April 1998

THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION & BROADCAST

EXCLUSIVES
Microtech Gefell UM900
Dartech DART Pro 32
Aphex FM Pro 2020
Summit EQP-200B
AMS Neve 1081C
Genesis DPM 101
Avalon VT-737SP
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LOUDSPEAKER TESTS
Introducing the definitive review

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

The

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Interview

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AMS Neve's digital dubbing consoles are setting new standards in quality and creativity at leading facilities worldwide.

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Lucasfilm Digital, San Rafael
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The Magnolia Studios, Burbank
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Todd-AO, Hollywood
Warner Bros, Burbank

UK
Boom Studios, London
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Videosonics CinemaSound, London

Europe
Cineberti, Belgium
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Exa, Spain
Filmtel, Spain
GLPIPA, France
The Icelandic Film Corporation
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Nordisk Film, Denmark
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Soundtrack, Spain
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Tremens Film, Austria

India
Gaurav Digital, India
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Below: Taking a shine to the Fleetwoods: plug in to processing: and two flavours of American outboard from Avalon and Aphex
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 purview 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 pre's 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 thrilling spreading mass of potential that is difficult to tie down with the main players resorting to convoluted 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.

Zazen 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 subject 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-ear-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 anachronistic 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
Great Studios Of The World

PRODUCTION NOTES

Olympic Studios, the prestigious London all-SSL recording facility, was one of the first UK studios to install an SL9000 | 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 SL9000 has remarkable capabilities. The bottom end is fantastic, the top end is really open and the automation is brilliant. The SL is the best sounding console that I have ever worked on."

Solid State Logic

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music week awards
BEST STUDIO
98

117 Church Road, Barnes, London, SW13 9HL, UK
Phone: +44 (0)181 286 8600
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 petrolometer and clocked up a frightening 13km 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 bug-snatchers on the public days. Frankfurt remains a phenomenon.

Zenon Schoeppe

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 deepset Buckinghamshire.

DEAF: www.interstudio.co.uk/lst/deaf.htm

Parsons Leaves Abbey Road

UK: After a remarkable brief period, Alan Parsons is leaving his job as EMI Studies 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.

Another fine Messe...

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.

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 Altman and a sound system from Autograph Recording including a 360° cluster of 11 Meyer MLS-4s, two tower arrays of 22 UM-1s and concealed UPBm-1s to lower the sound image. Om-1 systems also use Meyer PWS-1s and MLS-4s with mics being predominantly Sennheiser SK90s and MKE2as.

Reference and information

Recent publications of note include an instructional video, Thin Lizzy a Producer, from American Guy Marshall who teaches at the Musician's Institute and Salsa 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 audio activities including vocals, drums and multitracking. Thin Lizzy a Producer is available from Tutte & Babe Music in both VHS and VHS versions costing $24.

Philip Newell's new book, Recording Spaces (Heinemann, ISBN 0240519072), deals with the acoustics of recording rooms from the perspective of an author active in the music 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 pro to electronic instruments from a wide range of manufacturers—including HHB itself. Tutte & Babe Music.

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HHB Communications, UK.
Tel: +44 182 962 5000.

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April 1998 Studio Sound
UK: Provided and self-material. The earlier funeral had been moving affair attended by just some of Sanders’ many friends and, for the most part, the memorial shared the same loss held by those present—despite the humour most notably injected in the address made by Gyles Bradreth, 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 Orchestra lead by Christopher Warren-Green with a selection of soloists. The performances included a playback of Procol Harum’s ‘White Shade of Pale’ in which Sanders had been involved, and a piece entitled ‘For Golin’ composed by Nick Reis. 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.

**Collin Sanders Memorial**

**UK:** The Oxford University Church of St Mary the Virgin was almost full when it opened its doors for SSL-founder Collin 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 same sense of loss held by those present—despite the humour most notably injected in the address made by Gyles Bradreth, one-time business partner and self-proclaimed nemesis of Sanders’ otherwise remarkable business success.

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**Austria: Vienna’s MG Sound has installed an SSL 9000 series console in its Studio A.** Displacing the facility’s 4064G Plus console, which has been relocated to Studio B, 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 production operation of Martin Bohm and Steve Cross. Previous clients include Placidio 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 D01250 harddisk 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 1244. 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 4:2:3 news-gathering mixers. Fairlight, UK. Tel: +44 1474 821500. VW Marketing, UK. Tel: +44 137 72848.**

**Cumbrian-based British broadcaster, Border Television, has included a Cairec 5-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.**

**Germany’s PolyGram Recording Services (formerly part of Deutsche Grammophon) has taken delivery of four 48-track, 24-bit Aurgan digital recorder-editer systems. Following the recent purchase of a 24-track system, the new order is already set to be upgraded to Aurgan’s forthcoming 48-track remote control, the RC48-C, in the design of which PRS already has a stake. Sascor, 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 della Vittoria 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 in a variety of shows called Fantastico that reaches a 20m-odd viewing audience. EVII 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 AESA ADAT M20 ‘professional’ 24-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 MX3 Plus DAW system on its stock. Offering 72 track-hours of 24-track, 24-bit recording, the MX3 Plus is paired with an FED-V-MOD 100 for video playback. JTR expects the package to prove popular with California’s burgeoning post and film fraternity. FX Rentals, UK.**

**Tel: +1 818 755 9011.**

**West London’s new Gallery Studios, opened by ex-Roxy Music guitarist Phil Manzaneras, has added a Digidesign Pro Tools 24 system to its kit list. Running on a Mac 9600 server and using an 888 24-bit I/O interface, the system runs in conjunction with Emagic’s Logic Audio and a series of TDM plug-ins including the Focusrite D2-D3. Abbey Road studios has recently acquired a TL Audio M2-8-channel mixer, which is finding use in both classical and popular sessions, particularly with digital recorders.**

**Tel: +44 1753 683222.**

**California 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 capable of handling 5.1 discrete surround work that has already seen sessions from Ici lang and Chicago, while Blackhole’s reputation as a studio has been helped by its own WF, 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 taken delivery of a Digidesign Quantegy recording media. The move follows 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.**

**Tel: +31 24 373 0484.**

**Australia’s Sydney Opera House has installed a Terryn SuperDual system in its Drama Theatre. Based on S3000, B950 and T12 cabinets, the system is the only such system in the Opera House as the S44-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 McKnight’s Masterclass. Elsewhere in Sydney, commercial music facility Song Zu has bought two Fairlight MX3 Plus workstations, giving it full access to Fairlight’s system. Foxel has installed a second MX3 Plus in a new editing suite. Terryn, UK. Tel: +44 1238 402199.**

**Canada’s Cumm Television has chosen the Studer D950 for its unconventional broadcast facility. Located in the CummCity building in Toronto, Cumm’s philosophy uses ‘quick-track receptacles’ in lieu of a regular studio format. A popular approach to programme making has already proven successful in the MuchMusic and CableParis24 news channels. Cumm TV, Canada. Tel: +1 418 891 5757.**

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Tom Jung
Technical consultant for
Pro Audio Review
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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: PKO Events. Tel: +86 171 370 8231. Email: shanghai@eco.co.uk

22–24 Entach Halls 1 & 2, Sydney Exhibition Centre, Darling Harbour, Australia. Contact: Caroline Grafton Tel: +61 2 9876 3830. Fax: +61 2 9876 5715. Email: mail@conpub.com.au Net: www.conpub.com.au

27–28 DVD Forum The Berkeley Hotel, London, UK. Tel: +44 171 691 9191. Email: dvo@hqpc.co.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.prestitel.co.uk

26–28 TV 98 Thermal Hotel Hella, Budapest, Hungary.

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Contact: Scientific Society for Telecommunications Tel: +361 153 1027. Fax: +361 153 0451. Email: hiradatechnika@mttesz.hu Net: www.mttesz.hu/hiradatechnika


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@sisesexpos.com Net: www.sisesexpos.com

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

June

10–15 International Electronic Cinema Festival 1998 Chiba city, Japan Tel: +81 21 963 32 20. Fax: +81 21 963 88 51. Email: message@symposia.ch Net: www.montreux.ch/symposia

15–17 Meccon 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.meccon.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


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: +1 612 730 5699.

November
4–5 22nd Sound Broadcasting Equipment Show (SBES) Hall 7, National Exhibition Centre, Birmingham, UK. Tel: +44 1451 838 575. Fax: +44 1451 832 575. Email: dmcv@pointproms.co.uk Net: www.iway.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

World events

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I recommend everyone try one - they won't be disappointed.

MIRE KATING Stjng Tour Engineer

9

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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.prosound.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—Voting starts now

Fax your vote to:
+44 171 401 8036

SSAIRAs—Voting starts now

April 1998 Studio Sound

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<th>SSAIRAS</th>
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<td>Large scale console</td>
<td>Monitors</td>
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<td>2</td>
<td>Medium to small scale console</td>
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<td>Outboard Reverb</td>
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<td>7</td>
<td>Combined outboard device</td>
<td>Your Unique Reader Registration Number</td>
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SSAIRAS FAX: +44 171 401 8036
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 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 reverber 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 reverber 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-phatting 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
Introducing Finalizer™ Plus
Improving on the Multi-Award winning Finalizer platform, the Finalizer Plus delivers an unprecedented level of clarity, warmth and punch to your mix. With an all new set of advanced features and enhancements, Finalizer Plus puts the world of professional mastering within reach of every studio - large or small. Inserted between the stereo output of your mixer or workstation and your master recording media, the Finalizer Plus dramatically rounds out your material, creating that “radio ready” sound - previously unattainable outside a professional mastering house.

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Bernie Grundman Mastering
Six Time TEC Award Winner

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• Variable Slope Multiband Expander
• Variable Ceiling Limiter
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.

Unlike 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 Bruel & 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.

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 PI202 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 16384 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 100Hz 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 log-amplitude of the frequency response. The resultant cepstrum has the time of the impulse visible on the cepstrum (spectrum), cepstrum (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

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**Fig.1: System 600A on-axis frequency response and harmonic distortion**
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(4), 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.

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.

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 total 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, 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 power cepstrum is
Fig. 6: System 600A acoustic centre

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

**** 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 frequency axis of a spike in a power cepstrum represents the delay in seconds from the direct signal to its reflection, and the height of the relative strength of the reflection.

Fig. 8: Reveal horizontal off-axis frequency response

Fig. 9: Reveal vertical off-axis frequency response

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

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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

The 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

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”.

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mic and innovative force in professional audio.

HHB Worldwide

H HB 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.

H HB 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.

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Beware! All digital recording media is not the same. Professionals rely on HHB's award-winning Advanced Media Products for high performance and archival security.

WANT TO BE IN THE NEXT HHB NEWSLETTER?
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.
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 (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

**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.
Professional audio CD recorders just became affordable.

Recorder is a star 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." Mike Hedges

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 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 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 6-0s
- 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 160 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

20th Century Fox in 20-bit with the Genex GX8000

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..."

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, 96kHz 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 88.2kHz. 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 HDCC, 24 bit - 88.2kHz) 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 two 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."

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.

"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 6-track recorders."

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, MD 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.

Genex a major hit Worldwide

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 6-track recorders."

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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.

Denny Purcell mastering in 24-bit/96kHz with the Genex GX8000 and HHI M02.6GB Disks

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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

PORTADAT

Sounds great, handles superbly, bl

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 HH3 PORTADAT PDR1000TC for performance and quality sound. The conditions for 'Titanic' proved very harsh with water leaks to the timecode equipment but the HHB PORTADAT was considered vital not only for its sound quality, but for its design and easy audio effects team went out hunting for sound effects. "We used the HHB PORTADATs to gather effects for the 'Titanic' project," says Boyes. "We rented the ship Jerem an O'Brien because the engine design is similar to the Titanic's. The piston shall 15 feet tall and we wanted to place it in extremely tight and dangerous places. The PORTADAT was
It to last. No wonder Hollywood loves the PORTADAT.

with Skywalker sound
Boyse 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.’’

Boyse 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 ambiances and interesting animal sounds with the PORTADAT. The audio for the sequel was 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

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.

Africa, Outer Space, The North Pole — all in a day’s work for the PORTADAT.
High performance and unparalleled dependability

**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 shuffling.
- 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 Phthalo Cyanine 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 shutter 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.
MDD140  M02.6 GB  DA113  ADAT45

- 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 MDD140 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.

- 5.25" 1024 bytes per sector MD 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.

- 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.

- 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-line 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.

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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 quoted limits of 50Hz 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 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 1kHz. This is common to all designs with spaced drivers (as opposed to dual-concentric for example).

The step response (Fig.10) shows excellent time-alignment and crossover design with a sharp rise and steady decay. The power spectrum (Fig.11) shows a slight dip at about 100Hz, but is otherwise okay. The acoustic centre (Fig.12) shows a maximum group delay at low frequencies that corresponds to a shift of about 1.2ms; 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.

T he 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 sys-

SOUNDS.

When clean uncoloured sound with superb source separation is demanded, look no further than the 4011 cardiod microphones from DPA Microphones (formerly known as Danish Pro Audio). Proven and developed over years of service in professional audio environments, the outstanding performance of these high quality products is characterised by a flat on and off-axis frequency response combined with excellent phase response. The 4011 is just one of the high quality products from the renowned 4000 series, evolved from over 50 years of innovation and customer service - available now from DPA Microphones.

Series 4000 Microphones from DPA

HP no 4374 14/98

Studio Sound April 1998
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-200A Mic Pre-Amp/Comp-Limiter. It is a high quality and great sounding input device that will further enhance our music."

Hear the Warmth"
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 maintenance budget get 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 collectible 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 AKG is concerned, given improvements in manufacturing tolerances, but with something already as consistent as a Neve circuit the AMS Neve purity 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 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 remodelled 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 sound of 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 removed the wiring and found the sound changed, which considering the 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 14-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/switch stepped in 4dB increments. It has two distinct sections in its 6dB travel, one part selecting the line input with variable gain and the other the microphone input with a huge amount of gain available 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 >>>>

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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 1”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 plots. 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 MZ-R30 from Sony, the inventors of the MiniDisc format, and already established as a high-selling unit. Weighting just 2.5kg (!) knew a gill 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, or relating to amending or reading the table of contents, strobe indicating that Long-Play mode is enabled and almost doubling the recording time of 74 minutes possible for stereo recordings. The battery level >>>>
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 for setting the 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 pre-recorded sources to be recorded as digital or analogue clones. A pair of push-buttons acts as up-down arrow keys for hexadecimal 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 10W 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 44dBV 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-REC enables access to writing on 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 500 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 bag; but 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, which would be a welcome ‘semi pro’ start—it would be care- less 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 (£199 UK) or the Sony MZR-30 (£399 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.
"If the choice is left to me, I use BASF Studio Master 900 maxima. It is such a high-class analogue tape that I could not find a better one even after comparing several tapes with it. You get a super performance from BASF Studio Master 900 maxima even when you push up the level. The clarity is phenomenal. I don't use anything else now."

Ronald Prent has had success as a recording engineer working with such artists as David Bowie, Police, Elton John, Def Leppard, Iron Maiden, Peter Maffay, Jule Neigel, Rammstein, Guano Apes and Fury in the Slaughterhouse.

For more information contact 01295-227838 or visit EMTEC Magnetics' web site at http://www.emtec-magnetics.com
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 embazoned with the invitation 'restore your record collection on CD!' 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, nothing surprising to the user anyway.

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 heard on an existing WAV file opened it appears in a Soundfile window with waveform display and you are invited to register it as a 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 detector 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. Post-filtering 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, the manual warning that manual waveform redrawing is time consuming.

This Dartech 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 proceeding (left) or succeeding (right) ones. If a contiguous succession of defective blocks are manually marked 'left' or a 'snake eating' approach is used, then reconstruction may be applied, however there are obvious limits to this snake eating 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 similarly 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 Logfile 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 Culling is low-pass filtering. 

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www.americanradiohistory.com
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 (IR) type which introduce non-linear 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 impressive and very effective and is used to ‘mouthing around’ deciding what to do next this is irritating.

As with all restoration processes it is up to the reader 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 with a number of differing types of material and within the timeframe available 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. How 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 an 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 Lim-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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There simply is no better sounding or better value valve signal processors.

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 touches 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 makers of processor that come with factory presets usually put them in 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 LA 190 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, but 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.

Aphex Systems, 11068
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Studio Sound April 1998

ALSO AVAILABLE

Aphex Systems' 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.

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D-90 features...
- 8 track digital multitrack with no compression
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- 9' Virtual reels
- Versatile chase mode
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- 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 & FLEX 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 (+48V 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 converters, 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 slave to incoming MTC and on 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.
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 cleaner. The heavily 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 VU—very clever.

On the rear panel, there are XLRs and DIS 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 hypervis the input and output transformers for a subtly different sound, but still operate at ±14dB RCA phono sockets provide for compressor and de-Esser linking. Grid and channel 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 has serial number 001, so perhaps this is an oversight.

The input section includes a locking toggle switch for phantom power, switchable 80Hz and 120Hz bass roll-off, and a muting switch: centre position acclots 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 preamp. It can be switched in without a click. The opo-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 sustain 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 time characteristics of the original Manley Electro-Optical Limiter are

Studio Sound April 1998
Microtech Gefell UM 900

It's stylish, intriguing and packs a valve. Dave Foster 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 48mA strains credulity—many solid-state microphones drew 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 dialed capsule sitting in its own disc-shaped enclosure atop a chunky corrosion-proof body containing the electronics and switches. Maximum value is claimed from the two diaphragms, which have the familiar 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 cardiod 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.

The four diaphragms are made of Mylar, the most recent model being full Neumann mounts, while the others provide full car's radiator classic 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 those. 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 dodge 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 it's not because there's anything missing from the frequency response but because of smoothness and texture.

The polar patterns in the manual are fairly modest in their claims, showing all the patterns to be very similar at all frequencies. There are no nulls, except where 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 tend to ignore completely.

Typically, the most consistent pattern on paper is fig-8, but in practice it 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 D87, and warrants the price. There's nothing cheap about it, none of the dodgy build quality, raffish engraving or half-finished min-translated literature. The UM 900 shows itself to be a well thought-out design, and for all that Eastern microphones can compete on equal terms with the best.

NEW TECHNOLOGIES

——— PCI bus mixed to the 16 outputs. The two new pieces of hardware are a TDF-ADAT unit (currently known as S880-2) and a TDF to 8-channel unbalanced unit (currently S880-3) with 20-bit converters. The target price for each is US$600. Soundscape, UK. Tel: +44 1222 456120.

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 PA's 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 132dB. The coverage pattern of the UPA-1A is 100°x45° 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 front controls, 2 switches per channel for routing to the 2 stereo outputs, individual volume control and 10 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 Soni-comps, are switchable solid-state and LDR attenuators for maximum flexibility. Soni-comp I is targeted at the project studio and is a linkable 2-channel compressor with identical sonic performance as Soni icomp II with 120 metering. Soniicomp II complements the Oram Hi-End range in a bigger case with 12 meters. Both have an adjustable variable pots for input level, threshold, >
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Building on the deservedly solid reputation of the 200A, Summit has unveiled the EQP-200B. **Terry Nelson** tests the new EQ

**I**

**MUST CONFESSION** 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 yesterday, 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-8ce (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 pointers.

The rear of the chassis features a standard EC mains connector with 120V-250V 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 another EQQQ, with all the actual settings not visible to me. The first thing that grabs the attention with this piece of equipment is inserting into the signal chain with the EQ bypassed. Everything suddenly comes to life and has space around it. While this might be considered pushing a review's licence, 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 over or under powering at the frequency selected, whether the initial wish was to boost or cut.

As noted, the mid-high section is a 3-position 6dB-8ce low-pass filter and, again, is capable of introducing some quite 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, 50Hz 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. Figure 2 is a selection of programme equalisers and 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 tool 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.**

**NEW TECHNOLOGIES**

<<< ratio, attack, release and output level. Oram, UK. Tel: +44 1474 815300.

**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, Sabens, G1.2s, 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 parallelised 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 IFF 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 what console functions are reset. The 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 Cohut S650 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 MHz range.
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Watching the noughts and the ones can be a haphazard business.  

Dave Foister finds a metering solution

DIGIT IS DIGIT 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 precision 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 two 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, S/PDIF coaxial on XLR, 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 signal 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 bargraphs with PPM characteristics, and two corresponding 1-character 7-segment numeric displays. These show peak levels in dBFS with a resolution of 0.1dB, the first space showing the mains 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 converters 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 uncompassingly 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 AES standard of Over 2dB, the consecutive maxima, this can be set at the factory. At the lower end the numeric read-outs show 40, 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 brightness 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 painted scale which I understand will stand out better on future units.

The DPM101 is mains powered with an onboard supply and has a 5V output for connection of future Genesis products. As supplied it's a neat stand-alone box, but marking 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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Avalon VT-737 SP

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

ELEOPARDS 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 and 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. More than that, the usual course 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-pleasing 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 transformers, and a continuously variable gain pot is followed by a switchable high-pass filter. This leads into the compressor, which in common with the solid-state AD2014 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 centralvu meter can be switched to show resulting gain reduction. The compressor section is 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. IF and LF channel 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 42kHz. 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 smearing 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-hand 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 737’s 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 best of both worlds: diodehiclip 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 the L742, 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 six roles with something desirable of its own right. Add the flexibility that comes with integration and it’s a winner.
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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 $20,000 and S6000XL will all remain current—and went about designing a new generation of S1 for the foreseeable future. For the company's accompanying range of postproduction-oriented hard-disk systems derived from the DD1500. Akai also had to question 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, 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 and the flagships high S6000, although both sport a large display with keys surrounding it. You can also connect up a PS/2-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 multimodal operation; two pairs of MIDI In, Out and Thru ports, digital I-Q, 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 Jazz 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 crossover functions. There's also BPM match, 3-hand EQ, fade in 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 Modules is included together with new filters (15 different types), envelope generators and LFOs. A 8-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 $3,200 (UK, inc. VAT) with delivery expected in August, that of the S6000 is $3,600 (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.

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 GMS1000 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 SCSI 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 HS66 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 Minipart and adds support for momentary-action switches, multi-unit linking and knockout holes for permanent installation. Furman's first product the PQ3 parametric EQ instrument has been re/released 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.

April 1998 Studio Sound
Legendary Sound to Make Your Dreams...

The Studer V-Eight is an 8 channel 20 bit digital recorder based on the ADAT™ type II format, using S-VHS cassettes. The V-Eight is 100% compatible to all current ADAT formats with over 100'000 units sold. The professional design and reliability will give you a cost effective, faithful workhorse for all professional audio recording applications. The V-Eight features a professional S-VHS Tape Drive for extremely fast and gentle tape handling which leads to substantial time savings. The V-Eight also has the convenience of an integrated TC generator and chase synchroniser. Unique features are: 24 bit Studer converters based on the legendary D-827 DASH recorder technology, to improve the sound of your recordings and an On-board 9 channel monitor mixer to make live recordings without a mixing console.
**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 LOOKS 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-1, turning out to be outstanding at double bass. A companion range, the OM-series features 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 around 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-5b is the original, using the VLM Type B capsule in a transformerless arrangement that gives a quoted frequency response from 80Hz to 21k水肿 (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 5b's polar pattern is the standard hypercardiod, 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 Herz off the extremes of the frequency response but ups the sensitivity by 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 undersounded sound. Audix's brochure is scathing about nonminimum designs, talking about 'unnatural sound and excessive feedback, which might alterate some. The OM-5 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 both 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-1. 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 a similar sonic performance. The crucial characteristic is a crisp and smoothly extended top end which helps vocals cut through without making them unendurable. 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. Easier to check whether the qualities that had impressed me so much on the D-1 were carried across along with the capsule I put them up together, along with the OM-5, and was indeed equally impressed. While the 7 sounded like a good vocal dynamic, the 6 was effectively identical to the D-1, giving a sound that could easily have come from a condenser. There was no steam, no convection at the top and a smoothness right across the range, the bass in particular as on the D-1, 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.

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NEW TECHNOLOGIES

![Audio sources including analogue, AES-EBU, SPDIF, compressed MPEG1 Layer 2 can all be streamed through the ATM 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 multi-media audio production and in preparation for Internet delivery, providing loud and clean audio files. For webcasts, 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 multiformat conversion. This is made possible by sample-rate converters, 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 normalising 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 switches

Pro-Bel is launching a line of small routing switches with the 16x2 family satisfying the need for mixed signal formats. Typical configurations include 10x2 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 set-up buttons also provide destination and level control. For

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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 SWB 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 J. 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 problematic 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 bothishbone. When the software is used with the proprietary 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 -10dB 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 TriplelDAT'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-

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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 25cm 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 50Hz to 30kHz at a peak power of 146dB.

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

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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 handful of blintzy-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 LeBlanc and Minnie 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 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, making 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 is so huge, there are so many shots, and stuff is constantly being pushed back and forth. 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 300 shots missing, and we've often just had to make our best guess and then cross our fingers. We'll be remixing away and the director will come in and say, 'Oh 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 sounders 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, slicker 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 recognize.'

'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, 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 moving joints and interesting little noises as well... It's a fairly conventional-looking robot, a very heavy machine, and he runs on tracks, and we put an enormous amount of low end into the track sound so that it becomes an inanimate machine in itself. Then again, while we tried to give a servo to the truck and head movements, he has two regular arms and two smaller arms and for those we created little zippy sounds just to help identity how the different areas work... Sometimes you can use things that you never would have thought would work and accept it.

As for the outer-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 that 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 real sound, it's being made up by someone 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 all of the different sounds. You start with nothing and therefore you're scared that people are going to think that it's all a set. As a result you probably end up putting a bit more in 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, Audition, Pro Tools and Fairlight. To that end, the MMIR8 was the recording medium of choice because of its ability to interface with all of the aforementioned.

The MMIR8 is a great medium,' says Eddie Joseph. 'If you want to do changes on the stage the MMIR8 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 Iain Eng convinced everyone that the MMIR8 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 Red 5, for instance, and it probably took 30 minutes to do the form on the three masters as well as seamlessly fill in some backgrounds. In that way it's incredibly fast. Plus, I think editors will enjoy using the MMIR8 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 mods on DA-88.

Meanwhile, with the use of Todd-AO's new AMS Neve DFC consoles—installed on 5th December, operational from February—Lost in Space is the first all-digital Hollywood picture (discounting the 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 32 inputs, like everybody is—and the digital boards that we've used have just been terrible. The monitor bussing, the punting; 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 Logics—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 bussing structure and also the channel substructure. The bussing structure is completely solid 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 bussing 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

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Film photos: Jack English

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I shouldn’t worry about the noise - no one’s got a system that’ll pick it up anyway...

Mark Smith, music mixer

Eddie Joseph, sound editor

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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.

Technically, this is a two-voice piece. There are two sound tracks—one for the voice and the other for the music. The music is a bit of a compromise in that it’s not something that you can just cut in one place and pick up elsewhere. It has to flow from track 1 until the end, and so there are restrictions as to what you can do to it.

Meanwhile, for my part, having already pre-dubbed, 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 new 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 space 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

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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 gazed at the 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 corner 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 Olympic 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—unleashed a staggering total broadcast time of 115 hours and 30 minutes on domestic broadcaster NHK's general service alone. Over 51 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 debut at Nagano. Despite the cheap jokes at its expense, curling became one of the TV highlights, particularly late at night when its 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 case, 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 how broadcasters—producing the pool feeds—on certain events, in addition to programming 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 bobsleigh 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 how on the bobsligh 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 going 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; bobslagh and luge took place at the Spiral on the Iizuna Kogen Heights; alpine downhill skiing; super-G and other downhill events on the Happoone slope; 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 Yakehata, a peak in the Shiga Kogen Heights.

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

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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 been 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-5 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.

Routing 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 hucked-up with dual power supplies and control boards, were the HD and TM series configured as 12x16 analogue video and stereo analogue audio in-studio; 96x64 for distribution of analogue video and stereo analogue audio (plus 4x32 stereo audio); and 6x16 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 converters, were supplied by Leitch Technology. The IBC featured Digibase 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 large global distribution.

The events at the Winter Olympics can pose problems for producers, especially with such high-speed 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 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 60 km per hour. In designing the bobsleigh and luge, NHK 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 start and the Weight House, eight whip-pan units to capture the men's luge, and eight snowboarding, for the female luge, and eight airborne balloon-cams. 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 Koji Temple. Among the postproduction equipment was a Queenery H8 effects editing system, four Picturebox wall stores and three Paintbox Bravo graphics suites, plus 11 Chyron INFINITI graphic 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 by CBS New York audio/video post production specialist Howard Schwartz. Based around a 32-input Harrison TV950 console, with Tascom 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 NHK chief engineer Marty Newman oversaw its installation.

Three operators went over from America and worked 12-hour shifts to cover the
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 Digital Betacam, much of the audio post work was mixed straight back to the VTRs without 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 Hayabusa 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 CS9000 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 Omron 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 tracks. 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 exciting as they are. The microphones are set into the ice itself and are sensitive to both low and high 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 tones.

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 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 (Asah National Broadcasting)
Freestyle Skiing, Giant Slalom
TX (TV Tokyo)
Ice Hockey II
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
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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, so you also had to know how to deal with things such as magnetic stripe, not to mention learning how to mix 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 60s Es with Massenburg Eq crossovers, and then in terms of the effects... well, there were none.
'In that room we had three EMT 146s,' Scheiner recalls. 'We used an analogue tape machine to delay the send to the echo chamber, and that's what made the echo sound. Also, we had used quite a bit of an Emt 146 to delay the send to the echo chamber.' The echo machine delayed the sound by up to 200 milliseconds, creating a delay effect that Scheiner and Van Morrison used to create the famous 'Moondance' sound.

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 Malakh 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. That year's work on Van Morrison's 'Moondance, Band and Street Choir' has been brought to light by the composer. 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 'the occasional producer.'

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 worked 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 and reverb and everything was always cut dry. We didn't necessarily cut it that way, we used compression and EQ when we were cutting —but we never used reverb when we were cutting. When you hear one of those recordings now you see: 'Wow, man, there's nothing on this record!' But, you know, that's what we really didn't put much on it. We were so accustomed 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 E-1844 we used only very occasionally, hardly ever. Personally I was far more into flanging stuff. Apart from that, if we were asked to do a song 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 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 of the American, but 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 XAB 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 XAB 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 during a middle of a performance, but I remember 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 stuff 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, 'This is not 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, 'Hey, who do you want to work with?' I said, 'I want 25% 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 25% on time and 15% on tape, and that worked out pretty well.

However, when other guys saw what I was making then all decided to do the same, and that was the beginning of the demise of the A&R sound 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 'Juke' and in 1981 he scooped the same prize with the same band for 'Greatest Hits.' During the ensuing years, however, even though there had been nine further Grammy nominations, the awards themselves have eluded him. In 1998 these included ones for Best Engineered Album, Non-Classical ('Dwight Yorke's 'Turns For The Road' and Best Pop Album ('Bestwood 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 had a Grammy. 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, Snookey Robinson and Dan Fogelberg. At the same time, since 1976 production has also been a priority for Scheiner, and his credits in this field take in The Eagles, Mac. Fogerty, Glenn Frey, Jimmy Buffet, Bruce Hornsby, Donald Fagen, Toto and The New York Rock & Soul Review (featuring Fagen, Boz Scaggs and Michael McDonald). It's all exciting, but the cut 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 boring.'

'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.'

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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 analogously, lips on layer, 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 offering, there was a desire for these to be studio tracks rather than live cuts and, as Scheier was preoccupied with the MTV project, Henley brought in his own man, Ralph Jacobs, for Hell Freeze Over. As things turned out, with Jacobs and Scheier each in separate rooms at The Village, they basically ended up helping each other. Hence Scheier’s coproduction and engineering credits.

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 Scheier 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. Warrens again only hired Scheier 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 sidemen and so these were sort of difficult, especially with the marching band that Fleetwood used for Trojan. In that instance I was originally going to try and do coverage around the room as they marched 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.

**Fleetwood Mac: loving the I Love Lucy soundstage**

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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 (Zambisa) and Dave Grusin (Tico 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 strong 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 replies. ‘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?’
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...
the software audio plug-in market, which in essence has integrated the sort of effects capabilities previously only found in 'stand-alone' onboard 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 the developers.

Some of the established names in onboard 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 eL 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 multieffect 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 is 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 OS9 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 for 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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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?

Open plug-in formats

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

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.

The audio plug-ins market for MIDI + Audio software, then, is still relatively new, as is reflected in the profession 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 'narrowed' level.

While Pro Tools TDM is de facto common denominator for Mac packages, it is definitely not the 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 appears native real-time processing on the computer. In a significant late-breaking development 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.

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“I found the MLX2 performed very well. The FET DI is excellent, on bass guitar it saturates with valve-like warmth.”

Steve Fisher - Producer

LA AUDIO

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**The Manic Street Preachers’ 1996 album, Everything Must Go, virtually ensured the band’s reappointment in the mid-sixties, and it’s a sound that has 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 [Moorel’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 KM94 (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; BPM9’s on the overheads, tons are all Sennheiser 415, 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 MkI 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 hacking 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 hacking 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 hacking 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 at 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 bass, 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 5ips—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’ Château de la Rouge More 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 hacking 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’ Grimeble 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. If putting it on afterwards, says Hedges. The Copycat’s used quite a lot, but again you’d record the sound

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 1.2m 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 “loa” In your mix, and there’s nothing worse.’

‘On this project, we used Sennheiser MKH40’s 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 BPM44s 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 big gien 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, so when you come to mix, it's nice 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.

When we're recording keyboards, we don't just DI it into the desk, says Hedges. 'We'll be using Leslie 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 as 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 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 and 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.

'If I don't know whether I'll get involved with the band, I'm sure the band have their ideas of who they want to work with. So it's a very tight relationship. I'll 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 consoles—Bernie Grundman strikes a figure reminiscent of a vintage car mechanic. He appears at ease in an old-school manner, like a mechanic comfortable with old-fashioned engineering and values—the occasional personality quirks present a welcome relief from the horridly 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-era mastering engineers whose ranks include Bob Ludwig, Glenn Meadows, Doug Sax, Doug Levine, and Denny Purcell. They are all in their 60s and 70s and have careers which span eras in audio that started with the classic post-war big band records and which witnessed the radicalization of music from a culture into a multi-billion-dol-
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 the process, he recalls. It could be more innovative and hands-on about mastering because it was the one part of the business that Lester couldn't 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-spect 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.

Grundman, In one of the new studios under construction

He moved to A&M in 1969 and stayed for the next 15 years, working on a Decca 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 emerge. 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 characterized 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 1981, 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 5-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 a radical economic restructuring. The continued boom in Grundman's business allowed him to create a 13-room extension of his facility in Japan in 1999. 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.

"I became more and more passionate and wanted to apprentice myself 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, and it happens to be in the Shibuya district of Tokyo instead of

Studio Sound April 1998
Hollywood, California. Bernie Grundman, as is known, is a single-studio facility operated by Maeda as an employee of MGM, although he is lavish in his praise for Maeda’s talent, ambition and achievement.

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 recording 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 half-house over for each one.

Which brings us to Grundman’s newest venture, the complete replacement of his 11-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 steal stuff—I’m sitting here now with three Mison X-80 machines crammed around me and I can’t move.

The new facility will also have two Studio discque cutting lathes, both in the same suite—one fitted with analog components including a Marco cutting head, and the other with solid-state electronics designed by Grundman and Beschof. Other technologies include Harmonia Mundi BW 302 signal processors, Aputure converters and a Studer A80 mastering deck modified to accept 7-inch reels with custom discrete playback electronics. The all-discrete consoles will be custom-built, assembled from various components and are digital-analog hybrids, each having integrated onboard digital signal processing as well as analogue zooming.

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 editing and crossfades. Up till recently, BGM has been using Studer 8-track 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 great master out of a DAT, but what they’re not telling you is that they have to go through another transfer process to Excite 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 hit for hit 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 insulator 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 other solutions abound and proliferate. Grundman’s response, through his new facelift, is 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 realize the need for quality mastering will see that its cost remains a fraction—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 rates as high as $370 per hour. If it’s a bit less, 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.
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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, taggled 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 customers 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 recording-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 Dominos 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 peripheral process.

Things have changed yet again. As competition in many US markets evolves from simply right 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 It All. And Do It All.

Of course, 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 chimp's 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—because you can't do it all, and do it all well. There's no blame to be assigned nor any aspirations 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-wide 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

VD 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 catalogues, 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 famously 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 multi-channel 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 Meridian'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

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 Audio at the DVD depth of 0.6mm. Anyone licensed to press CDs or make CD
Digital radio promises improved services but so far has engendered confusion, fear and scepticism

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 troublesome. 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 bickering 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 test equipment and funded new services, made possible by the extra capacity of digital, being played down in the face of unwarranted publicity.

There were the technical considerations. Nobody could quite answer the question as to why layer 1, the 30 kHz frequency spectrum chosen to hold the multiplexes, the L1-band was to be the permanent home or merely a parking arrangement. It was linked into the future of FM, which would be freed up by the start of digital radio but would have to rely on at least 10 years to present disengaging these 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 steer 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, dealers and retailers, 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, the number of listeners 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 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 at a lower price.

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.
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 therefore 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
The system was first devised by the Neuroallience, both operation. Acoustic diaphragm where the chamber behind the 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 Neuromann company. The central block 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 figure-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.

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 radiator 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.
Speaker testing

REVIEWING LOUDSPEAKER systems 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 N10 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 ‘trials’ 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 grouping, 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 optimum 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-$5,200 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 1974 can still give an account of itself that the last 10 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 and transparent frequency response (or frequency response) at the listening position.

Largely due to work by Ohm and Helmholtz, it has been a widespread belief that the 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. 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.

In 1989, Keith Holland and I wrote a paper in which we asked more manufacturers to publish the impulse responses of their systems. 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 system.
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 Signal 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 had low-level detail and general openness, while seeming to be almost exemplary in other respects. After reading an AES paper by David Clark which, in which it discussed side-effect interactions in low-frequency motor systems, I wondered if the problem could be in the loudspeaker systems, whose inherent losses require a finite minimum drive signal to get them moving. It convinced me that these were purely LF problems, and that the frequencies of interest for transparency and openness were above the affected range. It was partly led astray by associating these problems with some new suspension systems, but Martin Galleys believes the problem to lie in one of the new cone materials, some of which, coincidentally were introduced with some of the new types of suspension systems. He also believed 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 systems. 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...
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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 dozen 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 visionsaries 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.

![Response of large widely used, 2-way studio monitoring system](image)

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

<table>
<thead>
<tr>
<th>Everyone Agrees!</th>
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<tbody>
<tr>
<td><strong>Earthworks</strong> Mics Sound Great!</td>
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<tr>
<td>But a few have mentioned that sell noise may be detectable when recording quieter sources, so...</td>
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<tr>
<td>...and because omnis are not for everyone,</td>
</tr>
<tr>
<td><strong>Earthworks</strong> introduces the Z30X cardioid mic!</td>
</tr>
<tr>
<td>The Z30X offers unusually smooth on-axis response to 30kHz, very low off axis coloration, extraordinary rear rejection, clean impulse response and great sound for vocals, drums and etc. Excellent for stage, studio, and broadcast.</td>
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<tr>
<td>...and just in case you haven’t heard</td>
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**Studio Sound** April 1998
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Compact Disc: the next generation

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

THE ATOMIC 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 1990, 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 encrypted 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 software or drivers. 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 SD/HD.

Video, with many of the same standardisation needs as audio, has a head start this time around. This is 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 bursting 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 multi-channel 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 itsponents claim (there are still those who will happily quote the ill-fated quadraphonic experience of the 70's in support of their fear).

An International Steering Committee (ISCI) consisting of the RIAA, the MHP and the RAI 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: that any new disc 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

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 data rates and high-quality 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

![Fig.1: Basic DSD signal chain](image-url)

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 flexible format, allowing a number of combinations of playing time, sampling rate and resolution: although it did not appear to have taken on board as primary features a number of the more advanced recommendations of associations such as the Acoustic Renaissance for Audio (ARAA) which concerned lossless coding and pre-emphasis in order to make use of linear PCM at either 96kHz or 48kHz but with 32bit resolution on 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 a 14-bit, 16-bit and 20-bit resolution. It was also a feature of the DVD-Video format for ---->
94 minutes on the channel mix could be included within the capacity of a single 4-GB layer, but this would mean adding another level, and this 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 GB 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 multi-channel 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 2.8224 MHz (originally the 2-channel section is left unspecified, but still claimed to have 100 kHz bandwidth and 128 kHz 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 100 kHz even if you reduce the primary 2.8224 MHz 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 filters can be modelled 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 disc 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 4.1 kHz 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 multi-bit 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.

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ABYRD 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 be 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 converters are actually quite advanced devices. You can even put a pair of headphones across the 1-bit digital output and hear a signal (albeit a rather 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 licencess 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.

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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, analogue 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 DAW’s 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 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 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.

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.
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