. His passing will clearly continue to be felt by his many friends.

Austria: Vienna's MG Sound has installed an SSL 9000J series console in its Studio A. Displacing the facility's 4064G Plus console, which has been relocated to Studio 2, the 80-channel 9000 will continue MG Sound's tradition as a top-flight Austrian music recording venue, as well as serving the in-house composition-production operation of Martin Bohm and Stevie Cross. Previous clients include Placido Domingo, Jim Steinman and Celine Dion. MG Sound, Austria. Tel: +43 1 535 6404 SSL, UK. Tel: +44 1865 824300.

■ Tokyo now hosts Japan's first Amek DMS digital console, courtesy of Answers Studio. The 24-fader, 72-channel DMS will run in conjunction with an Akai DD1500 hard-disk recorder and a pair of Tascam DA-88 MDM machines to provide post-production services to the likes of Fuji TV and TV Tokyo. Installation of the new desk took just four days from delivery to service.
Amek, Japan. Tel: +81 3 5707 0575.

■ New York-based All Mobile Video has chosen the AMS Neve Libra Live for its fully digital AV mobile, Celebrity. The 48-fader, 96-input digital console is central to what is claimed to be the first mobile of its kind, which includes both digital audio and video. With its launch planned for the Las Vegas NAB convention, the 53-foot Celebrity is intended to serve live music events and others where 'audio quality is of extreme importance'.
All Mobile Video, US.
Tel: +1 212 727 1234.
AMS Neve, US.
Tel: +1 212 965 1400.

■ China's CCTV is presently installing a 24-fader Fairlight FAME system in its main Beijing studio to meet the requirements of its foreign language dubbing and voice-over work. The installation is CCTV's third, accompanying FAME's now running in the post and overseas news departments, and accompanies the purchase of 10 TL Audio PM-1 4:2 news-gathering mixers.
Fairlight, UK. Tel: +44 1474 815300.
VW Marketing, UK.
Tel: +44 1372 728481.

■ Cumbrian-based British broadcaster, Border Television, has included a Calrec S-series console in its preparations for digital terrestrial broadcasting due to begin later this year. The 24-channel console is part of a complete refurbishment, occupying the evening news and daily magazine programme studio.
Border Television, UK.
Tel: +44 1228 25101.
Calrec, UK. Tel: +44 1422 824159.

■ Germany's PolyGram Recording Services (formerly part of Deutsche Grammophon) has taken delivery of four 48-track, 24-bit Augan digital recorder-editor systems. Following the recent purchase of a 24-track system, the new order is already set to be upgraded to Augan's forthcoming 48-track remote control, the RC48-C, in the design of which PRS already has a stake.
Sascom, US. Tel: +1 905 469 8080.

■ Italian state broadcaster, RAI, has purchased a Midas XL4 console to replace the existing two XL2s in its Teatro delle Vittorie television studio. Used for live productions involving live audiences, the studio's brief included the capacity for simultaneous mixes and ease of operation. The new console is already in use on a variety show called Fantastico that reaches a 20m-strong viewing audience.
EVI Group, UK.
Tel: +44 1562 741515.

■ International pro-audio rental companies have been responsible for several notable orders. The British FX Rentals, has taken delivery of the first 10 Alesis ADAT M20 'professional' 20-bit MDM machines. FX is part of PARN, the European alliance of rental companies. Meanwhile on the US West Coast, Tim Jordan Rentals now has a full Fairlight MFX3 Plus DAW system on its stock. Offering 72 track-hours of 24-track, 24-bit recording, the MFX3 Plus is paired with an FED V-MOD 100 for video playback. TJR expects the package to prove popular with California's burgeoning post and film fraternity.
FX Rentals, UK.
Tel: +44 181 746 2121.
Tim Jordan Rentals, US.
Tel: +1 818 755 9011.

■ West London's new Gallery Studios, opened by ex-Roxy Music guitarist Phil Manzanera, has added a Digidesign Pro Tools 24 system to its kit list. Running on a Mac 9600 platform and using an 888 24-bit I-O interface, the system runs in conjunction with Emagic's Logic Audio and a selection of TDM plug-ins including the Focusrite D2-D3. Abbey Road studios has recently acquired a TL Audio M-2 8-channel mixer which is finding favour in both classical and popular sessions, particularly with digital recorders.
Digidesign, UK.
Tel: +44 1753 653322.

■ Californian interest in Euphonix is high with A&M Studios opening a renovated Studio C complete with 96-fader CS3000, Blackhole Recording Studios opening its Studio A with a 48-fader CS2000, and Brandon's Way adding a new CS3000 to its existing CS2000. A&M is operating a flexible studio room capable of handling 5.1 discrete surround work that has already seen sessions from kd lang and Chicago, while Blackhole's reputation as a rap studio has brought in Wu Tang Clan and Coolio & Forty Cs among others. Brandon's Way's newly equipped main tracking room will continue the facility's practice of serving producer, artist and owner Kenny Edmonds's projects as well as outside work.
Euphonix, US. Tel: +1 650 855 0400.

■ The Amsterdam-based School of Audio Engineering has opted to use Quantegy recording media. The move follow the School's evaluation of the Quantegy range with SAE's wide range of courses in mind and will see various Quantegy lines in use in the SAE's 25 colleges spread over 13 countries worldwide.
SAE, The Netherlands.
Tel: +31 206 22 8790.
Quantegy, Europe.
Tel: +31 24 373 0484.

■ Australia's Sydney Opera House has installed a Tannoy SuperDual system in its Drama Theatre. Based on S300, B950 and T12 cabinets, the system is the only such system in the Opera House as the 544-seat Drama Theatre is the only auditorium to consistently use sound reinforcement and places much value in the ability of the SR rig to reproduce the human voice. To date, the Theatre has seen service on Moby Dick and Terence McNally's Masterclass. Elsewhere in Sydney, commercial music facility Song-Zu has bought two Fairlight MFX3 Plus workstations, giving it four, and cable TV network Foxtel has installed a second MFX3 Plus in a new editing suite.
Tannoy, UK. Tel: +44 1236 420199.

■ Canada's Chum Television has chosen the Studer D950 S for its unconventional broadcast facility. Located in the ChumCity building in Toronto, Chum's philosophy uses 'quick-plug receptacles' in lieu of a regular studio format. The mobile and flexible approach to programme making has already proven successful in the MuchMusic and CablePulse24 news channels.
Chum TV, Canada.
Tel: +1 416 591 5757.
Studer, Switzerland.
Tel: +41 1 870 7511.

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Tom Jung
Technical consultant for Pro Audio Review
President and recording/mastering engineer of Digital Mastering Products (DMP)

France	Greece	The Netherlands	Spain	United Kingdom
JBK Marketing	Autohellas S.A.	Holland P.A. Service, B.V.	Gedelson S.A.	Raper & Wayman
6, Rue Beaubourg	3, P. Tsaldari & 7	Kuipersergweg 12	Poligono Industrial	Unit 3, Crusader Estate
75004 Paris	Aristidou Str.	1101 AG Amsterdam-Z.O.	"Can Roca"	167 Hermitage Road,
(33) 1-44-54-50-03	15122 Athens	20740 Amsterdam	C/Mar Caribe No. 29	Manor House
(33) 1-44-54-50-08	(30) 1-8020777	(31) 20 691 2167	08130 Barcelona	London N4 1LZ, England
	(30) 1-6129964	(31) 20 693 2178	(34) 3-5741122	(44) 181-800-8288
			(34) 3-5742713	(44) 181-809-1515

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World events

April

14
AES Lecture: Source Independent Measure
Conference Room, Baden-Powell House, South Kensington, London SW7, UK.
Tel: +44 1628 663725
Fax: +44 1628 667 002.
Email: aesuk@aol.com

14–16
PLASA: Light and Sound Shanghai
Intex Centre, Shanghai, China.
Contact: P&O Events.
Tel: +44 171 370 8231.
Email: shanghai@eco.co.uk

22–24
Entech
Halls 1 & 2, Sydney Exhibition Centre, Darling Harbour, Australia.
Contact: Caroline Grafton
Tel: +61 2 9876 3530.
Fax: +61 2 9876 5715.
Email: mail@conpub.com.au
Net: www.conpub.com.au

27–28
DVD Forum
The Berkley Hotel, London, UK.
Tel: +44 171 691 9191.
Email: dvd@iqpcco.uk

29–30
20th ABTT Show
Hall One, Royal Horticultural Halls, 80 Vincent Square, London SW1, UK.
Tel: +44 171 403 3778.
Email: office@abtt.org.uk

May

10
The National Vintage Communications Fair
Hall 11, NEC Birmingham, UK.
Contact: Sunrise Press.
Tel: +44 1392 411565.

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

21–27
Expo Sound & Light 98
Romexpo Exhibition Centre, Bucharest, Romania.
Tel: +44 171 886 3103.
Email: info@otsa.prestel.co.uk

26-28
TV 98
Thermal Hotel Helia, Budapest, Hungary.
Contact: Scientific Society for Telecommunications
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.
Tel: + 1 914 993 0489.
Fax: + 1 914 328 8819.
Email: chris@ssiexpos.com
Net: www.ssiexpos.com

30–June 2
Nightwave 98
Rimini Exhibition Centre, Italy
Contact: Ms Gabriella de Girolamo.
Fax: +39 541 711243
Net: www.fierarimini.it

June

1–5
CommunicAsia 98
Singapore Suntec Centre, Singapore.
Contact: Singapore Exhibition Services.
Tel: +65 338 4747.
Fax: +65 339 5651.
Email: info@sesmontnet.com
Net: www.montnet.com/ses/

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

10–15
International Electronic Cinema Festival 1998
Chiba city, Japan
Tel: +41 21 963 32 20.
Fax: +41 21 963 88 51.
Email: message@symposia.ch
Net: www.montreux.ch/ symposia

15–17
Mecon 98
Medienforum NRW, KölnMesse trade complex, Cologne, Germany.
Contact: Musik Komm.
Tel: +49 221 91655.
Fax: +49 221 91655 160.
Email: mecon@musikkomm.de
Net: www.mecon.de

July

9
Biz Tech 98
Loew's Vanderbilt Hotel, Nashville, Tennessee, US.
Contact: SPARS National Office.
Tel: +1 800 771 7727.
Email: spars@spars.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

November

17–19
Digital Media World 98
Wembley Exhibition and Conference Complex, London, UK.
Contact: Digital Media International
Tel: +44 181 995 3632.
Email: digmedia@atlas.co.uk

December

9–11
5th Broadcast Cable & Satellite India 98
6th Commsindia 98
Pragati Maidan, New Delhi, India.
Contact: The Federation of the Electronics Industry
Tel: +44 171 331 2000.
Fax: +44 171 331 2040.

Studio sound audio industry recognition awards

THE AMSTERDAM AES Convention in May 1998 will be the setting for the first SSAIRAs—the *Studio Sound* Audio Industry Recognition Awards. Following our call for nominations there follows a list of products that have been put forward in the various categories at the time of going to press.

While the voting process is now effectively open, products can still be nominated and indeed will be should our readers choose to vote for products that are not currently listed. The only condition is that the product has to have been released onto the market since last year's European AES Convention in Munich. Nominations will be updated on the *Studio Sound* website www.prostudio.com/studiosound.

While anyone can nominate a product for a category, only qualified readers of *Studio Sound* are eligible to vote and this will be verified by the requirement for readers to quote their unique reader identification number.

Ways to vote

Readers can vote for one product in each category in four ways.

1. By filling in the form and posting it to:

THE UNIQUE READER IDENTIFICATION NUMBER IS THE NINE-DIGIT NUMBER STARTING WITH A ZERO THAT IS LOCATED IN THE MIDDLE OF THE TOP ROW OF YOUR STUDIO SOUND ADDRESS LABEL. IN ALL INSTANCES THE INCLUSION OF THE UNIQUE READER IDENTIFICATION NUMBER IS ESSENTIAL

SSAIRAs, *Studio Sound* Magazine,
Miller Freeman Entertainment,
8 Montague Close, London Bridge,
London SE1 9UR, UK

2. By faxing the form to +44 171 401 8036.
3. By emailing their unique reader identification number, the category numbers and their votes to zschoepe@unmf.com
4. By filling in the interactive voting form on the Studio Sound website: www.prostudio.com/studiosound

Readers will only be allowed to vote once. Readers may only vote for one product in each category.

The object is not necessarily to identify the best equipment in each category but to identify those items that genuinely warrant recognition as being special in some way.

It should be noted that in the case of outboard equipment, the categories describe a function rather than a product type. Thus a 'voice channel' may legitimately qualify under dynamics and-or EQ if you feel it excels in these areas. Readers are not obliged to vote in all categories and their attention is drawn to Special Category 13 which serves as a 'catch all' for any products not covered in the other 12.

Any questions can be directed to Zenon Schoepe and Tim Goodyer at Studio Sound. Tel: +44 171 921 5010.

Fax your vote to: +44 171 401 8036

Nominations

1 Large scale console
AMS Neve DFC
Calrec Q2
D&R Octagon
Lawo MC82
Soundtracs DPC II
SSL Avant
Stage Tec Cinetra
Studer D950S

2 Medium to small scale console
ATI Pro6
Audio Developments AD149
Allen & Heath GL3300
Amek Soho
Calrec C2
Summit TMX-420
Tascam TM-D8000

3 Outboard dynamics
Amek 9098 compressor-limiter
Aphex 661
BSS Opal DPR422
BSS Opal DPR522
dbx Blue 160s
Drawmer MX30
Drawmer MX40
Empirical Labs Distressor
Fairman TMC
Focusrite Green 4
Joemeek SC3
SPL Machine Head
Symetrix 562E
Prism Maselec MLA-2

4 Outboard preamp
AMS Neve 1081
Cadac mic pres
CLM Dynamics DB400
DACS Micamp
Grace Model 201
Martech MSS10
Oram Octasonic

5 Outboard equaliser
AMS Neve 1081
HSE EQ1
IC Vac Rac TEQ 1
Joemeek VC5 Meequalizer
Prism Maselec MEA-2
TL Audio Ivory EQ-5013

6 Outboard Reverb
tc electronic FireworX

7 Combined outboard device
Cedar CRX
dbx 1086
Junger Vamp 1
Manley Vox Box
Orban Optimod 9200
SPL Spectralizer
Symetrix 606
Symetrix 628
TL Audio O-2031

8 Monitors
DynaudioAcoustics C4
FAR AV5
Hafler TRM8
Harbeth Monitor 30
Harbeth Monitor 40
JBL LSR32
Meyer HM1S
Miller & Kreisel THX monitors
Quested F11

9 Microphones
AKG SolidTube
Audix D4
beyerdynamic MCD100
beyerdymanic MCE 82
BPM CR-10
Brauner VM1
Earthworks QTC1
Elation KM-201
Neumann TLM50
Oktava MK319
Pearl C-22
Rode NT1
Sondelux U195
Sonodore RCM-402
Trantec S4000

10 Convertors
Apogee AD-8000
dCS 972DDC D-D
Prism AD2/DA2

11 Audio editor
Creamware TripleDat 16
Digidesign Pro Tools 24
Orban Audicy
Otari DX-5050
Otari PD-80
SA&V Octavia 8/24
Soundscape v2

12 Audio recorder
DAR OMR8
Digigram PCXpocket
Fostex D160
HHB CDR800
Sonifex Courier
Sony PCM-R700
Tascam DA-98
Tascam DA-302
Tascam MMR-8
Zaxcom Deva

13 Special category
Antares Auto-Tune
Calistan Solutions Wave Safe
Dolby Surround Tools
Graham Patten Sound Pals
Klein & Hummel TRA60/NA
Lab Gruppen DSP24
Lexicon Studio System
Maycom ISYS ISDN

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SSAIRAS FAX VOTE

1 Large scale console	**8** Monitors
2 Medium to small scale console	**9** Microphones
3 Outboard dynamics	**10** Convertors
4 Outboard preamp	**11** Audio editor
5 Outboard equaliser	**12** Audio recorder
6 Outboard Reverb	**13** Special category
7 Combined outboard device	Your Unique Reader Registration Number

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Letters

Freeway delay

WE ARE A busy mobile recording and mixing facility based in Canada and have been subscribing to the Landline Delay Line Service for some 10 years and wonder if any of your readers would like to share their experiences with us as we feel we are approaching the time to reappraise the situation.

The Service acts as a delay by implementing the delays inherent in landline links to produce creative recording effects and while we have learned to live with the changeable audio quality, at times the degradation can be especially pleasing, but we are yet to conquer the logistics of setting delay times accurately and dependably. At present, we employ the recommended technique of driving the mobile up or down the landline in order to shorten or lengthen the loop and with it the delay time. At first this was a haphazard process but our driver has now identified a number of locations where desired delays can be captured.

The problem lies with the phenomenal mileage that can be required for a project and the fact that we have to be prepared to up the jacks and move on 400 miles for the sake of a double track on a bass guitar. Additionally, some of the modern delay effects like bpm matching are particularly difficult with our system and can involve crossing borders. Our truck covers more than 100,000 miles a year and service costs, gas and a physical inability to do the number of projects we would like means that the sums just haven't been adding up for years.

We have read about digital delay lines in your magazine but are concerned at how cheap they are. Your reviewers consistently praise them for their sound quality but are they really professional systems? Our feelings here are that we are reluctant to commit to anything that that won't be around in a few years' time. Have any of your readers had experience of using satellite in this capacity?

Yves Kordula, Big Beaver Sound, Canada

String driven thing

I AM AN enthusiastic recruit to audio recording and a great fan of British recordings from the sixties and seventies. Although I first set up my studio around ADATs and Mackies, I am now rebuilding it around collection of classic tube gear and winning quite a following here in West Virginia. I do seem to have run into a problem, though.

I have seen reference to an old artificial reverb system widely used in old recordings but can't track it down—it's called GBS. It took a while to find out, but I now know that GBS stands for Great British String. What I need from you is either a technical account of how you get reverb from string or a contact for someone who can supply me with it. I've tried, but I just can't work out how anyone can get reverb out of 'string'; is it some special kind of string or do you have to do something special with it.

I have to take my hat off to the Brits—when it comes to audio innovation, they're something else!

PS. Someone mentioned 'super string'. Am I getting closer?

A Hick, Vintage Studios, Virginia

De-light

I HAVE TO EXPRESS my growing concern over the emergence of ever increasing numbers of 'sound restoration' devices.

While—given the technical background to audio recording—the opportunities for profitable usage of scratch and hiss removal is self-evident, I am concerned that the obsessive search for devices to perform de-thissing and de-thatting is set to challenge the very foundations of our art. With hiss, snap, crackle and pop a thing of the past, manufacturers seem intent on enabling us to eliminate just about every aspect of a recording. How else can we explain de-hummers and de-phasers? And I read now that the British Cedar company has a de-buzz box—I ask if this is a device designed to remove unwanted door bell noises from South American soap operas or one to eliminate the Lightyear character from the *Toy Story* story. If this continues, what, I ask, might we expect from boxes called de-code (removal of time-code bleed?), or de-beers (restoration of intelligibility to an inebriated voice-over artist), or de-borah (injecting interest into corporate videos?), or de-brief (a domestic television filter rejecting reception of the great British farce?)?

Perhaps we will not be content until we have the ability to digitally de-everything we have recorded. Personally, I use the same method devised for analogue—the Erase button.

DB, Bolivia

Tannoy System 600A; Reveal

Beginning a series of objective loudspeaker reviews with Tannoy's System 600A and Reveal models, **Keith Holland** explains the test methodology and results, and explains how to interpret the measurements and translate them into operational considerations

UNLESS OTHERWISE stated, all of the measurements are carried out in the anechoic chamber at the Institute of Sound and Vibration Research (ISVR), University of Southampton. The chamber has a volume of 611m, and is fully lined with glass-fibre-filled wedges of 1m length. The measurement microphone is a Brüel & Kjaer type 4133 condenser microphone powered by a type 2609 measuring amplifier.

For **on-axis frequency response and sensitivity**, the loudspeaker under test is mounted on a flat-topped pole facing a boom-mounted microphone a distance of 1.7m away to reduce near-field geometric effects (note, this distance may be increased for larger loudspeaker systems). In most cases, the microphone is placed on a line normal to the drive-unit axes, passing between the geometric centre of the drivers (between the LF and HF driver centres for a 2-way system). Some loudspeaker manufacturers recommend an ideal listening position away from this line and in these cases their guidelines are followed in the placing of the microphone.

Tannoy System 600A

Pink noise is used as the test signal, ensuring a good signal-to-noise ratio throughout the audio bandwidth. For loudspeaker systems with passive crossovers, the input voltage signal is monitored at the loudspeaker terminals thus eliminating power amplifier and loudspeaker cable effects; the amplifier and cable then act as a perfect voltage source for the purposes of the measurements. For active systems, the line-level input to the loudspeaker system is used as the reference. The frequency-response function is then calculated between the output of the microphone and the reference input signal. The on-axis frequency response is scaled to represent absolute sensitivity in terms of dB SPL for 2.83V input at 1m distance for systems with passive crossovers; no attempt is made to give sensitivity figures for active systems as these are infinitely variable at the manufacture stage, and in many cases, depend upon the input socket used.

A complete frequency response measurement shows the variation of phase with frequency as well as amplitude. Phase plots are notoriously difficult to interpret, especially on a logarithmic frequency scale, so the phase response is omitted from these results, and is replaced by the more accessible acoustic centre measurements (see step response discussion below)

Two data-analysis systems are employed—a Diagnostic Instruments PL202 Realtime FFT Analyser is used for real-time monitoring of the measurements with a resolution of 1600 frequency lines 0Hz–20kHz. The microphone and reference signals are also stored on digital tape for subsequent high-resolution analysis. The frequency response functions as published are the result of averaging 160 records with a resolution of 16,384 frequency lines 0Hz–20kHz.

For **directivity** measurement, the setup is as for the on-axis measurement but the loudspeaker is angled away from the microphone. The frequency response function is measured at 15°, 30°, 45° and 60° off-axis horizontally, and 15° and 30° off-axis above and below vertically.

Harmonic distortion is measured by feeding a signal that sweeps logarithmically from 10Hz to 10kHz in 1 minute to the loudspeaker, and storing the microphone output on digital tape. The signal level is adjusted to an equivalent level of 90dB SPL at 1m, representing 10% of the maximum output capability of many small monitor systems. Analysis of the microphone signal yields levels of the fundamental and 2nd to 10th harmonics. Carrying out similar analysis on the test signal ensures that the harmonics present are below the noise floor and can therefore be ignored. For clarity, the results will only be presented for harmonics present at levels greater than -60dB (0.1%) relative to the nominal mid-frequency level of the fundamental.

Total power response measurements are carried out in the large reverberation chamber at ISVR. The chamber has a volume of 348m and has a reverberation time of 10s at 250Hz to 5s at 2500Hz. The loudspeaker under test is mounted on a stand 1m from the floor in two different positions, and a boom-mounted microphone is slowly swept throughout the chamber while the loudspeaker reproduces a pink noise signal. The results are presented as third-octave bands representing the total power response of the loudspeaker as a function of frequency.

The **step response**, **power cepstrum** and **acoustic centre** results are all derived from the high-resolution on-axis frequency response function. The step response is calculated as the time integral of the inverse Fourier transform of the frequency response and is plotted against time for the first 20ms.

The power cepstrum is the Fourier transform of the logarithm of the amplitude of the frequency response. The resultant quefrency response chart has the units of time and is plotted for the first 2ms. Note that the names of the quantities in cepstral analysis are anagrams of their frequency domain counterparts: cepstrum (spectrum), quefrency (frequency) and so on. The power cepstrum of the raw frequency response is dominated by the low-frequency roll-off of the loudspeaker, so this is removed to reveal more subtle detail. The low-frequency roll-off is removed by first finding the cut-off frequency and

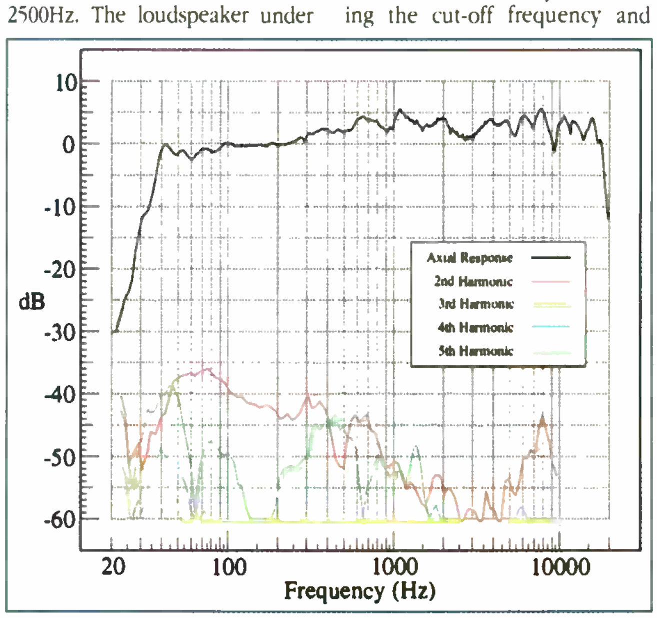


Fig.1: System 600A on-axis frequency response and harmonic distortion

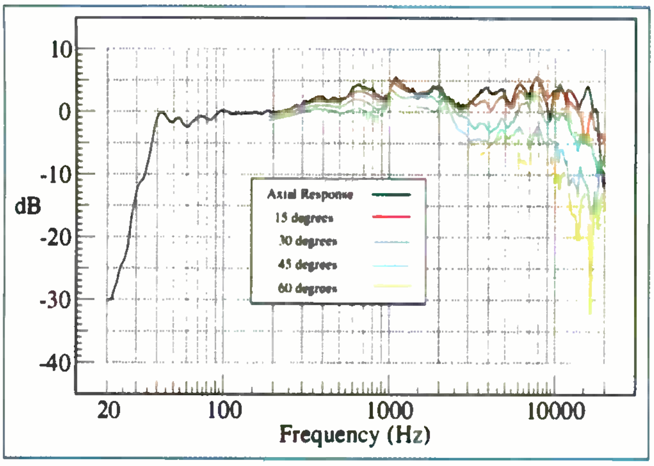


Fig.2: System 600A off-axis frequency response

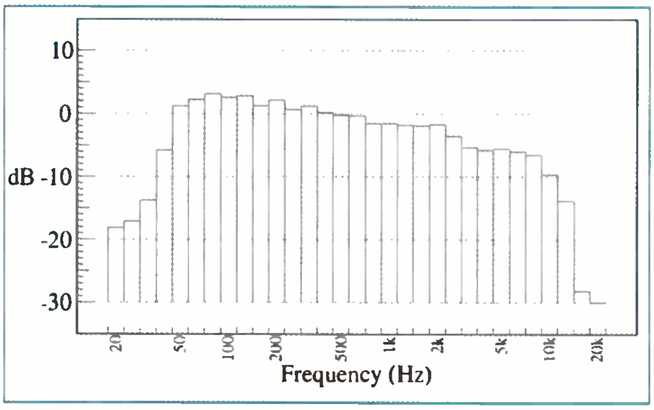


Fig.3: System 600A total power response

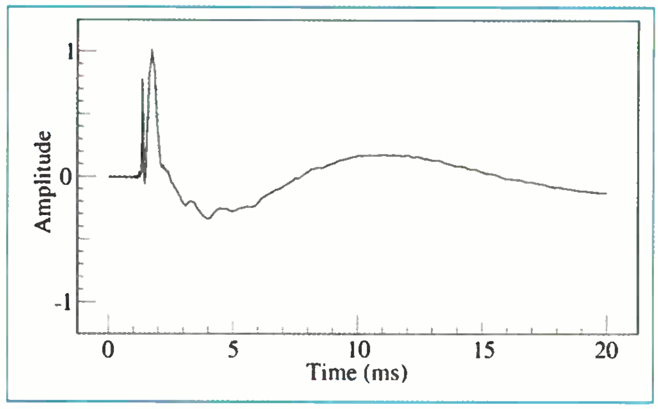


Fig.4: System 600A step response

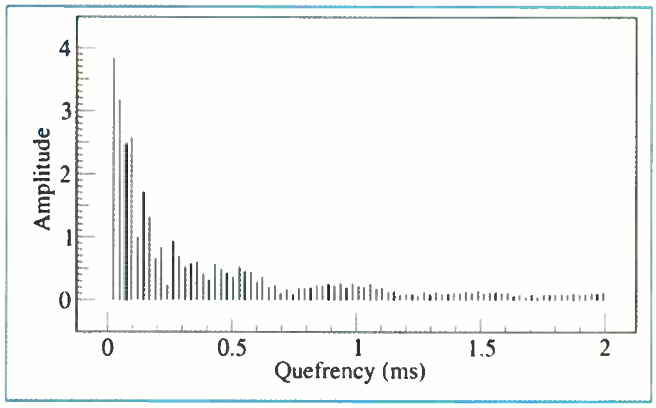


Fig.5: System 600A power cepstrum

order using a curve-fitting routine, from which an inverse filter is designed. Simply applying the inverse filter yields unacceptably wide variations in response at very low frequencies where the frequency response measurement is corrupted by noise, so the resultant response is multiplied by the envelope of the filter prior to calculation of the cepstrum. (Interested readers are referred to a paper published on this technique: 'The Use of Cepstral Analysis in the Interpretation of Loudspeaker Frequency Response Measurements', by KR Holland, *Proceedings of the Institute of Acoustics*, 15(7), 1993.)

The acoustic centre is calculated from the phase response of the loudspeaker. The group delay is first calculated as the slope (differential) of the phase response with respect to frequency; the equivalent position of the acoustic centre of the loudspeaker then results from multiplication of the group delay by the speed of sound. The acoustic centre is plotted as an equivalent distance in metres relative to the front baffle of the loudspeaker.

INTERPRETATION OF the measurements can be approached as follows: The **on-axis frequency response** is considered to be the most important of all loudspeaker specifications. It represents the level at which discrete tones of different frequency are reproduced by the loudspeaker under ideal (free-field) acoustic conditions. For studio monitoring purposes, it is widely accepted that this response should be as even and smooth as possible, and, ideally, should extend from the lowest audible frequencies (20Hz) to the highest (20kHz). Extending the high frequency response to 20kHz presents little challenge to loudspeaker designers nowadays, but the low end of the bandwidth is limited, practically, by the physical size of the loudspeaker; although larger monitor systems may be expected to reach the 20Hz target, it is not considered to be a fault if smaller systems roll-off at a higher frequency. Smooth variations in response, such as a gentle rise or fall with frequency are generally more acceptable than rapid variations such as peaks or dips, which are indicative of resonances or interference effects.

Although a vital specification for public address loudspeakers, **directivity** is of less importance in the studio environment where most (critical) listening is carried out on, or near, the loudspeaker axis. However, the off-axis sound radiated by the monitors may return to the listener via reflections from control room walls, and so on. The main directivity requirements for a monitor loudspeaker are therefore a reasonable coverage angle, both horizontally and vertically, over which the integrity of the on-axis sound is maintained, and a smooth response at all off-axis angles.

Harmonic distortion gives an indication of the degree of nonlinearity present in the loudspeaker. Such nonlinearities give rise to the generation of harmonics in the reproduction of tonal signals which were not present in the input signal, and the generation of sum, difference and other intermodulation products when reproducing complex signals such as speech or music. Some nonlinear behaviour is inevitable, especially at low frequencies, but as a general rule, the lower the harmonic distortion produced the better, and levels below -40dB (1%) are considered to be acceptable.

Listening rooms, such as studio control rooms, are neither anechoic nor fully reverberant, but have acoustic properties that lie somewhere between these two extremes. The direct sound from a loudspeaker to the listener is a function of the on-axis response of the loudspeaker, but any reverberation is a function of the **total power response**—the response summed over all angles. Thus the power response of the loudspeaker gives an approximation to the frequency distribution of the reverberant sound field in a listening room (the exact nature is, of course, very room dependent).

The **step response** measurement represents the response of the loudspeaker to a sudden rise in voltage input. Interpretation of the step response, in terms of its relevance to loudspeaker performance, is considered to be easier than the closely related impulse response, while still allowing transient rise-time, and so on, to be studied. The step response is particularly revealing of crossover alignment and drive-unit relative phase problems.

The **power cepstrum** >>>>>

Tannoy Reveal

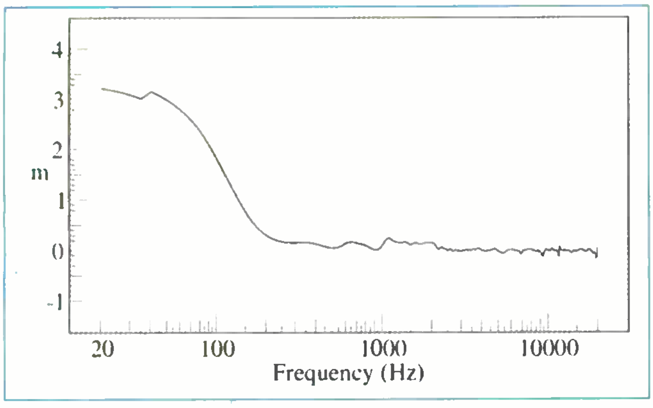


Fig.6: System 600A acoustic centre

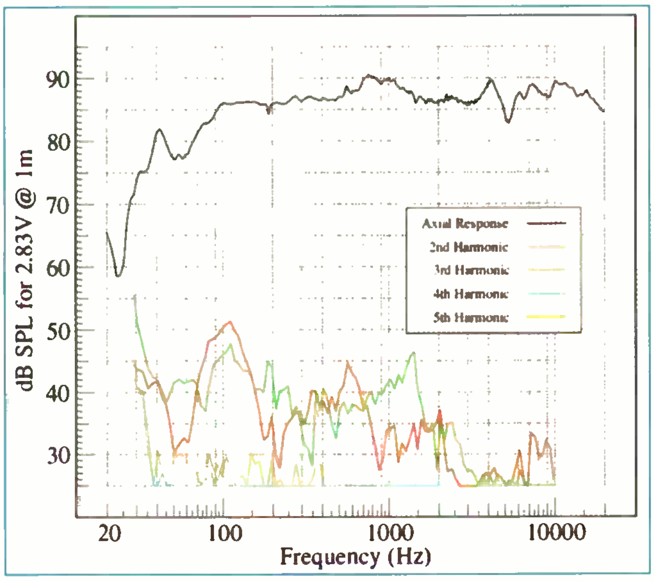


Fig.7: Reveal on-axis frequency response and harmonic distortion

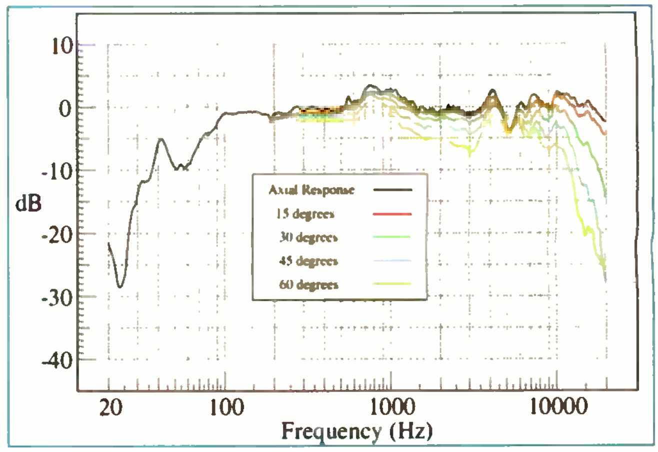


Fig.8: Reveal horizontal off-axis frequency response

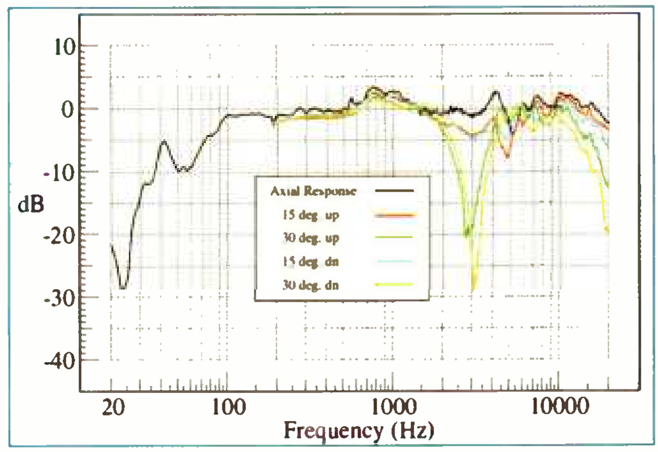


Fig.9: Reveal vertical off-axis frequency response

<<<<< is unusual in that it presents time-domain information about the response of the loudspeaker without considering the phase. Of particular value is the fact that reflections and echoes show up very clearly as spikes in power cepstra, even though these may not be apparent from the frequency response plot alone. In a loudspeaker system, these reflections may be due to such phenomena as cabinet edge diffraction problems, horn-mouth termination effects, drive-unit diaphragm termination effects. Broader features in the cepstrum are due to other response irregularities such as resonances or shelves in the response. The displacement along the quefrency axis of a spike in a power cepstrum represents the delay in seconds from the direct signal to its reflection, and the height, the relative strength of the reflection. Thus one can easily compute path length differences and hence possible positions for the source of the reflection.

The acoustic centre of the loudspeaker as a function of frequency is presented as an alternative to the phase of the on-axis frequency response function. The plot may be interpreted as the variation in effective loudspeaker position with frequency, in terms of the time-of-arrival of transient sounds. Some movement of the acoustic centre is expected at low frequencies due to the group delay associated with the low-frequency roll-off. Naturally, those loudspeakers possessing the smallest variation in acoustic centre with frequency will, all else being equal, preserve broadband transients better than others.

THE TANNOY SYSTEM 600A is an active loudspeaker system with an in-built amplifier and electronic crossover package. It is intended for use as a close-field studio monitor system with its 6-inch dual-concentric >>>>>

Romance

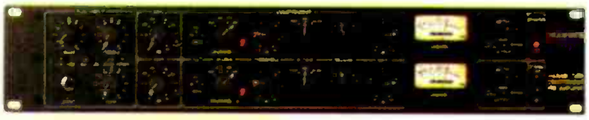
[but calculated]

A technical statement laced with passion. The Drawmer 1960s.

1962

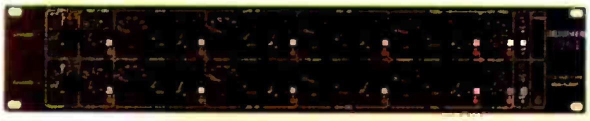
24-bit digital vacuum tube preamp

• Full 24 bit A/D conversion with 48K/44.1K sample rate. • Two ultra high-quality pre-amp stages providing 'clean gain' from -25dB to +60dB. • Dynamic range in excess of 133dB. • Comprehensive digital output formats include AES/EBU, SPDIF, Tascam™ TDIF-1 and Alesis ADAT™. • Creative processing including Fine Tune EQ, Dynamic Spectral Enhancement, Variable Tube Drive, Hi/Lo Pass filters and transparent Limiting can be inserted in the signal path. • Word Sync in/out.



1960

vacuum tube preamp/compressor



1961

vacuum tube equaliser

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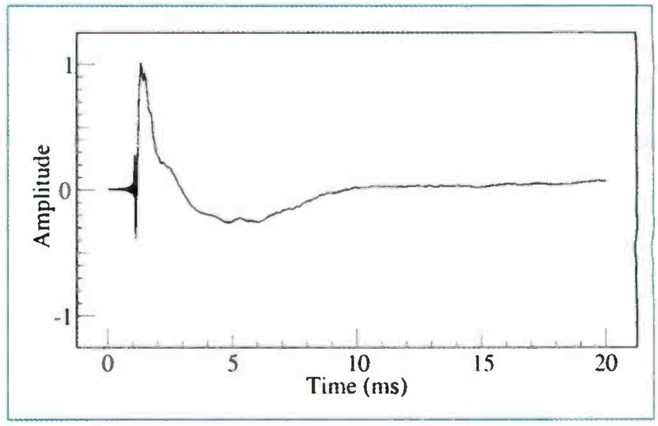

Fig.10: Reveal step response

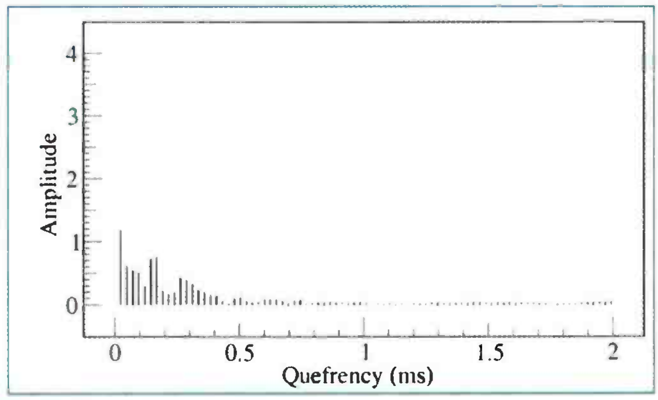

Fig.11: Reveal power cepstrum

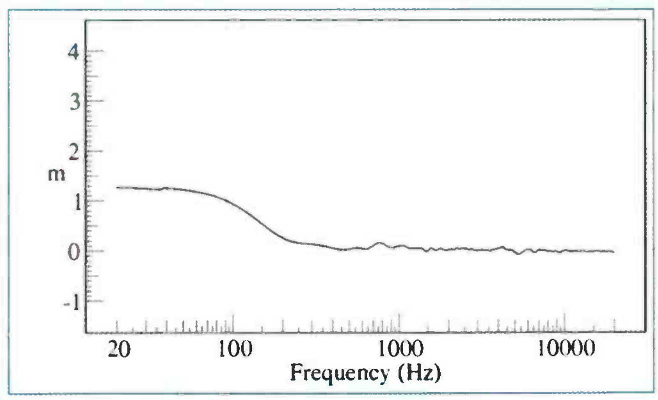

Fig.12: Reveal acoustic centre

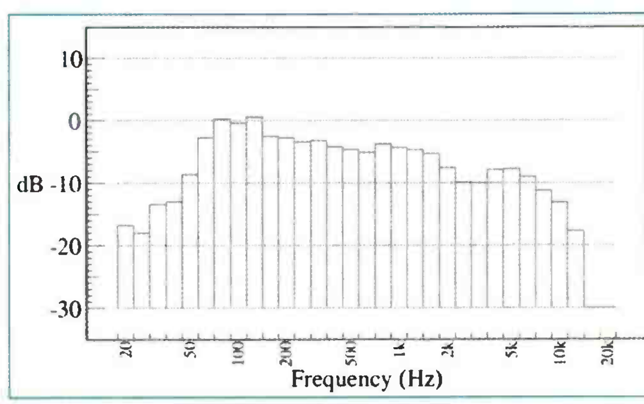

Fig.13: Reveal total power response

<<<<< arrangement allowing users to choose either vertical or horizontal mounting with no compromise in performance. The construction of the cabinet is to a high standard of finish, and the units feel very sturdy. The loudspeakers are quite heavy for their size at 9.5kg, so care should be exercised when mounting them on fragile monitor bridges. The cabinets have dimensions of 220mm x 360mm x 290mm front-to-back with an internal volume of 13 litres.

The bass driver has a polypropylene cone, and, as with other Tannoy dual-concentric

HHB CDR800

Professional Audio Hardware and Recording Media

Genex GX8000

PORTADAT

HHB Advanced Media Products

A Brief History

HHB Communications, established in London in 1976. Annual worldwide sales now exceed $30 Million.

Founded in 1976 in London, England, HHB Communications now occupies a unique position in the professional audio industry. Our involvement with digital audio began way back in 1983 when HHB pioneered the professional acceptance of digital mastering with the Sony PCM F1. Since then, we've grown to become Europe's leading supplier of professional audio equipment, playing a key role in the development and introduction of DAT, CD-R, MO and MiniDisc.

Our customers include leading professionals from all the major industry sectors – music recording, film sound, video post and broadcast. A long and close association with these customers has gained us a unique understanding of their working processes and equipment requirements, inspiring HHB to develop our own innovative range of HHB digital audio hardware and recording media which we now export all over the world.

In this publication, you'll read about the award-winning CDR800 CD Recorder, the ground-breaking Genex GX8000 hi bit, hi sampling 8 track MO recorder, the industry standard PORTADAT portable DAT recorder and the full range of HHB Advanced Media Products, widely regarded as the most dependable digital recording media currently available. You'll also read about the people who rely on our products day in, day out to deliver the highest possible standards of performance and reliability.

HHB wins prestigious Professional Recording Association Award for Technical Excellence

HHB is recognised internationally for its commitment to technical excellence. Our CDR800 CD Recorder is the winner of a PAR Excellence Award from Pro Audio Review, and both the PORTADAT Portable DAT Recorder and Genex GX8000 MO Recorder have been nominated for TEC Awards.

Awarding HHB the prestigious Professional Recording Association Award for Technical Excellence, UK APRS Chief Executive Mark Broad said: "HHB champions the cause of the end-user, significantly influencing the design of many major products in the process. Now, no longer satisfied with simply influencing other manufacturers, HHB have developed their own range of advanced digital recording hardware and media products. Such innovation, combined with a consistent commitment to technical support, makes HHB worthy winners of this, the first APRS Technical Achievement Award".

mic and innovative force in professional audio.

HHB Worldwide

HHB is a truly global organisation with headquarters in London, and its own distribution companies based in Los Angeles and Toronto. HHB digital audio hardware and recording media is available at more than 1000 specialist pro audio dealers worldwide, carefully selected for their experience and expertise.

HHB is also an active participant in many industry associations around the world, including the Audio Engineering Society (AES), the National Association of Broadcasters (NAB), the Institute of Broadcast Sound (IBS), the Association of Professional Recording Services (APRS) and the National Association of Music Merchants (NAMM). We are regular exhibitors at all the major international conventions, and frequently host our own seminars on subjects as diverse as DVD and the importance of choosing the right DAT tape.

CONTENTS

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Call HHB with details of interesting projects you're working on using HHB professional audio equipment or media, and you could be featured in the next HHB newsletter.

Make some space in your rack. Prof

HHB CDR800

The HHB CDR800 is the world's first truly affordable professional audio CD recorder. With a CDR800 in the rack, home recording musicians and broadcasters alike can produce top quality final copies of their work on a universal format that just about everybody has the facility to replay. The HHB CDR800 should not be confused with consumer CD recorders that use higher priced consumer CD-R media. The CDR800 is a fully professional device with balanced analogue inputs and an AES/EBU digital input, no SCMS and, most importantly, uses the widely available and less expensive conventional CD-R discs. The CDR800 also sets new standards in sound quality, ease of use and reliability.

- Designed specifically for the professional user
- Exceptional build quality and reliability
- Excellent sonic performance
- Straightforward operation
- Uses widely available, low-cost CD-Recordable media, not high cost 'consumer' discs
- SCMS free
- Comprehensive digital interfacing: AES/EBU input, SPDIF I/O (coaxial and optical), plus balanced analogue inputs
- Sample rate converter accepts any frequency from 32-48kHz and converts to CD standard 44.1kHz
- DAT, MD and CD IDs can be copied along with audio, enabling one touch cloning
- 5 simple record modes to cover all requirements

HHB's CDR800 CD big hit with produ

The Cure, Manic Street Preachers, Texas, The Beautiful South, Siouxsie & The Banshees – producer Mike Hedges' CV reads like a summary of everything that's good about British popular music over the past 20 years. From his own magnificent studio in the French countryside, where he's famed for combining vintage recording equipment with the best that modern technology has to offer, Mike told us about his latest acquisition, an HHB CDR800 CD Recorder. "I use the CDR800 as a direct replacement for my cassette deck. I simply compile the material on DAT and then go straight to CD. It's much easier to use than a computer CD writer."

Until now, Hedges felt that stand alone CD recorders have been too expensive. "The CDR800 is the first CD recorder to deliver the right combination of value and quality. It's performed faultlessly for us so far and is very easy to use."

CDR800 performs brilliantly for Grammy winner John Jones

Los Angeles based producer, writer, artist, engineer, programmer and Grammy award winner John Jones is another recent purchaser of the HHB CDR800. Jones received the 1996 Grammy Award as a producer, Album of the Year and engineer, Best Pop Album, Album of the Year for Celine Dion's LP 'Falling Into You'.

"I heard about the CDR800 and looked at all of the CD writers available, including computer CD-R drives, but I wanted something stand alone," says Jones. "As soon as I got the CDR800, however, I realised that it was much more. So far, I have made over one hundred CDs with the unit and have never had a problem or experienced a drop

Grammy award winner John Jones, in the studio with his HHB CDR800 CD Recorder.

Recorder is a
cer Mike Hedges

Throughout a career that has seen him go from studio tea boy to internationally acclaimed producer, Mike has always been a valued HHB customer. "I've been buying from HHB for 20 years. The service is the best and on the rare occasion when there is a problem, things always get sorted out immediately."

Big man – big talent. British record producer and CDR800 owner Mike Hedges.

out. Whether I am mixing, rough mixing or recording live, I record to the CDR800. I like it because it is immediate, I can't erase it and it becomes archived forever. Being able to playback any segment of a CD without waiting for the tape to rewind helps make finding sound samples and song ideas a cinch."

"The best CD-R on the market" John Jones with his CDR800

"The best thing about the CDR800 is the sound," continues Jones. "The internal A to D converters are as crisp and clear as any external converters that I have used in the past. The copy protection is another quality feature. When you make a CD for someone, you can make it so they can't copy it digitally or so they can only make one copy."

"As a stand alone stereo recorder, it is just brilliant, I think it is the best CD-R on the market."

Canadian Broadcasting standardises formats with a CDR800

The Canadian Broadcasting Corporation (CBC) is using the HHB CDR800 to transfer various antiquated formats, such as 45s, 78s, LPs, etc. into one common, universal format, the compact disc.

Adrian Shuman, user services librarian, says, "If we have something that is very noisy we can simply store that recording on the CDR800. We then expose it to CEDAR, clean it up and burn a new CD from there. It has allowed us to use parts of the library that had been unused for quite a long time because technicians were unfamiliar with the older formats."

CBC's broadcast library has been transferring between sixty to eighty hours of material a week and has found the machine to be highly reliable and efficient. "The CDR800 is wonderfully easy to use," says Shuman. "I opened the booklet, followed the instructions and was burning my own CDs within five minutes. It has been very flexible for us and worth the investment."

Presented at the New York AES Convention, the CDR800's prestigious Pro Audio Review 'PAR Excellence Award'

Genex In a world where 16 bit audio is no lor

GENEX GX8000

Developed by digital audio specialists Genex Research, the GX8000 sets new standards in digital 8 track recording. Only the GX8000 can record at up to 24-bit resolution and 96kHz sampling rate, to deliver a dynamic range and frequency bandwidth which out-perform any other tape or disk based digital multitrack. No other recorder is better equipped to meet the needs of new formats such as DVD. But the GX8000 is not just about sound quality. During development, we worked closely with professionals in all industry sectors from dubbing to mastering, to ensure that this recorder fits seamlessly into every application. Perfect as a film dubber, ideal as a video post production tool, ergonomic as a location music recorder and reliable as an archiving and mastering unit, the Genex GX8000 provides the ultimate solution in digital audio – right across the board.

- Up to 24 bits per sample recording for full compatibility with DVD, etc
- Sampling rate selectable up to 96kHz
- Simultaneous recording of 8 x 20 bit tracks with internal A-Ds
- Internal chase synchroniser with EBU / SMPTE timecode generator / reader
- Forwards, backwards and varispeed lock to timecode or bi-phase
- Sample accurate synchronisation of up to 8 machines (64 tracks)
- Built in 8 channel digital mixer
- Built-in ISO standard 2.6GB MO disk drive
- Seamless switching between internal and external drives for extended continuous recording
- RS422 (Sony 9 pin) and RS232 interfaces
- SCSI interface for links with external DAWs

New Genex remote control now available

20th Century Fox in 20-bit with t

20th Century Fox recently used the Genex GX8000 to mix and dub all of the songs and to dub the score for 'Anastasia', its first ever animated feature film. This full-length motion picture about a fabled lost Russian princess has received outstanding success among critics, as well as at the box office.

Academy award-winning Re-recording Mixer Bill Benton, who mixed the music on 'Anastasia', says, "It was very important to get the right sound on this project because

'Anastasia', 20th Century Fox's first animated feature film

24-bit mastering with Genex

Renowned mastering engineer Denny Purcell recently mastered George Strait's new album with the high resolution Genex GX8000 hi-bit, hi sampling multi-track MO disk recorder at Purcell's studio, Georgetown Masters in Nashville, TN. With the album's quality requirement of 24 bits and 88.2kHz sampling, the Genex was the perfect choice.

Purcell says, "The Genex unit is really the first magneto optical disc recorder that can record 24 bits and 88.2k. Genex was the logical choice as far as capabilities and sound quality. There is just a dramatic difference in sound with the Genex GX8000, and everyone was astounded when they heard it at 88.2kHz compared to 44.1kHz. Its reliability has also been amazing. We have worked with it for weeks on end and it has performed outstandingly."

The George Strait album used the Genex to record the mix master. Chuck Ainlay, Engineer for the album, says, "We had to go to some removable medium for our two mix master. Basically we went through outboard [Pacific Microsonics HDCD, 24 bit - 88.2k] converters using the Genex MO disk as our storage medium." By utilising the high resolution capabilities of the Genex GX8000, Purcell and Ainlay were able to create a master with the ultimate in audio fidelity on to a compact, rugged, removable and tapeless media: the MO disk.

mixes 'Anastasia' songs wo Genex GX8000s

the score is gigantic and wonderfully orchestral with a large choir in it. The songs are all like show tunes with huge band music, numerous lead vocals, and a large chorus – they needed a Broadway sound. We decided to use the GX8000 at 20 bits for audio clarity. Also it runs at a faster sampling rate, which makes quite a difference in the music. It is a great sounding digital system."

Bill Benton, John Kurlander and Doug Botnick with Genex GX8000s at the new 20th Century Fox Scoring Stage.

Grammy award-winning Scoring Mixer John Kurlander, who mixed all the songs for 'Anastasia' directly to two Genex GX8000 eight channel recorders locked together, says, "This film itself is cutting edge in terms of the technology behind it, and we wanted a recorder that would give us state-of-the-art sound. The GX8000, which represents new digital technology, gave us the exceptional sound quality we were looking for to match the quality of the animation."

Kurlander says, "I like working with the Genex unit and, most importantly, it is reliable and sounds great. Before Genex came along, it was very difficult to do multi-channel 20 bit recordings; the only way was to use bit-splitting technology, which is cumbersome and awkward. The quality of 20 bit is great and is becoming very popular in the film business."

Georgetown Masters also recently used the GX8000 on a trend setting first (for Country): A '5.1' surround sound release of Vince Gill's album 'High Lonesome Sound'. For Gill's album, Purcell used the Genex GX8000 unit for the DTS 5.1 surround format, both for the mix and the master.

"What is absolutely wonderful about the Genex unit is it works just like an analog tape recorder," says Purcell.

Denny Purcell mastering in 24-bit/88.2kHz with the Genex GX8000 and HHB MD2.6GB Disks

Genex a major hit Worldwide

Air Studios' Chief Engineer Geoff Foster with the GX8000

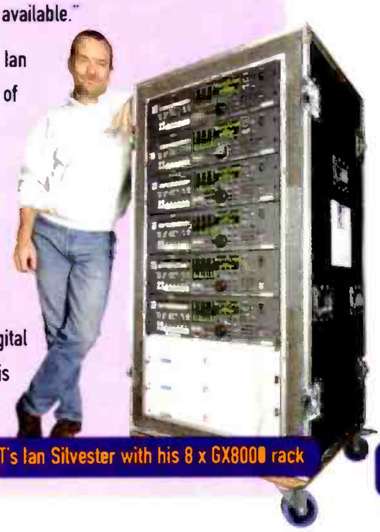
At George Martin's Air Studios in London, the Genex GX8000 is used in all kinds of work, including the recording of David Arnold's score for the latest Bond Movie 'Tomorrow Never Dies'.

"When we supply a finished master, we have to be absolutely confident in the excellence of our product" says Chief Engineer Geoff Foster. "We can relax in the knowledge that the sound quality of the Genex is vastly superior to that of all the tape-based digital 8-track recorders."

Alain Roy (left) and Francois Deschamps (right) at CINAR

Meanwhile in Montreal, Canada, the major multinational TV production company CINAR uses 15 GX8000s for mixing, re-recording, recording and dubbing. VP of Studios Francois Deschamps says: "Unlike hard disk, MO is a removable medium and unlike analogue tape, its archiving stability is amazing – more than 100 years. By recording at the maximum possible dynamic range we are preparing for new formats as they become available."

Pictured right is Ian Silvester, owner of Digital Audio Technology, a London based company specialising in the rental of leading edge digital systems, with his 48 track Genex system.



DAT's Ian Silvester with his 8 x GX8000 rack

PORTADAT

Sounds great, handles superbly, bu

PORTADAT

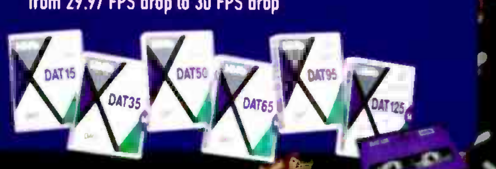
HHB's PORTADAT range of professional portable DAT recorders combine superb sound quality, rugged reliability, excellent handling and a full complement of professional features, making PORTADAT the industry standard in location DAT recording. There are two models in the range, the PDR1000 and the timecode-equipped PDR1000TC, both developed in consultation with leading location sound recordists to ensure convenient operation in the field. The PORTADAT is light and strong, with a wide variety of power options. A full range of professional accessories is available, including AC adaptors, battery chargers / dischargers, hard and soft cases and cables.

PDR1000

- Developed in consultation with leading location sound recordists
- 4 heads for confidence monitoring
- Rugged 4 motor direct drive transport
- High quality mic preamps and DACs ensure exceptional sound quality
- 32, 44.1 and 48kHz recording via analogue inputs
- Balanced XLR analogue inputs
- SPDIF and AES/EBU digital I/Os
- Advanced Nickel Metal Hydride rechargeable batteries
- Phantom power
- Built in monitor speaker
- Optional M/S monitor matrix selects stereo, mono left, mono right, mono sum and M/S (mid-side) modes

PDR1000TC

- As PDR1000, plus..
- Records, generates and references to timecode in all existing international standards
- Jam sync facility
- Converts absolute time to timecode
- Optional Master Sync module for maximum 1 frame / 10 hours drift. Aaton camera compatibility and pull up from 29.97 FPS drop to 30 FPS drop



Light and compact, the PORTADAT was ideal for use in tight paces gathering sound effects for 'Titanic'.

At work on 'Titanic recordist Chris

This year's blockbuster movie 'Titanic' is one of the most spectacular movies ever made in the history of film making. Starring Leonardo DiCaprio and Kate Winslet and directed and written by James Cameron, 'Titanic' combines epic action-packed romance and drama with cutting edge audio and visual effects technology. Recently the winner of 4 Golden Globe Awards, including best picture, the film cost over $200 million dollars to produce, making it the most expensive film ever made.

When confronted with demanding directors and arduous shooting conditions, Sound Designer for Skywalker Sound Chris Boyes relied on his HHB PORTADAT PDR1000TC for performance and quality sound. The conditions for 'Titanic' provided a real world trial by salt water for the timecode equipped PORTADAT [illegible].

The HHB PORTADAT is considered vital not only for its sound quality, but for its design when the audio effects team goes out hunting for sound clips. "We used the HHB PORTADATs to gather effects for the 'Titanic' project," says Boyes. "We rented the ship Jeremiah O'Brian because the engine design is similar to the Titanic's. The piston shaft is 15 feet tall and we wanted to place mics in extremely tight and dangerous places. The PORTADAT was

lt to last. No wonder Hollywood loves the PORTADAT.

vith Skywalker sound Boyes and his PORTADAT

the recorder of choice because it is well designed. With its low noise preamps and A to D converters built into the unit, we didn't have any extraneous wires dangling about or extra equipment to carry. The unit is compact, performs well and is highly reliable... very important for the type of opportunistic work we do."

The PORTADAT performed faultlessly for Chris despite extreme humidity on location for the film 'The Lost World: Jurassic Park'

Boyes also worked on last summer's action-packed sequel 'The Lost World: Jurassic Park'. He and an audio effects team went to Costa Rica to record ambiences and interesting animal sounds with the PORTADAT. The audio for the sequel was slightly more complex than the original due to the fact that there were more dinosaurs in this film and the dinosaurs had more onscreen time. And capturing the right sound for a T-Rex or a Brontosaurus was made much easier thanks to the PORTADAT.

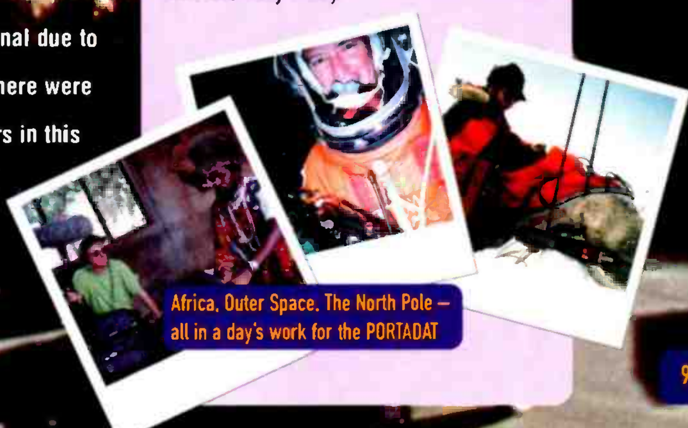
PORTADAT always in the action

Egyptian Radio and TV standardises on the PORTADAT for all OB work

ERTU, Egyptian Radio and Television Union, has purchased 300 PORTADAT PDR1000s, standardising on the popular HHB portable DAT recorder for all OB and ENG applications for both radio and TV broadcast. The PORTADAT was chosen after extensive evaluations of all available DAT portables. A 4 head transport, sound quality and proven broadcast reliability placed the PORTADAT on top.

On the subject of reliability, HHB's service department reports that with over 4000 PORTADATs now in use worldwide, not a single unit has yet been returned with a worn head drum. "The combination of a uniquely robust head chip material and individual motors for supply and take up reels mean that even machines that have clocked up over 3000 hours of use still show no signs of head wear" reports HHB Service Controller Gerry Glancy.

Not one single PORTADAT has yet been returned with a worn head drum



Africa, Outer Space, The North Pole – all in a day's work for the PORTADAT

HHB ADVANCED MEDIA PRODUCTS

High performance and unparalleled dependab

DAT

- Independently proven to be the most dependable brand of DAT tape.*
- Exceptional archival stability (>30 years).
- Available in 6 convenient lengths – 15, 35, 50, 65, 95 and 125 mins.
- Specially formulated binder ensures consistently low block error rates, even after 100 passes.
- Flexible base film minimises head wear.
- Fast discharge anti-static lid resists dust contamination.
- Cassette shell will withstand a temperature of 107°C (224.6°F) without warping.
- New hub lock assembly (Jan 98) improves braking action, reduces tape slack when ejecting, and hence provides better loading.
- Shatterproof Polypropylene library case.
- Professional labelling.

DDS90M

- DDS tape specially developed for DAW data back up.
- Exceptional archival stability (>30 years).
- High output.
- High density binding polymers ensure consistently low block error rates.
- Heat resistant shell withstands high temperatures resulting from continuous high speed shuttling.
- Controlled distribution of special lubricant reduces head wear.
- Positive locking hubs for improved tape handling.
- Embossed friction sheet eliminates vibration during high speed search.
- Improved slider design reduces loading and jamming problems.
- Professional labelling.

CDR74 Gold & P

- The World's first CD-R disc optimised for professional audio
- The most dependable CD-R disc on the market (archival stability >100 years).
- Exceeds Orange Book standards in all critical areas.
- Also available in printable form (CDR74 Gold P) with new improved white printing surface, ideal for use with Afex, Copytrax, Fargo and IMT printers.
- Specially formulated Phthalocyanine dye results in exceptional recording accuracy, and is less susceptible to the harmful effects of UV exposure.
- Professional labelling.
- *The HHB CDR74 Gold should not be confused with consumer CD-R audio discs which can only be used in consumer CD recorders.*

MD74

- Developed specifically for professional audio use.
- Block error rates 10 times lower than consumer MiniDisc media.
- Exceptional archival stability (>10 years).
- Can be used reliably for more than 1 million read / write cycles.
- Advanced sputter coated recording layer achieves exceptional recording precision.
- High carrier to noise ratio.
- Disc protected against extreme environmental conditions by a tough UV coating.
- Advanced foil shutter provides greater protection from dust contamination than conventional metal shutters.
- Special lubricating agent ensures optimum head contact.
- Professional labelling.

* Studio Sound 'DAT On Trial'

BEWARE! ALL DIGITAL RECORDING MEDIA IS NOT THE SAME

Unlike its analogue counterpart, digital audio is recorded as a series of discrete numbers. But this doesn't make the choice of recording media any less critical. Inferior quality tape and disc based digital recording media can put your valuable recordings at risk – if not today, then maybe tomorrow. Take DAT tape for instance. If the bond between the magnetic particle recording layer and the base film isn't strong enough, you're losing more and more of your recording every time you play the tape. The Block Error Correction that's part of the DAT format will cover things up for a time, until one day there's not enough data left to re-construct the recorded material and you're left with a damaged master. Similarly CD-R discs. The long term stability of the organic recording dyes used varies greatly from brand to brand, affecting the long-term security of the data recorded on the disc. Even issues such as the rigidity of a DTRS cassette shell are critical, as flexing can lead to loose tape packing and, ultimately, tape snapping.

ity make HHB Advanced Media Products No.1 with the pros.

MDD140

- MD Data format disc developed specifically for use in MiniDisc multitrackers (Sony, Tascam, Yamaha, etc.)
- Exceptional archival stability (>10 years).
- High carrier to noise ratio.
- Low block error rates, even after repeated passes.
- Professional labelling.
- *If you try and use an MDD140 in a standard audio MiniDisc recorder it will not work. If however you try to use an HHB MD74 MiniDisc in a MiniDisc multitracker designed for use with MD Data format discs, it will work, but will only allow 2 tracks to be recorded or played.*

MO2.6 GB

- 5.25" 1024 bytes per sector MO disk.
- Each disk individually tested and certified.
- Carries a lifetime (100 years) warranty.
- Developed specifically for hi-bit, hi-sampling digital audio recording.
- Exceptionally stable recording layer.
- Carrier to noise ratio in excess of 45dB.
- Consistently low block error rates.
- Specially compounded polycarbonate substrate functions dependably under extreme variations in temperature and humidity.
- Anti-static hard coating repels dust and minimises scratching.
- Professional labelling.

DA113

- Metal particle DTRS tape developed specifically for use with the Tascam DA88 and its derivatives.
- Designed to withstand the notoriously harsh conditions of DTRS 8-track recording.
- Exhaustively tested in a wide variety of applications from music recording to film dubbing.
- Exceptional archival stability (>10 years).
- Hi-packing recording surface delivers 54dB carrier to noise ratio.
- Flexible base film eradicates stretching and snapping.
- Specially formulated binder ensures negligible drop out, even after 100 passes (-0.5dB).
- Exceptionally rigid, heat resistant cassette shell.
- Professional labelling.

ADAT45

- Developed specifically for use with the Alesis ADAT 8-track recorder and its derivatives
- Exhaustively tested in a wide variety of typical ADAT applications.
- Exceptional archiving stability (>30 years)
- Ultra-fine cobalt ferric oxide magnetic surface delivers enhanced high frequency response.
- High output.
- Consistently low block error rates.
- Special high density binder stops oxide shedding, even after 100 passes.
- Rigid, high performance cassette shell ensures precision tape handling.
- Professional labelling.

With HHB Advanced Media Products, we set out to develop a complete range of digital recording media, delivering the highest possible standards of performance and long term dependability in every major format. Drawing on 15 years experience at the forefront of digital audio, we turned first to our customers – leading professionals working in all areas of audio production from music recording to broadcast – and sought their opinions on issues ranging from convenient tape lengths to label designs. Next, we went in search of the best possible manufacturing partners, specialists who could consistently deliver to our exacting standards. And it doesn't stop there. As a professional audio company with a worldwide reputation for technical excellence, we test all of our HHB Advanced Media Products on all the major recording hardware as it becomes available to ensure consistently high performance.

designs, the tweeter consists of a compression driver, mounted behind the bass driver, the cone of which acts as the horn flare for the tweeter.

Discussion of measurements: The on-axis frequency response (Fig.1) sits between the usually accepted limits of 3dB from about 40Hz to 19kHz, with, at each end of the spectrum, a rapid fall-off beyond these frequencies. The harmonic distortion is shown to be below -40dB (1%), at all frequencies except the lowest octave of the loudspeaker bandwidth. As the loudspeaker is a dual-concentric design, only one set of directivity measurements are presented (Fig.2). These show that the off-axis response falls smoothly with increasing angle from the axis; a fact that is borne out by the excellent total power response (Fig.3).

The step response (Fig.4) shows a short delay between the initial leading edge and the rest of the response, indicating some time-alignment problems; the delay is only about 200µs however, and is therefore probably inaudible. The power cepstrum (Fig.5) is fairly well controlled, except for a spike at about 140µs and another, smaller spike at 280µs. These may be due to discontinuities in the horn of the HF unit (the cone of the LF unit), as they represent distances of about 50mm and 100mm. The acoustic centre measurement (Fig.6) is dominated by an effective shifting of the source of low frequencies to up to 3m behind the loudspeaker. This represents a group delay of some 10ms, probably due to the rapid 6th order roll-off below 40Hz. This delay can also be seen in the step response; the low-frequency part of the response appears to begin later than the high-frequency part.

Overall, the loudspeaker system performs well, with no serious problems. The low-frequency time problem is probably a direct consequence of the sharp LF cut-off. The off-axis performance is particularly good.

THE TANNOY REVEAL is a small, 2-way monitoring loudspeaker with an integral passive crossover. By means of a specially shaped frame, the drive-units are mounted as physically close together as can be reasonably achieved. The two drivers are a 6-inch bass driver and a 1-inch soft-dome tweeter, mounted in a cabinet of 12 litres volume with a tuning port at the rear. The drivers both have shielded magnet assemblies. The cabinet face is sculpted in an unusual way to reduce cabinet edge diffraction problems. Overall, the loudspeakers are quite light in weight and are well finished. The rear-mounted terminals are substantial enough to allow for adequate gauge of loudspeaker cables to be attached.

The Reveal is rated at 50W continuous programme, and has a quoted sensitivity of around 90dB at 1m in typical listening situations. The nominal impedance is stated to be 6Ω.

The manual supplied with the loudspeakers is thorough, and has several pages devoted to positioning recommendations and potential problems. Tannoy (sensibly) strongly recommends that the loudspeakers be used in the upright (portrait) orientation, as the off-axis response problems (see measurements), which are inevitable with spaced-driver systems, are then confined to the vertical plane.

Discussion of measurements: The on-axis response of this loudspeaker (Fig.7) sits between 3dB from about 80Hz–20kHz. There is a peak in response at 40Hz which may be due to port tuning. The harmonic distortion is below -40dB except for a peak in the second harmonic at about 110Hz. The horizontal off-axis response (Fig.8) is extremely well controlled except for a peculiar widening of the coverage angle at 5kHz. The frequency coincides with a dip in the off-axis response, suggesting a dip in the polar distribution of sound radiation on-axis, rather than peaks off-axis. The vertical off-axis response (Fig.9) shows the expected severe dip at 30° above and below at the crossover frequency of 3kHz. This is common to all designs with spaced drivers (as opposed to dual-concentric for example).

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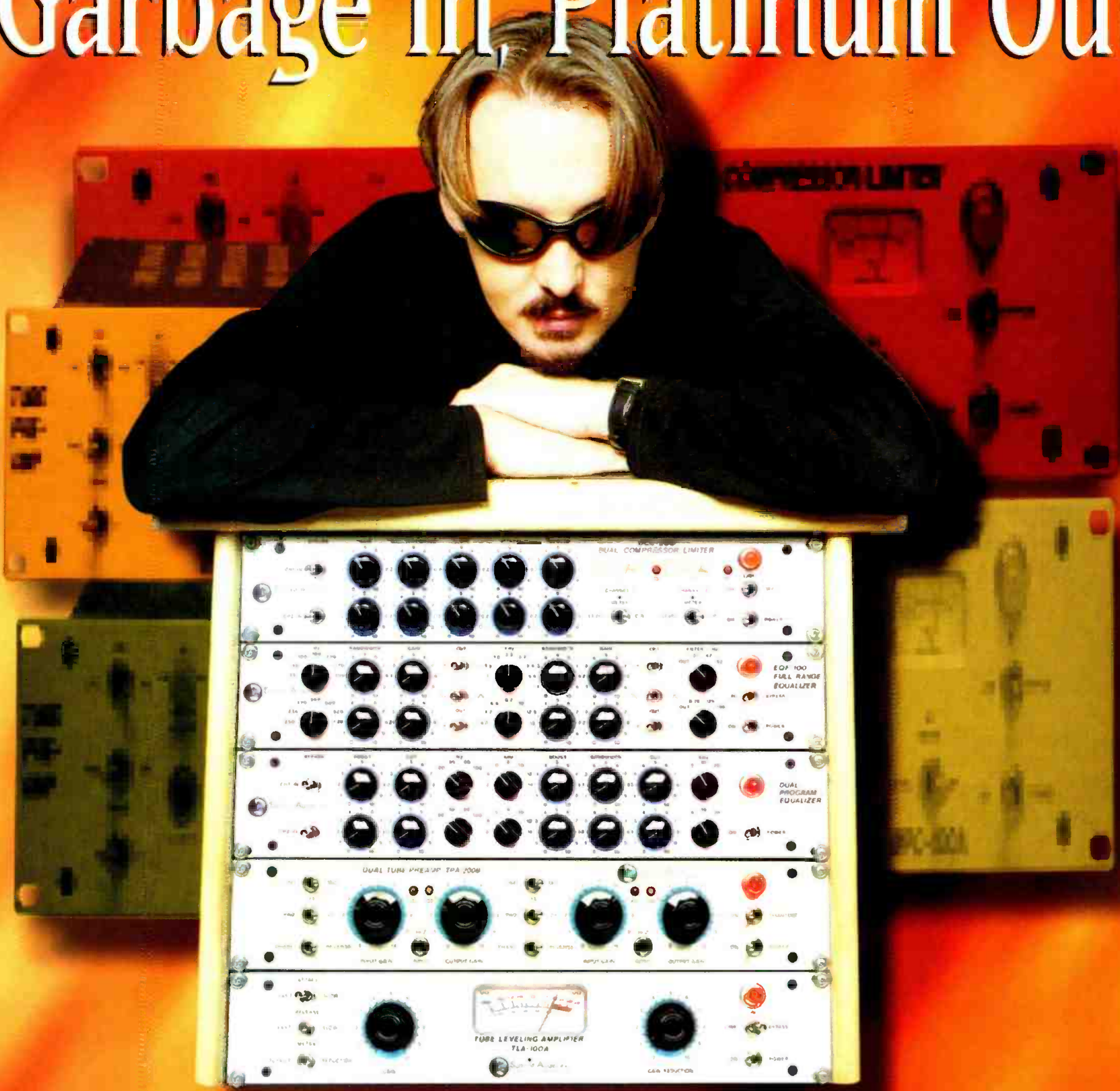
Europe: Tascam.
Tel: +44 1923 819630.
Fax: +44 1923 236290.
US: Tannoy/TGI.
Tel: +1 519 745 1158.
Fax: +1 519 745 2364.

The Step response (Fig.10) shows excellent time-alignment and crossover design with a sharp rise and steady decay. The power cepstrum (Fig.11) shows a spike at about 160µs, but is otherwise okay. The acoustic centre (Fig.12) shows a maximum group delay at low frequencies that corresponds to a shift of about 1.2m; this result confirms the good transient response shown in the step response plot. The total power response is fairly smooth, with a dip at crossover frequency of 3kHz due to driver interference (see vertical directivity).

Overall, the loudspeaker works well. The design has no obvious flaws other than the off-axis radiation at crossover; a necessary result of spaced drivers. ■

Garbage In, Platinum Out



Summit Audio Success Stories

Butch Vig, engineer, producer, co-owner of Smart Studios and the drummer for Garbage, relies on Summit gear for all his work. Vig engineered the group's latest platinum album, "Garbage," nominated for three Grammys this year, as well as producing albums for Smashing Pumpkins, Nirvana, Soul Asylum and Sonic Youth.

"Whether I'm working at Smart Studios or I'm on the road touring, I always use Summit tube gear. I particularly like using the DCL-200 Compressor Limiter for tracking vocals. It colors the sound very subtly, while retaining its warmth and transparency. Often I will compress a vocal performance quite a bit. This allows me to place it exactly in the mix while maintaining a lot of presence and natural dynamics without sounding too loud. This works especially well when the mix is very dense."

"Summit just keeps coming out with great gear. We can't wait to get our hands on the new MPC-100A Mic Pre-Amp/Comp-Limiter. It is a high quality and great sounding input device that will further enhance our music."

Hear the Warmth™

Summit Audio

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www.summitaudio.com E-mail: sound@summitaudio.com

AMS Neve 1081C equaliser

Few sound chains command as much respect as the vintage 1081. Frequently copied, **Dave Foister** toys with a replica from the originators

THE REVERENCE with which vintage Neve consoles are regarded puts them in the same bracket as classic microphones. Large numbers of 20–30-year-old Neves are still going strong, many of them in Los Angeles, where a big Neve and a maintenance budget to match is seen as better value than many new consoles. The sound's the thing, and a major defining factor in that sound is the input-EQ module, so that refurbished Neve console equalisers have appeared in various third-party rackmount guises. Faced with a situation where people will pay silly money for an old equaliser that then has to be virtually rebuilt before it's useable, AMS Neve took the step taken a few years back by AKG with the C12: producing a replica, built to the same specification but new, warranted and similarly priced to the collectable originals.

The dedication with which this has been achieved is remarkable. It was decided that the new 1081C (for Classic) would be as exact a copy of the original as possible; there would be no thought of 'improving' the electronic design with modern components or production methods as there was every chance that the sought-after sound would be changed in the process. This contrasts strongly with the line taken by AKG, among others, where it was felt that modern techniques could achieve identical results with better consistency and reliability, given sufficient understanding of what made the original so desirable. This strikes me as perfectly valid where a transducer is concerned, given improvements in manufacturing tolerances, but with something already as consistent as a Neve circuit the AMS Neve purism is equally appropriate. The lengths to which this was taken, and the painstaking research involved in realising the project, show that Neve perfectionism is alive and well. In this case it came in the person of Neve veteran Robin Porter, for whom the 1081 has become something of a passion.

The first job was to find out how many of the original components were still available. Basic electronic parts were not a problem, even though original types of capacitor were to be used where others might have ditched them. Similarly there was no thought of updating the potentiometers to newer types, but getting hold of the right carbon-track pots was less easy. The original suppliers had stopped making the specific type used on the 1081, but further enquiries revealed that they had sold the tooling for them on to another company, who were therefore able to make new ones up. The use of the original types means the pots will have familiar idiosyncrasies, but authenticity, in this case in the feel of the controls as they turn as well as the electrical characteristics, had to be the overriding priority. The knobs for turning them were more of a problem, as they were not available and the original supplier no longer had the tools for them. The look and feel of an old Neve equaliser depends heavily on the distinctive knobs, both the dual-concentric arrangements for frequency switching and boost-cut, and the skirted ones with pointers for the input gain, so it was vital that they be the same. Most of the tools were eventually tracked down elsewhere, but one knob-skirt had to be remoulded from the originals.

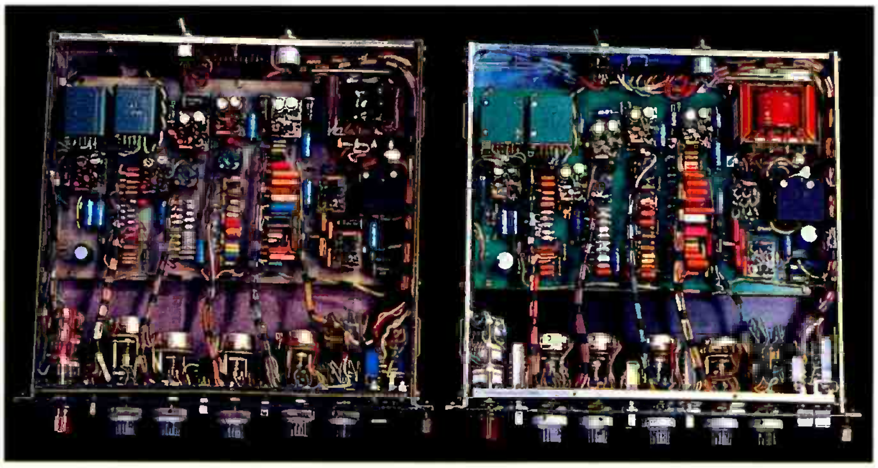
The biggest challenge was the transformers. The input transformers were originally custom wound by two different suppliers, and, although one of them had gone out of business, the other was still around and able to produce more. The output transformer suppliers however had disappeared completely, so the specs had to be found and the input transformer supplier (who had, in fact, made samples for Neve in 1973) charged with the task of matching the originals. Even this apparently simple task took four attempts before a match close enough to satisfy Porter was achieved.

The PCB layout had to be retained exactly as it was, complete with the piggyback boards containing discrete gain blocks—although the connection pins were made thicker to eliminate the occasional problem of the things falling out. Again, modern techniques could have been used, but if it is true that layout optimisation improves the sound of a device then changing the 1081's layout would inevitably alter its character. Likewise the wiring looms between boards, controls and connectors had to be wired and routed in exactly the same way as on the originals. Some users had rerouted the wiring and found the sound changed, which considering capacitance in the wiring and the inductors on the board near the looms is perhaps not surprising.

The 1081C is, mechanically, a direct replacement apart from a slight improvement to the side panels to stop them bending out, so it will fit into an original console frame where, perhaps, the modules supplied with it have been cannibalised or sold off as outboards. For this reason the 18-way connector on the back had to be identical, even though it was hard to find and disproportionately expensive.

The front is instantly recognisable. Everything from the colour to the print is exactly as before, and when you start turning the familiar knobs it's hard to believe this is a new piece of equipment as these controls simply are not fitted to new kit any more. The design is old enough that there will be many reading this who have not encountered one before, so a quick rundown of the controls is in order. First, as the module was originally the input section of the channel strip, there's a dual-function INPUT GAIN switch stepped in 5dB increments. It has two distinct sectors in its 345º travel, one part selecting the line input with variable gain and the other the microphone input with a huge amount of gain avail-

able by today's standards. The EQ itself has four bands, with a range of adjustment that today looks a bit basic. High and low frequency sections can be shelving or peaking, and the turnover frequency is switchable with the knurled aluminium outer ring of the control pair. Two overlapping mid bands again have switched frequency settings, and >>>>>



<<<<< bandwidth is controlled only by a HI-Q button. The controls are completed by a final concentric pair for HF and LF filters, again with switched turnover frequencies.

There is a set of four square push-buttons (two illuminating) beyond the filters, two of these perform the obvious functions of switching the EQ in and out and inverting the phase. It is worth noting that the filters can only be used with the whole equaliser switched in, but that each band can be switched in and out independently. Besides these there is a SOLO switch intended for use when several 1081s are racked together: the signal and closing contact appear on the rear panel connector. The final switch is unmarked and available for custom functions.

Part of the appeal of the new 1081 is the provision for assembling a collection of them in a rack. For this reason two versions are available, with printing horizontal or vertical to allow for mounting orientation. AMS Neve offers two racks, one accommodating eight 1081Cs mounted vertically and the other, as supplied to me, carrying a horizontal pair. The dimensions don't allow them to fit end to end, so the rack is 3U-high and includes a substantial power supply, which also delivers 48V phantom via its own front-panel switches. These sit alongside output level controls which can back off up to 20dB. A hinged back panel carries I-O XLRs and a 25-pin D-connector for the Solo functions, and so on, and opens for access to the backplane and connections plus a mic input impedance switch and a gain trim pot.

It's some time since I used real vintage Neve EQ, but I'm sure those who are intimately familiar with it will find it just as hard to believe that the 1081C is anything other than an old EQ refurbished and given a good clean. Its mic amp is quiet, impressive at both ends and crystal clear, and the EQ just oozes character. It's always salutary to be confronted with the apparent limitations of a classic old EQ design only to realise that it's easier to get what you want out of than most modern fully-variable circuits. The innate sound, the musical detail, and how kind it is when the boost is cranked right up, are more important than homing in on exactly 357Hz with a Q of precisely 2.3. Devotees of Neve EQ will know that and will be hoping to hear those qualities in the 1081C. They won't be disappointed. This is the real thing, put together with a care bordering on reverence—the reverence accorded the originals by the cognoscenti. At the same time it's comparably priced, it'll work right out of the box and it'll carry on delivering that sound without maintenance being permanently on standby. All the good bits without the bad—it can't fail. And there's apparently a possibility of using them in a modern vintage console. ■

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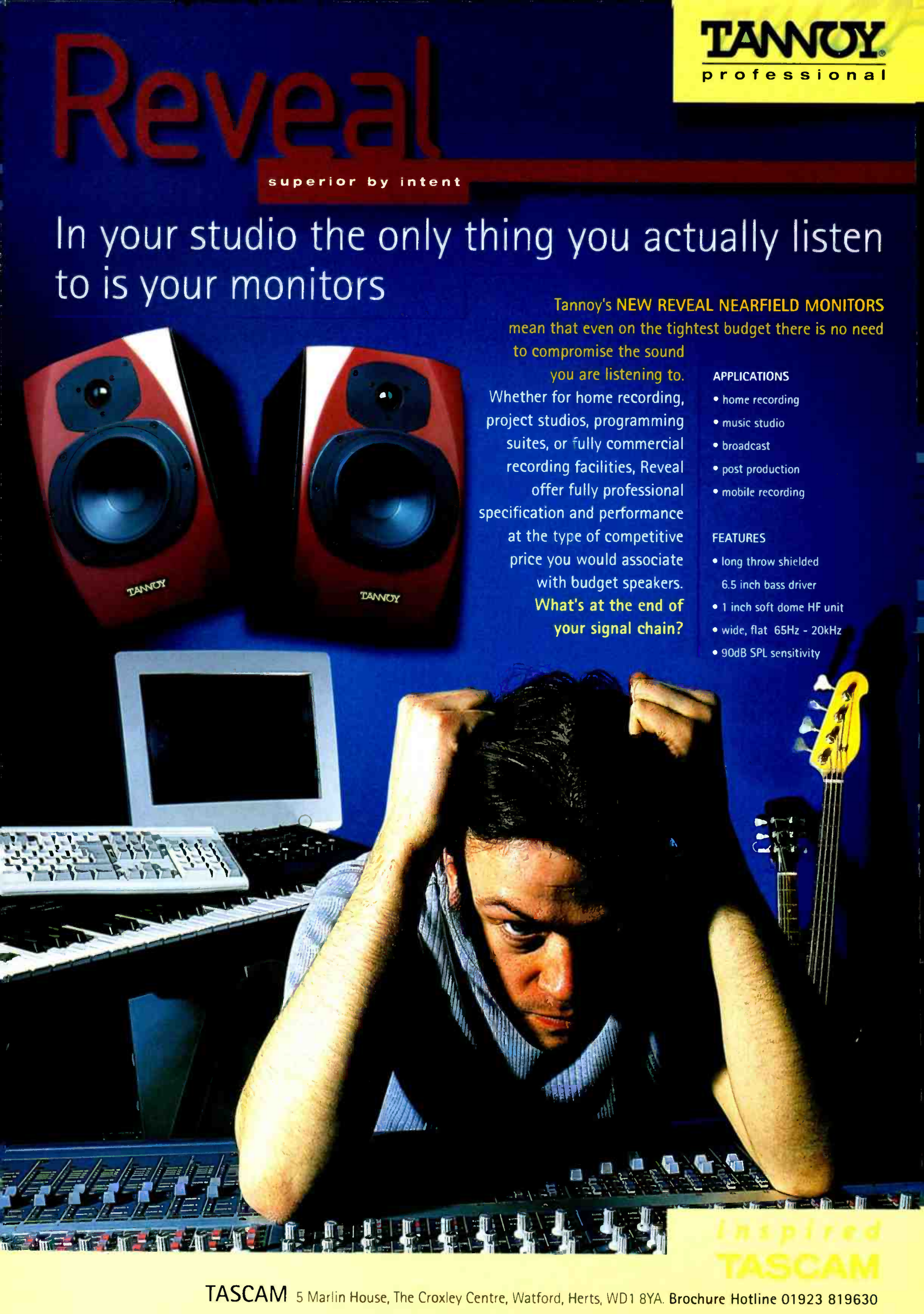
LX20

M20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Denon DMP-R70

Having achieved pre-eminence in the studio MiniDisc market, Denon has spotted the potential for the location recordist. **Neil Hillman** pockets Denon's first portable MD machine

YOU DIDN'T HEAR IT from me you understand, but MiniDisc is alive and well, and the subject of Chinese whispers—'MiniDisc is okay, pass it on... MiniDisc is okay, pass it on...' As we dare to talk of the format that may not speak its name, it is gaining an acceptance among a usually discriminating audience. How can this be?

Amid the high ideals of 96kHz sampling rates and hard-disk location recorders stand a few considerations rooted firmly in practicality: namely media cost, power consumption and weight. Apart from good-old ¼-inch analogue tape (which we may nowadays eliminate on the grounds of noise performance), every recording device available to a location recordist is seemingly compromised by one or other of these pertinent conditions. Put simply, the 44.1kHz sampling MiniDisc has taken on the role as the audio version of the DVC camera; and rather like the conflict between the DVC and Digital Betacam protagonists, there are strong voices expressing both sides of the argument. We, as engineers striving to achieve the best results possible, know that we should reject it on a total-quality basis, but those results ultimately must be judged by ears and not by chart plotters. It is precisely this truth that makes audio engineering the hybrid black art that it is. What alchemy is at work when a particular microphone sounds so good, so right, on an application that logic would never have used it for in the first place? The analogy to DVC also continues with the fact that the format offers the possibility of capital cost-savings; although the rumours that the bean-counters in British public-service broadcasting are renaming the nations' favourite channels DVC 1 and DVC 2 are, at present anyway, unfounded.

With the MiniDisc format readily available to the domestic consumer—and it is being taken up deceptively quickly—and its availability in many outlets worldwide, there can be some reassurance of back-up while on location; at worst, total-replacement cost is not going to ring the bell at Lloyd's of London when the director's 4-wheel drive goes off-road over it.

So into the portable, recordable arena comes the Denon DMP-R70 recorder, standing shoulder to shoulder with the MZR-30 from Sony, the inventors of the MiniDisc format, and already established as a high-selling unit. Weighing just 275g (I knew a girl who smoked heavier cigarettes) and with roughly the width, height and depth of a cigarette packet, this is a beautifully styled and tactile device of burnished aluminium that owes much of its external design to a Ronson cigarette case of the 1950s and 1960s; and yet as you move it around in the palm of one hand its clear, uncluttered lines dare you to imagine that this consumer product will double as a professional workhorse recorder.

The top face of the Denon carries the slim LCD window that indicates track number, duration, and symbols associated with being in record; a spinning disc showing REC, TOC relating to amending or reading the table of contents, MONO indicating that Long-Play mode is enabled and almost doubling the recording time of 74 minutes possible for stereo recordings. The battery level >>>>>

<<<<< indicator is a bar-graph of four segments inside a battery outline that reduces as the voltage available from the battery falls. The recording level is shown as a single bar-graph representing both left and right channels, with calibration points of -40dB, -12dB, -4dB, 0dB and Over. A double-dot around the -12dB point would indicate that this is perceived as a line-up level; although I set the line input to one segment below this point for a 0dB input from my mixer, and this seemed to correspond to where I would normally expect to set the input to at around -18dB.

THE CONTROL buttons run down the right-hand side of the top face and start at the top with the recessed REC button, made more obvious by being coloured red, but this is a small button, and gloved hands would find it awkward to operate with any degree of accuracy or precision. Below this are a pair of push-buttons for character selection of the alphanumeric font for labelling tracks while in Edit mode or showing date, time and duration of recording while in Record mode—and DISPLAY, which, while in Stand-by, toggles the read-out between the time used on a disc and the volume name of the disc or, when in Record mode, switches the elapsed time display of the track being recorded on or off.

A single double-function button below that pair acts as an ENTER key for edit operations or as a synchronous record prompt for line inputs taken from external prerecorded sources to be recorded as digital or analogue clones. A pair of push-buttons acts as up-down arrow keys for headphone level and create the ability to scroll through the alphanumeric font to name tracks in Edit mode.

The final four dual-function switches are mounted as two pairs, one above the other, and represent conventional transport keys of Rewind-Skip Back, Fast Forward-Skip Forward, Stop and Play-Pause. Their other functions being Record Level up and down in Record mode or cursor movement left or right in the display window while in Edit mode; the bottom two keys of STOP and PLAY doubling as the OFF and ON switches.

The front face below the display window houses the shuttered entrance for the disc itself, this being a rather better arrangement I feel than the lid of the Sony MZR-30. Also on this face is the microphone input, a 3.5mm socket that has a sensitivity selectable for 0.25mV (high) or 2.5mV (low) and an input impedance of 10kΩ. A HOLD key disables the operating buttons preventing the recorder being knocked out of record or playing to itself until the battery runs out and a large spring-loaded EJECT sliding key is sited alongside the disc slot to the right.

The left-hand side carries in line the 5V DC external input power-charge socket, the 3.5mm headphone socket delivering 10mW per channel at 16Ω and the 3.5mm optical-line in socket rated at 100mV with an input impedance of 20kΩ and an output of 350mV at -12dB with an output impedance of 50kΩ.

The right-hand side of the recorder carries three slim switches barely prominent above a recess and mounted horizontally in line, providing bass boost for Playback; Mode to provide random play, continuous play, single track repeat play and Mono mode. The third switch, EDIT-AUTO MARK, enables access to allow writing to the disc and naming it in Stop or Pause modes, combining tracks together, erasing a track, moving a track, naming a track, erasing all tracks and name stamp data-reading from another disc.

The machine requires 150mA to power it and the 3.6V lithium-ion battery lasts for seven hours in record or over nine hours when solely used for playback, and the battery is rated over 300 charge cycles, taking three hours to charge from empty to full. An external lock-on battery pack holds AA-cells with a seamless take-over by the internal battery when they fail.

In use, it's hard to fault the DMP-R70. It does what it should, with the minimum of fuss and its ability to withstand shocks and knocks is impressive.

It's not often that we can steal a march on equipment manufacturers by using a cheaper product for a more demanding job, or to predict and anticipate the way a market is going; but quietly and steadily MiniDisc is gaining acceptance among the location recording fraternity—by both the originating recordists in the field and the practitioners of the dubbing suite; and like all the best subversive movements this change is being applied slowly, with care and caution.

The Denon recorder clearly has the build quality, ruggedness and pedigree to sit in the front pocket of a location recordist's mixer-bag and while there are improvements that would quickly change this device into a 'professional' product—initially little more than upgrading the input and output connectors and adding a big RECORD button would be a welcome 'semi-pro' start—it would be careless to forget that we are judging a consumer product, here, against criteria outside of its design brief and that, in engineering terms, there is a huge gulf between the performance of the MiniDisc system and its superior rival DAT; but its meagre thirst for power, its lack of weight and its low cost, robust and easily available recording media make it a compelling and viable option.

In the absence of a purpose-designed, time-coded location recorder, and given the alternatives of either the Denon DMP-R70 (at around £299, UK) or the Sony MZR-30 (£199 UK) it comes down to a matter of paying your money and taking your choice; because let's face it, at this level we are talking pocket, not rocket, science. ■

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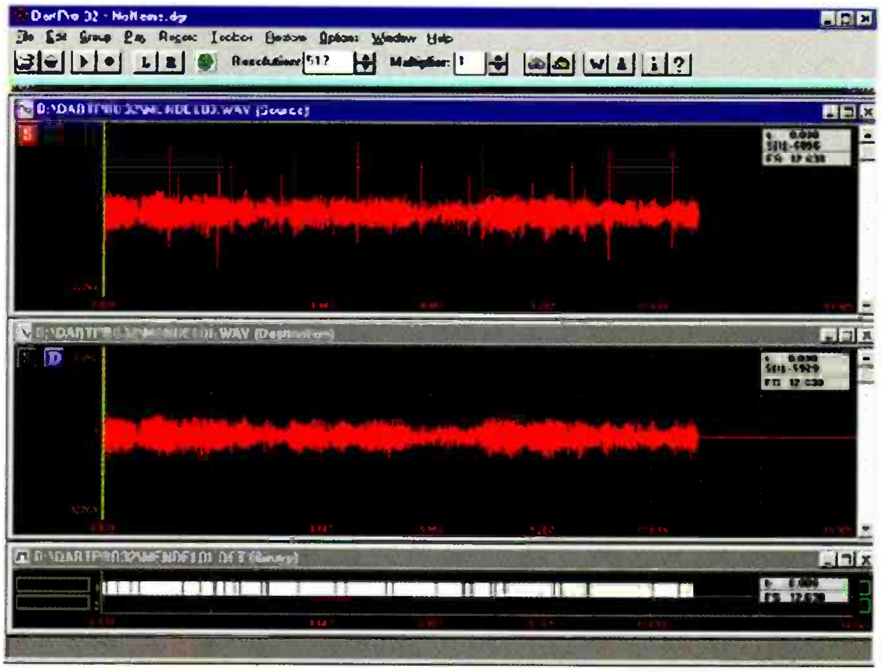
Dartech DART Pro32

Sound restoration is becoming less a specialised area and more one that appeals to the masses. **Rob James** investigates a new audio restoration package for the PC



DARTECH'S DART Pro32 (Digital Audio Restoration Technology) is a digitise, clean-up and CD-R burning software package for the PC. It would appear to be primarily aimed at people with a vinyl collection since both the pack and Dartech's website are emblazoned with the invitation 'Restore your record collection on CDs!' There are obvious copyright implications to this but I doubt the record companies are going to lose too much sleep over the possible drop in 'replacement' CD sales. The manual carries a stern warning about copyright, neatly passing the problem to the user.

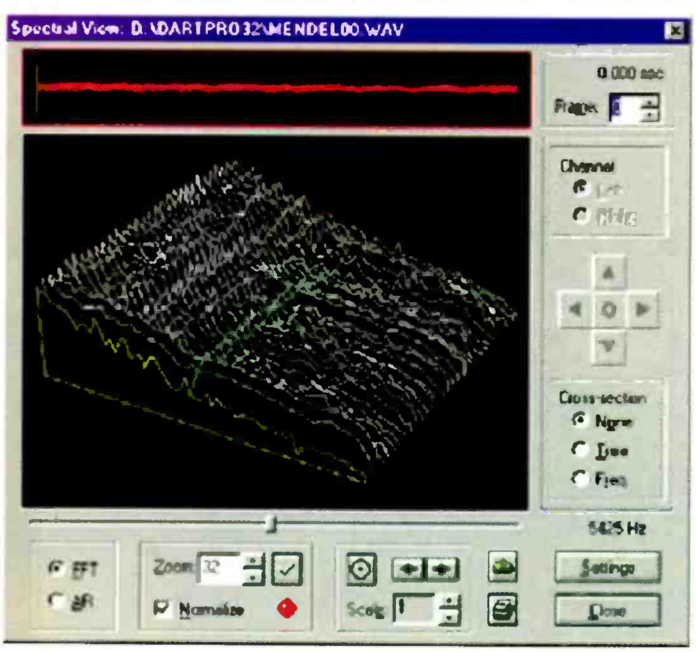
The software runs under Windows 95 or NT. Dartech reckons it will run on a 486 with any 16-bit soundcard. I tried a Creative Labs SoundBlaster and a TripleDAT card and it works well with both. As usual with any PC audio package, the higher the spec of the machine, the faster and more robustly it will run.

The manual is well written and informative. Also included is a comprehensive 'guided tour' tutorial that runs for some 40 minutes.

In use, once a track has been recorded or an existing .WAV file opened it appears in a Soundfile window with waveform display and you are invited to register it as a root soundfile. DART's register will then keep track of all files associated with the restoration of a root file. Both amplitude and time are scalable in each window. Windows can be synchronised to enable easy visual comparison of before and after soundfiles and to enable inspection of samples marked for processing in a Binary Detection window. The Binary window displays the contents of a compressed binary detection file as markers. The file is created by the Outlier detection process used to define where impulse disturbances are present.

Before any restoration can be undertaken, a file should be created and registered to contain the treated result. Two buttons select whether a file is to be considered a Source or Destination and the software warns if you are about to do something you might regret.

Simple restoration is best attempted by dealing with impulse noise first and then attacking the hiss or background noise. DART Pro32 will allow you to tackle both processes in one pass but the manual correctly warns this is undesirable. As a first attempt you should accept the default values of the various parameters for the processes and audition the results. For each process a Test facility is provided that allows a user-defined section of the Soundfile to be processed and auditioned before committing to processing the entire file.



For more advanced use DeClick provides adjustments for a variety of parameters: Smoothing Factor controls the action of an adaptive Kalman filter. Postfiltering Factor controls an adaptive wide-band noise-cancelling algorithm. Detection Threshold sets the sensitivity of the Outlier detector and a Maximum length of Detection Alarms determines the maximum number of samples which can be scheduled for reconstruction in automatic mode. There is also a switch to select Music or Music and Speech modes that alters the response of the Outlier detector to accommodate the impulsive nature of speech.

The Retouch option is used to edit, define or redefine areas to be processed by DeClick, not manual waveform redrawing as the name might imply. Using this option very large blocks of samples (up to 1500 per block) can be scheduled for reconstruction. DART Pro32 replaces the defective samples with copies of the preceeding (Left) or succeeding (Right) ones. If a contiguous succession of defective blocks are manually marked 'Left Hand' reconstruction may be applied, however there are obviously limits to this 'snake eating it's own tail' approach and for very large problems it is better to resort to manually editing in a substitute block from elsewhere in the file if a suitably similar one exists.

The DeNoise and DeHiss tools are based on the noise-print model. DeNoise requires a sample of noise, free of wanted signal, to be taken from the subject soundfile whereas DeHiss is based on a standardised noise model. DeHiss should be tried first since it is more likely to produce a good result where the noise varies over time. Noiseprints can be saved in a Rogues Gallery and applied to other soundfiles. Controls are Gain, which sets the level of noise the software attempts to remove in DeHiss, Weight which does approximately the same thing in DeNoise, Smoothing Range is approximately the same thing as Smoothing Factor in DeClick. Frequency Carving is low-pass filtering. >>>>>

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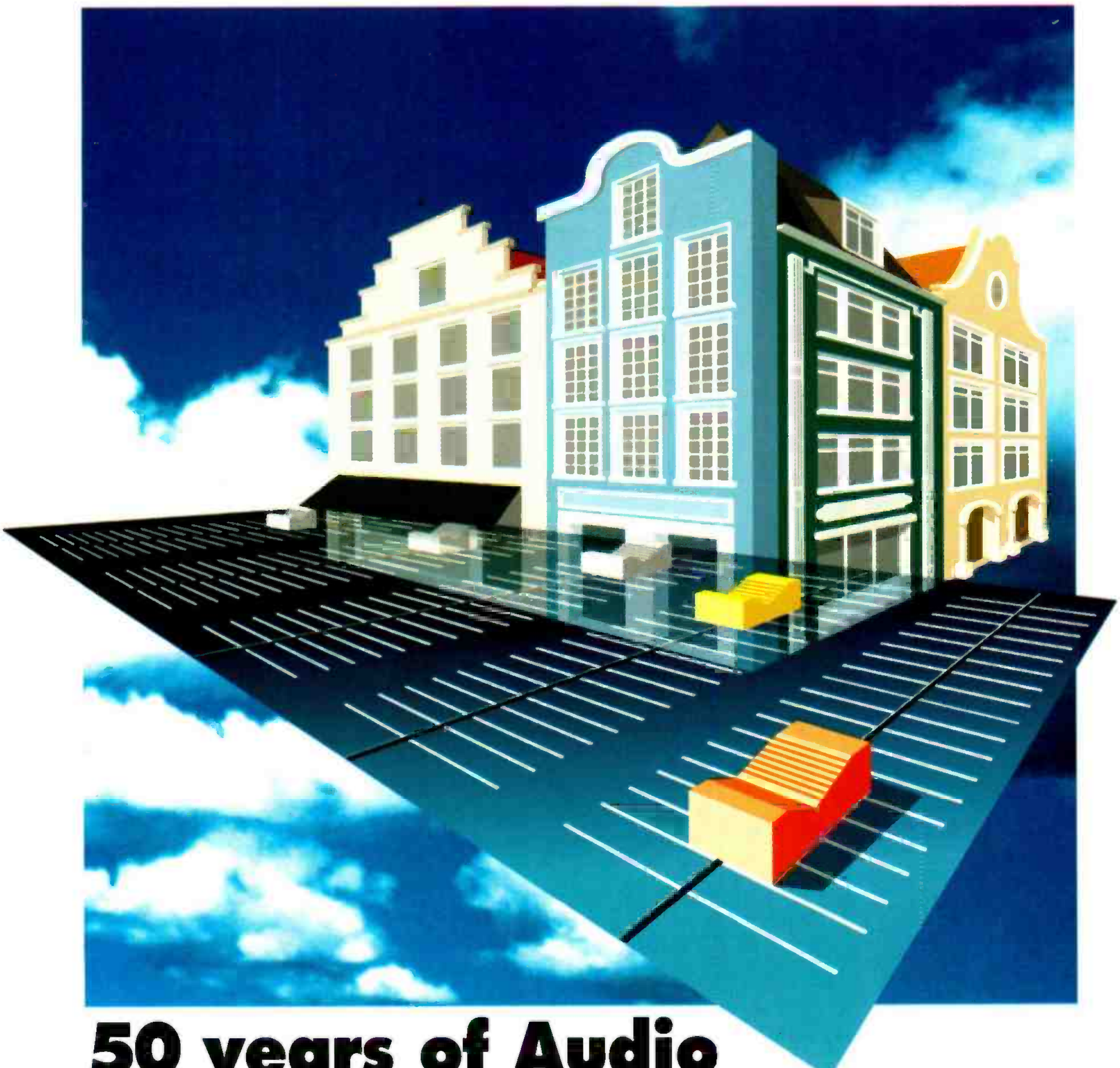

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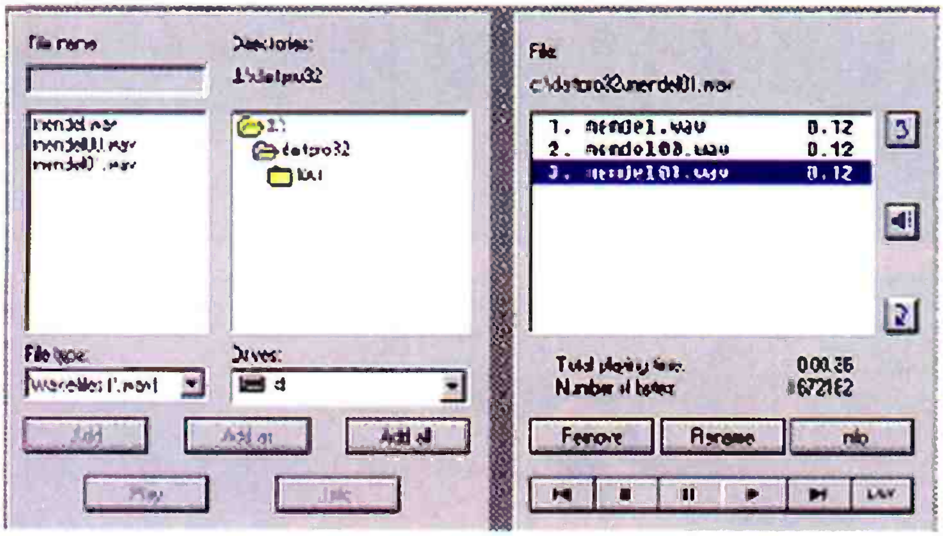

<<<<< Frame size defines the size of the analysis frames used in analysing the sound and Overlay defines how much these frames overlap. Decreasing the Overlay speeds the process but may result in artefacts.

The object when setting all these parameters is to achieve a compromise between optimal 'improvement' of the material with the minimum degradation. Some experimentation and very careful listening is required to decide what is an acceptable result.

For advanced use there is a toolbox of extra facilities, filters and so on, which enable the design of multistage processing and processing in reverse which can be very effective with certain types of impulse noise. Other tools include spectral analysis, resampling, speed changing and a graphic equaliser to compensate for the effects of noise reduction.

Filters and equalisers are Butterworth Recursive Infinite Impulse Response (IIR) type which introduce nonlinear phase distortions. An option is provided enabling filtering to be carried out twice, once forward and once backward which preserves the phase; although processing takes longer.

DART's wave manager allows playlists to be constructed and manipulated in order to produce DAO (Disk At Once) CD-Rs with a suitable recorder.

The software is loaded with nice little touches which aid the processes and, with practice, DART Pro32 provides a good introduction to serious audio restoration and provides a consistent interface for the whole process from loading material to burning CD-Rs. My only real moan is as soon as you hit any on-screen button or pull-down menu, preview audio stops playing. If you are used to 'mousing around' deciding what to do next this is irritating.

As with all 'restoration' processes it is up to the operator to make compromises—there is no such thing as a free lunch. Don't get me wrong, I am an enthusiastic user of this type of software, but it is all too easy to overdo it or even make things worse if the tools are not used with some sensitivity.

I experimented a number of differing types of material and within the time available I was able to produce results which 'improved' on the original without introducing unwanted artefacts or losing the intrinsic character. This product will appeal to the computer literate collector but for professionals with limited restoration requirements who would rather keep the process 'in house' DART Pro32 may well be all they ever need. ■

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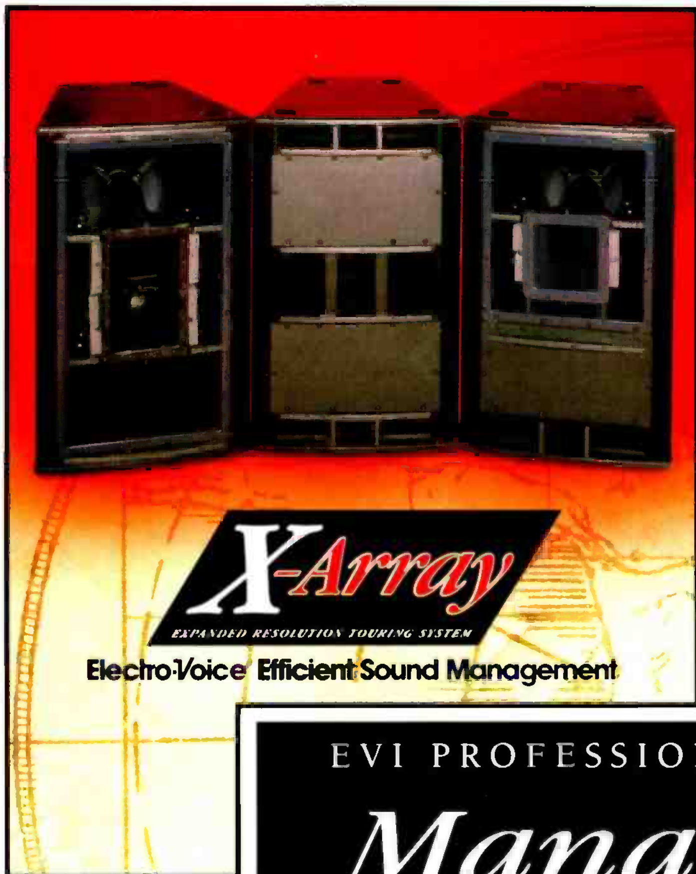
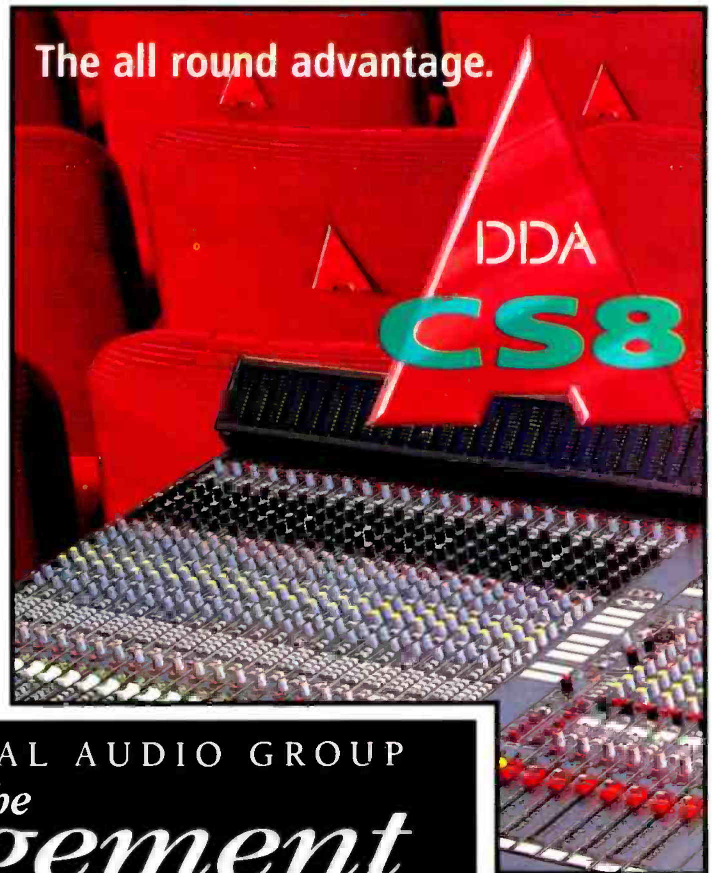
Aphex FM Pro Model 2020

Giving a radio station an audio signature involves more than signing a presenter's large cheque. **Rob Budding** investigates a new broadcast signal processor

WHAT MAKES ONE RADIO station sound different from another? Is it the presenters, the programming or the chips in the equipment room? Ignoring the first and assuming the same playlist, the last could make more of a difference than you might think—and we are not talking about the remains of the engineer's burger here. Audio processing is what it is all about and processors get bigger each year. What started life as a basic compressor-limiter has evolved into a complex affair with multiband limiters, intelligent gating, soft clipping and RS232 interfaces, to name but a few developments.

Why the need for all this added complexity? Well, when a station has been operating with little more than a compressor-limiter in its output chain, the results of fitting even a basic processor with limited controls can be astounding—sometimes prompting listeners to phone in and ask how it can sound so different. That is a rewarding moment, but what happens when the competition discovers the same box as you? Do you put on a £1m jackpot competition, hire a dizzy blond to giggle and fluff her lines on the breakfast show or buy a box with more knobs. In other words, in a fiercely competitive market, you have to do something to set yourself apart from the others—and that just might be achieved if you can sound different. Note that we are talking 'different' as opposed to 'better' since the latter is highly subjective and the probable >>>>>





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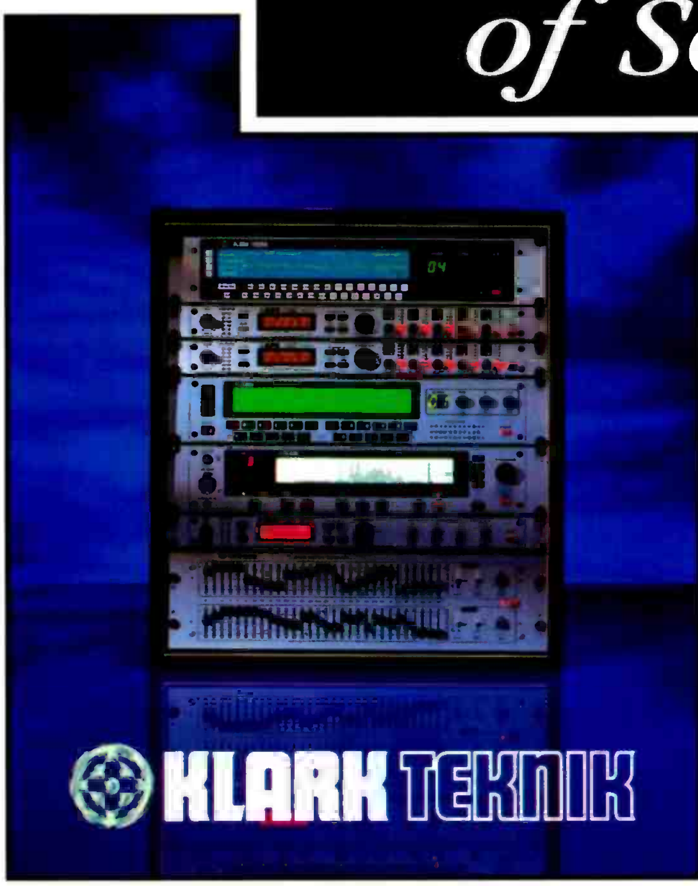

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<<<<< cause of more fights in the Knob Twiddlers Arms than any other topic. Logically, in order to create a 'different' sound, you need to find a means of modifying the audio signal significantly—and a recently released audio processor is claiming to do just that.

For 25 years, Aphex has been making equipment for the audio industry. Initially an audiophile approach was adopted with the emphasis on a transparent and natural sound, but the reality quickly dawned that if a station's output was to fight its corner, then it needs to get tough. And in 1997 Aphex launched the FM Pro Model 2020.

The FM Pro is promoted on its ability to act 'as a sound designer's palette', and there is certainly a huge number of available parameters to alter. In a bid to help you create a unique on-air sound Aphex has come up with no less than five new patented audio processing innovations including the Sticky Window, which is something to do with aural excitation, no doubt.

The FM Pro is well presented in a 2U-high case with a smart brushed aluminium front panel bristling with indicators and meters. Starting on the left-hand side there are eight 10-segment LED bar-graph meters for the following functions: Input Level (LR); Levelling Gain; the 4-band Compressor; and the last for the Limiter. A total of 20 LEDs indicate the status of various selectable modes. Then there is the spin wheel, LCD window and, to its right, six buttons that navigate through the various menus. Finally, there is a monitor jack that is selectable between input and output allowing a quick comparison of input to processed sound.

The input AGC circuit has the aforementioned Sticky Levelling which is really an operating window that the signal must exceed before an AGC action can happen. The main benefit of this is to prevent the AGC tracking small signal-level changes thus reducing distortion and improving signal dynamics

Both analogue and digital audio connectors are provided on XLRs for input and output, a BNC for multiplex (stereo) and a 9-way DIN for the RS232. There are some good fail-safe features built in such as power-off internal bypass relays for digital and analogue audio I-O circuits and the analogue input will be automatically selected if the digital signal becomes corrupted. There is also an internal digital stereo coder-generator available as an option. All the front-panel controls can be password protected to prevent unauthorised adjustments.

The audio signals follows a fairly typical processor path of input signal conditioning (filters), automatic gain control, multiband compressor and output limiter. However, Aphex has added some nice touchs to nearly all of these stages. The input AGC circuit has the aforementioned Sticky Levelling which is really an operating window that the signal must exceed before an AGC action can happen. The main benefit of this is to prevent the AGC tracking small signal-level changes thus reducing distortion and improving signal dynamics. The AGC is also frequency discriminating, from 200Hz the attack time gradually slows down as the frequency drops which prevents bass notes from being 'pulled back' by the AGC.

The multiband compressor boasts lots of adjustments—the band filters cover a wide range, the release time is adjustable for each band and the output mixing is adjustable with up to 6dB of boost and infinite cut for each band. Aphex has added interband coupling controls which link together elements of the compressor function on adjacent bands, and is said to reduce the long-term equalisation effects of the multiband compressor—sounds like a health warning. The limiter next: this has a couple of bass equalisers with the aim of expanding the bass rather than restricting it as often happens with the use of heavy compression and limiting when trying to increase loudness.

Another feature is the comprehensive day parting scheduling. Four events per 24-hour period can be assigned, each event comprising a designated preset and takeover time defined by the hour and minute.

So that is a rundown on what goes on inside, but what does it actually sound like?

Firstly, you have to get the thing working—which is very straightforward. Just switch the unit on and it kicks into life recalling the last saved setup. There are eight factory presets which is probably the best place to start. You can choose from Classical to Oldies to Urban or to something called CHR. All sound quite acceptable yet they also sound quite similar. Some other makes of processor that come with factory presets usually put in some extreme settings to give you a real taste of what is available. I suspect Aphex has avoided this route as the current fashion in the States seems to be veering away from heavy compression and back to something where detail can be heard again. I spent about a couple of hours coming up with a sound I liked then compared it to the unprocessed input and found I preferred that... In some instances the high frequencies appeared to be distorted, but this was discovered to be caused by the input level being too high and the Leveller incorrectly set. Several more hours, a dozen CDs later and I felt I was getting somewhere. This is clearly a box that needs gentle adjustment and time to acclimatise to the new sound before going further. Thankfully there are plenty of user presets so you can always get back to something you were happy with—assuming you saved it first.

Aphex also provides remote-control software for a PC that uses Windows to provide full reproduction of the bar-graph meters and status indicators plus graphical display of the adjustment controls such as Leveller Gain and Crossover Frequency. The Windows interface is probably easier to navigate around than the front panel as several menu screens are assembled onto a single page on the PC monitor. The remote connection can be established directly or via a modem and password protection applies as per front panel controls.

All in all, the FM Pro is an attractive piece of equipment. Aphex has done more than produce an updated version of an earlier model—it has made a unit that incorporates a lot of innovative new ideas and wrapped them in a package that is logical and intuitive to use. The 2020 will provide hours of fun for anyone interested in creating a different sound for a radio station. In fact, the applications for this and other audio processors do not end here—TV stations, cable networks, recording studios, live performances and wherever there is a need for limiting, compressing or processing of audio signals present potential sales. Aphex' 2020 is not the cheapest device on the market, and some smaller users might find it hard to justify the additional expense over other units. Where it could come into its own is where there is a highly competitive market, for instance, in the London FM music radio arena, where the likes of Radio 1, Virgin, Capital and others are vying for the same listeners and need to find a way of making their music more attractive to the audience than simply sounding the loudest. ■

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- 44.1 or 48kHz (selectable)
- Copy, paste, move & erase editing with undo & redo
- ADAT™ Digital Interface (simultaneous 8 channel)
- ±6% pitch control with no loss in audio quality
- MMC & Fostex Exclusive Message for controlling transport from external MIDI device, e.g. sequencer
- Approx. 30 mins recording to 1.3GB drive @ 44.1KHz
- 9 'Virtual reels' ■ Versatile chase mode
- Tempo mapping - create up to 64 tempo & signature changes per song
- Midi clock with song position pointer
- Analogue & Digital I/O (S/P-DIF & ADAT interface)
- SCSI-2 interface option for fast backup of sessions

D-160 features..

- 16 track digital multitrack with no compression
- 8 further 'ghost tracks' for additional takes
- ADAT™ Digital Interface (simultaneous 16 channel)
- ±6% pitch control with no loss in audio quality
- SCSI-2 interface as standard for fast back-up
- 44.1kHz & 48kHz sample frequencies
- Up to 99 'Virtual reels'
- Tempo mapping - create up to 64 tempo & signature changes per song
- Midi clock with song position pointer
- MMC & FEX implemented for external MIDI control
- Copy, paste, move & erase editing with undo & redo
- Analogue & Digital I/O (S/P-DIF & ADAT interface)
- Optional LTC Timecode board with Word & Video sync
- Balanced I/O option (+4dBu I/Os on D-sub 25-pin)

Digital Multitrack Recording - you have a difficult choice

Choosing to 'go digital' is fast becoming one of the easier equipment decisions you have to make.

But choosing the right digital multitrack can be a little more taxing as you have to be sure your chosen recorder excels in four critical areas: audio quality, expansion, synchronisation and editing.

Both the Fostex D-90 and D-160 offer industry standards in digital recording. Using both 18-bit & 20-bit convertors, they provide for CD-quality audio with a choice of 44.1kHz & 48kHz sample rates.

And being Fostex, the audio remains uncompressed meaning no compromises.

An ingenious caddy-held hard drive system means that increasing recording time is simply a matter of popping in a larger hard drive.

SCSI back-up of recording sessions is available too.

Sync facilities are as you would expect from Fostex. Both models are equipped with the ability to chase to incoming MTC; MTC plus S/P-DIF or ADAT™ (optical); or run free after MTC lock.

In addition, timecode sync facilities can be added to the D-160 via an optional board.

Finally, being non-linear machines, full copy, paste, move and erase (with undo & redo) editing is available across all tracks.

So maybe the choice isn't so difficult after all.

At least you know it'll be a Fostex.

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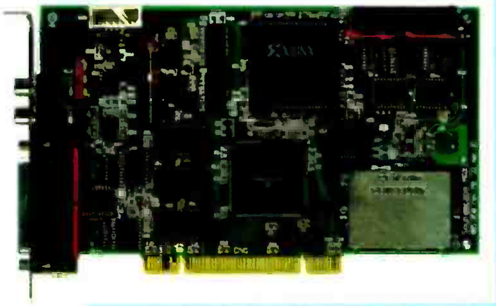
4-channel DI

The Radial JD4 is a 4-channel rackmounted direct box that uses Jensen audio transformers and Mogami cable, and was designed with input from the live and recording studio communities. It uses the circuitry from Cabletek's JD1 DI which is claimed to be flat to 80kHz, has a phase response that is said to be 'spot on' and is also said to be virtually impossible to overload. Entirely passive, input and through connectors have been paired on the front and back of the unit and supersonic filters on each channel reduce the noise from keyboards and computers. A PHASE REVERSE switch is included along with a STEREO SUM switch that takes a stereo output from a keyboard and sums it to mono plus a ground lift and 15dB pad. The Radial Convertible is a 50-channel audio snake that is set up in metric rows of ten. The last ten channels are paralleled with male output connectors allowing the snake to be used in a 40x10 or 24x8 configuration. The device is also equipped with a Ground Test Circuit that self-tests the 50 channels for abnormal ground hum.

Cabletek, Canada. Tel: +1 604 942 1001.

Soundscape PCI card

A PCI digital-audio interface card and two less expensive audio I-O interfaces, the SS8IO-2 and SS8IO-3, will be released by Soundscape by the middle of the year. The card will be able to run 16 tracks of digital audio simultaneously, in and out of the computer, using 2 standard TDIF ports and will be supplied with the company's v2 Mixer Software from the SSHDR-1 Plus DAW. This will allow the use of high quality third-party plug-ins developed for the SSHDR-1 Plus, such as TC Reverb, Wave Mechanics Reverb and the Soundscape Audio Toolbox. Soundscape does not intend to provide any editing-recording software for this product—as it is intended for use with PC software, such as Sound



Forge, Cakewalk, Emagic Logic Audio, Cubase Audio & VST, SAW, Samplitude, and Cool Edit. Additionally, there are packages for the video post industry that could also use the card, including Adobe Premiere, In-Sync Speed Razor, Ulead Media Studio, Softimage DS, Montage, and DVision. The PCI card has a projected retail price of US$700. As well as the twin TDIF ports, it has MIDI connections and optional SPDIF I-O and wordclock. The Mixer will support 32 channels of audio from the PC's >>>>>

Manley Voxbox

Not all manufacturers are targeting the low-cost, large-sales market. **George Shilling** evaluates Manley's 'Rolls-Royce' voice processor

AFTER THE HUGE number of voice channels launched in recent years, Manley arrives in this market gunning for the Rolls-Royce slot with its Voxbox. All signal paths use only valve amplification, and circuits have been carefully designed for optimal signal integrity, combining elements from previous designs but taking them further. The four main sections are mic preamp, compressor, EQ and de-esser. The extremely thick, polished metal, front panel includes the power switch, accompanied by a Ready LED that lights after about 20 seconds when the protection circuitry has done its stuff. There is a large, stylish VU meter that is accompanied by a 5-position switch to display line input, outputs and gain reductions. The remaining controls are grouped in their respective sections onto black etched panels which are bolted on to the main panel to form a huge VB—very clever.

On the rear panel, there are XLRs and TRS jacks for Line Input and Insert Input, an XLR mic input, XLR and unbalanced jack sockets for EQ Out and Preamp Out. The jack connectors bypass the input and output transformers for a subtly different sound, but still operate at +4dB. RCA phono sockets provide for compressor and de-Esser stereo linking. Circuit and chassis grounds are provided, and a fuseholder accompanies the IEC mains socket. The operating voltage is factory preset. No mains lead was provided but this review unit bears serial number 001, so, perhaps this is an oversight.

The input section features a locking toggle switch for phantom power, switchable 80Hz and 120Hz bass roll-off, and a PHASE switch; centre position selects line input. A front panel 100kΩ instrument jack overrides the rear line input. There is an INPUT pot, and a GAIN control has five positions (40dB to 50dB). Manley points out that this is not a pad, but controls the amount of negative feedback. At lower settings the sound is more clean and transistor-like. At higher settings a warmer, more 'valvey' characteristic is in evidence.

The compressor is highly unusual in that it occurs in the signal path before the mic preamplifier. It can be switched in without a click. The opto-isolator is able to work at extremely low signal levels, and can actually prevent mic signal clipping before the first tube. The compression approximates to a ratio of 3:1. The ATTACK and RELEASE controls each have five positions, which look simple enough, but there is some clever circuitry behind this. Four pairs of time constants provide a high degree of control. The manual suggests settings for different uses: on the fastest attack and release, the time characteristics of the original Manley Electro-Optical Limiter are achieved. With slower release settings, the unit takes on a degree of the auto characteristics of some units. Certain settings emulate LA-2A and LA-3A, and the overall character of the unit is smooth and gentle. The only other control is a THRESHOLD pot; there is no make-up gain control for reasons of sonic purity.

The passive EQ section includes three bands, each with 11 switchable frequencies. However, the High and Low sections are (bell-curve) boost only, and the Mid is cut only, each up to 10dB. There are no ICs, transistors or valves in the EQ section; although a tube circuit follows to make up gain lost in the EQ and De-Esser sections. This passive approach involves far fewer components than more common active designs and therefore in theory provides a cleaner signal path. The character of the EQ seems beautifully subtle and precise. You can make large adjustments without introducing distortion and phase changes that you hear with lesser designs. The lack of any shelving bands may be off-putting, but this design prevents you from ruining a good signal. The EQ can be fed from Line In, Preamp Output or Insert, meaning that you can use it for a separate signal while something else passes through the compressor.

The De-Esser provides four frequencies which control a sliding filter and a THRESHOLD pot that controls another opto-isolator. This tames sibilance much more gently than most VCA-based de-essers, and leaves the fine detail of the original sound intact. A fifth setting provides a 10:1 limiter that is not a perfect brick-wall limiter but rather pleasantly squashes the signal with its opto circuit.

The manual appears to have been printed on recycled parchment which all adds to the rather special image that this unit projects. It goes to great lengths to explain all aspects of conception and operation—slightly long-winded in places, but revealing an expert yet open-minded approach to the design, which on paper, and in practice, outperforms most comparable units.

Criticisms? Well, none of the pots are damped, which makes them easy to knock by accident, and the mesh-type top of the unit rattles like snare-drum springs if disturbed. Apart from these niggles I found it hard to fault this unit. The compressor section, in particular, is very useful for signals other than vocals. Considering the amount of thought and care put into the Voxbox, the admittedly high price looks reasonable for such a truly high-class unit. ■

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Microtech Gefell UM 900

It's stylish, intriguing and packs a valve. **Dave Foister** investigates the latest in German mic chic

OF ALL THE COMPANIES to have emerged from the Eastern Bloc, Microtech Gefell stands out as having caught the eye of the upper end of the market. I use the term advisedly, as the looks of its microphones have often attracted as much attention as their performance, helping the range to establish itself more quickly than it might otherwise have done. Recently a new model has been jumping out of the ads with a particularly outlandish appearance, which turns out to be a curtain-raiser for the similarly unexpected internal design.

The UM 900 is as far as I know the first valve microphone to hit the market that can run from 48V phantom power—no bulky power supplies, no special cables, no hum loop problems, just an apparently normal microphone built round a valve. That all the circuitry including the valve's heater can run off just 4mA strains credulity—many solid-state microphones draw more than that.

The bold styling makes the UM 900 unmissable, but according to Microtech it's for practical acoustic purposes rather than cosmetic, minimising the reflection of sound off the body into the capsule. Of course it is.

The design leaves the 1-inch dual-diaphragm capsule sitting in its own disc-shaped enclosure atop a chunky curvaceous body containing the electronics and switches. Maximum value is obtained from the two diaphragms, with the full set of five polar patterns available, including the less frequently provided wide cardioid. The switch for these is on the front, and is a continuously rolling thumbwheel with no endstops. Two other similar switches on the back deal with bass roll-off and attenuation, the latter offering not just the usual 10dB pad but an additional setting giving an extra 4dB of gain in cardioid mode.

All this is mounted in a cup-shaped body whose base incorporates its connector—not the expected multiway but a conventional 3-pin XLR—and a screw collar for attachment to the various stand-mounting options. The standard version as supplied to me has a suitably elaborate swivel arrangement offering forwards-backwards tilting, and relies on internal shock mounting, but there are two other options. One is an even simpler swivelling base ring, very reminiscent of Neumann mounts, while the other provides full cat's cradle elastic suspension. I wasn't aware of any LF problems with the supplied mount.

There are a lot of microphones around whose image is supposed to impress us so much that we don't notice that they're not really terribly good, but the UM 900 isn't one of these. In fact I have a horrible feeling its appearance may count against it as the cynical will assume that its looks are there to compensate for its sonic deficiencies and we're not going to let them pull the wool over our eyes are we? In fact such an assumption would be a long way wide of the mark; whatever Microtech Gefell's reasons for making it look the way it does, it's not because of any reservations about its sound.

Neither is the powering arrangement a bodge to sell a compromise-riddled idea; the concept works extremely well, and I would hope to see it taken up by others. Here we have a microphone that connects to the console in the conventional way yet manages to convey the essence of a good valve design: a smooth fluid sound lacking nothing in the way of detail, focus or spectral completeness. I think you'd know it was a valve microphone straight away, yet that's not because there's anything missing from the frequency response or because of noise—this is a remarkably smooth and clean microphone. It's clear and crisp at the top and has a full and extended bass, and strikes a particularly good balance between flat neutrality and valve character. Its SPL handling—something you'd think perhaps the powering setup might have compromised—is also impressive, making it perfectly at home inches from a trumpet's bell. I also found it useful on double bass, where its big low end combined with good figure-of-eight side nulls, rejecting drum spill pretty well. The polar patterns in the manual are fairly modest in their claims, showing all the patterns to be pretty similar at very high frequencies, somewhere between fig-8 and hypercardioid. This is not unusual; what is unusual is a willingness to show a microphone's shortcomings at 16kHz, something most people keep quiet about. Typically, the most consistent pattern on paper is fig-8, but in practice all come over well with as good an off-axis response as this type of all-purpose switchable design can generally achieve. This is certainly in the same league as the industry standards in this respect.

And this is true in general. Where some new-found microphones appeal by virtue of silly prices, the UM 900 is as seriously expensive as a U87, and warrants the price. There's nothing cheap about it, none of the dodgy build quality, naff engraving or half-baked mis-translated literature. The UM 900 shows once and for all that Eastern microphones can compete on equal terms with the best. ■

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NEW TECHNOLOGIES

<<<<< PCI bus mixed to the 16 outputs. The two new pieces of hardware are a TDIF-ADAT unit (currently known as SS8IO-2) and a TDIF to 8-channel unbalanced unit (currently SS8IO-3) with 20-bit convertors. The target price for each is US$600.
Soundscape, UK. Tel: +44 1222 450120.

Meyer powered

Based on its original UPA speakers, Meyer is introducing the self-powered Ultra-Series with the first products the UPA-1P and UPA-2P speakers, the USW-1P subwoofer and UM-1P stage monitor intended for small PAs in clubs, studios, churches and theatres. Despite the inclusion of an amplifier and control electronics within the enclosure, the UPA-1P and UPA-2P are compact and lightweight, and the same size, but only 10lbs heavier than the non-powered versions. The speakers have 12-inch low-frequency cone drivers and a 3-inch diaphragm high-frequency compression driver and both offer a claimed maximum SPL of 132.5dB. The coverage pattern of the UPA-1A is 100°x40° vertical, that of the UPA-2P is 45°x45°. The speakers have two channels of biamplification and an electronic crossover-processor card. All powered Ultra Series speakers employ limiting technology that predicts power dissipation and include Intelligent AC, a power supply that protects the amplifier and drivers by auto-selecting voltage and minimising in-rush current, filtering EMI and performing power surge protection.
Meyer, US. Tel: +1 510 486 1166.

Oram MWS

The Oram Microphone WorkStation of two channels of mic pre and 4-band EQ has been enhanced. The MWS mkII has an improved noise floor and an insert between the preamp and the EQ. Octamix is a rack mountable 8-channel mixer with 8 pan controls, 2 switches per channel for routeing to the 2 stereo outputs, individual volume control and LED metering together with balanced XLR outputs. A second stereo output has a headphone jack output for cueing. The BEQ Series Four, is a small format console with 4 sub-masters, stereo and cue outputs. The input section is identical to the Oram BEQ Series 8 console. Available with

8, 10 or 12-input channels and optional PPMs it is also available in a flightcase. Mains or battery-powered, the internal battery pack will run for 10 hours on a recharge of less than 30 minutes. The first Compressors from the company, the Sonicomps, have switchable solid-state and LDR attenuators for maximum flexibility. Sonicomp I, is targeted at the project studio and is a linkable 2-channel compressor with identical sonic performance as Son icomp II with LED metering. Sonicomp II complements the Oram Hi-End range in a bigger case with vu meters. Both units have variable pots for input level, threshold, >>>>>

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PROFESSIONAL DIGITAL AUDIO

Summit Audio EQP-200B

Building on the deservedly solid reputation of the 200A, Summit has unveiled the EQP-200B. **Terry Nelson** tests the new EQ

I MUST CONFESS to having found it immensely satisfying to plug in a piece of equipment, turn it on and to have everything working straight away. Such is the case with the American Summit gear: there are no Page Up or Page Down functions, no File Error messages, and, correspondingly, little operational stress. But I digress.

Summit Audio is already well known for its range of hybrid valve/solid-state processors—which use valve stages for the audio, solid-state circuitry for the rest—an arrangement that appears to give the best of both worlds. One of the latest offerings from the Summit camp is an updated version of the popular EQP-200A, the EQP-200B Dual Program Equaliser. And, although the name and control layout of the new box owe much to the ubiquitous Pultecs of yesteryear, it has to be admitted that someone got the concept right from the start. As the term programme equaliser implies, the original intention was to be able to give a bit of tweaking to the final programme output rather than heavier processing. While the modern Summit version certainly provides more EQ power to the user, it does remain faithful to the original concept.

The EQP-200B is a fully-independent, dual-channel unit with controls for LF boost and cut, mid-HF peak boost, and HF cut. In addition, a -6dB/8ve (at 50Hz) high-pass filter can be switched into the main EQ path. All controls have a pleasant vintage look and feel comfortable—meaning that there is something you can get hold of—together with a very precise position markers.

The rear of the chassis features a standard IEC mains connector with 120V–230V selector switch and XLR I-O connectors for the audio. I have one beef here. Why is Pin 3 Hot and not Pin 2? Chassis construction is as solid and smart, as you might expect.

In terms of signal flow, the EQP-200B features an electronically balanced input to a passive equaliser. The gain is then restored by valves running in class-A and feeding the 990 discrete op-amp balanced output stage.

In use, the equaliser interfaced with no problem at all—either in insert on a Neotek Elite console or as part of a component signal chain (preamp—EQ—recorder/monitoring) with no noticeable nasties such as hum or noise. The first thing that grabs the attention with this piece of equipment is inserting it into the signal chain with the EQ bypassed. Everything suddenly comes to life and has space around it. While this might be considered pushing reviewer's license, I can assure you that several people at different stages of the test (even non-audio people) immediately asked 'What did you do?', the effect is that noticeable.

Putting this bonus aside and using the EQ function, it is almost intuitive in telling you what frequency to select and what to do with it. Starting with the LF section, the fact that you can boost and cut at the same time opens the way for some very subtle frequency tailoring. Though this operation may appear to be contradiction in terms, this 'dual control' has the effect of sliding (or tilting) the corner frequency so that you can compensate for some overpowering at the frequency selected, whether the initial wish was to boost or cut.

As noted, the mid-high section is boost only and the amount of Q can be varied with the BANDWIDTH control. This starts with the tightest filter and turning the control clockwise introduces a broader response. In the maximum bandwidth setting, boost is limited to about +10dB against +16dB for peak, but in practise I found this to be totally adequate.

The high frequency cut section is a 3-position -6dB/8ve low-pass filter and, again, is capable of introducing some very subtle frequency tailoring when using quite high amounts of boost with the mid-high section.

The icing on the cake with the EQP-200B is the high-pass filter. This does exactly what it sets out to do—introduces a gentle roll-off at the low-frequency end without removing the body from the overall signal. It is also possible to warm up the low-end by boosting the LF (say, 30Hz or 60Hz) while using the filter to contour the response.

Programme equalisers do provide a different way of tailoring the signal and this unit is certainly no exception. I tested it on masters of live recordings, individual tracks and a selection of programme material—all with great success.

If you have not experienced this method of EQ'ing, I strongly recommend that you try it. Whereas the vintage look and feel are doubtless attractive, it would be easier to miss the point. The larger knobs and switches make the equaliser easy to use and provide the precision that fiddly buttons lack. The drawback with this type of equipment is its cost, but in return, you are getting a hand-crafted unit that does its job superbly well and that, surely, is the bottom line.

The Summit Audio EQP-200B finds applications in mastering, final programme sweetening and as a channel equaliser. Ultimately, there was no way I could let it go—I just had to buy it. ■

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A&H association

The Independent Allen & Heath Association has been formed to provide members with information and advice on A&H consoles, and to provide a forum for sharing ideas. Membership benefits include discounted technical services, cut-price equipment insurance, exclusive merchandise, free classified ads, and a quarterly newsletter giving in-depth coverage of A&H. The Association has also set up its own website at www.allenandheath.com. and owners of System 8s, Sabers, GL2s, CMCs and all other A&H mixers can contact the Association by e-mail: iaha@allenandheath.com
IAHA, UK. Tel: +44 1209 214 147.

Stereo source mixer

Latest in the LA Audio Millennium series of processors is the SPX2 stereo source selector and preamp. It has six selectable inputs and two independently controlled outputs

in addition to headphone monitor and record outputs. The input one trim is on a control knob with the other inputs accessible via screwdriver trimmers. Connectors are on XLR, TRS jack and phono with input one featuring a paralleled set of TRS jacks on the front panel. Source selection is via momentary push-buttons. Outputs have overall BALANCE, DIM and MONO buttons.
LA Audio, UK. Tel: +44 171 923 7447.

Libra Live enhanced

Improvements to the AMS Neve Libra Live digital desk include an enhanced IFB matrix that makes an output available for every fader with talkback and AFI facilities plus a split console mode that allows global changes to be applied independently to the left and right sides of the desk. The desk's snapshot automation now incorporates a 'scope' tool for giving the user control over which console functions are reset while on-air logic has been extended to safeguard the desk against any action that will take the desk off-air. Hardware options now include stand-alone I-O units and fast reboot from Flash RAM. The 55 Series analogue board now includes VCA faders that permit the creation of eight VCA sub-groups via a compact master controller section, an input preselector system and new bar-graph meters with programmable vu or PPM ballistics, variable reference level and a range of scale types.
AMS Neve, UK. Tel: +44 1282 457011.

Telex mics

Telex has debuted its Cobalt SE60 electret condenser cardioid mic with a claimed 40Hz to 20kHz frequency response and maximum SPL of 140dB. The news coincides with the release of the ProStar UH12AD UHF wireless hand-held mic with Audix OM-3XB dynamic hypercardioid capsule. The system works in the 690 >>>>>

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ALLEN & HEATH

Genesis DPM101

Watching the noughts and the ones can be a haphazard business. **Dave Foister** finds a metering solution

DIGITS IS DIGITS and numbers is numbers, innit? Wrong, as anyone who has watched the meters on two adjacent digital recorders supposedly showing the same thing will know. Happily the days when digital recorders showed analogue signal on their meters are gone, but building a meter that can respond fast enough to the digital datastream to show all the peaks and translate the numbers into a display recognisable as a meter is not as straightforward as it might sound. Hence the growing number of outboard digital meters on the market, intended to bring the same pre-

cision to the digital domain as specialist metering did for analogue.

The latest contender comes from Genesis Systems, whose pedigree in digital to analogue conversion is unquestioned. The stimulus for the meter project came from a major London recording establishment that felt the need for a standardised accurate meter for use throughout its various recording, mastering and multimedia facilities. Many of the existing specialist meters have a lot of facilities on them to assist with the specific requirements of mastering, perhaps in the process sacrificing a little of the accessibility of a straightforward analogue meter, and taking the price beyond what is needed for general level monitoring. Here the requirements, in no particular order, were reference accuracy, legibility and simplicity: a plug-and-play meter shedding some of the complexity of existing models. The result is the DPM101, a straightforward and unpretentious tool fulfilling all these criteria.

The simple front panel is dominated by the two principal displays, with only a small section at the left-hand end devoted to status indicators and switches. Only three switches are needed: one selects the digital source, one determines the display mode, and one resets held displays. The input selection makes the unit particularly flexible, as it offers rear panel inputs for AES-EBU on XLR, SPDIF coaxial on phono, and optical on Toslink. One of these will meet anyone's needs, and for those with multiple sources more than one could be permanently connected and then selected on the front panel. All inputs have corresponding buffered loop-through outputs.

Nine small LEDs show various useful items of status information. The incoming sample rate is shown, and the possibilities include 88.2kHz and 96kHz although the hardware for dealing with these is not yet fitted. Professional status is shown, as is Emphasis, and a final pair show proper lock and the presence of errors.

The display area itself comprises a pair of long green bar-graphs with PPM characteristics, and two corresponding 4-character 7-segment numeric displays. These show peak levels in dBFS, with a resolution of 0.1dB, the first space showing the minus sign. There are two basic modes of operation and one more specialised, selected by a second 3-position toggle. In the basic modes the numeric windows show the highest peak so far, and in Hold mode the meters, too, retain peaks. In Normal mode the meters only hold peaks for 3s, and in both modes a single RESET button clears all the held maxima. This button is augmented on the back by a 3.5mm jack to which a remote reset switch can be connected.

A third, Fast, mode dispenses with any peak holding features to show instantaneous levels. This is designed for use with calibration tones, where the delay in reaction of a conventional peak-holding meter can be so frustrating. The immediate response and fine detail make it easy to check and calibrate inputs to convertors and recorders, and indeed to check what's actually coming out of a device that's telling you it's delivering a certain level.

At the top of the scales is a pair of over LEDs, which uncompromisingly light when a single sample reaches maximum. This is for absolute safety, and effectively leaves any arguments about how big an over is audible to those dealing with the finished product. For those who prefer the 1630 standard of Over=3 consecutive maxima, this can be set at the factory. At the lower end the numeric read-outs show -LO- when the signal is below -60dBFS, and, when digital silence is present, which again can be very revealing.

The meter scales can be set to light brighter above a predetermined level, and the threshold for this is adjustable on a screwdriver rotary switch on the back panel—the only tweak on the unit. With this the point at which the 'bright-up' begins can be set anywhere from -15dBFS upwards, or the feature can be turned off, leaving the whole scale at high brightness. This makes it very easy to see at a distance what's going on, despite the rather faint grey printed scale which I understand will stand out better on future units.

The DPM101 is mains powered with an onboard supply and has a 5V outlet for connection of future Genesis products. As supplied it's a neat stand-alone box, but racking kits are available for mounting a single meter or a pair in 1U of rack space. It does a useful job extremely well with the minimum of fuss, and there are few facilities that could not benefit from it. ■

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NEW TECHNOLOGIES

<<<<< to 725 RF range with a frequency stability of 0.005%. The receiver comes in a half rack space unit.
Telex, US. Tel: +1 612 884 4051.

VCA chips

That Corporation has introduced the 2002 that it claims has the lowest noise, lowest distortion and widest dynamic range of any 202-series VCA ever made. Of particular interest to owners of SSL, Neve, Sony, MCI and Harrison desks, the chips are available e as pin-for-pin upgrades for all modular 202 VCAs.
That Corporation, US.
Tel: +1 508 229 2500.

Neotek MicMax

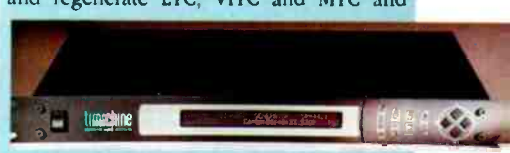
Based on the same circuitry as that found in the Neotek Elite console, the MicMax mic preamp is balanced from input to output and employs discrete transistors and high voltage op-amps. Features include a subsonic filter, bar-graph metering, polarity reverse, switchable input impedance with 500Ω, 1.5kΩ and 10kΩ settings, output ground lift, and phantom power. Gain is controlled in 5dB increments on nudge buttons with a 2-digit read-out plus a ±5dB detented trim on the output.
Martinsound, US. Tel: +1 626 281 3555.

Digital adaptors

Neutrik's NADITDBNC-F and M AES-EBU digital audio adaptors with digital audio impedance transformers allow for longer cable runs via unbalanced coaxial lines rather than twisted pairs. The adaptors provide impedance matching between 100Ω and 74Ω, transition of balanced and unbalanced circuits, electrical isolation, attenuation for use of analogue video distribution equipment and the reduction of hum and noise.
Neutrik, US. Tel: +1 732 901 9488.

Synchroniser

C-Lab's TimeMachine universal clock convertor can synchronise digital sources and sequencers to tape machines, and film and video recorders. The box can read, generate and regenerate LTC, VITC and MTC and



burn time code into video picture. It works as a stand-alone unit with operation via a 4-key pad and text display. Two option slots permit ADAT machine control and video sync pulse generator boards to be fitted. The box uses a special algorithm to render MTC acceptable as a sync source.
C-Lab, Germany. Tel: +49 40 69 44000.

ARX amps and cabs

ARX launched the AmbiDrive 3-channel power amp at Frankfurt with built-in electronic crossover, ISC speaker processing, XLR and jack inputs and Speakon outputs. Channels A and B deliver 160W into 4Ω while the mono subwoofer channel >>>>>

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Avalon VT-737 SP

A next move from the 'purist' camp to 'classic' sees Avalon launch a valve channel strip. **Dave Foister** reports

LEOPARDS CAN CHANGE their spots—if only slightly. Previous experience (all of it pleasurable) with Avalon equipment has ineluctably led to the conclusion that Avalon stands for discrete solid-state class-A just as Nimbus stands for Ambisonics. Equalisers, compressors and microphone preamplifiers all lean heavily on this area of technology with an evangelical fervour, so it comes as a slight surprise to find an Avalon box full of valves. The VT-737 contains no less than four valve stages in a path from microphone to line output, taking in compression and EQ along the way. Not that the traditional cause is forgotten: this is a hybrid design, and the non-valve bits are class-A, solid state, and discrete.

This is not altogether a new model, but the original version has been joined by the SP variant, replacing garish purple knobs with the custom type fitted to the existing range. These are long, fluted aluminium controls, big enough to end up almost too close together and protruding far enough to risk obscuring the panel graphics. The graphics, like the knobs, are more subtle than on the original, making do with simple charcoal legends in place of purple panels. The transformation is startling, turning an eye-poking statement into a model of functional elegance.

And functionality is something the 737 has in abundance. However esoteric the circuit topology, there's nothing minimalist about the facilities. The input stage has three inputs, for balanced microphones and lines and unbalanced instruments, and a continuously-variable gain pot is followed by a switchable swept high-pass filter. This leads into the compressor, which in common with the solid-state AD2044 uses an optical gain control element, eliminating VCAs and ICs. Two of the valves sit either side of this, and the ensemble gives a simple effective compressor. Continuously variable Threshold, Ratio, Attack and Release controls allow virtually any type of compression characteristic to be set up, and the dominant central vu meter can be switched to show resulting gain reduction. The compressor section, like all the other blocks, can be bypassed when not in use.In normal use the equaliser follows the compressor.

No valves are used in this stage, which is based on the same class-A ideas as found in the 2055 EQ. HF and LF shelving controls each have four switched turnover frequencies, and the two mid bands are fully swept with x10 ranging switches. Bandwidth is only variable to the extent that each mid has a HIGH Q button to narrow it down, but in practice the two options meet most needs. All the bands have a more than generous frequency range, the HF in particular throwing down the gauntlet by having a setting above 20kHz, at 32kHz. Of course, this frequency represents the 3dB down point or something (it's not specified) so its effect can be heard in the conventional audio band as a subtle smoothing or an extra sheen.

One of the things that marks down the 737 as out of the ordinary is a number of thoughtful details that make the unit even more flexible than it at first appears. For starters, the order of the blocks can be swapped around with a single switch, so that the EQ comes before the compressor. Personally this is how I prefer to work in most instances, feeling that the compressor should be given as finished a version of the signal you want before doing its job, so the ability to lay it out like this where others force you the other way is much appreciated. But an even more useful touch is the ability to switch the mid-band swept equalisers into the compressor's side chain for frequency-related effects like de-essing, leaving the bass and treble bands for signal sweetening.

Besides the expected ins and outs on XLRs, the rear panel has a 2-pole jack for linking the compressors on two 737s for stereo use—quite a powerful combination. The equalisers, of course remain completely separate. There is also a link on a barrier strip for separating audio and chassis grounds.

It's hard to fault the VT-737. Not only does it give you the kind of audiophile signal handling that has made Avalon's name, but it does it with a breadth of facilities rarely found on a device with such high sonic aspirations. The sound quality is indeed superb, with real transparency and very low noise. Avalon likens the behaviour of its equalisers to that of a passive design, and that seems to be true in that cranking it up, even with the high gain ranges available, doesn't seem to contribute any additional noise or other by-products. The compressor too, with its unconventional approach (shared with Joemeek but not many others), seems to do nothing to the signal except compress it—not as obvious a point to make as it might seem. Some of these all-in-one signal paths trade heavily on one strength, but the VT-737 is equally convincing in all its roles, with every block desirable in its own right. Add the flexibility that comes with integration and it's a winner. ■

Contacts

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Fax: +44 171 2312 9111.

NEW TECHNOLOGIES

<<<<< delivers 200W into 4Ω. AmbiSub is a compact subwoofer accompaniment to the Ambience 1 and 5 loudspeakers. EC-4 is a 4-channel 2-way electronic crossover designed for monitor applications with 24dB Linkwitz Riley filters, low and high level controls on all outputs, balanced I-Os and user-variable crossover points while the 122SK loudspeaker system is a mid-high loudspeaker designed for club installations with a 12-inch mid-range driver complete with a phase plug and a 2-inch compression driver mounted in a rigid flying frame. Dispersion pattern is stated as 60° horizontal and 40° vertical.
ARX International, UK.
Tel: +44 181 742 0350.

Patchbay

Switchcraft's TTP96 audio patch panel is available as a patch panel, patch kit, with EDAC connectors and new front access version. Offered in full normal, half normal and

open circuit configurations, fanned solder terminals make soldering easier while co common bus ground connections are aided by offset ground terminals.
Switchcraft, US. Tel: +1 773 792 2700.

DVD-ROM developer

Available for Mac OS and Windows NT, Sonic DVD Vobulator is an authoring tool that enables multimedia producers to create content for DVD-ROM and digital broadcast use. Designed to convert video, audio and still images into the data formats required by DVD, the package includes software-based MPEG2 variable bit rate video compression, Dolby Digital surround audio compression and Video Object multiplexing and demultiplexing. QuickTime, AVI or OMF video files can be converted into MPEG2 variable bit rate or constant bit-rate video. Compression parameters and group of pictures structures can be varied and DVD Vobulator also converts uncompressed digital audio files in AIFF format into Dolby Digital digital audio.
Sonic Solutions, US.
Tel: +1 415 893 8000.

HHB CDR74 Gold P

The CDR74 Gold P is a printable version of HHB's CDR74 Gold disc optimised for professional audio CD recorders. It features a white paper-like surface for text and graphic printing directly onto the disc surface using a CD-capable ink-jet printer or for hand-writing with a water-based ink marker pen.
HHB, UK. Tel: +44 181 962 5000.

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Preview

Akai S5000; S6000

The original champions of the sampler have upped the ante with two next-generation units. **Zenon Schoepe** previews them

THERE CAN BE FEW people who haven't encountered an Akai S-series sampler in the course of their work. Be it through the musical instrument applications they originally found, or as the fast and powerful capture, edit and fly-in tool for theatre sound, post, sound design and broadcast applications that they have expanded into. Akai samplers are now ubiquitous. Thus it is with intense interest that visitors to the Frankfurt Musikmesse welcomed the announcement of the S5000 and S6000 samplers—not just because of their lineage but because they represent a new generation of the technology that Akai claims effectively reinvents the sampler.

The complete rethink was brought about by the fact that previous-generation sampler technology had hit the end stops for further development—it's worth pointing out that S20, S2000 and S3000XL will all remain current—and went about designing a new generation of LSI for the function. With the company's accompanying range of postproduction-orientated hard-disk systems derived from the DD1500, Akai also had to questioned where the role of the sampler ends and that of the hard-disk recorder-editor begins. There was also the consideration that, when the S900 and later the S1000 were launched, they effectively defined what a sampler should do as no true precedent existed, by starting with a clean sheet of paper these ideas had to be challenged.

A completely new flash ROM operating system has been devised that is multitasking and the demands of polyphony, outputs, memory and disk limits were met with the new processor to run it all. There is some modularity possible with the hardware and the software is now substantially more open ended for easier addition of future features. Much attention has been applied to the user-interface as the requirement to add more and more features to the original S1000 through newer models did render what was originally a fairly straightforward system more complicated over the years. One of the most significant developments is the implementation of .WAV files as the native sample format in the new units. This was requested by users and means that sound design can be performed on the samplers and when the drive is hooked up to a PC work can be continued. It also opens up avenues of convenient downloading of sounds into the machines. Similarly while the previous boxes looked at stereo signals as two mono signals the S5000 and S6000 deal in true stereo. The machines will, however, read S3000 and S1000 sound libraries.

Cosmetically, they look great with a detachable front panel on the flagship 4U-high S6000; although both sport a large display with keys surrounding it. You can also connect up a PS2-type computer ASCII keyboard for text-entering purposes. Specifications include 128-voice polyphony, which is standard on the S6000 and upgradable from 64 voices on the S5000; the ability to install 256Mb of RAM; 32-channel multitimbral operation; two pairs of MIDI In, Out and Thru ports, digital I-O; 16 individual outputs configurable as stereo pairs; wordclock connection and two SCSI ports. Both machines can replace their onboard diskette drives with a Zip, and the S6000 will also be able to house a Jaz drive. An ADAT digital interface is planned.

Timestretch has been improved and has been taken from the DD1500 together with various derivations of pitch shifting. Looping has been improved and includes new loop crossfade functions. There's also BPM match, 3-band EQ, fade up and down and resample. Up to 128 multi-programs can be loaded into RAM at a time while a quickload function loads programs directly into parts. Akai's Assignable Program Modulation is included together with new filters (15 different types), envelope generators and LFOs. A 4-channel, 20-bit multieffects processor is being developed and this will be standard on the S6000 and optional for the S5000.

Price for the S5000 is £2,000 (UK, inc VAT) with delivery expected in August, that of the S6000 is £3,000 (inc VAT) with arrival expected a month later. There can be no doubt that Akai has raised the ante significantly with the unveiling of these two devices and it remains to be seen what its competitors are going to respond with. In the S6000 Akai would seem on paper at least to have created the definitive sound designer's professional sampler for the late 1990s. ■

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NEW TECHNOLOGIES

CD jukebox

Described as a 'universal jukebox' system, Grundig's GMS3280 CD jukebox can accommodate 280 slots via eight exchangeable caddyless magazines and a maximum of six drives for library and archive applications. Mastering and CD reproduction is possible with the company's GRS1000 software and a maximum of 280 discs can be produced using four CD recorders in parallel mode and a printer module for automatic CD labelling is also available as an option. Other options include a mail slot for media exchange in on-line mode, fast SCSI2 interface and applications for Windows, Novell, Mac and Unix platforms.
Grundig, UK. Tel: +44 181 324 9488.

Furman expands range

New additions from Furman include the HR6 headphone personal 6-channel headphone mixer which clamps to a mic stand and allows musicians to customise their own mix; the HDS6 headphone distribution

system for driving a chain of HR6s; and the IP2B Iso-Patch dual transformer isolator. The PLH15 power and light centre combines a power conditioner with lights while the MiniPort20 power relay is an upgrade to the original MiniPort and adds support for momentary-action switches, multi-unit linking and knockout holes for permanent installation. Furman's first product the PQ3 parametric EQ and instrument preamp has been rereleased with its familiar green panel and red knobs but with the addition of a front-panel input socket.
Furman, US. Tel: +1 707 763 1010.

Netshow 3.0 supported

Waves' Audio Transmission Processor (ATP) software package, v2.0, now supports the Microsoft NetShow 3.0 multimedia server. NetShow enables Internet content providers to deliver high quality audio and video across LAN and ATM networks. ATP is a real-time signal processing tool that allows for optimal sound control when transmitting live events. It can handle eight or more simultaneous channels and is well suited for Internet broadcasters transmitting multiple streams of audio. The system was developed with broadcasters in mind, delivering a single rackmount box for netcast, FM, AM, TV or cable audio processing. Currently, ABC Radio in New York City is beta-testing the ATP software. The ATP system controls the listening volume and prevents overflow and distortion of broadcast audio. >>>>>

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Studer Professional Audio AG, Althardstrasse 30, CH-8105 Regensdorf - Zurich Switzerland. Tel. +41 1 870 75 11, Fax: +41 1 840 47 37, http://www.studer.ch

Audix OM-series

Retaining many of the benefits of the D+series, the OM also do well as dynamics, as **Dave Foister** discovers

A RECENT LOOK at the Audix diminutive D-series studio dynamic microphones revealed unsuspected gems. The little known and still expanding range gave surprising results on a wide variety of sources, with excellent transient response and smooth extended bass making them particularly suitable for percussion, and one in particular, the D-4, turning out to be outstanding on double bass. A companion range, the OM-series, shares many of the same features in bodies intended primarily for stage vocal use, and the addition of a model with studio aspirations makes them worth a further look here. Like the Ds, the OM-series comprises four closely related models. All are built round various versions of the Audix VLM (very low mass) capsule that uses a very light diaphragm to obtain a particularly fast response. The capsule comes in three flavours offering slight differences in sensitivity and impedance but similar overall performance, and the use of the different variants is the chief distinction between the models. This central capsule design is credited with providing a combination of accuracy, extended HF and low distortion alongside the traditional virtues of dynamic operation.

From the outside the four models are virtually identical apart from the numbers in white below the heads. They are neat, slim and unobtrusive, yet appear highly robust and likely to withstand stage use well. The hard, black coat finish too smacks of quality, capable of taking a knock or two, and the whole appearance gives off an impression of quiet class.

The microphones come with a simple but adequate stand mount (complete with thread adaptor!) and the whole lot is contained in a canvas-style pouch. It would, perhaps, have been useful if one of the variants was offered with an on-off switch since some singers find them so reassuring, but, as it is, all the models are completely devoid of switches and controls.

Three of the microphones have been around for a while and Audix literature makes it clear their intended environment is the stage. The OM-3xb is the original, using the VLM Type B capsule in a transformerless arrangement that gives a quoted frequency response from 38Hz to 21kHz (no tolerances specified). This is the standard response in the specs for nearly all the Audix dynamics, and unless the limits are the -20dB points it suggests a high level of performance for a dynamic. The 3xb's polar pattern is the standard hypercardioid, again common to all the capsules and all the models.

The Type C capsule differentiates the OM-5, this time used with a transformer. The arrangement knocks a few Hertz off the extremes of the frequency response but ups the sensitivity by 3.5dB, giving a significantly higher output level. Alanis Morissette apparently loves the way it sounds, from which you can draw your own conclusions, and the aim remains the same—to deliver a natural undistorted sound. Audix's brochure is scathing about neodymium designs, talking about 'unnatural sound and excessive feedback' which might alienate some. The OM-7 is effectively a transformerless version of the 5, which restores the full frequency range but leaves it a full 10dB less sensitive. According to Audix it features a 'controlled low-gain output stage', but further clarification is not given.

The final model, and the latest addition to the range, is the OM-6. This states its intention to be useful in the studio as well as on stage, and uses the top of the capsule types, the VLM Type D as used in the newest of the D-series, the D-4. The capsule has similar electrical characteristics to the Type C but manages a considerably more refined sound.

To set it in context, the other OMs are excellent stage vocal microphones. I've used them on several voices with favourable responses from the singers, and have even put them on violins on stage with very good results. The chief difference under noncritical conditions seems to be the sensitivity, because all deliver quite similar sonic performance. The common characteristic is a crisp and smoothly extended top end which helps vocals cut through without making them unduly harsh. At the same time the lows are well handled and the proximity effect easy to control.

But the OM-6 is almost in a different league. Eager to check whether the qualities that had impressed me so much on the D-4 were carried across along with the capsule I put them up together, along with the OM-7, and was indeed equally impressed. While the 7 sounded like a good vocal dynamic, the 6 was effectively identical to the D-4, giving a sound that could easily have come from a condenser. There was no strain, no constriction at the top and a smoothness right across the range; the bass in particular, as on the D-4, was full and extended.

The roster of artists said to be using this Audix range is already impressive, and the addition of the OM-6 should add to it considerably. All the OMs are capable of holding their own in much more familiar company, but the OM-6 stands out, live and in the studio, as a particularly good find. ■

Contact

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NEW TECHNOLOGIES

<<<<< Audio sources including analogue, AES-EBU, SPDIF, compressed MPEG1 Layer 2 can all be streamed through the ATP system. Meanwhile, WaveConvert Pro software v3.0 now supports the Microsoft NetShow 3.0 multimedia server. WaveConvert Pro is a multimedia audio mastering application that allows users to prepare audio files for transmission over the Internet. It can preview and batch process all Waves' Native plug-ins, plus convert PC audio files between sample rates, word lengths, channels (stereo-mono) and file types (AIFF, snd, QuickTime and .WAV). It includes special preprocessing filters for streaming Internet audio. WaveConvert Pro is suited to multimedia audio production and in preparation for Internet delivery, providing loud and clean audio files. For websmiths, it allows the user to tailor the sound of the resulting encoded file. WaveConvert Pro features low aliasing, maximum audio level, more intelligible speech, reduced background noise, and one-step multifunction conversion. This is made possible by sample-rate conversions, audio level maximisation, presence enhancement, 16-bit to 8-bit conversion, automatic gain enhancements and batch processing.

Waves, US. Tel: +1 423 689 5395.

New Isopatch

Signex has replaced its Isopatch with a fully redesigned model. Retaining many of the features of the original model, the 1U-high rackmount now has 48 sockets which are of a fully-enclosed design and help keep out contaminants. All sockets are mounted on two horizontal PCBs which eliminate internal wiring and add rigidity. Supplied with all sockets isolated, half or full normalling can be achieved by soldering across pads on the top PCB. The use of flexible jumper cables to carry normalising signals between top and bottom sockets allows full access for servicing. The new Isopatch is available with jack, phono or direct solder terminations at the rear.

Isotrack, UK. Tel: +44 1202 247000.

Smaller switchers

Pro-Bel is launching a line of small routeing switchers with the 16x2 family satisfying the need for mixed signal formats. Typical configurations include 16x2 analogue video and stereo analogue audio, 16x2 serial digital video and AES audio, and 16x2 serial digital video and stereo analogue audio all in 1U-high rackmount frames. Front-panel control provides 16 source push-buttons with dual colour LEDs for indicating video or audio selection. Dedicated buttons also provide destination and level control. For >>>>>


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1038A
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Power Technology DSP FX

As the platforms prosper, the wealth of plug-in effects modules grows daily. **Rob James** presents an exclusive preview

THE AMERICAN Power Technology's DSP FX package is a collection of Direct X (and SAW format) effects plug-ins for PC-based DAWs. With one or more of Power Technology's ISA DSP cards, it can also turn a PC into an effects rack, which is the genuinely interesting part of the proposition. Unfortunately I was only able to get hold of the Virtual Pack—read the plug ins—for this preview.

On the upside, the software algorithms are claimed to be identical for both packages. Further hardware offerings are the DSP FX AES-SPDIF interface and the DSP FXConverter which offers 20-bit conversion, and +4dBu balanced connections. Both of these connect direct to the processor card. The other interesting item is a JL Cooper CS-10 compatible hardware controller.

If you have any experience in attempting to add multiple cards to a PC you will appreciate Power Technology's approach. There are essentially three compatibility issues when adding cards, IRQ (Interrupt ReQuest), DMA channel (Direct Memory Access) and I-O address. Power Technology has avoided the two most problematical issues—IRQ and DMA—by designing around them leaving only the I-O address, which is relatively simple to resolve by setting of three DIP switches on the card.

The Virtual Pack is currently dongle protected against software piracy. However the manual promises a future version of the software protected by a registration process for those, like myself, who find dongles loathsome. When the software is used with the proprietory card(s) no dongle is required. Up to 8 cards can be installed (if you have enough spare ISA slots) for a total of 8 simultaneous effects. The cards are equipped with unbalanced -10dBv analogue I-O. Power Technology recommends the cards are patched via the external connections. All audio processing takes place on the card, so performance is largely independent of the host PC.

With the Virtual Pack, plug-ins can also be run 'stand-alone' to process .WAV files. Real-time preview is available. I can see no good technical reason why this should not be extended to real-time effects processing like TripleDAT's Warp mode, assuming a suitable duplex soundcard is installed, although performance would obviously depend on the host PC, but no doubt Power Technology wants to sell its processor cards.

The effects all use a similar virtual interface. Some components are common to all the mod-

NEW TECHNOLOGIES

<<<<< smaller applications in existing Pro-Bel systems, the 16x2 routers can be controlled from remote panels connected to System 2 or System 3 controllers.

Pro-Bel, UK. Tel: +44 118 986 6123.

Soundcraft SM20

Soundcraft has revealed the SM20 monitor console which is available in 40, 48 and 56-input sizes with 20 outputs globally switchable to mono or stereo. Features include built-in mic splitting, sweepable high-pass filters and MIDI control of BSS Varicurve equalisers. Also new is the Series Five Monitor desk with a feature set similar to the FOH variant but with more elaborate routeing and output control. Types include

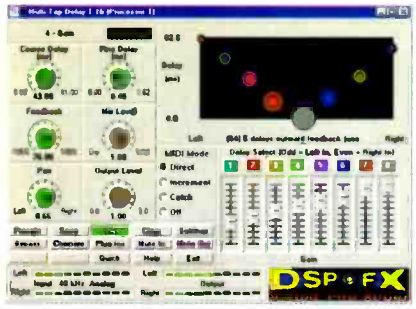
NEW TECHNOLOGIES

24-bus and 32-bus versions with the former including a 23x12 output matrix. The established K2 8-bus desk is now available in a 48 mono channel version with an additional four stereo inputs.
Soundcraft, UK. Tel: +44 1707 665000.

SA monitors

Stage Accompany has expanded its Master studio monitor series monitors with the M57 which uses Ribbon technology and has been developed with what the company terms 'linear response' design. The SL range of cinema bass cabinets from the Screen series are only 23cm deep and complement the Screen top designs. The P2-29.SB touring system from the Performer series uses SA's plug and play and comes with four top cabinets, four sub cabinets, four digital amps and the SA-net system. Frequency response is claimed as 30Hz to 30kHz at a peak power of 148dB.

Stage Accompany, Netherlands.
Tel: +31 229 282930.



ules and others vary according to the specific process. There are a bunch of 'knobs' on the left of the screen with a block of 'buttons' below. Horizontal input and output meters are bottom left. A vertical row of 'radio' buttons select MIDI mode and there is a graphical representation of the effect top right. Bottom right is blank in some effects or may have a row of virtual sliders on, say, a graphic EQ or the delay module. Each effect has a comprehensive library of presets which can be added to by the user and a QUICK PICK button allows one of five favourite presets to be selected without scrolling through a list. The effect can be Bypassed or the output Muted if you do something dangerous to hearing or speaker cones. The Compare function allows a direct comparison of current settings with the original preset. If you want to start from scratch press CLEAR. The EDIT or MORE buttons access extra parameters by changing the function of some of the knobs. Left and Right parameters can be set together by pressing LINK. Alternatively some weird and wonderful sounds can be obtained with independent settings.

The algorithms currently in the pack are Reverb, Delay, Chorus, Flanger, Pitch shifter, Parametric EQ, Autopanner and Tremolo. 32-bit floating point arithmetic is used throughout the processing chain.

This is a thoroughly usable set of effects which would not disgrace any studio.

Effects are always a very personal thing and my favourite of this bunch is the delay. With up to eight elements and comprehensive controls some really huge and outrageous effects are simple to set up. The reverb is fine, but a Lexicon 480 it isn't and I have to confess to preferring some other plug-ins. The pitch shifter reminds me of early Eventide (910) but maybe my memory is being kind to the Eventide. One annoyance is the shift is calibrated only in cents and I always work in percentages. Again, there are better stand-alones around but nothing close to this price range. In fact for the price of the only truly competent pitch shifter I have ever found you could have the whole Power Technology setup including the PC. The EQ is fine with an unusually large ±30dB on offer. Flanging, chorus and Tremolo are all good fun.

If the price is kept reasonable the whole concept is a good one. Dedicated processor cards offer a more robust and predictable solution than software only. The idea of turning last year's PC into a serious effects rack, that can be updated with new effects comparatively cheaply, has a lot to commend it.

Contact

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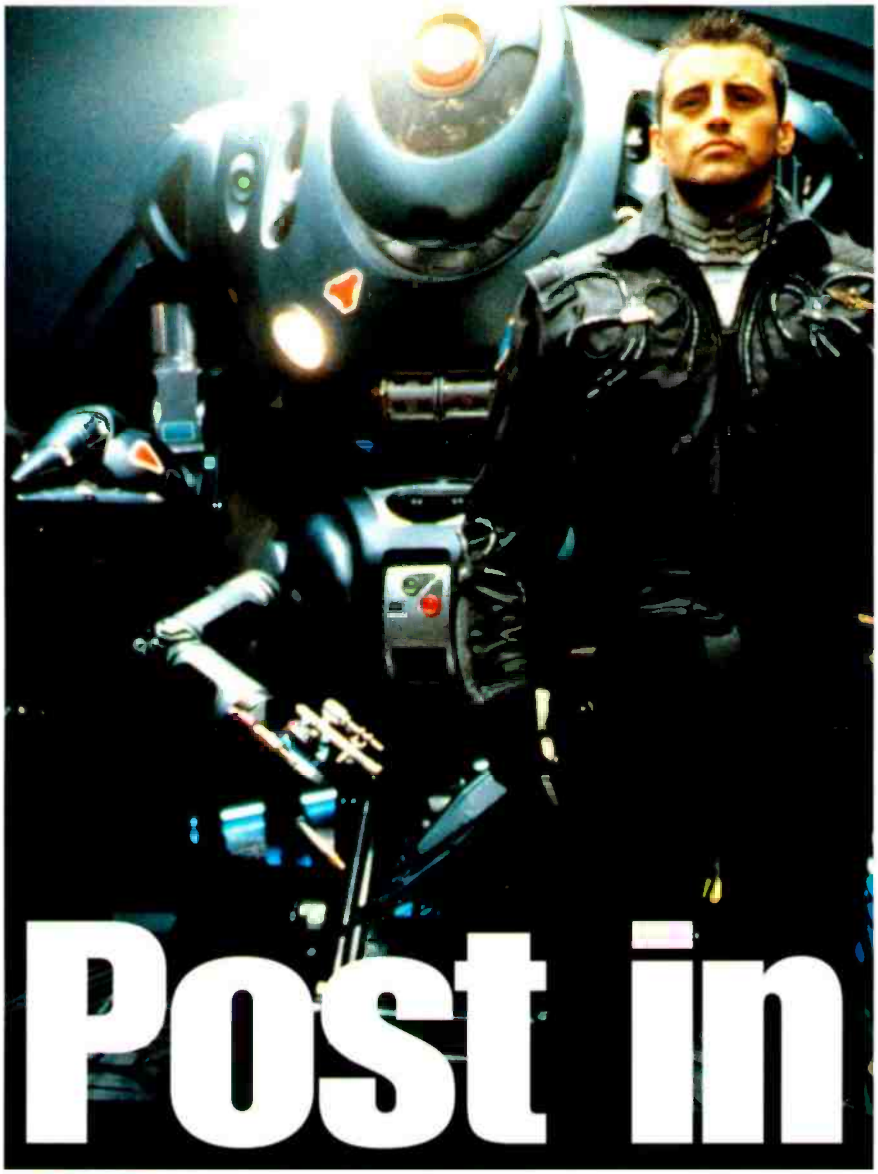

Post in Space

On the eve of the release of the first all-digital Hollywood picture, **Richard Buskin** speaks to the crew behind a new sci-fi blockbuster, and discovers a postproduction schedule that would exhaust even *Lost in Space*'s Robbie the Robot.

NOBODY EVER PRETENDS that film work is easy, but even by Hollywood's manic standards, this takes some beating. The location is Todd-AO in Los Angeles, the date is 10th March 1998; a bunch of bleary-eyed professionals are working from 9am until midnight seven days a week to complete the final mix on *Lost in Space*. Shot entirely at England's Shepperton Studios between March and July of 1997, directed by Stephen Hopkins and starring William Hurt, Gary Oldman, Matt Le Blanc and Mimi Rogers, this is a high-tech, big-screen update of the 1960s TV series about the trials and tribulations of the space family Robinson together with Dr Zachary Smith and Robbie the Robot.

The past few months have seen three temp mixes, multiple editing, the predubs and now the final mix. Today it is Tuesday. On Friday the team is scheduled to fly to London to commence work on the mag mastering. Just over two weeks later *Lost in Space* will open in the US. Nevertheless, at the time of my visit to Todd-AO's refurbished and equipped Stage 1 (more of which later) and nearby Studio C, there are still half a dozen London facilities cranking out the visual effects and sending them to LA via ISDN, where they are converted to PAL 25fps Beta SX. That means the post guys are, in numerous cases, mixing the sound without pictures and then remixing after either seeing the visuals or simply being corrected by the director.

'We've done a lot of manic projects but this is the worst,' says Chris Jenkins, the president of Todd-AO who is also mixing the film's dialogue. 'There's been no time for preparation, and let's just say that the fault for that lies with the nature of the movie. It's so huge, there are so many shots, and stuff is constantly being pushed back and back. Yet we have a release date to meet and we've got two studios up and running to somehow make that deadline.'

'Things have definitely been extremely compressed,' adds effects mixer Ron Bartlett. 'The lack of visual shots has been the hardest thing—it's not easy trying to do this job with about 500 shots missing, and we've often just had to make our best guess and then cross our fingers. We'll be mixing away and the director will come in and say, "Oh, no, no, no! In this sequence a huge rock comes down...", and we'll say, "There's a rock?" So it really has been crazy and a lot of times our predubs are useless. Chopping up the sound of a moving object, creating holes in it and trying to fit it to a new effect is just wrong. It sounds bad. So, in those cases what we've done is retain the pieces that are good, cut a mass amount of sweeteners to fix around that, and then make that all move in conjunction with what's supposed to be there.'

'Sometimes it feels like we're doing too much,' says supervising sound editor Eddie Joseph, 'but somehow we keep it all together. We have to. We've received the last visual effect and the film is completely locked just three weeks before we open, whereas normally you'd be locked before mixing. In this case, however, due to the complexity of the visual effects it just hasn't been possible.'

'Poor Eddie could hardly do any sound design because he was always chasing temp dubs,' adds Chris Jenkins.

Joseph commenced work on the project in his native England, and, as initially described to him, the overall idea was to aim for that sci-fi movie staple: 'something different'.

'We've had *Star Trek* and *Star Wars* and so on, and the objective for this film was to go another stage,' he continues. 'Stephen [Hopkins] actually wanted to make the spaceship more organic—even though everything is high-tech it's only set about 60 years in the future, so a lot of the things are still going to be the same as we know them. The robot, for instance, would be servo-constructed but would still have the cyber equipment in there to give it a more snappy, sleeker sound. That, therefore, is what we were generally going for: a combination of sounds that you'll know and understand together with others that you won't recognise.

'I've done big films before but this is the first sci-fi movie that I've worked on, and so whereas normally I would never have thought of employing a sound designer *per se*, in this case I've brought in people who could do a specific job for me. For instance, there's a musician who I met at Pinewood Studios who

helped me design some 'zaps' and moves for the robot. He could use music samples to achieve this, rather than me simply utilising a real sound, and the robot therefore ended up with servo parts and movable joints and interesting little noises as well... It's a fairly conventional-looking robot, a very heavy machine, and he runs on treads, and we put an enormous amount of low end into the tread sound so that it becomes a formidable machine in itself. There again, while we tried to give a servo to the trunk and head movements, he has two regular arms and two smaller arms and for those we created little zippy sounds just to help identify how the different areas work... Sometimes you can use things that you never would have thought would work, and people accept it.'

As for the oyster-shaped spaceship that the Robinsons travel aboard, Joseph describes this as, 'a very beautiful, sleek beast. It has its own kind of life force in that it has this throbbing sound, instead of a constant noise which a lot of other movie spaceships have. Again, it gives the impression that, while it's a machine, there's also something organic about it, and we don't always achieve that because we can detract from what's going on by being a bit too clever with the all of the different sounds. You start with nothing and therefore you're scared that people are going to think that it's a set. As a result you probably put in more than you should to start with, and you then start taking things away. If the music is good—which it is—and if the dialogue is important—which it often is—then of course the backgrounds and effects have to be supplementary; the normal mixing procedure, not always adhered to, but we've at least tried to do that.'

WHILE EDDIE JOSEPH works with a DAR Soundstation, other sound crew personnel were employing Waveframe, Audiovision, Pro Tools and Fairlight. To that end, the MMR8 was the recording medium of choice because of its ability to interface with all of the aforementioned.

'The MMR8 is a great medium,' says Eddie Joseph. 'If you want to do changes on the stage the MMR8 will plug into a Waveframe or whatever, you can do a file edit, put it back again and it's done, and you haven't actually transferred anything out.'

According to Joseph, sound editor Ron Eng convinced everyone that the MMR8 route was the way to go on the *Lost in Space* project: 'We could do a master and he could then unplug the drive, put it into his Waveframe on the stage and edit, whereas a Fairlight doesn't interface with the editors unless you're using Fairlight. We had a change yesterday on Reel 5, for instance, and it probably took Ronnie 15 minutes to do the form on the three master stems as well as seamlessly fill in some backgrounds. In that way it's incredibly fast. Plus, I think mixers will enjoy using the MMR8 more and more because they can hear backwards and it locks in instantly. I wish we had more here, we haven't got enough. We've only got about eight and we really need 15. We hang the rest of the predubs on DA-88.'

Meanwhile, with the use of Todd-AO's new AMS Neve DFC consoles—installed on 5th December, operational 6th February—*Lost in Space* is the first all-digital Hollywood picture (discounting the mag mastering). In this respect the Americans have thus far lagged behind the Europeans, and Chris Jenkins doesn't only put this down to cost-effectiveness.

'Until now, nobody has built a digital desk that can handle film mixing,' he states. 'We've used everything that's out there, because mixing has become so convoluted that we're running two consoles. Even with our bigger boards we're running extra consoles—another 72 inputs or another 38 inputs, like everybody is—and the digital boards that we've used have just been terrible. The monitor busing, the panning; they're all geared towards making records, whereas the DFC is the first all-digital desk that we've seen that can conform to realistic film needs. We had its predecessor—the Logic 2—in here two years ago and it was terrible; we actually had two of them and they were an absolute nightmare to work with.'

Todd-AO now houses a 3-man DFC on Stage 1 and a 2-man version in Studio C. 'The DFC is based on the same processing engine as the Logics, but we've redesigned the surface so that it's more user-friendly in the film environment,' explains Hugh Gwilym, the AMS Neve product manager who was nearing the end of a babysitting stint at Todd-AO when I was there. 'Basically, that consists of

the busing structure and also the channel substructure. The busing structure is completely soft in this environment, customised so that it's organised in stems as the film mixers are used to, and on each of the assignable channel strips the busing is presented accordingly. Then, in terms of the layout of the channel strip, the parameters and the facilities that film mixers consider to be the most important have been laid out closer to the surface, and we've also redistributed them on the same page. At the same time we've introduced some additional facilities, such as linking parameters, while the console can be run in multiple sections.'

'This board is just remarkable,' Chris >>>>>

FILM PHOTOS: JACK ENGLISH

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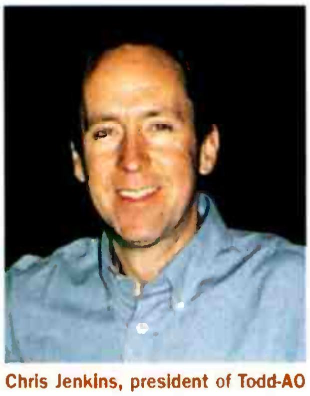
Ron Bartlett, effects mixer



Chris Jenkins, president of Todd-AO

TECHNICIAN PHOTOS: RICHARD BUSKIN. FILM PHOTO: JACK ENGLISH

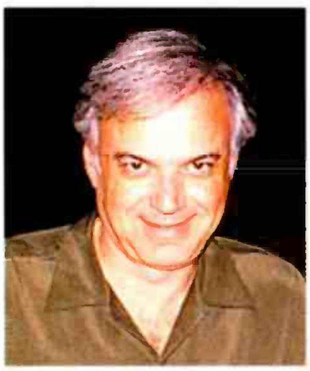
<<<<< Jenkins agrees, with regard to not only its practicality but also the digital sound. 'Compared to where you are with analogue boards and analogue outputs the difference is just black and white, and I wasn't quite prepared for that. I don't think anyone was. I don't think they were aware of what the gains would be as far as closing the loops on things. It happened so fast in this company: between December and February the world completely changed with regard to the work process, how stuff sounds, how it gets around the place. Nobody could have foreseen us doing things on the scale that we now are for this picture, because we reinvented everything. I'm constantly running around and yelling at people, saying, "You still put tones on the head of a drive because it's a drive". We're going to take all this stuff to England without lineup tones, the labelling isn't right, and this all because we truly have thrown out 75 years of professional audio discipline. It's exhausting to see what people are going through to impose all of the old practices on the new technology.

'The shift in culture is just stunning. We have one guy who does everything upstairs, and the reel changes that used to take an hour will now take him maybe 45 seconds or a minute to set up all of the audio for the show. He'll pop in half a dozen drives and the videotape picture and it's there. Nevertheless, it's funny how it's perceived—I remember, we were temp dubbing and we had everybody in Studio C, and because it took two-and-a-half to three minutes to change a reel instead of one minute people suddenly became really tense and uptight! Never mind that it took a twentieth of the time that it used to take.'

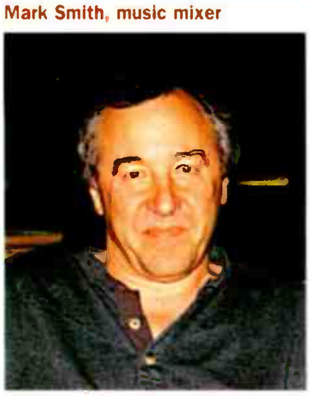
'Using the digital console is such an advance,' adds Ron Bartlett. 'What you put in you get out. There are, of course, some really nice things about mag—the flow of the way you work—but now at least we don't have to worry about machines being lined up properly, Dolby SR cards and the headroom on that. The speed is lightning fast versus rolling down a 2,000-foot roll of mag and trying to cut a predub, and not being able to do a cross-fade or anything to fix it with. You know, we would get rolls of mag chopped up and then a bunch of sweeteners to fix it, whereas the guys will now just fix it in their workstations. So, that's a huge help.

'We bought the console and bought all of these A-D converters together with some digital—definitely more analogue than digital—assuming that people were going to come in on 2-inch analogue mag and some digital workstations. Eventually, we thought, things would cross over totally to digital, but the very first show, whoosh, they were gone. We didn't touch one piece of mag. It was unbelievable, but who could have known? Really it was down to having to put this show together in the time required that meant we had to do things this way. We called around town and got all of the MMR8s we could find, we had two stages going in the same PAL setup, and hooking that all up was an amazing task. The engineering department deserves a hand, because just to get all of this gear to work within our compressed schedule was shocking.'

Bruce Broughton composed the film's bombastic music score which was recorded at Abbey Road in London and then mixed by Mark Smith in LA. Indeed, Smith, who worked on the effects predubs together with Ron Bartlett, really got himself out of jail when—as per the two men's usual procedure at Todd-AO—he switched to the music mix while Bartlett continued to deal with the effects. After all, it wasn't the music which was thrown into turmoil by the constant flow of visual effects being sent >>>>>



Mark Smith, music mixer



Eddie Joseph, sound editor

TECHNICIAN PHOTOS: RICHARD BUSKIN. FILM PHOTO: JACK ENGLISH

<<<<< via ISDN from London.

'I'm out here in the sunshine talking to you while the others are sweating it out in there,' is how Smith succinctly summarises the situation. 'You know, what with all of the dialogue changes, ton of ADR that has come in since we predubbed, sound effects literally being spotted as we go along, and all of the picture changes and conceptual changes on top of that, these poor guys are going crazy. Meanwhile, I'll lower the intro a little and relax. As any of the music guys in town will tell you, we've got a great gig. Because, as important as the music is—it's the emotion of the movie, often driving the film along and linking scenes together—by its very nature it's not something that you can just cut in one place and pick up elsewhere. It has to flow from Reel 1 until the end, and so there are restrictions as to what you can do to it.

'Meanwhile, for my part, having already predubbed the sound effects I know where all of them are, and so I know what's working with the music and what isn't. The music is a fact of life, whereas in most cases the sound effects aren't. The dialogue is also a fact of life, and so it's really good to know what does and doesn't work with the music.

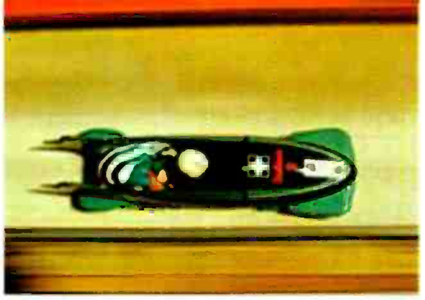
'The music came to me on eight tracks with a left-centre-right orchestra, left-centre-right synthesiser, choir tracks and two tracks of percussion. The percussion mainly comprised close-in mics from the live orchestra as well as the occasional overdub; the synth parts were very, very minimal; and the choir performed a very important part of the score, with female voices split from male voices.'

Stressing the importance of carefully programming the Neve DFC in order to achieve the desired results, Smith asserts that, in this respect too, the music mix represents the relatively easy option. 'With music you have a beginning and an end, and you have spaces to set up the board,' he explains. 'The sound effects, on the other hand, are going constantly, and so you have to be very careful in setting up the console, while dialogue is marginally easier; there are pauses in the dialogue where you can actually go offline, set up your mix and punch it in. Still, music is the easiest thing to work on with this console because you do have the time, and it sounds fabulous.'

Still, if the *Lost in Space* dialogue has been easier to mix than the effects, Chris Jenkins could be forgiven for not noticing. Thanks to the usual extraneous noises caused by locations, sets and costumes, not to mention the more-than-usual number of film revisions that have been deemed necessary throughout the months, about 90% of the dialogue has been looped.

'Everyday we loop,' says Jenkins. 'We'll have the actors in all day on the stage, and we'll finish mixing at one in the morning and then somebody will come in because another ten lines need looping. Still, I have to say that Stephen Hopkins is just the most enthusiastic, hard-working guy. He's doing this and he's doing visual effects as well and keeping all of these balls in the air, and he's working his tail off to do it. He's a great guy to work with because he never lets it get him down, but we're quickly coming to the realisation now that we have three more days to work on two artist reels. We usually have three days a reel, but we keep cutting time off the schedule. There again, this is not a crew that likes to compromise at all because they are trying to do their best work, so it's a constant struggle—not to mention the fact that we've got an entirely new system: new workstations, new consoles, new everything. That's been the nature of *Lost in Space*.

'We really are in the trenches on this project,' concludes Mark Smith, 'but we'll all go on to other movies and look back on this as a tremendous bonding experience. Everybody's got a great attitude and we are having fun, even though things have been—how should I put it—very, very labour intensive.' ■



Cutting the ice

Kevin Hilton reports on the technical innovation and unprecedented airtime at the Nagano Olympics

IT IS SAID that one can learn a lot from television. Over two weeks in February, many people gleaned that during a curling match, the ice is brushed not to speed the stone up but to help it to slow down less quickly. We also learned that snowboarders are both closely related to hippies and a cornet short of an ice cream van. If all this were not enough, it became apparent that no matter how many highly paid, cynical, professional ice hockey players the US threw into the ring, the amateur Olympian spirit was destined to win through.

These shimmering facts were to be found in the learning zone that was the 1998 Nagano Winter Olympics. As the worldwide television appetite for sport apparently increases exponentially, this year's ice and snow Games—held in a remote Japanese city, approximately 200 miles north of Tokyo—amassed a staggering total broadcast time of 115 hours and 36 minutes on domestic broadcaster NHK's general service alone. Over 31 hours more than that of the last Winter Olympics in Lillehammer, Norway. More significantly, from the future technological angle, there was 272 hours of Hi-Vision (the Japanese high-definition format) coverage, two-and-a-half times more than the 1994 event and the most extensive use of the system at a Winter Games so far.

For 16 days, no matter where you were, it was virtually impossible to avoid at least some of the 68 events in the seven sports that now comprise the Winter Olympics, with curling, women's ice hockey and snowboarding making their official debuts at Nagano. Despite the cheap jokes at its expense, curling became one of the TV highlights, particularly late at night when its near hypnotic qualities no doubt soothed many a viewer off to sleep.

An event of this size and complexity, with a potentially huge worldwide audience and a large number of sports spread around several venues, will stretch any broadcaster, and in these circumstances the overall co-ordination of coverage is dealt with by a designated body. This, the Olympic Radio and Television Organisation (ORTO), is a division of the event's organising committee (NAOC) and is usually headed up by the national public service broadcaster of whichever country is hosting the Games: in this way ORTO 94 at Lillehammer was largely staffed by NRK (Norwegian Broadcasting Corporation) personnel. This is in keeping with the convention of international broadcasting events revolving around a host broadcaster, which arranges facilities and equipment.

Large-scale outside broadcasts will stretch even the biggest broadcaster company and bringing in independent mobiles and scanners is not unusual. An event of the magnitude of the Winter Olympics pushes even these extended resources and so ORTO 98, primarily representing NHK, followed the lead of ORTO 94 and contracted other countries' broadcasters to concentrate on some of the sports. This produced the vaguely confusing situation of Canadian, Finnish and British companies being 'host broadcasters'—producing the pool feeds—on certain events, in addition to organising their own coverage for consumption in their home countries.

This arrangement reduced—slightly—the pressure on NHK, which, despite its stature, is a relatively small organisation, and put certain events under the direction of broadcasters with proven expertise from a high proportion of coverage on their domestic output. This explains the involvement of Finland's YLE on the biathlon and CBC of Canada, the country credited for 'inventing' ice hockey, covering some of that sport and the debuting curling, another event it is dominant in. Less obvious is the reasoning behind the BBC being host on the bobsleigh and the luge, although the Corporation had distinguished itself at Lillehammer, winning the Silver Rings Award for best host coverage, while this year the Great Britain 4-man bob team won the Bronze Medal, the country's only gong of the proceedings.

Each single event, and its relevant host broadcaster, were housed at the various venues around the Nagano Prefecture: the White Ring for figure skating and short-track speed skating; ice hockey slammed into action at Big Hat; bobsleigh and luge took place at the Spiral on the Iizuna Kogen Heights; alpine downhill skiing, super-G and other downhill events on the Happo'one slopes; the M-Wave Arena for speed skating; ski jumping at the Hakuba Stadium; cross-country skiing on the Snow Harp fields; and super-G, giant slalom, slalom and snowboarding on Mount Yakebitai, a peak in the Shiga Kogen Heights.

Core to ORTO's coverage of these Games was the International Broadcasting >>>>>

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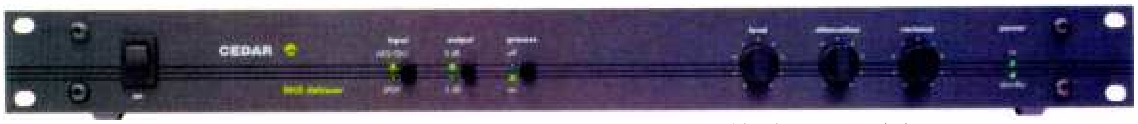

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<<<<< Centre (IBC), equipped by Matsushita-Panasonic in its capacity as both a sponsor and official broadcast equipment supplier and systems integrator. This is the third consecutive Olympics that the Japanese conglomerate has filled this position, having being heavily involved at both the Atlanta (1996) and Barcelona (1992) Summer Games. This year Panasonic supplied a range of its products, including the digital DVCPro. D-3 and D-5 video formats and digital cameras. The IBC featured 120 cameras, 630 VCRs and 2,000 monitors, while 250 DVCPro systems were loaned out to international film crews.

Routeing and switching are crucial to any broadcast event, particularly one of this size. Panasonic selected 14 Pro-Bel routers and a System 3 controller to handle all audio and video feeds originating in the IBC for worldwide distribution. The routers, which, like the control system, were backed-up with dual power supplies and control boards, were the HD and TM series, configured as 128x16 analogue video and stereo analogue audio in (with an extra 48x16 stereo audio); 96x64 for distribution of analogue video and stereo analogue audio (plus 48x32 stereo audio); and 64x16 analogue video and stereo analogue audio, plus 32x32 serial digital video, for transmission.

Further routing, switching and processing equipment, plus A–D and D–A convertors, were supplied by Leitch Technology. The IBC featured Digibus products to process signals, while the company's Digital Glue converter units and Sync Pulse Generators were also used to carry signals from the venues to the IBC for global distribution.

The events at the Winter Olympics can pose problems for producers, especially with such high speeds disciplines as race skating and the bobsleigh. For some of these digital technology was used to enhance picture quality and produce blur-less images.

What was hailed by Panasonic as the world's first digital slow-motion camera was used on the figure skating and ice hockey, with 20 units placed at the White Ring and Big Hat. In a sport like figure skating, being able to see every twist and sequin of a jump is vital to fans and TV commentators alike and this system enabled the split-second flash to be lengthened and analysed.

Specific sports required specific technology or a particular application of existing techniques. NHK developed Linear Cam, which was used for the cross-country skiing. Designed to give more dynamic pictures, the Linear Cam is radio-controlled and suspended from two parallel overhead wires, allowing it to literally zoom along at speeds of up to 40km an hour. In designing its bobsleigh and luge coverage, BBC Outside Broadcast Production Resources found that it would be using 33 cameras to cover the tightly curving Spiral run. These split into four hand-held Sony BVPT70s at the starts and the Weight House, eight whip-pan units to capture the near-blinding approach and pass of the contestants (a split variant of the BVPT70), eight fixed, operatorless BVP550s, six tall rostra cameras (BVP700s and 901Ts), five cameras on overhangs looking up the track (BVP700s or Panasonic AQ-2000 Super Slo-mos), one tripod unit, a tracking camera and an airborne balloon-cam. This amount of cameras meant that there were a high number of cuts, some as short as 1.5 seconds between each shot or angle.

OF THE MANY world broadcasters attending the Games, CBS's presence was among the most extensive. As it was the exclusive American network television coverage provider, it built its own TV studio inside the Ze Ko Ji Temple. Among the postproduction equipment were a Quantel Henry V8 effects editing system, four Picturebox still stores and three Paintbox Bravo graphics suites, plus 11 Chyron INFINIT! graphics systems. The complex featured nine video editing rooms, all of which relied on a single audio post studio for over-dubbing and sweetening.

This facility was provided for CBS by New York audio-for-video postproduction specialist Howard Schwartz Recording. Based around a 32-input Harrison TV950 console, with Tascam DA-88s and CD players, Sony time-code DAT, 360 Systems Digicart II, and a SSL ScreenSound V5, the room was built in the US and then shipped to Japan, where HSR chief engineer Marty Newman oversaw its installation.

Three operators went over from America and worked 12-hour shifts to cover the

24-hour a day demands of the Olympics. While there was extensive equipment available for full sound postproduction, tight turnaround times dictated how involved this could be. With most of the footage coming in on DigiBeta, much of the audio post work was mixed straight back to the VTRs' empty track.

CBS also shipped over two of its OB vans, each featuring a Euphonix console. One covered the Opening Ceremony at Nagano Sports Park before moving to the Habuka ski jumps and then back for the Closing festivities; the other was used to mix sound effects on the downhill ski courses. Euphonix was widely used during the Games: all Olympic broadcast audio for the Australian market was provided by Network 7 through a Euphonix CS3000 system based in the IBC, while the brand featured heavily in NHK's coverage, with five being used in total.

Three portable desks were used during the downhill ski events, while a NHK mobile was present at the Opening and then worked for ORTO during the rest of the tournament. The fifth Euphonix was part of the Hi-Vision forum, an independent facility within the IBC established by NHK and five Japanese commercial TV stations. This coverage also involved 100 Hi-Vision cameras and nine OB trucks. Hi-Vision, along with Linear Cam, was another example of new technology being implemented by NHK for these Games. On the audio side, the broadcaster's R&D department developed an ice-zone microphone, designed to capture the sounds that help make such events as skating so exciting. The microphones are set into the ice itself and are sensitive to both high and low frequencies, which travel through the ice without interference from extraneous noise like cheers or shouts. As it is disc-shaped and only one centimetre thick, the mic does not disrupt the surface of the ice and is not susceptible to its conditions as it is housed in a water-proof casing that can withstand pressures of up to eight metric tonnes.

The fourth NHK innovation takes the concept of fantasy sports—great athletes being pitched against either figures from the past or contemporaries who compete at different times—and makes it virtually possible. Virtual Competition is based on the use of computer-controlled cameras, with each competitor's run being filmed with identical camera control timing. An image processing technique is then used to 'cut out' each competitor's image, which is then recorded in a video file. This allows two or more of the stored images to be retrieved and multiplexed for simultaneous display, creating a virtual race on a TV screen.

Each passing Winter Olympics brings innovations such as these and each is hailed as having the largest number of hours of coverage since the last one. This was undoubtedly true of Nagano but despite the technical and production expertise of all the broadcasters involved, surveys have revealed low viewing figures for the whole event, prompting some to question whether such sports as curling, ice hockey and even downhill skiing have the pulling power programmers thought they did.

One thing is for certain—this Games proved that there is something more ludicrous than the luge, when one person goes down the side of a mountain on a baking-tray: the two-man luge, when two people go down the side of a mountain on a baking tray.

Host broadcasters during the Nagano Olympics

NHK (Nippon Hoso Kyokai)
Men's/Ladies' Downhill, Ski Jumping, Cross Country, Opening Ceremony
NTV (Nippon Television Network)
Speed Skating, Half Pipe
TBS (Tokyo Broadcasting System)
Figure Skating, Short Track, Closing Ceremony, MPC
CX (Fuji Television)
Slalom, Snowboard, Giant Slalom
ANB (Asahi National Broadcasting)
Freestyle Skiing, Giant Slalom
TX (TV Tokyo)
Ice Hockey B
BBC (British Broadcasting Corporation)
Bobsleigh, Luge
YLE (Finnish Broadcasting Corporation)
Biathlon
CBC (Canadian Broadcasting Corporation)
Ice Hockey A, Curling (3 feeds)
Nagano Local TV Station Consortium
Victory Ceremony

Quick work

Quick work on big artists established Elliot Scheiner's reputation as a producer. **Richard Buskin** tracks a fast-moving career

'I DON'T LISTEN to my old records anymore,' says veteran producer-engineer Elliot Scheiner, discussing a career that stretches back to 1967. 'I can't go back that far. It depresses me. It's just too long ago. I mean, it doesn't feel that long ago but I know it was, and so I try not to think about it.

'I have to say, however, that I had a great time making records back then. It was so much fun. Generally it was more fun than now, because everybody was live. There were so few overdubs. We made records very quickly.

'The primary thing was the music and not the sound of it. We went for as good a sound as we could get, but nobody worried about that. Everybody was just concerned with the music: "Did we get the take? Did we get the performance?", and that was an approach that I could really relate to.'

It still is, as reflected in some of the projects that Scheiner has been involved with of late: The Eagles, Fleetwood Mac, John Fogerty, Steely Dan, as well as a slew of other, lesser-known live-performance bands. Resident on America's East Coast, Scheiner often finds himself working out West with many of the aforementioned artists, and it was while he was involved in a Fogerty mix session at Capitol in Los Angeles that I caught up with him for this interview. We'll get to the recent work in a moment, but first here's a brief recap of the man's illustrious past.

Elliot Scheiner started out as an assistant to producer Phil Ramone at the latter's studio, A&R Recording, located on 48th Street in New York City. There he remained until 1973, during which time he learned how to cut discs and how to work with film on his way to becoming a fully fledged engineer.

'They believed in well-rounded engineers in those days,' he now recalls. 'A&R was a full-service facility and back then people didn't make tape copies, they had reference discs cut, and so you had to know how to cut a disc. On top of that there was a lot of film work being done, and so you also had to know how to deal with things such as magnetic stripe, not to mention learning how to mic practically everything that came into the studio.' At the start of the seventies A&R Recording was equipped with a relatively new 32-input 16-output Neumann console.

'By that time we had in-board EQ, but there was nothing beyond that,' says Scheiner. 'There were no in-line compressors and there were no gates or anything like that. There had already been a console with all of that in-line, but this one just didn't have it.'

While the control room measured about 18ft x 15ft, the recording area was wrapped around it in an L-shape and measured about 40ft x 20ft, with an additional 20ft x 20ft at the tail of the L. Fabric covered all of the walls, there was carpeting on the risers and in the vocal booths, a composite was used for the floor in the basic part of the studio and the ceilings were decorated with acoustic tiles.

'Back in those days they built rooms as much for appearance—and maybe sometimes more for appearance—as they did for sound,' Scheiner muses. 'In the case of this particular room, however, I think they lucked out, because the sound was good.'

In the fall of 1969 much of Van Morrison's *Moondance* album had been recorded in this room using a Scully 8-track machine, whereas the subsequent *Van Morrison, his Band and Street Choir* sessions upgraded to 16-track. The monitors were Altec 604 Es with Mastering Lab crossovers, and then in terms of the effects... well, there were none.

'In that room we had three EMT 140s,' Scheiner recalls. 'We used an analogue tape machine to delay the send to the echo chambers and that's about all we had. I mean, there might have been a Cooper Time Cube and there might have been an old Eventide digital delay, but that was it. Whatever processing you did, if you were going to flange something you used machines for it. That's what I ended up doing on *Band and Street Choir* and I remember using a couple of different machines to do it, but the primary outboard gear consisted of echo chambers and delays and that was all we used.'

Van Morrison had technically been the producer of *Moondance*, but when the time for mixing arrived he wanted to return home to Woodstock in upstate New York for Christmas. Thus he asked Elliot Scheiner and drummer Gary Malabar to take care of the mix and then send him copies of their work. This they duly did, now prompting Scheiner to ask, 'What does a producer do?'

Back in 1970 he thought he was about to find out, for on *Van Morrison, his Band and Street Choir*, Scheiner was supposed to be coproducing with the artist. During the course of the project, however, the two men had a disagreement and so the task fell to Morrison and his new drummer, Daud Shaw, while Scheiner ended up being intriguingly credited as 'Production Coordinator'.

'Back then we didn't know enough and we weren't discerning enough,' he says. 'You know, we were making rock 'n' roll records and obviously we'd listen to the sound of these records, but there really wasn't much thought put into it because we were limited as to what was available in the studio. You just went in and did it. There was no such thing as renting mics from rental companies.

'That wasn't done. You worked with what you had, and if that was what the studio owned then that was what you used. You put anything else out of your mind.

'Mixing amounted to balancing, EQ, reverb and echoes. Everything was always cut dry. We didn't necessarily cut it flat—we'd use compression and EQ while we were cutting —but we never used reverb when we were cutting. When you hear one of those recordings now you go: "Wow, man, there's nothing on this record!" But, you know, you forget that we really didn't put much on it. We were so unaccustomed to echo and reverb that when you put some on it sounded alien, because that's just not the way instruments sounded. You didn't hear that stuff live, there was nothing sophisticated with regard to effects and in general people just wanted their instruments to sound on record the way that they sounded in a room, and so that's what you went for.

'Things like double-tracking with the Eventide we used only very occasionally, hardly ever. Personally I was far more into flanging stuff. Apart from that, if we wanted an effect going for live stuff we'd sometimes employ a room mic sparingly, or I'd face a guitar amp into a piano and then pick up the harmonics off of the strings. Doing stuff like that we thought was very arty... it turned out to be a crock of shit. People couldn't hear it anyway. You'd say, "Oh, you know what I did here?" and they'd go, "What?", "Oh, really?" Nobody cared, but it was just a case of who could be cooler than the next guy, and in that respect I think that the English were definitely more adventurous than the Americans. I remember the first time I heard Elton John's records over here, I thought, "Geez! How did they record those strings and those drums?" It was unbelievable, and what it turned out to be was the difference between the CCIR curve and the NAB curve. Because when I eventually went over to England and worked in a studio I thought, "Well, gee, this doesn't sound so great," but when I brought it back to the United States and played CCIR-recorded stuff on an NAB curve it was a totally different thing: a phenomenal sound. You have to remember that in those days even the cuts didn't make the sound great. You know, we went for a vibe and we cut only when something was really bad. So, if we liked the body of a take and there was one section which we weren't at all happy with, we'd try to cut it in. We'd look for a take that had the right part and just try to edit it. You definitely could punch-in back then—you couldn't punch-in in the middle of a piano part, but we were pretty good at vocals. We wouldn't even attempt punching-in single syllables, but we'd punch-in a word or two—and pray. Those machines were slow getting out.'

It was during the mid-seventies that, with a solid apprenticeship under his belt, Elliot Scheiner took the then-innovative step of going freelance as an engineer in New York City. 'As part of the A&R studio staff I had been bringing in more than a million dollars a year in business for them,' he explains. 'That was a phenomenal amount of money. Meanwhile, they were paying me $40,000 and I thought, "Gee, this isn't right." So, I said, "Look, I don't want to do this anymore. I want a commission for all of my clients," and they said, "Sorry, but no". I then called all of my clients and told them I wasn't working at A&R anymore, and everybody cancelled their sessions, so the studio called me back and said, "Okay, what do you want?" I said, "I want 20% of everything," and they said, "Okay," but when my cheque for the first week was like $5,000 or $6,000 they called me back and said, "This is not working." So we ended up with a deal comprising 20% on time and 15% on tape, and that worked out pretty well.

'However, when other guys saw what I was making they all decided to do the same, and that was the beginning of the demise of the staff engineer.'

The reciprocal agreement with A&R lasted until the start of 1977, by which time Phil Ramone had sold his interest in the studio, and Scheiner then began to spread his wings further afield. That year he won a Grammy Award for Best Engineered Recording with Steely Dan's *Aja*, and in 1981 he scooped the same prize with the same band for *Gaucho*. During the ensuing years, however, even though there have been nine further Grammy nominations, the awards themselves have eluded him. In 1998 these included ones for Best Engineered Album, Non-Classical (Dave Grusin's *Two For The Road*) and Best Pop Album (Fleetwood Mac's *The Dance*, which he coproduced with the band).

'I'm getting used to it,' he says in a resigned tone. 'Still, it's pretty good recognition; although I have to admit that there were a couple of years when I thought I definitely should have won. Whatever. Life's good. I'm not complaining.'

Other engineering-mixing clients have included Aerosmith, Bonnie Raitt, Barbra Streisand, Billy Joel, George Benson, Natalie Cole, David Sanborn, Luciano Pavarotti, Ricky Lee Jones, Smokey Robinson and Dan Fogelberg. At the same time, since 1978 production has also been a priority for Scheiner, and his credits in this field take in The Eagles, Mac, Fogerty, Glenn Frey, Jimmy Buffett, Bruce Hornsby, Donald Fagen, Toto and The New York Rock & Soul Review (featuring Fagen, Boz Scaggs and Michael McDonald).

'My career was doing fine, but the call from Glenn [Frey] and Don [Henley] to do The Eagles' MTV Special really changed everything,' he says. 'When that thing took off I got a lot of credit for it...' including an Emmy nomination for Outstanding Achievement in Sound. I won't bother to tell you the result; it's too upsetting.

'It just happened to be circumstance,' he continues regardless. 'They played great, it looked great and it sold a lot of records. The shoot took place on the soundstage at Warner Brothers in Burbank, using Le Mobile, >>>>>

Van Morrison: to produce or not to produce, that was *Moondance*


Fleetwood Mac: loving the *I Love Lucy* soundstage

<<<<< and we then moved to The Village Recorder [in Santa Monica] for editing as we had to chop between two nights. Le Mobile houses an old Neve 8068 and it's small: it's like 36 or 40 inputs, and I was using 76 inputs excluding the string section.

'With the string section it was well over a hundred. We did it analogue, 15ips Dolby SR, and; although it sounded really good in there it was a little bit of a nightmare because the truck was so small. You know, I was surrounded by outboard mixers, and it was a hassle trying to get everything on tape while doing a live mix at the same time for these guys to listen to.

'They were playing on the old *I Love Lucy* soundstage—it's the biggest one at Warners—and it's a great-sounding room. I've since gone on to do Fleetwood Mac there and it's wonderful: very controlled, so we were able to whatever we wanted... had we wanted to do anything.'

As for the mix sessions, these took place both at The Village Recorder and in New York while the band was embarked on its reunion tour. However, with new songs in the offing there was a desire for these to be studio tracks rather than live cuts, and, as Scheiner was preoccupied with the MTV project, Henley brought in his own man, Ralph Jacobs, for *Hell Freezes Over*. As things turned out, with Jacobs and Scheiner each in separate rooms at The Village, they basically ended up helping each other. Hence Scheiner's coproduction and engineering credit, not to mention the pair of Grammy nominations that he received for *Hell Freezes Over*... Oh well, never mind.

'We didn't win anything but I started getting calls to do these kinds of projects,' he says. 'Fleetwood Mac had seen the Eagles show and said, 'Well, why don't we work with this guy?' Then, John Fogerty saw the Fleetwood show and said, 'That's the guy I want.''

While Scheiner had previously worked with individuals from both The Eagles and Fleetwood Mac, he'd never collaborated with the actual bands before. Now he had the perfect opportunities and he didn't waste them. 'Having already worked with Steely Dan these projects enabled me to complete a trilogy of the Southern California seventies scene,' he says. 'It was great. I had a great time doing the Fleetwood show. I was brought in just five days before they were due to shoot it, and by then they'd already hired the truck, so again I found myself working in Le Mobile.' For the John Fogerty TV special Warners again only hired Scheiner a few days before the shoot, and so guess which truck he ended up using. 'I'm really enjoying doing this,' he states, 'and now Warner Brothers wants me to do four more of these projects this year. I don't know who they're with, and, although I've been concerned about getting out of the mainstream a little too much, I'm having such a good time that it's hard to refuse.

'With John the recording setup was pretty straightforward as he was playing as part of a small 5-piece band, but with Fleetwood and The Eagles there were a lot of sidepeople and so those were sort of difficult, especially with the marching band that Fleetwood used for *Tusk*. In that instance I was originally going to try and do coverage around the room as they march in, but we then decided that they really wouldn't play until they'd get on stage as otherwise it would be too much of a nightmare.

So, as they came marching onto the stage I had mics set up at the back as well as three mics in front for the drummers. It was wide coverage and it wasn't great, but there really wasn't much of a choice. There was no other way to get around the situation. I wasn't going to individually mic 80 people.'

During the course of our interview at Capitol in LA Scheiner touches on his other recent album projects, such as those with Natalie Cole (*Stardust*), Toto (*Tambu*) and Dave Grusin (*Two for the Road*), and all share several things in common. For one thing, they were all nominated for Grammys, for another

they all enjoyed the same fate, and for yet another they all saw Elliot Scheiner sharing the engineering duties with—among others—his longtime friend and collaborator, Al Schmitt, whose string sound Scheiner enthuses about.

'I like having his name on my records,' he says. 'It's important to me. He's the king. I learned from Phil [Ramone]; he was my mentor, but Phil doesn't do it anymore. Al's still doing it, he's still in the thick of it, and I hope I can look like he does when I am his age.'

Timely words, being that, at that precise moment, who should walk in from the next-door studio, but the man himself; ladies and gentlemen, Mr Al Schmitt.

'He was asking about our relationship,' Scheiner tells Schmitt. 'Oh sure, he'd like to be my boyfriend,' Schmitt retorts. 'We hold hands whenever we get the chance!'

Joking aside, the two men, together with producer Ed Cherney, are starting their own label, as yet unnamed. 'The three of us will produce a few projects a year for the label,' says Schmitt. I ask if they have a distribution deal.

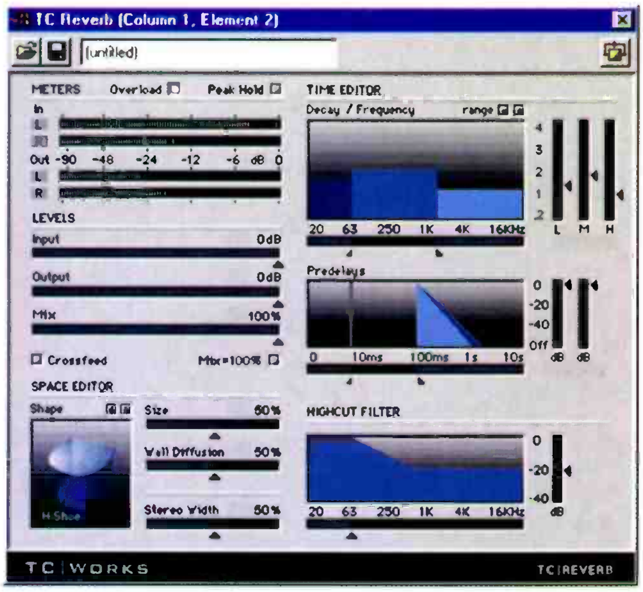
'That's almost locked up,' responds Scheiner, who, in addition to the aforementioned prospective sessions for Warners, is working at Clinton in New York on a new Steely Dan album and scheduled to mix three old Eagles albums in 5.1 Surround.

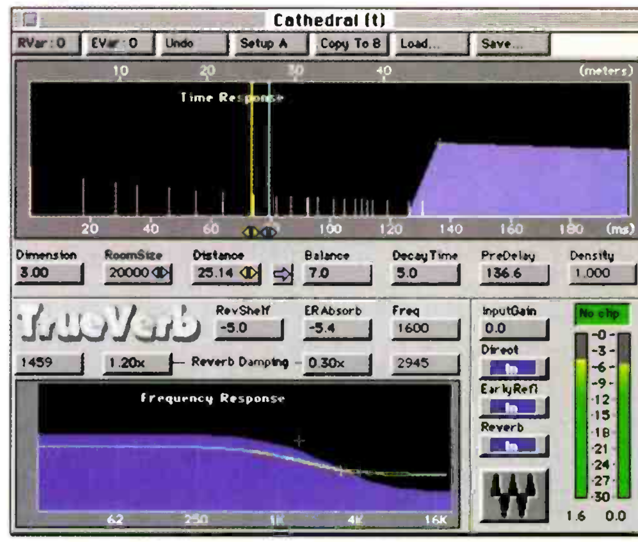
'Why, have you got somebody?'

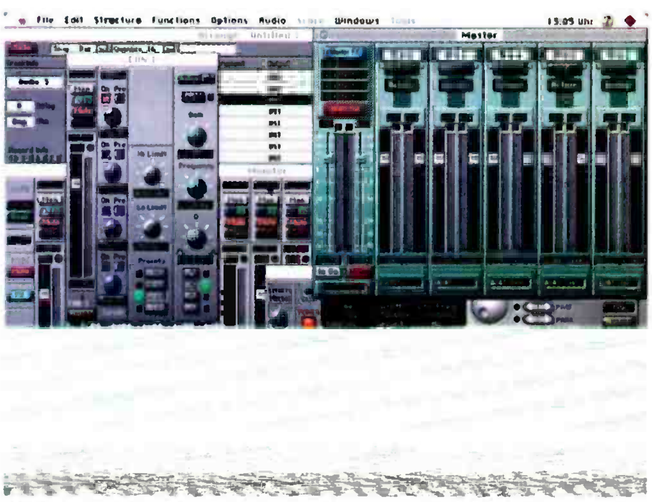
The KRK Connection

ELLIOT SCHEINER may well love the KRK E8 monitors that he now carries around with him from session to session, but he wasn't always overly enamoured with that company's product. 'Keith Klawitter [KRK Systems president] was trying for years and years to get me to use his speakers,' Scheiner asserts, 'and every time he'd send me a pair I'd listen to them and think, "This guy's dreaming." Let's just say that it was a different sound from what I was used to and I basically stuck with the NS10s. Then he sent me a pair of E7s when I did a live Westwood One show with Fleetwood Mac back in October, but I was ready to go on and so I wasn't about to do a mix on a pair of speakers that I didn't know. I did, however, use them a little bit during that weekend and I thought, "Well, these aren't anything either." So, Keith said, "I've got another pair that I want you to try." Those were the E8s, and he flew to New York for the very first morning of the new Steely Dan album project in November, he brought them in, and they sounded remarkable to me. Walter [Becker] and Donald [Fagen] came in and they normally use Meyers, and one of the guys in the studio had a copy of *Aja*, so we A-B'd *Aja* between the E8s and the Meyers and there was no comparison. The E8s just blew them away. You can crank these things and they don't distort, and you can listen off-axis and they're okay. I'm amazed at what these speakers can do, and they've now basically replaced the NS10s for me. They're non-fatiguing, the relationships are very accurate, and I can listen to them all day long.'

Plug 'n' play







Latest generation MIDI + Audio sequencing packages are becoming ever more flexible and sophisticated thanks to a burgeoning market in software plug-ins. **Simon Trask** plugs in

GREATER DESKTOP COMPUTING power and larger data-storage capacities combined with falling prices are making computer-based, multitrack, digital-audio recording a practical proposition for a growing number of musicians. And, increasingly, it is the MIDI sequencing package developers who are benefiting.

For the many musicians who have grown up with the likes of Cubase, Logic or Cakewalk, and who live and breathe the computer-based MIDI sequencer approach to music-making, having multitrack digital audio recording capabilities built into a multitrack MIDI sequencing environment is a logical progression. This is, of course, exactly what computer-based sequencer developers like Steinberg and Emagic have been developing over the last few years; although typically with separate MIDI-only and MIDI + Audio packages which have required users to pay a premium price for audio capabilities.

This situation is now changing with, for instance, Emagic revamping its cross-platform (Mac-Windows) Logic product line to provide integrated MIDI and audio capabilities throughout the range, beginning with the entry-level MicroLogic AV (at £99 in the UK) and moving up through Logic Audio Silver, Gold and Platinum. These four upcoming packages will offer progressively more audio tracks, editing capabilities and software effects buses through the range, along with progressive support for more sophisticated hardware—the Platinum version supports 24-bit Pro Tools in its Mac incarnation and Soundscape's SSHDR-1 in its Windows incarnation, allowing it to act as a front end to these systems. Digidesign worked with the sequencer companies from the outset of the nascent MIDI + Audio software to ensure software and hardware compatibility, with the result, nowadays, that all four major software packages for the Mac platform (Emagic's Logic Audio, Steinberg's Cubase Audio VST, Opcode's Studio Vision and Mark of the Unicorn's Digital Performer) support Digidesign's TDM hardware. As of writing, Steinberg has announced that it, too, will be supporting Pro Tools 24 format, with the upcoming Cubase Audio VST 24, which will replace the company's current flagship package, Cubase Audio XT.

These liaisons have proved important in another, related, audio development. Along with the increasing power of computer hardware and increasing sophistication of MIDI + Audio software has come the rapid growth of the software audio plug-in market, which in essence has integrated the sort of effects capabilities previously only found in 'stand-alone' outboard gear into the computer-based digital-audio recording environment. Digidesign can take credit for having introduced the plug-in concept to the pro audio world, first of all with Sound Designer and, then, with Pro Tools. However, it is a concept that has spread to MIDI + Audio software through the MIDI software companies supporting the various Digidesign plug-in formats (details of which, to its credit, the company has always made available to third-party developers). These companies have also developed their own plug-in formats, though not all with the openness of Digidesign; Steinberg has been foremost in making its VST plug-in format broadly available, with the result that Cubase Audio VST has built up the most impressive and wide-ranging collection of audio plug-ins outside of the Digidesign formats.

The plug-in concept is a powerful one because it harnesses the skills and enthusiasm of a broad range of people around a core 'project'—the main software package. The company responsible for developing that package benefits because it gets enhanced functionality without necessarily having to do the work itself, third-party plug-in developers benefit because they have a ready-made market they can, erm, plug into, and last, but not least, customers benefit because they get more of the features they want sooner than they would if it was all left up to the main developer. The plug-in model could be labelled 'selective openness', in that companies provide selected hooks into their proprietary code; this way they retain control of their commercial property while also providing limited access to it. This approach also allows them to control what aspects of their programs can be altered or augmented. If Steinberg, for instance, provided hooks into all the colour settings for the graphical layout of Cubase, you can bet someone would come up with a plug-in that would let you have a purple virtual mixing desk fascia and an orange Record button if you wanted. As it is, the existing plug-in formats in the computer-based Digital Audio Workstation and MIDI + Audio worlds allow third-party developers access to the internal digital-audio streams, while leaving the decisions on how to process the audio up to the imaginations and commercial acumen of developers.

Some of the established names in outboard effects processing have climbed aboard the audio software plug-in bandwagon in recent years (you have read the reviews of their plug-ins in the pages of this very magazine), names like Aphex, Drawmer, Focusrite and tc electronic. But equally there have been new names entering the field, such as Arboretum, DUY, Prosoniq and Waves, software-only companies rather than more traditional hardware manufacturers. The range of audio processing capabilities that have emerged from the growing plug-in fraternity is truly impressive, covering not only the more traditional, familiar effects such as chorus, delay, flanging, phaser, reverb, EQ, auto-pan and wah-wah, but also venturing into such areas as crackle and hiss removal, vinyl surface-noise simulation, analogue tape-saturation effects, surround encoding, 3D spatialisation, spectral reshaping, and valve amp simulations. Some plug-in effects are available individually, others come as part of multieffects packages, with prices ranging from nominal through modest to expensive.

This brave new world of plug-in software effects may have its advantages in terms of convenience, flexibility and digital integration, but it it probably will not make life any less expensive for the studio that has to be flexible and well-prepared (for one thing, there is no hire-in model for commercial plug-in effects as yet). And, it has to be said, there are a lot of factors which need to be taken into account when considering plug-in effects that simply do not apply with stand-alone hardware units and standard analogue audio patching. If you buy or hire a traditional rackmount effects unit, you know that you can plug it into your patchbay and it will work with your setup. But once you enter the world of software plug-ins and MIDI + Audio recording packages you have to consider the compatibility of plug-ins with your computer-based recording system on a number of levels.

Most obviously, there is the matter of whether they run on your computer platform and operating system of choice: MacOS or Windows 3.1/95/NT. Some Mac plug-ins will run on both 68k and Power Macs, others on Power Macs only, while, for instance, the advent of OS8 for the Mac means that you need to be sure you get plug-in versions with the appropriate installers for the system you are running (7.x or 8.x). On the Windows platform, a growing number of plug-ins, but not all, will run under Windows NT. There are other questions you need to ask before handing over your credit card (or wasting time and money downloading free plug-ins). For a start, you can not presume that any plug-in you come across for your computer platform will be compatible with your recording package of choice.

WHILE SOME plug-in formats have entered into common use, others are manufacturer-specific and not supported by rival sequencer companies, either for political-commercial reasons or because the format simply has not been published. Emagic, for instance, has not published its own format, while, as mentioned ear- >>>>>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lier, Steinberg has made its VST format available. Both companies support Digidesign's TDM format, which requires Digidesign hardware in order to work —another issue to bear in mind when looking at plug-ins (some formats operate in software only, others require external hardware). Of the four new Logic Audio versions mentioned earlier, only the Platinum version supports TDM. Unusually, the format situation is more complicated on the Mac platform than the Windows platform, where Microsoft's assumed role as standard setter means that the only open format is the company's DirectX protocol—although there are several proprietary formats.

On the Mac platform, Digidesign's old Sound Designer II plug-in format still has some support from the likes of DUY and Waves, but it is worth bearing in mind that you will need an Audiomedia II (NuBus, 68k) or III (PCI, PowerPC) card, while plug-ins in this format are also typically available in one or more of the newer Digidesign Pro Tools TDM plug-in formats. SDII is also a non-real-time format, as are two of the other open formats, Adobe Premiere and Digidesign's AudioSuite—meaning that plug-ins available only in these formats are not suitable for live mixing applications (where plug-ins function as insert or master send-return effects in a virtual mixing desk), only for file-based processing. AudioSuite allows batch processing, meaning that AudioSuite plug-ins can work their way through any number of audio files that you give them.

If you want real-time processing, you need to look at VST and TDM on the Mac platform and DirectX on the Windows platform, along with plug-ins in the proprietary format or your system. A more recently developed real-time plug-in format is Mark of the Unicorn's MAS (MotU Audio System), which has so far only attracted third-party support from DUY. MotU's newly-announced 2408 MacOS and Windows PC computer-based hard -disk recording system will support third-party plug-ins in both MAS and Adobe Premiere format, and will also work in conjunction with the company's Digital Performer MIDI + Audio software package. The proliferation of Mac plug-in formats means that the major third-party plug-in providers find themselves, as a matter of commercial necessity, having to support multiple formats. This is good news for users, if something of a headache for the developers.

> How many simultaneous effects can your computer system support? How freely can you configure your effects? How quickly will the file-based effects work on your system? Will you need plug-in DSP cards to augment the onboard processing power of your computer?

The audio plug-ins market for MIDI + Audio software, then, is still relatively new, as is reflected in the profusion of formats. For pro users on the Mac platform, widespread Digidesign compatibility in the MIDI + Audio market is a definite plus, but given the wide range of users and associated price expectations in this developing market the sequencing software companies will need to ensure that a plug-ins market can thrive at a less 'rarified' level.

While Pro Tools TDM is the *de facto* common denominator for Mac packages, it is decidely not a lowest common denominator, as you need to use Digidesign's hardware as well. This is where Steinberg's cross-platform VST format comes in, as it allows native real-time processing on the computer. In a significant late-breaking development as this article was being finished, Emagic announced that the upcoming versions of Logic Audio will support VST plug-ins. Emagic and Steinberg are, of course, long-standing rivals, so the announcement is a first sign of a new maturity taking shape in the audio plug-ins market: it is also an acknowledgement that Steinberg got it right by publishing details of their VST format from the outset.

Of course, for users there are many issues other than which format to get. How many channels or tracks can you run your real-time effects across as insert effects? How many simultaneous effects can your computer system support? How freely can you configure your effects? How quickly will the file-based effects work on your system? Will you need plug-in DSP cards to augment the onboard processing power of your computer? These are all issues which spring from the computer/plug-in approach, but of course the small matter of the quality and suitability of these effects applies just as it does with stand-alone boxes. Here, one potential advantage of the software-based approach is that developers of commercial plug-in effects could provide time-limited demo versions for auditioning purposes, perhaps with the option for you to 'unlock' the limit once you have paid up and been e-mailed a registration number (a system already employed by some companies elsewhere in the software world).

All in all, 'MIDI sequencer-based' audio recording and its attendant audio plug-in expandability represent a significant new area of development in pro audio recording, it is an area which *Studio Sound* will be covering on an ongoing basis. ■

Open plug-in formats

Mac: Adobe Premiere
Digidesign AudioSuite
Digidesign Pro Tools TDM (III, 4, 24)
Digidesign Sound Designer II
MotU Audio System
MAS Macromedia Sound Edit 16
Audio Xtra Steinberg VST
Windows: Microsoft DirectX

Manic impressions

After a two-year hiatus, the Manic Street Preachers are back in the studio recording a follow-up to their acclaimed album Everything Must Go. **Tim Goodyer** talks to producer Mike Hedges and engineer Ian Grimble

THE BENCHMARK SET by the Manic Street Preachers' 1996 album, *Everything Must Go*, virtually ensured a rematch between artist and producer. And while the album as a whole benefited from the work of several production teams, it was Mike Hedges and Ian Grimble that graced the high profile hit, 'A Design for Life' as well as most of the remaining songs. To assume that their reappointment was a direct result of a success that Hedges terms 'a phenomenon' may be to miss the point, however, as the one thing the band has sought to avoid is a direct successor to their greatest success.

'They've done that, and this is another direction,' he elaborates, having spent parts of the last six months ensconced with them in his studio in northern France. 'It's a bit more introspective and, for me, has a lot more atmosphere. I think it's too easy to say, "that's a formula that does very well, lets have another big hit record" and this band don't think like that at all.'

Empowered by some 20 years in the recording game, Hedges periodically refers to the Manics' wish to develop, and its manifestation in every aspect of their work from song-writing and performance to their choice of equipment—a consideration that impacts directly on his job as producer and Grimble's as engineer. The drums that sounded so imposing on *Everything Must Go*, for example have been completely rebuilt.

'I think the drums are going to sound much better on this one,' Hedges confirms. 'It's a different kit and the drums sound better to begin with. Sean [Moore]'s playing better every day. And Ian's refined the way he records the drums.'

'We've gone for a different approach here,' Grimble agrees. 'The last album was quite ambient and this one's a lot drier and bigger because we deadened off the drum area and built bass screens around it to stop the bass going into the rest of the room...' ('old-fashioned style, real sixties,' chips in Hedges) '...so the actual kit sound is quite dead and we used the omni mics to add any attack that was needed. The mics were an SM57 and a KM84 for the snare—I used more of the 84 than the 57 and it works very well in this context. There's a very old D12 on the bass drum; BPM95s on the overheads; toms are all Sennheiser 41s, and for ambience I used MKH20s which sound terrific. It was basically the sound we used on "Yes" by McAlmont & Butler—it's the sound you get with those mics in that room.'

'The final 'Yes' sound was 90% MkH room omni and 10% of everything else mixed in with it,' Hedges adds. 'There's more of the EMI compression that's on the desk too.'

The desk in question is the famed EMI TG vintage affair that previously hosted Pink Floyd's *Dark Side of the Moon* at Abbey Road. Additionally, Hedges has a 24-channel model he takes with him when working in other facilities. 'The desk in France has 40 compressors built into it which is pretty amazing for a desk designed in the mid-sixties,' he observes.

'My major problem has been with some of the old keyboards we've used,' Grimble continues. 'We've used a Wurlitzer piano, Rhodes, Hammond M100, and a double-manual Vox Continental organ. Individually they may sound okay, but when mixed in with the others they don't, so we spent a lot of time getting them to work and stay in tune. But it sounds good for it. Very individual.'

The console is indicative—but not representative—of Hedges recording philosophy. The main multitrack is a Mk.1 Studer A80 2-inch, 16-track running at 15ips with Dolby A, while microphones can be selected from a sizeable collection containing many 'classics'. Yesterday's technology benefits greatly from the company of today's however, and a Pro Tools system neatly adds much of the flexibility common in more modern studios.

'The recording process is to try to catch the band as live as possible in as few takes as possible because they are a very, very good live band, as anyone who's seen them live will know,' explains the producer. 'And they create a very good atmosphere on a backing track. What we've been trying to do is capture the first possible take we can in the playing process so they're not having to overwork it in the studio. That way we can keep it as fresh as possible with as much excitement as possible. To do that we've been going for probably only three takes of a backing track—until we have one that we're 95% happy with. The first take is usually very good; then we'll do a couple more just in case.

'When you're doing backing tracks these days, people are very conscious of timing. Obviously the feel is important but we feel that people are too conscious of timing and they'll discard a take because there's a fill wrong in the second verse or there's a missed bass beat in the second chorus. To get rid of those problems we'll use Pro Tools—we'll go with the best take, so it's a really good, fresh, exciting take, and then either repair it with Pro Tools or take it from one of the other takes we've got. That way you've kept it pretty much as they've played it live.

'We use a lot of Pro Tools but we always use the 16-track as the front end because it's an amazing-sounding machine. Using a lot of analogue as we do, I think you have to record everything digital or everything analogue. If you record on analogue with digital bits, when you come to put them together it's like a square peg in a round hole.

'For the mix we occasionally run things live from Pro Tools, but not generally. Generally, everything is compiled on the 16-track and a 24-track running at 30ips—so we've got a 40-track master; although typically I'd say we end up using about 22 tracks.

'There's a tight line between going for that so-called perfect take and yet keeping the freshness of what the band are playing. After seven or eight takes, or nine takes, or 10 takes you can get a take where all the fills are right and the timing's right—everything's right—but then when you compare it to an earlier take, the earlier take always has something. It's that "first take syndrome". This happens with nearly every band that exists, I think.'

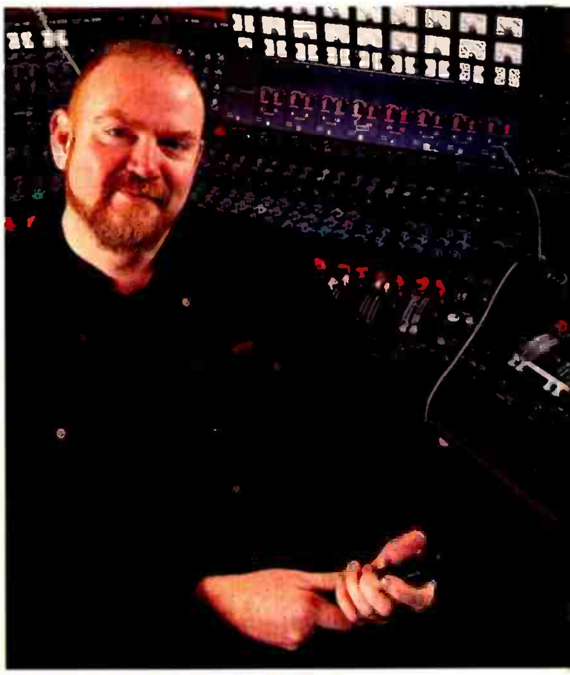
THE BAND ARRIVED at Hedges' Chateau de la Rouge Motte studio with the songs written and the arrangements largely in place, having written a couple of final tunes upstairs from the main session on *Everything Must Go*.

'We had demos of a few of the songs but with some of them we just got a play-through,' Hedges recalls. 'That's normal with them—the demos are in their heads. The arrangements were not set in stone but most of them were already done. They're very conscious of what makes a good song; they're not the sort of band who go off into little jams and lose direction. James [Dean Bradfield] is probably the most concentrated on the arrangements because he's the singer—he knows how he wants the vocal to sit with the track.'

The backing tracks consist of the drums, bass, guitars, keyboards and guide vocal, and contain the essential atmosphere of the songs including much of the signal processing.

'I like things with their own space,' Grimble explains. 'If it's recorded with a reverb and delay or whatever you're using, it's got its own sound and its own space. A lot of the sounds are quite dry, quite up front. If we are going to use reverbs and delays and things we tend to decide on the sound beforehand and stick them through the amp. We might use a spring or a Quadraverb, for example, but we put it through the amp.'

'It sits pre the amplification, which is quite a different sound to putting it on afterwards,' says Hedges. 'The Copycat's used quite a lot, but again you'd record the sound >>>>>



Mike Hedges with Sennheiser MKH 40 and EMI TG console

Cupboard love

'IN THE STUDIO in France we've built three "cupboards" that you put amps inside,' says Hedges. 'They're about 1.5m by 1.20m deep—two of me could sit inside one of these little cupboards. They're very soundproof and have different surfaces inside: stone, absorbent and semi-absorbent. Using them, we can put guitars and bass and keyboards in the same room and we don't get any spill at all. They're good for overdubbing as well—especially the absorbent one. It gives a kind of presence because you're not getting any room on the close mic at all, and so it sounds right in your face. The pre-delay time is very short—shorter than short—otherwise it would be a problem. Once you get above a certain size you can hear a "box" in your mix, and there's nothing worse.

'On this project we used Sennheiser MKH40s for the closer mics in there, especially when it's very loud, because quite a few of our mics wouldn't take that sort of battering. With the guitar going full blast through an amp with the doors closed, it's nasty, but you can put an MKH40 an inch or two from the speakers and they don't "squash" very easily. We were finding with the guitar sounds we'd set up in France using MKH40s, we would then come to somewhere like Abbey Road and go through a whole set of different mics and they just wouldn't sound as good. The MKH range is very high gain and really pure, they don't have any nasty little peaks in the middle. We used BPM94s a lot for the distant mic in the cupboards.

'The stone box is a very irregular surface and a very strange shape. It used to be an oven—it was a doorway originally but it got bricked up and they put a great big oven in it to heat the house. The oven was scrapped and we ended up with a hole with 3ft-thick walls and the chimney.'

<<<<< through the Copycat, rather than add it later. It's part of the recording process. The mix progresses as the recording progresses. With most of our clients, unless everything sounds great all the time, you're in trouble.'

'It's partly determined by the studio as well,' Grimble continues, 'because it's a 16-channel desk with one send per channel, so when you come to mix its nicer to have everything down—if not on its own track, maybe with the effects on their own track.'

With the backing track caught complete with much of its final treatment, minimal overdubs remain to be done.


Engineer Ian Grimble: 'I like things with their own space.'

'When we're recording keyboards we don't just DI it into the desk,' says Hedges. 'We'll be using Leslies or amps to get room sounds. Occasionally all the sounds work with one setup, but not very often. Usually there's one sound that works great but there's usually one that you don't want to go through that particular amp in that particular room. It's exactly the same for the guitars.'

'If there are a lot of sound changes, for example if you've got the keyboards playing three sounds in the song, it's very difficult to get all three sounds just right to be able to keep all of them, so we usually end up concentrating on one of the sounds and being able to keep that, and then overdubbing the other two using the others for guides. You tend to optimise for the main thing.

'So there are overdubs on keyboards, extra guitars and sometimes the bass. Sometimes you redo the bass, or you may decide you want to change the bass sound or the bass guitar itself at a later stage. The performance is important and that is the decision you have to make; can I keep the feel of the original performance or is the sound going to make a dramatic difference, because it is a very, very tight relationship between the bass and the drums. And it depends where in the song it's placed. If the bass is a major element in the song, it is much harder to replace than if it's "sunk in".

'The hardest part has been that the rough mixes have been so easy to do that they've come back and said "perfect but could you just make that guitar a bit louder in the second chorus", and then we've got to try to copy the rough mix exactly—which takes hours. And we'll be doing it with the old console and the SSL at Abbey Road side-by-side.'

The new Manic Street Preachers album had no title at the time of writing, but with a September release date. Hedges expects it to do 'very well—it's a very strong album'. Asked whether he expects to be involved in a third collaboration with the band, he is surprisingly cautious.

'I don't know. I think maybe two albums is enough. If they ask me next time I'd be delighted to do it, but things must progress. We've got exciting things up and coming and it doesn't hurt to progress, it really doesn't.

'I'd be delighted to do another one—I'd love to—but by the time I've finished this I'll have been working with the band for three years. Which is a long relationship.' ■

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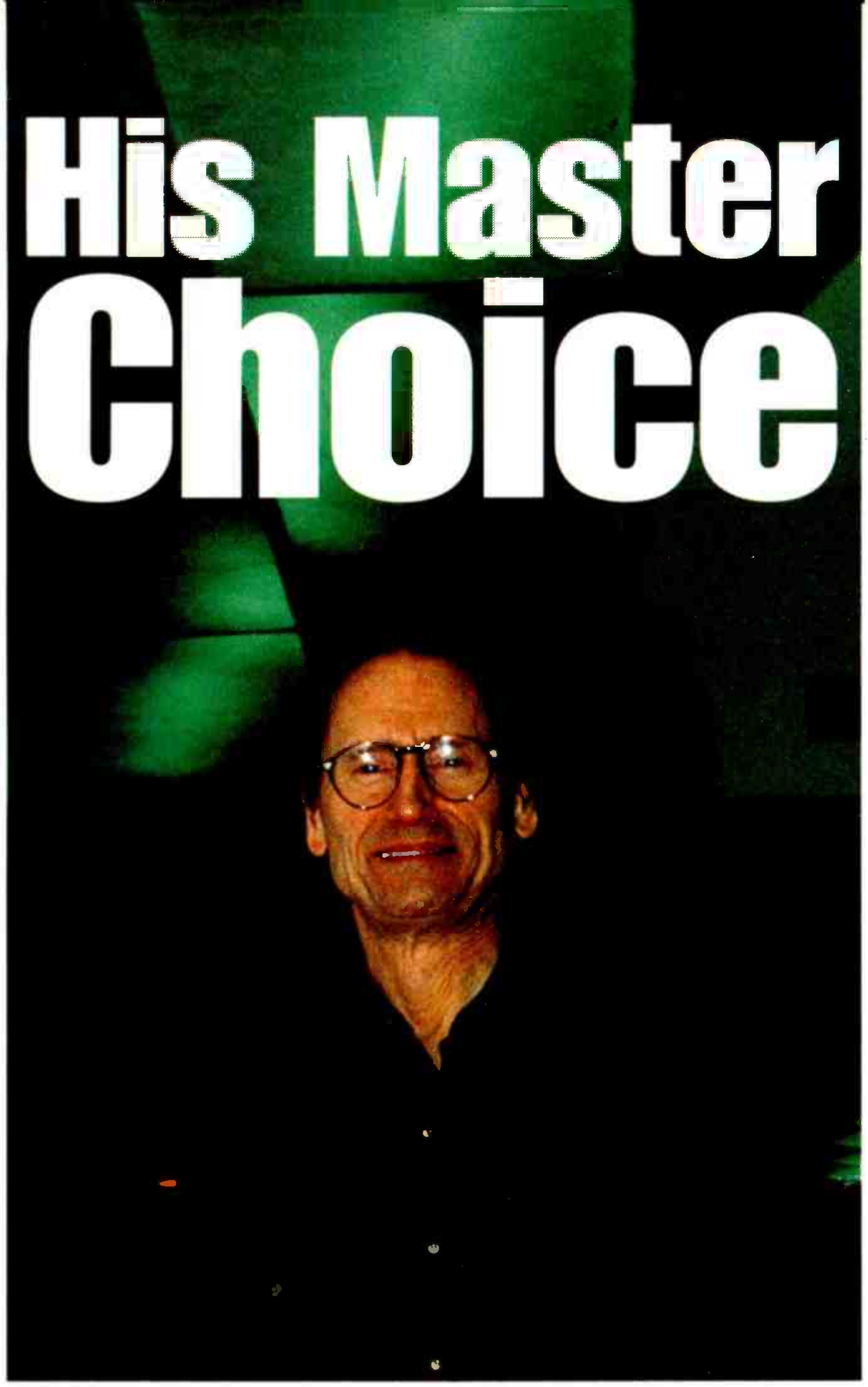

Bernie Grundman expands a facility and describes a mastering career that runs from cutting lathes to DAWs and back. **Dan Daley** listens

TAKING A BREAK from working on a cutting lathe—a machine whose sharp edges and rotating plates are a world apart from the virtual cosmos—Bernie Grundman strikes a figure reminiscent of a vintage car mechanic. He appears at ease in an old-order manner, like a mechanic comfortable with old-fashioned engineering and values—the occasional personality quirks present a welcome relief from the boringly predictable digital diagnostics of modern microprocessor control.

Grundman is a member of a small, elite cohort—one of the Masters Of The Universe: the handful of a dozen or so golden-eared mastering engineers whose ranks include Bob Ludwig, Glenn Meadows, Doug Sax, Doug Levine, and Denny Purcell. They are all in their 40s and 50s and have careers which span eras in audio that started with the classic post-war big band records and which witnessed the radicalisation of music from a culture into a multi-billion-dollar business, the introduction of digital, and the advent of a towering Babel of formats and technologies that increasingly have less to do with audio and more to do with marketing schemes. The difference between these Masters and someone now entering the business is the difference between someone who lived through two World Wars and someone who has lived through software revisions v2.0 and v2.0.1.

Grundman is hardly an antique himself; the Minneapolis native still talks about his work with the kind of buoyant enthusiasm that he brought to it 33 years ago as a college student interning at a local studio in Phoenix, Arizona, in 1965. But even his newest facility—a 16,000ft², 5-room mastering palace that opened in March of this year, six months after the opening of a branch operation in Tokyo—still adheres to the precepts of self-built mastering consoles (constructed by Grundman's long-time partner and technical wizard, Karl Bischof) and a heavily customised, passively amplified monitoring system comprised of vintage Tannoy components, such as the high-impedance tweeters the company stopped making around 1970, and a woofer transducer magnet that was the last Alnico-5 magnet used before the switch to ceramic magnets.

'I was brought up in the business during a time when you made a lot of your own equipment,' Grundman asserts. 'I came out of wanting to do this because of the music, not the technology. That came later. When I was a kid I would listen constantly to the huge collection of 78rpm records my father had. They were big band records, like the old Capitol stuff, and they sounded great. Then I started buying my own. One day I spotted an album and it looked interesting. I bought it, took it home and then, my God, that was it for me—it was Clifford Brown and Max Roach and I knew this is what would do it for me. I was always hanging around the hi-fi store on Saturday afternoons. I spent every penny I had on records and hi-fi gear. My dad had an old jukebox and I tore out the amplifier and speakers and hooked it up to a turntable.'

Grundman worked some more electronics into his background during a stint in the Air Force as an Electronics Counter Measures (ECM) technician, refining ways to track Soviet airborne radars instead of jazz combos. But his time as a (barely) paid gopher at Audio Recorders in Phoenix after his discharge and during college was the first link in a sequence of events that would eventually put him on top of the Mount Olympus of audio mastering two decades later.

'I went to the studio to look for a job and they told me I should talk to the engineer they had recently hired—Roy Dunann,' he recalls. 'When I heard that name I went through the ceiling. He was the guy I had been idolising for years as the engineer on great jazz records with people like Sonny Rollins and Art Pepper. He had been doing those records for years at this legendary West Coast jazz label, Contemporary Records in Hollywood, which had a small studio. Before that he had been the manager of Capitol Records' studio in Hollywood. He had moved to Phoenix in 1964 for his wife's health. Roy had set up that studio and Contemporary's. But in the process of building a studio on a budget, he built a truly innovative studio, one that had incredibly short signal paths and very little amplification, and it shows in the recordings that came out of it—they were incredibly clean and pure.'

Through Dunann, Grundman met studio and equipment designer and builder Howard Holzer in Van Nuys, California, where Grundman drove out to meet him at his house-cum-laboratory. It was Holzer who told him that Contemporary Records' founder and owner Lester Koenig was still without an engineer after Dunann's defection to Arizona. He not only suggested to Grundman that he go for the job, but before the young engineer knew it, had set it up as a sure thing with Koenig.

'I didn't think I could handle it at that point,' muses Grundman. 'But it was Howard who told me, "if you truly love what you do, you'll find a way to do it". And he was right, and that's how I've done everything I've ever done ever since.'

He's done quite a bit. Grundman spent two years as the engineer at Contemporary's small but busy studio in Hollywood, during that time watching as A&M Records built their new music and mastering facility nearby on La Brea

PHOTOS: DAVID GOGGIN

Grundman, in one of the new studios under construction

Avenue on the lot of the former film studio owned by Charlie Chaplin. In those two years, Grundman had honed his mastering skills: at a time when many mastering sessions were done according to specifications outlined by the record labels themselves and based more on level than anything else—so much for a single, so much for an album cut or juke box product—Grundman was learning to affect the fundamental sound of the records he worked on. 'Around that time, if you wanted to add EQ, top end, say, you had to make a transfer and do it in that process,' he recalls. 'I could be more innovative and hands-on about mastering because it was the one part of the business that Lester could bring in clients that weren't on the label. So I got exposed to a lot of different kinds of music and records and productions, and I wasn't rushed there, so I really got to learn how to manipulate sounds. That formed the basis of me as a mastering engineer: people come to you because you've accumulated years worth of knowing what sounds good for different types of music. But the equipment I was using at Contemporary was older—the cutter was a D-spec that had been used on the [first sound-tracked film] *The Jazz Singer* with Al Jolson. It had been made in the 1920s. A&M was putting in a state-of-the-art mastering facility.'

HE MOVED TO A&M in 1969 and stayed for the next 15 years, working on a Haeco console and Westrex lathes (both of which he still uses at his current facility), running the company's mastering studio at a time when independent producers were gaining more clout on productions, right through the mastering stage. It was, he says, the period during which the modern concept of mastering—in which that stage became as critical to the finished sound as tracking, overdubs and mixing—came of age.

It was also the time during which the cadre of mastering luminaries began to coalesce. The recognition that they possessed something that was unavailable elsewhere was becoming more widely understood.

'The reason that there have been so few mastering engineers at the top over the years, and the reason they last as long as they do, is because it takes so many years to accumulate the depth of exposure to so much music, as well as the talent to be able to understand what each needs,' Grundman explains. In a very real sense, he and the others became a living reference point, the ultimate arbiters of sound in an extremely arbitrary pop music culture. As a result, they could not, by nature, become the very thing that characterised many of the artists' and producers' records that they worked on: overnight sensations. For a mastering engineer, as for a head monk, longevity was part of the job description.

'You spend time with recordings the same way someone spends time with art in a museum,' he continues. 'If you go enough, and look long enough at the right things with some knowledge, you eventually see the depth and the meaning of the aesthetic. We bring a level of objectivity to something that's very subjective.'

In 1984, after several years of (relatively) quiet planning and building, Grundman and Bischof, who had been working with Grundman at A&M as head of technical maintenance, moved into Bernie Grundman Mastering (BGM) in Los Angeles. Bischof had built the consoles and other gear, and Grundman's wife had supervised the construction while the two partners continued to work at A&M. Grundman says that A&M didn't make any serious attempts to retain him only because, he says, 'I simply didn't make that an option. A&M had become a lot more corporate in recent years and I didn't have the flexibility I once had. I found myself having to fight to get budgets for new digital equipment. It was doing well financially, and I was making a percentage of that, but it was time to move on. So we set it up that when we left work at A&M on a Friday afternoon, the following Monday morning we were able to walk into the new studio and never miss a beat.'

Grundman's considerable following of clients came with him, and over the next decade his work won him six awards as mastering engineer and six more for his 3-room mastering facility as Outstanding Mastering Facility. The economics of the mastering studio have never been an issue, he says, even as the rest of the studio industry underwent radical economic restructuring. The continued boom in Grundman's business allowed him to create a 1-room extension of his facility in Japan in 1997. But that move was also facilitated by an increasingly rare phenomenon: Grundman had taken on a Japanese would-be mastering engineer, Yasman Maeda, in 1991.

'He came to me and in essence said he wanted to apprentice himself to me,' recalls Grundman. 'I'd never heard anyone ever ask that before in 25 years. So I took him on. How could I not?'

The move proved to be a wise one. Maeda not only proved he had an aptitude for mastering, but for hustling, as well, bringing in a significant number of new clients for the mastering facility—mostly Japanese. It was getting to the point where Grundman began to consider building a studio for Maeda, as he had done with two previous proteges, Brian Gardner and Chris Bellman. Eventually, he did just that; the studio, however, just happens to be in the Shibuya district of Tokyo instead of >>>>>

PHOTO: DAVID GOGGIN

<<<<< Hollywood, California. Bernie Grundman Tokyo, as it's known, is a single studio facility operated by Maeda as an employee of BGM; although he is lavish in his praise for Maeda's talent, ambitiousness and follow-through.

It also gave Grundman an opportunity to assess the state of mastering in Japan, which he says is about where it was in the US three decades ago, not in terms of technology but in attitude.

'It's considered a routine part of the record-making process, but it's not given the same emphasis that we've come to place on it,' he explains. 'Mastering studios in Tokyo are simple and stark, not built out like they are here, or like the beautiful studios they have in Tokyo. Mastering is subordinated to recording in the pecking order there. But the desire to change that is also there. They've been very responsive to our approach that Yasman brought over there, and when Bruce Swedien and I did a mastering seminar over there the summer before BGM Tokyo opened, the response was overwhelming—we sold out a hall twice over for each one.'

Which brings us to Grundman's newest venture, the complete replacement of his 14-year-old 3-room facility with a new \$2.5m-plus one. The design is based almost completely on the previous facility's layout and acoustics. The changes largely reflect amenities, such as larger seating areas for clients behind the consoles in mastering suites, and, as Grundman puts it, 'More room to stash stuff—I'm sitting here now with three Mitsu X-80 machines crammed around me and I can't move.'

The new facility will also have two Scully disque cutting lathes, both in the same suite—one fitted with analogue components including a Haeco cutting head, and the other with solid-state electronics designed by Grundman and Bischof. Other technologies include Harmonia Mundi BW102 signal processors, Apogee convertors and a Studer A80 mastering deck modified to accept ¼-inch reels with custom discrete playback electronics. The all-discrete consoles will be custom-built, assembled from various components and are digital-analogue hybrids, each having integrated onboard digital signal processing as well as analogue processing.

'We want to be able to combine digital EQ and analogue compression or vice versa on any project,' says Grundman. Each console will also have integrated into it a digital audio workstation for edits and crossfades; up till recently, BGM has been using Studer Dyaxis systems, which Grundman says sound good, but which are no longer meaningfully supported by the manufacturer.

Grundman will continue to use both Sony 1630 and Yamaha CD-R formats as final master delivery formats to manufacturing. However, he cautions, the replication plants, in a scramble to remain competitive, have been less than stringent in their own premastering requirements. 'They want the business, so they say they can make a glass master out of a DAT, but what they're not telling you is that they have to go through another transfer process to Exabyte to do that. And my point has always been that anything you do to a master will affect its sound. Any changes in a circuit or in a process changes the sound. And bit-for-bit analysis doesn't reflect those changes accurately. That's another reason why the few top mastering engineers remain in place: the human component is still the primary one in mastering.'

The size and scale of the new Grundman mastering facility keeps it firmly planted in the upper echelons of the mastering pantheon. However, its instigator is aware of the fast-growing niche of small and mid-sized mastering facilities, a response to the massive growth in low-budget and mid-budget independent records in the last few years and enabled by the same types of technology as the so-called project studios. Mastering facilities based on Pro Tools and Sonic Solutions abound and proliferate. Grundman's response, though, shows no trace of concern. He's pleased that mastering is finally being considered part of the normal routine for this stratum of records—too many have been released without even being passed through a stereo compressor. He also believes that those who realise the need for quality mastering will see that its cost remains a fraction—albeit a larger one—of their overall record budgets. Finally, at \$280 per hour, Grundman has kept his hourly rate somewhat below that of his cohorts, who often have card rates as high as \$350 per hour.

'It's a bit low, but we want to keep the rooms full, and we don't want to be elitist about things,' he says. 'We really enjoy getting a variety of stuff in here, and we learn something from all kinds of records. Always have.' ■

Contact
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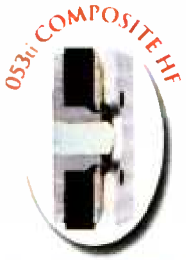

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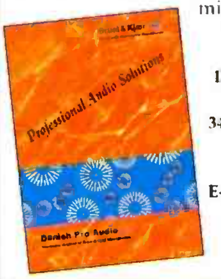

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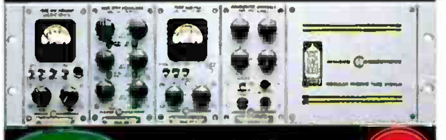

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US: Having it all

The American Dream meets up with reality in the new order of studio business writes **Dan Daley**

IN AMERICA, there is a long-distance phone company commercial running in that shows a young executive riding an ascending elevator on a meteorical, metaphorical rise to the top of the corporate heap, tagged with the line, 'You can have it all'. Having it all is an American tradition, the battle cry of a consumer culture for which less is never more. And why should consumers of professional audio services be any different?

What clients have come to expect from recording studios has changed over the years, but most radically in recent times. In the beginning, studios provided all the fundamental services required of the record-making process, from basic tracking to mixing. You went elsewhere for mastering and then to the pressing plant to get your vinyl. Then it became fashionable to use studios less as places of work than as comprehensive creative environments. In the early 1970s, Criteria Studios in Miami was a perfect example of how a studio became an organic part of records, where Derek & The Dominoes and the Bee Gees spent as much time writing the songs as recording them. Then, around 1979, somebody started looking at the budgets, and that was the end of that.

In the 1980s, specialisation became the norm, as studios looked for niches to deal with increasing competition. You went one place to track, another to overdub, another to mix. Mastering and duplication were still other components of an increasingly peripatetic process.

Things have changed yet again. As competition in many US markets evolves from simply tight to outright cut-throat, studios now try to group together new services in an attempt to create as many revenue streams as possible. It's not uncommon now to see recording studios advertise tracking, overdubbing, mixing, digital editing, format conversion, ISDN/T-line link-up, mastering and a smattering of postproduction services under a single roof, whether it's the capacious mansard of New York's The Hit Factory, or the less palatial space of mid-sized studios using affordable technologies hoping to beat project studios at their own game with.

The concept of Full Service is upon us, in spades, driven by real competition and this vague cultural notion that You Can Have And Do It All.

Or can you? With a siren-like allure, the technology seduces you to the affirmative, no less than does the aforementioned commercial. (And from a purely cultural point of view, the two are inextricably linked.) In many cases, the studios are simply trying to implement a service before clients try to do it themselves. And encouraged by the development of user-interfaces that increasingly suggest chimps as control subjects and technology prices that are now affordable even by Ugandan economic standards, a 'why not?' mind-set has developed.

So I'll tell you why you can't have it all—it's because you can't do it all, and do it all well. There's no blame to be assigned nor aspersions to be cast in this turn of events; what's actually changing are the expectations of quality. The bar has not been lowered, but the floor has been raised—wider access to technologies that can putatively perform more functions leads to the belief (a careful choice of word) that with access comes ability. The Golden Dozen of leading mastering engineers know that their long lock on that end of the business won't continue indefinitely as time and technology

Europe: Teething trouble

The long-awaited announcement of DVD's audio provision has assured continued confusion writes **Barry Fox**

DVD FINALLY LAUNCHES in Europe in April. It's a 'soft' launch, with no official kick-off day and different software companies coming on-stream over a six-week period. The movie titles are all back catalogue but the studios promise a 'hard' launch later in the year, with more exciting titles. The delays are due to squabbles over the audio format. The shakedown is exactly as Ray Dolby predicted at the Berlin IFA show last August when Warren Lieberfarb and the DVD Forum members firmly announced that the format for Europe would be MPEG2. It won't be.

The no-MPEG news broke at a briefing staged in London by the UK's DVD Launch Committee, with all the hardware and software companies supposedly presenting a consolidated front to launch the new format. 'Supposedly' because Warners failed to attend, for fear of stealing thunder from a glitzy bash secretly planned to mark Warner's software launch in Europe on 22nd April. By December, the studios were still waiting for working MPEG2 encoders, and there was no sign of a consumer decoder. In December, the DVD Forum relaxed the standard requirement to allow Dolby Digital AC-3 or MPEG2. There are still no MPEG decoders and all but one of the studios has abandoned the format. Even PolyGram, a subsidiary of Philips which backs MPEG2, has gone for AC-3. Only Columbia-Tristar is hedging its bets and putting both MPEG2 and AC-3 sound on the same disc. This is likely soon to end as, in a bizarre twist, Warner Vision will release music videos with AC-3 multichannel on one side and PCM stereo on the other. Not one of the studios even mentioned DTS.

With MPEG2 now flushed down the pan, Philips is concentrating all efforts on trying to make Super Audio CD, and DSD, the standard for an audio-only version of DVD. This puts Philips (and partner Sony) in direct conflict with 'super' PCM which uses 24-bit coding and 96kHz sampling. This approach is backed by most Japanese companies, the mainly European Acoustic Renaissance for Audio (Chaired by Merdian's Bob Stuart) and the Advanced Audio Disc Working Group of the American Consumer Electronics Manufacturers Association. The DVD Forum in Japan has now published v0.9 of the DVD-Audio standard which leaves room for either sound system in a bewildering range of new audio, video and mixed media discs.

The advantage of the 24-96 approach is that the audio-only disc is playable on existing DVD video players. Early models will resolve only 20 bits and even players that claim 24-bit chips may well fall short of full and accurate decoding—just as early CD players came nowhere near the 16-bit standard.

The Super Audio CD proposal is motivated by commercial politics as well as any sincere belief in Sony's Direct Stream Digital

> But if making hybrid discs is such an easy fix, how come none were available as demo discs that would prove the principle by playing either on a DVD player or CD player?

1-bit technology. Indeed, Sony's faith in DSD will be easier to believe when the company uses the system for archiving. The 24-96 system is based exclusively on DVD technology and cuts CD out of the loop. Philips and Sony want to keep CD alive. And this they can do by making Super Audio CD a hybrid disc, with Red Book CD audio recorded at a depth of 1.2mm and DVD Super Audio at the DVD depth of 0.6mm. Anyone licensed to press CDs or make CD march on. But in talking with them, it becomes clear that they are not anxiously anticipating the loss of income as much as the dilution of wisdom that accompanies this trend.

The myth that access equals ability, compounded by the need to stay afloat in difficult times, is seductive and overwhelming. But who would you rather have doing open-heart surgery on you: a general medical practitioner or a cardiologist? The medical analogy is apt: as technology increased in complexity, specialisation became more necessary, not less so.

Yet the opposite has happened in the studios. Facility operators and owners are encouraged to believe that they can offer soup-to-nuts service, from rehearsal to CD-R replication, because the technology permits them to. And a combination of changed expectations as to what constitutes quality, and the attraction of a lower, all-in price for an entire project, has many clients readily agreeing.

There will be a few geniuses out there who can do it all, reasonably credibly. But most can't. It's unrealistic to ask studios not to test new waters even as they continue to swim in familiar pools. But it's appropriate to suggest a caution in the process, for their own long-term good as well as that of the client and the industry. As Clint Eastwood, an American with a long-lived career of cautiously chosen paths once said, 'A man has to know his limitations'.

players can make hybrid Super Audio CDs, with no extra royalty payment due to Philips and Sony.

At first this sounds a very attractive proposition and it was hard-sold at the promotional week staged by Sony and Philips at Abbey Road in February. But if making hybrid discs is such an easy fix, how come none were available as demo discs that would prove the principle by playing either on a DVD player or CD player? This is how Philips tests hybrid discs made at its pilot plant in Eindhoven. Weeks after the Abbey Road event, I am still waiting for promised samples. Panasonic has already warned that dual layer hybrid discs are much more expensive to make than DVDs. They are also more likely to deform in humid conditions because they are made from an asymmetrical 2-layer sandwich. And artists may well ask for extra royalties on the double recording. All this puts a big question mark over the idea of single inventory releasing, at no extra cost to the consumer.

The Abbey Road demonstrations can only be described as odd. Sony had recorded a jazz piano trio, with—boasted Sony—'10 to 20 mics and contact-miked acoustic bass'. As a result, the piano sounded as wide as a battleship. Philips' multichannel offerings were no better, with five mics placed around an orchestra in the same position as the five speakers, and feeding five discrete channels. I amused myself by imagining what Michael Gerzon would have said when he heard this hark back to 1970s quadraphonics and the disconcerting effect of sitting in the middle of a band.

The sound of uncertainty

Digital radio promises improved services but so far has engendered confusion, fear and scepticism writes **Kevin Hilton**

THERE IS AN UNLOVELY part of the human psyche that takes comfort from the misfortunes of others, particularly if our own situation is not that tremendous. It really is a case of, 'Well, everything's going down the tubes but at least we're better off than they are'. While putting together a series of articles on digital radio recently, a person involved in the promotion of the format commented to me that although it was experiencing problems, at least it was in a better position than digital television. This would almost be a justifiable comment if digital radio itself were on firmer ground, but as it is not, it is an indefensible slight.

Broadcasters and manufacturers, while enthusiastic about the technical and commercial benefits that digital will bring, have behaved like two people heading for the same narrow door at the same time who end up hesitating and dithering without thinking to say, 'Please, you go first'.

Eventually, the broadcasters, led by the BBC in the UK and its counterparts in Denmark and Sweden, rolled their eyes, tutted loudly and barged through the door, while the manufacturers became embarrassed and went back inside. Despite this emphatic move, the BBC in particular made a number of serious public relations errors when it introduced its pilot service in 1995. The phrase 'launch' was dropped for the more prosaic 'switch-on', with the preparation of the much-lauded new services, made possible by the extra capacity of digital, being played down in the face of unwanted publicity.

In the face of Eureka conquering most of the world, the US was still dithering between the European contender and the IBAC-IBOC alternative

Then there were the technical considerations. Nobody could quite answer the question as to whether the frequency spectrum chosen to hold the multiplexes, the L-band, was to be the permanent home or merely a parking arrangement. This was linked into the future of FM, which would be freed up by the start of digital radio but would have to simulcast for at least 10 years to prevent disenfranchising those listeners who did not immediately make the switch. These were questions the broadcasters could not readily answer, as was the matter of the preferred DAB system in Europe, Eureka 147, being one of two worldwide standards. In the face of Eureka conquering most of the world, the US was still dithering between the European contender and the IBAC-IBOC alternative, although these were withdrawn from testing during 1996.

The biggest problem faced by digital radio has been the lack of suitable domestic receivers; there have even been stories of broadcasters having to wait several months to get prototype equipment that would enable them to internally monitor their own DAB output. There was much optimism at the IFA trade fair last August, but; although 17 manufacturers showed a range of products, many of these were still preproduction versions. Broadcasters have become frustrated but this is reciprocated by organisations representing manufacturers, dealers and retailers, who say that they have heard nothing from the leading broadcast companies since IFA.

Television has faced similar trials, but, despite a line of questioning designed to see if those involved in DTT would sneer in the same way at digital radio, those involved in this sector have been more contained. Perhaps this is a realisation that they have problems of their own to deal with. The DTG, a UK cross-industry body representing manufacturers, retailers and broadcasters, has said that its main priority is to hit the air date of this autumn, starting transmissions in a simple way and later adding the functionality and interactivity that, as with digital radio, has received so much attention.

Those close to the project have admitted that hitting the start date will be tight. Even then, there will be problems as reception of the services will be graduated. In the UK, it is intended to have 90% coverage on the 'best' multiplexes, decreasing down the scale, to the point where some areas will not be able to receive services. As with digital radio, simulcasting will be a crucial factor; the DTG's technical director, Peter Marshall, says that switching off analogue is a political decision, not a technical one, a comment that undoubtedly applies to radio as well.

The manufacture of digital set-top boxes is said to be underway, with the roll-out of services being engineered to start by this Christmas, when consumers may be thinking of high-tech presents for their sofa-bound loved ones. This timing may further convince conspiracy theorists that this new technology is nothing more than a way to sell new radios and TVs sets to people who already have perfectly acceptable analogue models. Matters have not been helped by a report from a UK consumer watchdog that advises viewers to wait until either the price of set-top boxes stabilises or integrated TV-receivers appear, offering better quality and value.

Consumer reservations regarding this technological upheaval are understandable, but it is a less cynical situation than what happened with CD and vinyl back in the mid-eighties. There are more benefits to be enjoyed this time around, but as it is being handled in such a muddled way, it is not surprising that the market is confused, fearful and sceptical.

Dr John

Microphone principles: 1

The transducer at the front of the audio chain is regarded by many as the most important single piece of equipment. **John Watkinson** begins a guided tour of microphones

SOUND CONSISTS of both pressure and velocity variations, and microphones can use either or both in order to obtain various directional characteristics. Fig.1a shows a true pressure microphone that consists of a diaphragm stretched across an otherwise sealed chamber. In practice a small pinhole is provided to allow changes in atmospheric pressure to take place without causing damage. Some means is provided to sense the diaphragm motion and convert it into an electrical output signal. This can be done in several ways (these will be considered in a later article). The output of such a microphone is independent of direction except at high frequencies as Fig.1b shows.

Unlike human hearing, which is selective, microphones reproduce every sound which reaches them. Placing a microphone near to a hard wall means that the microphone receives a combination of direct and reflected sound between which there is a path length difference. At frequencies where this amounts to a multiple of a wavelength, the reflection will reinforce the direct sound, but at intermediate frequencies cancellation will occur, giving a comb-filtering effect. Clearly a conventional microphone should not be positioned near a reflecting object.

The path length difference is zero at the wall itself. The pressure zone microphone (PZM) of Fig.1c is designed to placed on flat surfaces where it will not suffer from reflections. A pressure capsule is placed facing and parallel to a flat surface at a distance that is small compared to the shortest wavelength of interest. The acoustic impedance rises at a boundary because only half-space can be seen and the output of a PZM is beneficially doubled.

Fig.1d shows the pressure gradient microphone in which the diaphragm is suspended in free air from a symmetrical perimeter frame. The maximum excursion of the diaphragm will occur when it faces squarely across the incident sound. As Fig.1e shows, the output will fall as the sound moves away from this axis, reaching a null at 90°. If the diaphragm were truly weightless then it would follow the variations in air velocity perfectly, hence the term velocity microphone. Unfortunately, it isn't, and a pressure difference is required to make it move, hence the more accurate term pressure-gradient microphone.

The pressure gradient microphone works by sampling the passing sound wave at two places separated by the front-to-back distance. Fig.1f shows that the pressure difference rises with frequency as the front-to-back distance becomes a greater part of the cycle. The force on the diaphragm rises at 6dB octave. Eventually the distance exceeds half the wavelength at the critical frequency where the pressure gradient effect falls rapidly. Fortunately the rear of the diaphragm will be starting to experience shading at this frequency so that the drive is only from the front. This has the beneficial effect of transferring to pressure operation so that the loss of output is not as severe as the figure suggests. The pressure gradient signal is in phase with the particle displacement and is in quadrature with the particle velocity.

Polar coordinates are used for microphone directionality plots such that the magnitude of the response of the microphone corresponds to the distance from the centre point at any angle. The pressure microphone has a circular polar diagram as it is omnidirectional. Omni microphones are good at picking up ambience and reverberation which makes them attractive for music and sound effects recordings in good locations. In acoustically poor locations they cannot be used because they are unable to discriminate between wanted and unwanted sound. Directional microphones are used instead

The PG microphone has a polar diagram that is the shape of a figure-of-8. Note the null at 90° giving rise to the term dipole. The fig-of-8 microphone (sometimes called simply 'an eight') responds in two directions giving a degree of ambience pickup; although the sound will be a little drier than that of an omni. A great advantage of the fig-of-8 microphone over the omni is that it can reject an unwanted sound. Rather than point the microphone at the wanted sound, a better result will be obtained by pointing the null or dip in the polar diagram at the source of the unwanted sound.

Unfortunately, the pressure gradient microphone cannot distinguish between gradients due to sound and those due to gusts of wind. Consequently, PG microphones are more sensitive to wind noise than omnis.

Various useful results can be obtained by combining the omni and eight principles. Fig.2a shows that if the omni and eight signals are added equally, the result is a heart-shaped polar diagram called a cardioid. This response is obtained because at the back of the eight the output is antiphase and has to be subtracted from the output of the omni. With equal signals this results in a null at the rear and a doubling at the front. This useful polar response will naturally sound drier than an eight, but will have the advantage of rejecting more unwanted sound under poor conditions.

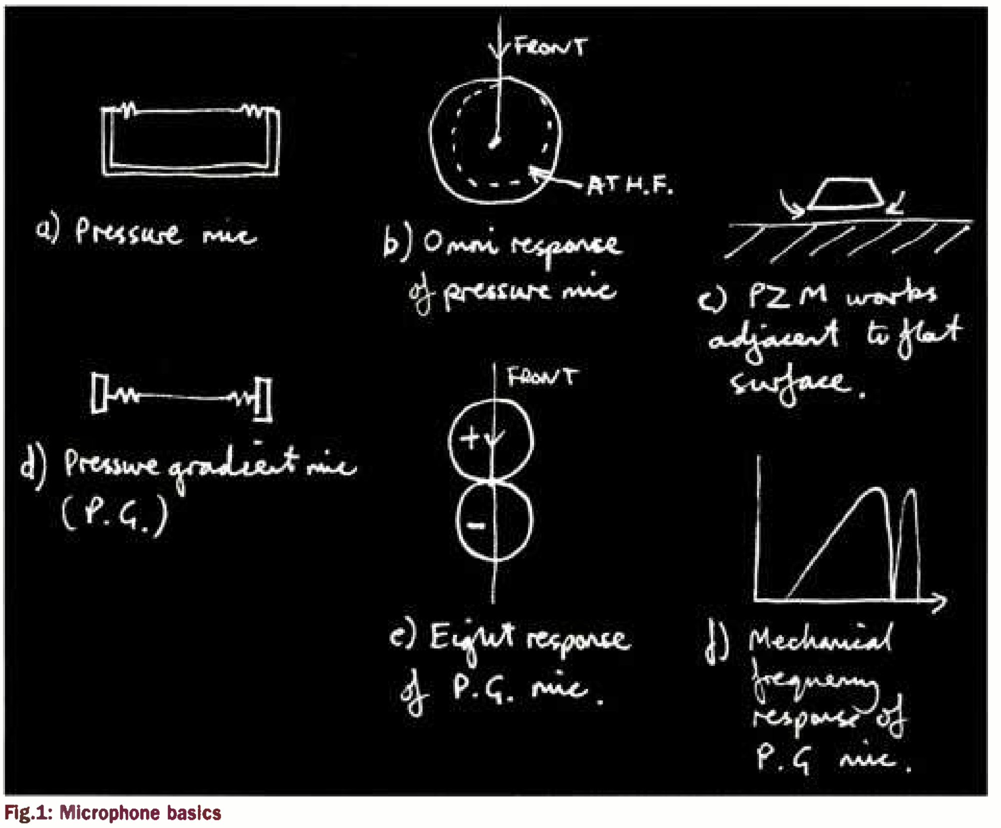

Fig.1: Microphone basics

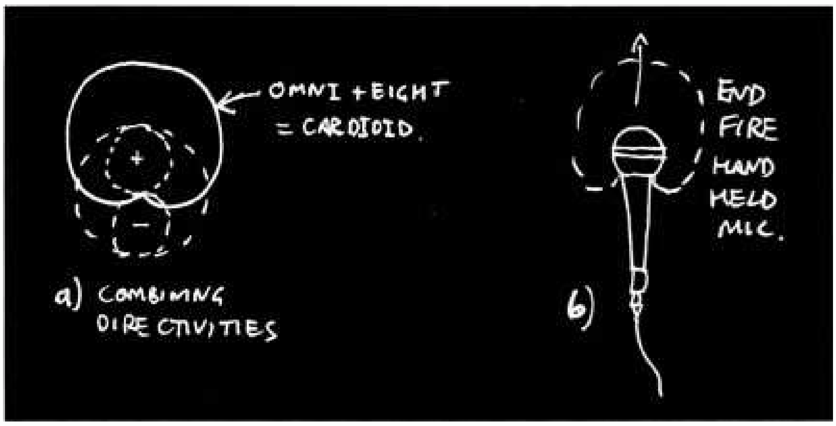

Fig.2: Derivation of cardiod polar pattern and 'end-fire' construction

In public address applications, use of a cardioid will help to prevent feedback or howlround which occurs when the microphone picks up too much of the signal from the loudspeakers. Virtually all hand-held microphones have a cardioid response where the major lobe faces axially so that the microphone is pointed at the sound source. This is known as an end-fire configuration shown in Fig.2b.

Where a fixed cardioid-only response is required, this can be obtained using a single diaphragm where the chamber behind it is not sealed, but open to the air via an acoustic labyrinth. Fig.3a shows that the asymmetry of the labyrinth means that sound which is incident from the front reaches the rear of the diaphragm after a path difference allowing pressure gradient operation. Sound from the rear arrives at both sides of the diaphragm simultaneously, nulling the pressure gradient effect. Sound incident at 90° experiences half the path length difference, giving a reduced output in comparison with the on-axis case. The overall response has a cardioid polar diagram. This approach is almost universal in hand-held cardioid microphones.

In variable directivity microphones there are two such cardioid mechanisms facing in opposite directions as shown in Fig.3b. The system was first devised by the Neumann company. The central baffle block contains a pattern of tiny holes, some of which are drilled right through and some of which are blind. The blind holes increase the volume behind the diaphragms, reducing the resonant frequency in pressure operation when the diaphragms move in anti-phase. The holes add damping because the viscosity of air is significant in such small cross-sections. The through-drilled holes allow the two diaphragms to move in tandem so that pressure gradient operation is allowed along with further damping. Fig.3c shows that sound incident from one side acts on the outside of the diaphragm on that side directly, but passes through the other diaphragm and then through the cross-drilled holes to act on the inside of the first diaphragm. The path length difference creates the pressure gradient condition. Sound from the 'wrong' side (Fig.3d) arrives at both sides of the far diaphragm without a path length difference.

The relative polarity and amplitude of signals from the two diaphragms can be varied by a control. By disabling one or other signal, a cardioid response can be obtained. Combining them equally results in an omni, whereas combining them with an inversion results in a fig-of-eight response. Unequal combination can obtain the sub-cardioid or a hyper-cardioid response.

Where a flexible polar response is required, the end-fire configuration cannot be used as the microphone body would then block the rearward access to the diaphragm. The side-fire configuration is used where the microphone is positioned across the approaching sound, usually in a vertical position. For television applications where the microphone has to be out of shot such microphones are often slung from above pointing vertically downwards.

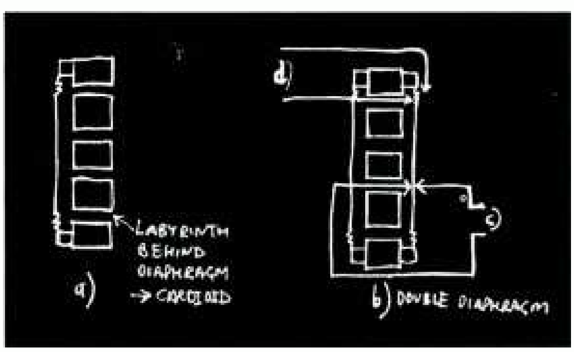

Fig.3:

In most applications the polar diagrams noted above are adequate, but on occasions it proves to be quite impossible to approach the subject close enough and then a highly directional microphone is needed. Picking-up-ball sounds in sport is one application.

In the shotgun microphone a conventional microphone capsule that is mounted at one end of a slotted tube. Sound wavefronts approaching from an angle will be diffracted by the slots such that each slot becomes a reradiator launching sound down the inside of the tube. The radiation from the slots travelling down the tube will not add coherently and will be largely cancelled. A wavefront approaching directly on axis will pass directly down the outside and the inside of the tube as if the tube were not there, and, consequently, will give a maximum output. ■

Advertisers Index

Speaker testing

Reviewing loudspeakers is an increasingly contentious issue. **Philip Newell** describes the problems facing a reviewer and the goals that can be realistically achieved

DESIGNING A MEANINGFUL loudspeaker test is a considerable challenge. The problem is one of reconciling an authoritative and repeatable test with subjectively widespread perceptions. What, for example, can we measure about an NS10 that relates to its decade-and-a-half of popularity? However a test is done, it is unlikely to identify outright winners, as even in a purely objective world, different loudspeaker responses suit different circumstances.

Subjectively, user expectations and preferences also vary, as do the demands of different types of music and recording techniques. In fact, some of these requirements appear mutually exclusive, and will probably remain so as long as we do not have perfect loudspeakers. What we have are performance 'truths' that are relatively incontrovertible in their objective-subjective relationships. If we can show how the perceived characteristics of groups of loudspeakers relate to their objective groupings, then we can identify connections and form a wider picture. But before discussing our truths, we should list the realities that the marketing departments of certain loudspeaker manufacturers would rather we forget.

There is no loudspeaker that is optimal for all rooms. There is no loudspeaker that is optimal for all music. There is no loudspeaker that can perform optimally in an acoustically bad room. High-definition monitor systems, with low distortion, good time-response accuracy, and linear amplitude and phase responses, cannot, as yet, be made cheaply (£2,000–$3,200 per pair would seem to be about the minimum). Computer-aided design does not mean that the product is superior to a non-computer-aided design. New technology cone materials are frequently subjectively inferior to older ones. Too many loudspeakers are designed by computers, measured by computers, and sold on this basis to an industry whose staff are now so computer-orientated that they would rather believe a computer than their own ears. The Quad Electrostatic Loudspeaker of 1957 can still give an account of itself that the last 40 years of development have not been able to significantly better, at least not within its axial SPL capability.

If this appears a negative list, here are some subjective-objective truths.

It is generally considered that the single most important aspect of the perceived accuracy of a loudspeaker, or an entire monitor system, is that it should provide a uniform acoustic pressure amplitude response (or frequency response) at the listening position.

Largely due to work by Ohm and Helmholtz, it has been a widespread belief that phase response uniformity is not particularly critical. Unfortunately, the above work was carried out using sine waves, which bear little relation to music signals. Briefly, the phase response, in combination with the pressure amplitude response, defines the time response, and the three are inextricably linked by the Fourier Transform[1]. In the days of analogue dominated recording, phase inaccuracies inherent in the recording process rendered high degrees of phase accuracy in the monitoring systems to be somewhat less important, but digital recording has put a new emphasis on the phase accuracy. In order to perceive very low-level detail in digital recording, modern control rooms tend to be acoustically dead, and these conditions happen to be exactly where phase inaccuracy becomes more noticeable[2].

In 1989, Keith Holland and I wrote a paper in which we asked more manufacturers to publish the impulse responses of their systems[3]. The integration of this—the step-function response—is, perhaps, a more visually representative form, and step functions themselves are actually realistic test signals (Fig.1). The 1989 argument was that, in the past, the time responses of all but a very few loudspeakers (such as the Quad Electrostatic) were simply too far away from time accuracy to be compared on such a basis. In 1997 people are publishing time responses, largely in response to the demands of digital recording. A great deal of the natural sound character of an instrument is dependent upon the integrity of the leading edge of its waveform, and this can only be preserved if a loudspeaker sys-

Step Function Responses

The five plots show the electrical input of a step function (a) followed by the acoustic output from four different monitor loudspeakers. The tail-off of the responses, to the right, is a function of their limited low frequency responses such as those shown in (b) and (c), but it may come as little surprise that those two loudspeakers are generally considered to be very 'fast', transparent and open-sounding. Their superior transient performance is clearly seen from the step function responses. The responses of (d) and (e) are more typical of loudspeakers in general. All the plots are from professional monitoring systems in widespread use. All exhibit acceptably linear frequency response plots from which no clear distinctions or inferences can be made.

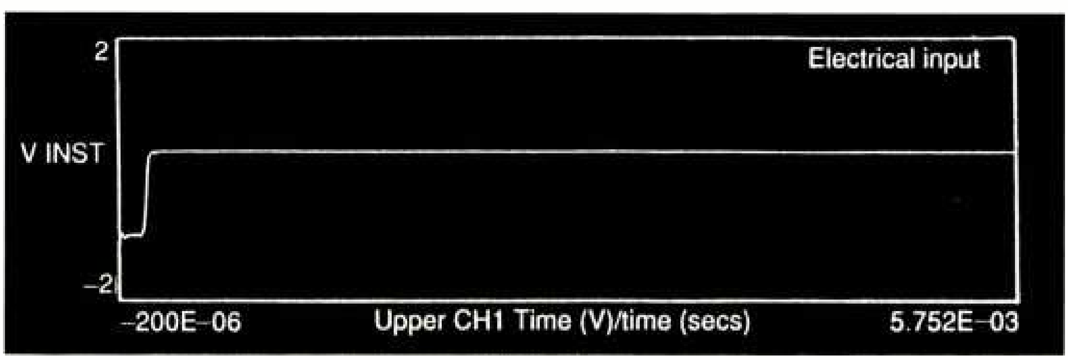

Fig.1a: Step function electrical input

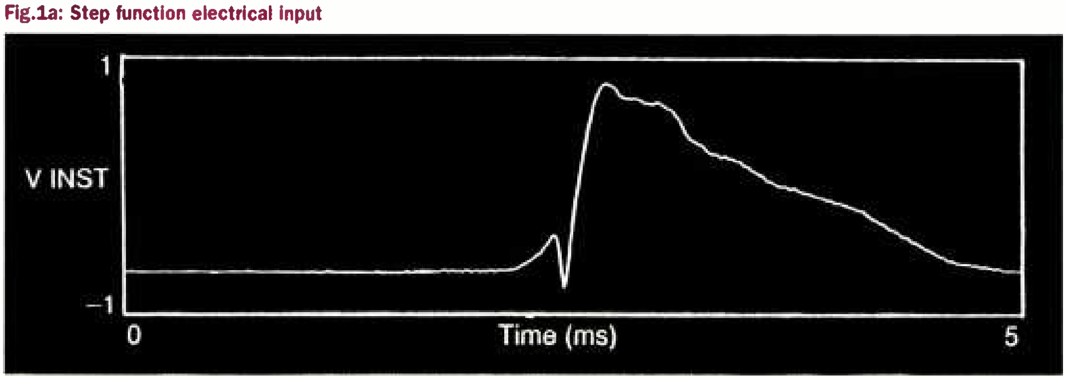

Fig.1b: Response of system using a 'D'Appolito' vertical array

tem maintains time integrity. Transparency and openness are also very dependent on the accuracy of a loudspeaker's time response.

Harmonic and intermodulation distortions produce sounds which are not a part of the input signal. Higher harmonics generally lead to harshness and aggression, making any given Sound Pressure Level (SPL) seem subjectively louder, but masking some of the low-level detail in the signal. It is also difficult to produce an open, expansive, transparent sound with significant levels of nonlinear distortion. Second harmonic distortion can produce richness and warmth, but if this exists in the monitor loudspeakers, it is not necessarily on the recording, so will not travel with it. If such a sound is wanted, then one should use some valve microphones or valve electronic processors, and record the sound of the harmonics. It is no good just to listen to it on the recording-mixing monitors.

Over a period of years of listening, it became evident to me that some 'advanced' types of loudspeaker lack low-level detail and general openness, while seeming to be almost exemplary in other respects. After reading an AES paper by David Clark[4] in which he discussed stiction effects in low-frequency motor systems, I wondered if the problem could be in the suspension systems, whose inherent losses require a finite minimum drive signal to get them moving. He convinced me that these were purely LF problems, and that the frequencies of interest for transparency and openness were above the affected range. I was partly led astray by associating these problems with some new suspension systems, but Martin Colloms believes the problem to lie in some of the new cone materials, some of which, coincidentally were introduced with some of the new types of suspension systems[5]. He also believes that some of the new, rigid, high internal loss cones exhibit hysteresis losses that are difficult to quantify, yet that can drastically effect the perceived naturalness of their response. Part of this problem may have arisen as a result of the over use of computer-aided design, where solving one engineering problem (such as creating ultra-damped cones) has caused other problems, but the computers have never heard them. In listening tests, designers can, all too frequently, be drawn into listening so specifically in the area of interest that they often fail to notice side-effects that result from the 'improvement'. Technical developments must be carefully subjectively assessed—they do not always bring the overall advance in audible performance that they suggest from their specific performance improvements.

In some of these areas, we are still in trouble when using direct measurements, and perhaps the best insight into the origins of such problems can be gained from the grouping of similar materials or similar design concepts to similar some performances. Careful listening in acoustically controlled environments may be the only reliable way to pursue the facts. Although this is time-consuming and costly, it is, perhaps, the only way that >>>>>

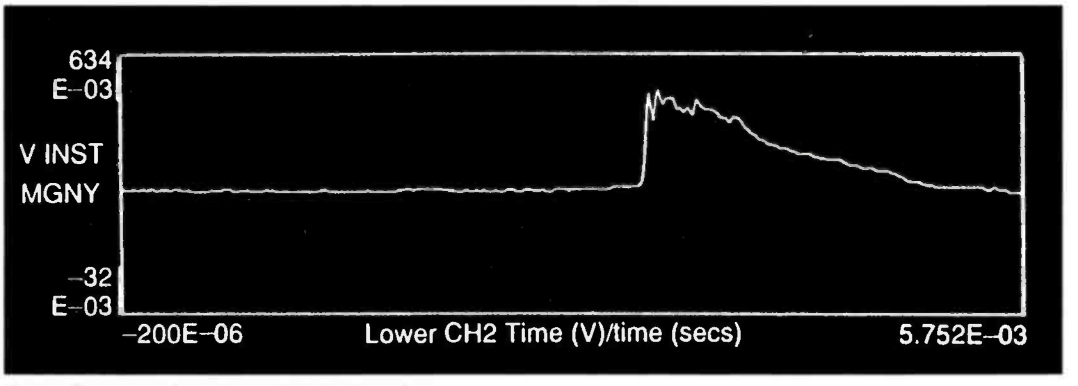

Fig.1c: Response of an electrostatic loudspeaker

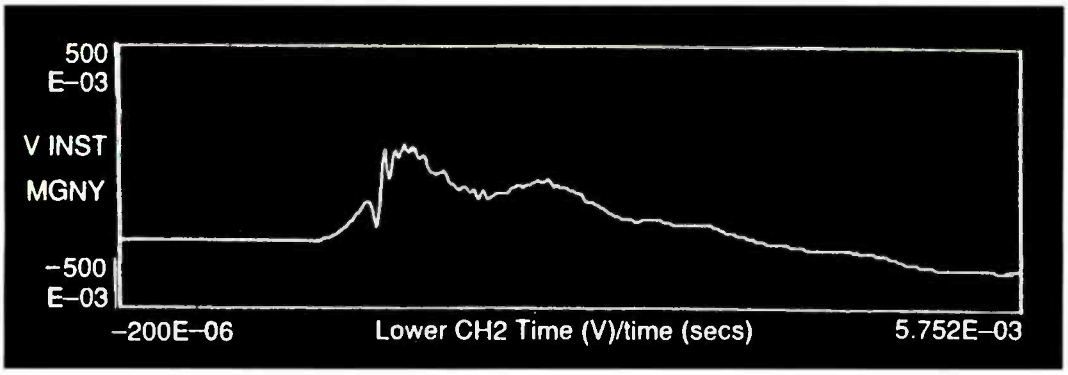

Fig.1d: Response of a dual-concentric

<<<<< we can currently find the sources of some anomalies in the objective-subjective comparisons. In order to make these comparisons, we must have the objective measurement results and sufficient meaningful data to attempt to make cross-correlations. Hopefully, we could all learn something from such a programme of tests.

There would seem to be little doubt that one of the greatest problems facing the designers of loudspeakers for critical listening purposes is the human-perception factor. Over the years, loudspeaker design has largely advanced in bursts, but these bursts have frequently been due to new audiological-psychoacoustic information becoming available to the designers, as opposed to the use of 'revolutionary' new materials. In fact, the latter have frequently not lived up to their technological promises once a period of critical listening has been completed. We are still a long way from being able to specify or quantify many of the parameters which impart on one loudspeaker the quality that gives it an edge.

Furthermore, the situation is complicated by many people, even experienced professionals, having different ideas of what to expect from a loudspeaker; especially in terms of their hierarchy of performance demands. This situation exists with other subjective senses, though, and is not exclusively an audio-related phenomenon.

For example Kodak, Fuji, Konica, and other film manufacturers all have partisan followings. Personally, I am a Kodak user because Kodak films and processing give me photographs that I see as being the best representation of what I saw at the time of taking the pictures. No Fuji fanatic is going to convince me otherwise, yet equally, I realise that if Fuji fans see the beauty that they are looking for in the blues and greens, then far be it from me to argue with them. This situation exists despite the fact that a photograph can be held up in front of the original object, and, given the same light as at the time that the photograph was taken, the colours can be directly compared. This also exists despite the fact that the optic nerves carry a relatively complete signal to the brain, and that these signals can be measured from person to person, and that differences can be assessed. On the other hand, no sound reproduction system can ever be compared, side-by-side, with an actual event, except perhaps in an anechoic chamber, and even this is fraught with problems. No loudspeaker ever radiates sound in the manner of an acoustic instrument (no instrument squirts all its high frequencies out of one point, as do most loudspeakers). The soundfield distortion that is thus produced means that each one of us will hear different differences between the live and recorded sounds. The reason for this is that human pinnae (outer ears) are as individual as fingerprints. The way that they collect the sound is therefore unique to each one of us, and soundfield differences will, in turn, be differently perceived from one person to another.

Worse, there is no parallel in the auditory system to the optic nerve which carries a comprehensive electrical analogue of what the eye perceives. Nerves from the inner ear disappear into about six different parts of the brain, and there is nowhere to measure a complete electrical signal relating to what we hear. The final blow comes from the fact that our perception of music is highly unstable in terms of our moods. This kind of problem called for the creating of a whole branch of science called psychoacoustics. It exists

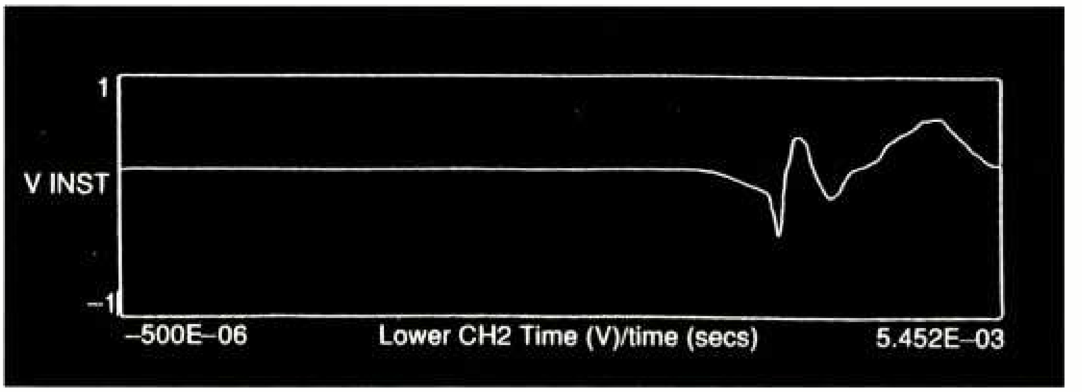

Fig.1e: Response of large widely used, 2-way studio monitoring system

because of the extraordinarily complex ear-brain relationships, and often attempts to define areas which allow equipment designers to take most advantage from what the brain needs to receive for a maximisation of realism with a minimum of complication. This is the only path that we can currently follow if we are seeking the highest fidelity of reproduction that we can achieve, because the cost of truly recreating an original soundfield would be enormous.

The late Michael Gerzon suggested that if we wanted to reproduce an accurate soundfield of a musical instrument, then we would have to surround it by about a million microphones, going to a million-track recording device, and reproduce the recording via a million amplifiers and loudspeakers. Even this system would assume perfect microphones, and that these one million perfect microphones did not reflect sound back towards the instrument or the other microphones. In other words, we cannot do it.

The above paragraphs should have made the point abundantly clear that loudspeaker reproduction of sound is not even close to recreating a 'real' sound except, perhaps, if that original sound was itself generated by a loudspeaker in anechoic conditions—an unlikely set of circumstances for a realistic recording process. Given these limitations, it should be realised that objective assessment of loudspeakers can only go a certain way towards describing the subjectively perceived aspects of their performance. The fact was highlighted over 20 years ago by another doyen of audio, Richard Heyser, who said that in order to fully enjoy the intended illusion of a recording, it is necessary to suspend belief in reality. He continued: 'All recording and reproduction via two loudspeakers is illusory: it will take powerful signal-processing systems and multiple-loudspeaker technology before we will ever be likely to see the three main groupings of classical, rock, and cinema in absolute agreement'. With such visionaries as Gerzon and Heyser seeming to agree completely on such an issue, it would take genius of greater understanding or a fool to argue with them.

In coming months, *Studio Sound* will endeavour to perform a set of measurements, some of which will be new to many readers. From here we will look for groupings that can then be correlated with the subjective opinions of the loudspeaker users. The experiment will, hopefully, be as enlightening for the experimenters as for those reading the results. ■

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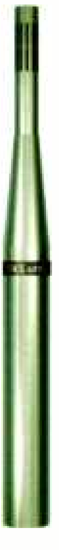

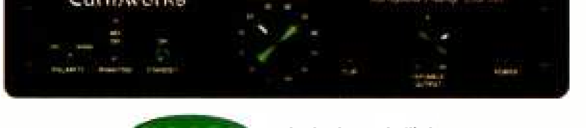

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Compact Disc: the next generation

Technology

The course of CD's development has taken in a glorious entrance, a seminal performance and graceful overture to its successor. **Francis Rumsey** investigates its understudy

THE AUDIO INDUSTRY has been tantalised by the possibility of a new consumer audio-disc format based on the DVD for some time, but because of the politics of the record industry and various electronics companies, the appearance of an audio-disc format has lagged behind the various DVD video and data disc formats. This is the opposite situation to that which prevailed with the CD, where audio was first off the mark, while data and video applications followed. In fact, as it turned out with CD, the market for CD-ROM players became much larger than that for audio players (around 70% to 30%), possibly leading the industry to tackle the launch of a new low-cost optical disc format the other way this time.

In any case, the requirements for an audio disc and player are rather different to those of a data-disc format. Whereas data discs are intended to be used for a wide range of different types of data in a random-access structure, audio discs are expected to offer a standardised method of data representation. Audio discs should be playable on anything from the cheapest Walkman to the most expensive hi-fi system without the need for particular operating systems, drivers, application software, and so forth. Data discs are really just flexibly formatted, general-purpose stores for data, whereas audio discs typically require the specification of exactly what data is stored on the disc. That said, it should be noted there is also a sizeable group of potential users that believe a new audio disc should be more of an 'open standard', providing for audio in a wide range of different formats including compressed options such as MPEG, DTS and SDDS.

Video, with many of the same standardisation needs as audio, has a head start this time round. This may well be due to the fact that video can be handled now, whereas at the time of the CD launch the data-rates required to store video in a digital form were simply too high for the optical disc technology of the time. Now that MPEG2 video compression has made it possible to reduce data rates to a very high degree, it becomes possible to store high-quality wide-screen pictures on improved consumer optical discs, together with new interactive features.

Ironically, while the video industry has been busting a gut trying to squeeze high-quality video into the relatively small space available on a DVD, the audio industry has been rubbing its hands in anticipation of what it might do with all the extra data capacity. Few in the audio industry seem prepared to accept a new format that uses only lossy data compression. Quite the contrary in fact: the audio industry wants to go up-market, adding more audio channels, more dynamic range and more bandwidth.

A new audio-disc format involving multichannel sound has considerable implications for production techniques and technology, and the situation is further complicated by the appearance of a new approach to digital recording, in the form of Direct Stream Digital (DSD) from Sony. This contrasts with the promotion of conventional high-resolution PCM approaches by other companies. These issues have been giving the industry headaches because of the uncertainty expressed by many, firstly, as to whether we need anything better than CD in the first place (some suggest the record industry needs a new format like a hole in the head), and, secondly, whether multichannel sound is really the important 'big idea' that its exponents claim (there are still those who will happily quote the ill-fated quadraphonic experience of the 1970s in support of their fear).

An International Steering Committee (ISC) consisting of the RIAA, the AFPI and the RIAJ has taken the initiative in laying down criteria which a new audio disc should attempt to satisfy. These include numerous 'usability', copyright management and security issues, but also make very clear that multichannel audio is an important feature of a new disc, and that any specification should allow for two high-quality stereo channels in addition to six channels of surround sound. It is stated that the six channels should be of as high a quality as possible within the remaining space on the disc.

It is taken as read that a higher sound quality than that of the CD format is desirable, and considerably greater flexibility is allowed in the DVD format compared with the CD (which only really offered one sampling rate and resolution). It is this potential for trade-offs between sound quality, method of coding, multichannel capability and other disc features which makes the format so hard to tie down.

Compatibility of the discs and players was also raised as an important factor. DVD players should play CDs, and DVD-Audio discs should have a CD-compatible layer playable by CD players. This latter requirement is proving to be one of the biggest points of contention, as the only way to fulfill this requirement is through the construction of a dual-layer 'hybrid' disc, one layer of which is a CD Red Book layer and the other a high density DVD layer. A serious technical concern among some parties is whether the hybrid disc can actually be made to work satisfactorily, and whether it will play properly in the majority of CD players. Sony and Philips are convinced that it is a practical proposition while others seem to be more sceptical. There can be little doubt that a reliable hybrid disc would be the key to a successful market introduction for the format, as it would provide a virtually seamless migration path from CD to the new format. The attraction of single inventory stock is considerable for record shops and companies, and consumers would presumably treat the disc as if it were a CD until such time as they bought a new player.

A certain capacity for integrating data and graphics is also a recommendation of the ISC, although it is not specified what the extent of such features should be.

THE DVD VIDEO FORMAT went a long way towards providing many of the high-quality audio features for which audio engineers were looking. It already allowed for 48kHz or 96kHz sampling rates and 16-bit to 24-bit PCM resolution, as well as up to eight channels of audio for surround sound. If you did your sums, though, it was possible to see that even if the disc was used only for audio you could not accommodate the highest resolution and multichannel sound at the same time as providing a 2-channel mix of the same material (which is desirable for systems not offering surround capability), while also offering a playing time comparable to that of CD.

The outline DVD-Audio proposal ('Pure Audio' DVD) introduced in New York at the AES Convention was clearly based on the audio aspects of the DVD-Video, and promoted by companies such as JVC, Pioneer and Panasonic. It was presented as a fairly flexible format, allowing a number of combinations of playing time, sampling rate and resolution; although it did not appear to have taken onboard as primary features a number of the more advanced recommendations of associations such as the Acoustic Renaissance for Audio (ARA) which concerned lossless coding and pre-emphasis in order to make possible 96kHz recording of both 2-channel and surround information. Lossless coding has been mentioned as an option for future extensibility, but not as a primary feature.

In summary, then: although the exact format is still confidential at the time of writing, its main audio features are expected to involve the use of linear PCM at either 48kHz or 96kHz, but with 44.1kHz and 88.2kHz also allowed (this was not a feature of the DVD-Video for- >>>>>

Fig.1: Basic DSD signal chain

<<<<< mat), and 16-bit, 20-bit or 24-bit quantisation. Eight channels of audio are provided for. A number of flags have been included to allow for future extension of the format, including lossless coding and hierarchical encoding of surround sound (such as Ambisonic techniques to handle signals in a format other than simple speaker feeds, in order to enable full periphonic reproduction). There is also a 'data definition' area to enable the flagging of 'future use' applications such as Sony's DSD (see below), which leaves the format quite open in many senses. Both 3-inch and 5-inch discs are proposed. It appears unlikely that existing DVD Video players would be able to play Pure Audio DVDs, but new 'universal' players would handle both types.

It is clear that without lossless coding some form of mixed sample-rate structure may be required if one is to include 5-channel or 6-channel surround sound and a 2-channel mix on the same disc (if the 96kHz sample rate is to feature at all). It has been proposed that surround channels could operate at 48kHz with the front channels at 96kHz, or that surround mixes might be limited to 48kHz sample rate while an 'audiophile' 2-channel mix could be provided at 96kHz. In the former configuration one could expect to obtain a playing time of 64 minutes on a single-layer disc (shorter than a CD) and 117 minutes on a dual-layer disc (both layers DVD, rather than one Red Book).

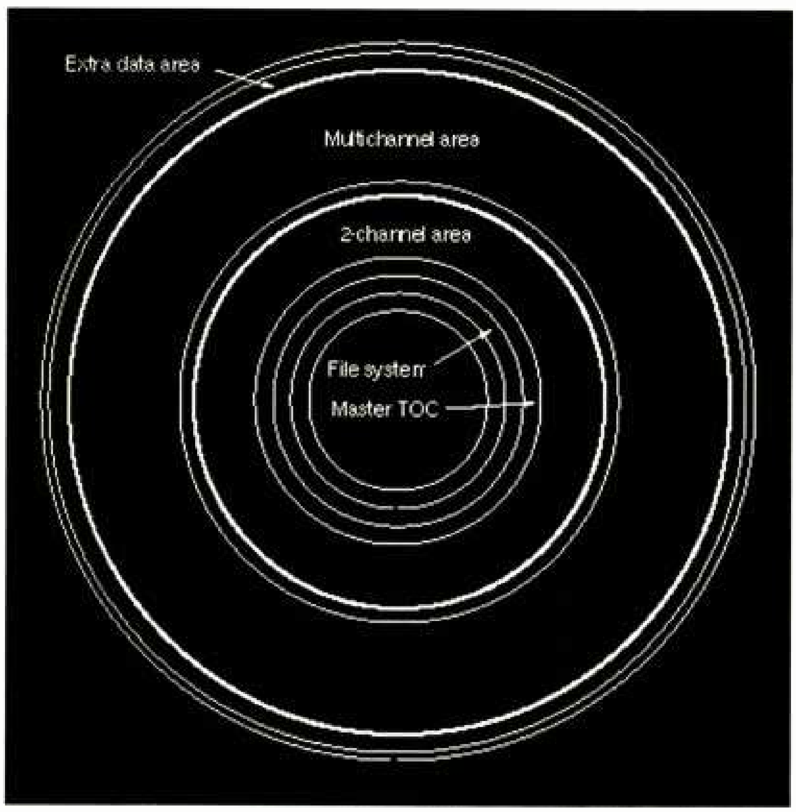

Fig.2: Super Audio CD is divided into a number of areas for different content types. Disc access is controlled using either a conventional TOC (Table of Contents) or filing system (UDF or ISO 9660). Data is grouped into blocks of 2064 bytes, of which 2048 bytes represent the main data. (Not to scale and based on preliminary specification.)

Whether or not this format can be combined with a CD-hybrid disc structure is not clear, but it does not appear to be proposed. The reasons for this can only be speculated upon, and a number of possibilities have been suggested. The first is that a hybrid structure involving a Red Book layer would involve two royalty or licensing charges—a conventional CD license fee to Sony-Philips and another for DVD-Audio. The second is the question of technical feasibility.

For some time now, Sony and Philips have been proposing an alternative solution for the next generation audio disc based around Sony's Direct Stream Digital (DSD) technology. This is based on the principle of recording the 1-bit output of a sigma-delta A–D convertor directly, without decimation and filtering for conversion to PCM (Fig.1). A very high sample rate of 2.82MHz is employed, delivering a usable audio bandwidth of around 100kHz, together with 120dB dynamic range in the audio band. The new disc was presented recently at a series of demonstrations at EMI's Abbey Road Studios in London.

Philips and Sony are as much a party to the negotiations over the DVD-Audio disc as any of the other companies, and they hold a number of the key patents relating to the technology of DVD, but there is a seemingly widening gap between them and those in the DVD Consortium who are promoting the solution described earlier. This gap is now so obvious that even though Sony and Philips would like their proposal to be adopted as the new audio disc format, they will almost certainly go ahead with their own system even if another approach is standardised by Working Group 4 (WG-4) of the DVD Consortium.

The Super Audio CD is the Sony-Philips title for a hybrid disc format that employs a standard Red Book CD layer, compatible with ordinary CD players, and a semi-transparent high-density layer seemingly based on DVD technology. This way the disc will play in either a CD player or a new Super Audio CD player. Rather significantly the literature does not mention DVD anywhere; although the capacity of the HD layer appears to be the same as a single DVD layer (4.7Gb) and is probably physically similar.

The disc is claimed to offer many of the features required by the ISC, remarkably including 6-channel surround in addition to a 2-channel mix on the HD layer, with a playing time the same as CD (74 minutes). In the last year or so Philips has developed lossless coding for 1-bit audio signals that is capable of reducing the data rate of DSD sufficiently to enable both mixes to be included within the capacity of a single 4.7Gb layer, while also leaving space for some data and graphics information. Philips is understandably cagey about how it is achieving this at the moment, because the sums do not appear to add up if one assumes that a maximum of around 2:1 saving is available from lossless compression. (Indeed only a year ago, many were claiming that lossless compression would be ineffective with DSD data). One must assume that the sample rate of the multichannel section of the disc is lower than that of the 2-channel section; although this would not produce a directly proportionate gain with lossless coding, because the degree of redundancy in the data gets smaller as the sample rate gets lower. The preliminary specification suggests, by omission, that something like this must be the case, as the 2-channel 'Super Audio' section of the disc is quoted as having a sampling rate of 2.8224MHz, while the multichannel section is left unspecified, but still claimed to have 100kHz bandwidth and 120dB dynamic range. There is much greater flexibility in terms of how one quotes bandwidth and dynamic range with highly oversampled 1-bit signals, since the Nyquist frequency is still half the sampling frequency (which is well over 100kHz even if you reduce the primary 2.82MHz frequency considerably) and the limits of the system are more likely to be determined by the point at which the signal-to-noise ratio becomes too poor to be useful. The order and shape of noise-shaping filter can be modified accordingly.

The disc appears to be arranged in such a way as to divide the area between 2-channel, 6-channel and Extra data (video, graphics and so on), as shown in Fig.2. This differs somewhat from the proposal described earlier which appears to multiplex such data in one DVD transport stream. In a Super Audio CD player, therefore, different sections of the disc would be played depending on which mix was desired. Cheap Walkman players would not have to be able to decode the multichannel section at all, looking only at the central 2-channel section which might be easier to track than the outer section.

The Red Book layer of the disk will be no different from a normal audio CD, but Sony has developed a modified form of Super Bit Mapping called SBM Direct that will enable high-quality down-conversion of DSD signals to 44.1kHz, 16-bit PCM, using high-precision digital filters to preserve as much as possible of the original signal information.

Numerous technical discussions have raged over the relative merits of DSD and multibit PCM, and there is not really space here to analyse them in detail. What is important, though, is the implication that a move to 1-bit technology would have for the industry as a whole, since substantial re-equipping would be required throughout the signal chain. Of course, recordings could be made using conventional PCM and then upconverted to DSD just to put it on the disc, but that would somewhat negate the claimed advantages of using DSD in the first place.

DSD simply eliminates the digital filtering-decimation that takes place in con- >>>>>

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<<<<< ventional PCM systems between a sigma-delta A–D convertor and the rest of the system, thereby retaining the oversampled signal in as raw a form as possible. The resulting 1-bit signal requires different forms of signal processing to multibit PCM, thereby potentially making current digital mixing and processing technology largely redundant. The race that is on with Sony at the moment is to try to develop DSD processing applications as fast as possible, in order that enough of the signal-chain components are available by the time that the new consumer disc is launched (proposed for Spring 1999 in Japan). Some Sony development engineers like to present DSD signal processing almost as long-term research, as many of the techniques have not yet been developed to a sufficiently high degree. At the same time, the pressure to get the new disc format standardised and accepted by the industry is being driven by the competition from other members of the DVD Consortium, whose proposal uses technology that is already available in the industry. By introducing the new format now the industry may be persuaded to hang on and see what happens long enough for the system to be perfected.

A HYBRID DISC that can be made to work would be a very persuasive element in favour of the Sony-Philips Super Audio CD. It is really the primary key to commercial success in the launch of any new audio disc format, since the majority of consumers would not notice the difference, whereas audiophiles would be able to gain access to higher quality material in multichannel form. It effectively enables a seamless migration from one technology to another. The video market, on the other hand, has no need of this compatible migration path because CD-sized optical disc formats are not used very widely in the current video market place.

It is problematic from the recording and mastering industry's point of view that DSD is more of a revolutionary rather than an evolutionary step. Linear PCM with higher sample rates and resolutions is somewhat easier to accommodate and some existing equipment and interfaces can be modified to handle it. Nonetheless, the change from analogue to digital audio was a major stimulant to the industry and it is possible that another step change such as a move to DSD could have a similar effect if enough people jump on the bandwagon. Of concern, though, is that DSD technology may lie solely in the hands of one or two companies, potentially taking us back to the days when if you wanted to make CDs you had to use their equipment. Such a monopoly is probably not in the broader interests of the industry, as most people welcome a choice of supplier and competition between manufacturers is a healthy thing. It will be interesting to discover what is patented and licensable about 1-bit audio, since the concept is actually rather a simple one, but signal processing algorithms are probably new intellectual property.

There are those who will point out that all this is not actually about the high end of the market at all, but about the very low end of it. DSD signals are actually extremely easy to convert back into the analogue domain (at least with basic quality). The simplest conversion can be achieved with a low-pass filter on the 1-bit output; although good 1-bit convertors are actually quite advanced devices. You can even put a pair of headphones across the 1-bit digital output and hear a signal (although not a very pleasant one). It is possible that the ease with which such signals can be replayed may knock a few dollars off the manufacturing cost of Walkman-like players, which is not to be sneezed at. In this respect one can see that DSD-based consumer equipment could range from the very highest to the very lowest quality, potentially with a much greater range between the best and the worst than currently seen with PCM equipment.

Sony and Philips have proposed to licence Super Audio CD to existing CD licencees for the same royalty level as currently charged for CD. It is proposed that new licences would also be issued under the same terms. Methods of visible and invisible watermarking have been shown, which makes the surface of the disc difficult to counterfeit. The two companies between them have a substantial section of the record industry sewn up, which is not something that many other members of the DVD-Audio Consortium can claim. Altogether then, they seem to have put together a rather persuasive package that challenges the whole industry to decide which way it wants to go. The last thing we really need is a format war right now. ■

Software: expectation and reliability

Don't be deceived by the apparent conciliation taking place between analogue and digital technologies. **Todd Wells**, chairman of Soundtracs, gives the inside track on hard and soft, analogised and digitised

EACH TIME I open a trade magazine, there is news of yet another digital mixing console. I always look for the shipping date and then have a quiet laugh—not because I disbelieve the intent of my industry colleagues, but because I have a good memory of the history of DAWs in the mid-eighties (remember Wave Frame and Synclavier-NED) as well as Soundtracs' own substantial digital experience.

The reality is that getting an analogue product out the door in perfect condition was always difficult because of the permutation of usage (inputs, outputs, knobs, switches...). No matter how well you designed or tested it, there was always a particular situation where the console would oscillate or be prone to RF interference, or the crosstalk was unacceptable with certain cabling, and so on. That's assuming you design and build them well with good components and know what you're doing.

As for digital consoles, well, let's look at the simplistic view of the way it works. You will have a vast number of switches, turning on and off at amazing speed. The DSP at machine-code level is responding to a zero (no) or one (yes) and given the speed (millions of millions of operations per second) and the length of the instruction 'words' of zeros and ones, then it's not surprising that if you loose just one 'one', then there may be a problem. Add to this statistic, gremlins, lock ups and the severe complexities of code then its very very impressive our industry has got as far as it has.

There is no software that is bug free (ask Bill Gates or fly in an Air Bus). Our industry is very small and under funded compared with the PC world, civil aircraft or the medical profession which all make heavy use of software. They have the resources, but they also have the problems. Yet

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there are many in our industry who grudgingly accept digital, but expect it to be as reliable as analogue (which suffered its own faults).

Well friends, I've news for you. You cannot have your cake and eat it. Digital in consoles brings cost effective automation, sonic purity (if you want that) headroom and memory of what you've done. You'll never be able to do those things in the analogue domain at the prices now being offered. So, Luddites, you'll sadly have to accept (as those of us using Windows 95 have) if you want to use a software-based product in pro-audio, it will probably contain bugs. The important thing is that it is stable and you can work around it. If you want perfection stick to analogue (but don't expect it to make your life easier).

As for shipping dates and the actuality that most manufacturers need the customer to assist them in discovering the bugs over a period. This may seem unfair, but very few manufacturers have the internal test and interfacing facilities for the eventualities and environments. Here, again, it's important the manufacturer is trusted and willing to respond quickly to problems. One thing's for sure—never buy from a manufacturer who claims the software product is bug free—there's no such thing. More so for the date of shipping, add a year to 18 months for a stable expansive product especially for those manufacturers announcing their first digital product.

Enjoy the digital world and its benefits, but don't expect analogue stability and perfection—it's impossible. Instead, I'd advise you to look for Digital Competence (DC) and experience in your dealings.

And as a post script, I'd point out that there are 150-odd Virtua users worldwide who have experienced Digital Competence first hand. ■

MS1202-VLZ 12-CH • MS1402-VLZ 14-CH • COLLECT 'EM • TRADE 'EM WITH YOUR FRIENDS

VLZ MICROSERIES: SMALL MIXERS WITH WHAT IT TAKES TO HANDLE SERIOUS PROJECTS.

MICRO SERIES 1202 VLZ 4 MONO CHS. 4 STEREO CHS.

MICRO SERIES 1402 VLZ 6 MONO CHS. 4 STEREO CHS.

BOTH MIXERS ARE QUITE AFFORDABLE AND ARE NOW IN STOCK AT YOUR LOCAL MACKIE DEALER. CHECK THEM OUT TODAY!

BOTH MODELS HAVE:

Studio-grade mic preamps with discrete circuitry for high headroom low noise (–129.5 dBm E.I.N.) and wide frequency response (over 300kHz!).

Low Cut filters (18dB/oct. @75Hz) on mono mic/line channels allow use of low-frequency EQ on vocals without boosting room rumble, mic thumps, P-pops and wind noise.

Trim Controls on mono channels have 60dB total gain range for boosting weak sound sources and a 10dB "virtual pad" for taming hot digital multitrack outputs.

2 Aux Sends per channel (one globally switchable pre/post, one post-fader), each with 15dB of gain above Unity to boost weak effects.

3-band equalization with 12kHz High shelving EQ, broadband musical 2.5kHz peaking Midrange & 80Hz Low shelving EQ.

Constant loudness pan controls.

Stereo in-place Solo.

Mute button routes signal to "bonus" **Alt 3-4 stereo bus** outputs & Control Room matrix. Handy for both recording and live applications.

MS1402-VLZ ONLY: 60mm logarithmic-taper faders based on our exclusive 8•Bus design. Long-wearing wiper material and tight polymer lip seals to protect against dust & other crud.

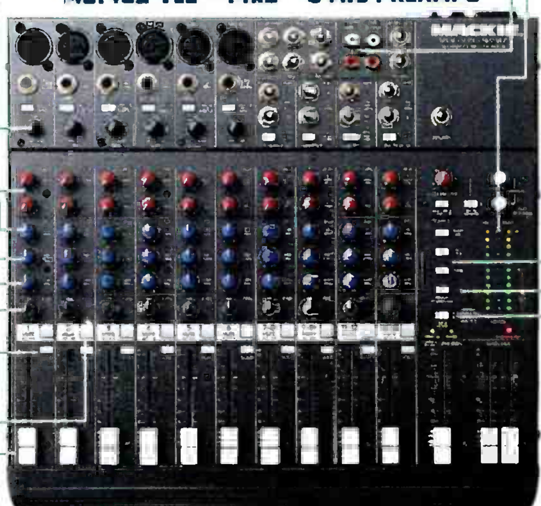
Built-in power supplies — no outlet-eating wall warts or hum-inducing line lumps.

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).

MS1202-VLZ • 12x2 • 4 MIC PREAMPS

MS1402-VLZ • 14x2 • 6 MIC PREAMPS



All inputs & outputs are balanced[1] **to cut** hum & allow extra-long cable runs, but can also be used with unbalanced electronics.
[1] except RCA tape jacks, headphone jack & inserts.

VLZ (Very Low Impedance) circuitry first developed for our 8•Bus console series dramatically reduces thermal noise & crosstalk in critical areas.

Control Room outputs feed monitor speakers without tying up the headphone jack.

BOTH MODELS HAVE:

Radio Frequency Interference protection via metal jacks & washers plus internal shunting capacitors.

High-output headphone amp can drive virtually any set of phones to levels even a drummer can appreciate.

Aux 1 Master level control & pre/post switch.

Effects Return to Monitor switch folds Aux 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/ Phones Matrix adds monitoring, mixdown & metering flexibility. Select any combination of Main Mix, Tape In and Alt 3-4 signals 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 **Solo Mode** selects PFL or AFL solo modes.

Solid steel chassis & thick fiberglass internal circuit boards resist abuse.

Channel inserts on mono channels.

Not every production project requires dozens of input channels and boatloads of buses.

But doing ANY audio job well requires a mixer with superb specs...and the right combination of useful features.

Our MicroSeries 1202-VLZ and 1402-VLZ might have small footprints, 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 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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The compressor that forgives, but never forgets.

50 factory presets, 50 user programmable presets

Dual mono or true stereo linked operation

OverEasy® with VariKnee™ or hard knee thresholds

All the classic dbx trademark sounds in gating, compression, limiting, de-essing, and sidechain EQ.

Compression controls: threshold, ratio, gain, OverEasy®, auto, attack, hold, and release

Variable Transient Capture Mode™.

Wide-ranging gain control allows for +4/-10 operation

8-step analog input and output metering

Digital input and output meters measure internal digital processing levels

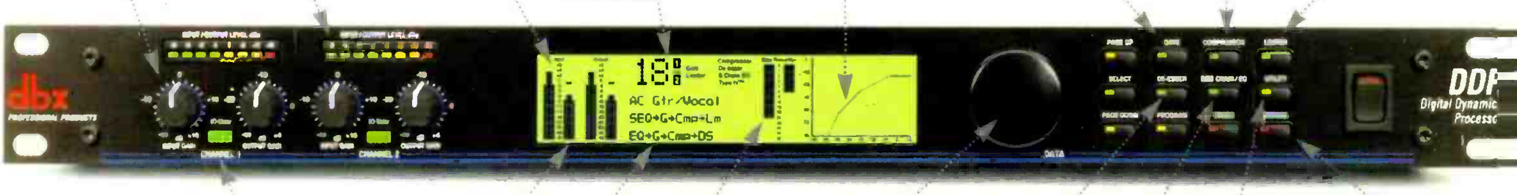
Hybrid display technology. Combines the best of graphics, character and icon based displays

Hi-res graph shows composite output vs. input plot of audio signal

Gate controls: threshold, ratio, attack, hold, release

Ultra-smooth continuous Auto mode.

Limiting from - 60dB to 0dB, wit gain, attack and release.



Ultra wide dynamic range 24 - Bit A/D and D/A converters with TSE™ Tape Saturation Emulation on board

Digital meters show both peak and average levels

High resolution gain reduction metering

Precision control over every parameter

De-ess from 800Hz to 8kHz, vary the amount.

Utils: sample rate, A/D input and output, Midi functions, Sysex functions.

Midi bypassable v midi program changes.

Optional AES/EBU or S/PDIF output for assured compatibility and flexibility. 48 / 44.1 kHz output

TYPE IV™ output available when equipped with digital output option

Build your own presets using your favourite building blocks

Change programs, parameters, and bypass via midi controllers

XLR and 1/4" balanced ins and outs.

Sidechain functions for advanced filtering applications. Sidechain monitor included

EQ offers 20Hz to 20kHz, Q, ±12dB, and SCMonitor.

Gate

Set parameters for threshold, ratio, attack, hold, release, and output gain. See the effect of your settings on the graphical display, as well as on the gain reduction and audio level meters, they all interact in real time with your manipulation of the parameters. Start with a threshold setting of about -60dB to clean off the noise in between the vocal takes. You can save your final gate settings as a "gate preset" building block and recall it into any other setup you do.

Compressor

The effects of the gate settings are visible on the graphic display to help you determine where to set your compressor threshold. Move through all the regular parameters (displayed in real time), like threshold, ratio, attack, release, and output gain. For vocals use a threshold of about -25dB, a ratio of about 3:1 or 4:1, and a slow attack and fast release for the most natural sounding effect. Your compressor settings can also be saved as a building block to be called up into any other preset.

Limiter

Changes you make to the limiter settings are also seen on the graphical display. You can adjust the level and also the speed at which the limiter lets go of the signal as it goes below the threshold. This is truly smooth limiting, with patented dbx PeakPlus™ algorithms, so rest assured that wherever you set your threshold level, your tape will not distort. And like the other parts of the processor, your limiter settings can be named and saved for later recall.

De-esser

De-essing works the same way; see the effects of your settings displayed on the graph. Parameters here are the common ones: threshold (800Hz to 8kHz), and amount (%). Other processing includes EQ - both in-path and sidechain - for special-effect types of processing. When you are editing any of the building blocks, its icon is visible on the display, and the parameters are shown on the graph, so it's always easy to know where you are.

and More

You can also work in stereo, or set up a completely different and independent processing chain for the other channel. Optional digital output with the TYPE IV™ Conversion System with TSE™ (Tape Saturation Emulation) provides up to 24-bit output in either AES/EBU or S/PDIF formats with the trademark digital processing of TYPE IV™. The DDP also has full MIDI/Automation capability, with separate midi in and thru jacks.

It Forgives

• New dbx technology, the TYPE IV™ Conversion System with TSE™ (Tape Saturation Emulation) gives you the pleasant overload characteristics of analog tape without the harsh distortion of most digital input systems. No more dancing around with the input levels to protect the integrity of your audio.

• Ultra-wide dynamic range 24 bit A to D converters with TYPE IV™ make your signal sound better than you ever thought possible.

• With the extensive metering of the DDP, you can see EXACTLY what is going on with ALL parts of your signal.

• And speaking of stereo, you can work in stereo with dbx's True RMS Power Summing™ for phase-coherent tracking, or in dual mono mode, without the two channels interacting at all.

It Never Forgets

• The DDP works right out of the box. It comes with 50 factory setups that are guaranteed to knock your socks off. There are presets for every application you can think of, and then some.

• Want to duplicate that perfect compressor set-up? Each processor in the chain has all the parameters you would expect. After you set the parameters the way you want them, save it as a processor preset, available to be recalled any time.

• When you make changes to any parameter, you can see where your adjustments are effecting the signal, simply by looking at the Hi-Res graphical display, which shows the processing curve in real time as you make your adjustments.

Check out the DDP at your local pro audio dealer, and experience DIGITAL performance you'll never forget.

dbx PROFESSIONAL PRODUCTS

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