EXCLUSIVES

SD Systems LDM94; LCM89
LA Millennium SPX2
Frontier Design Zulu
Tube Tech MEC1A
TL Audio C-5021
Sony Freedom
Studer D950S
Purple MC76
Alesis XT20
DAL V8

SSL AXIOM-MT
Digital music mixing desk exclusive

The
NORMAN
SMITH
Interview

ABBEY ROAD AND BAUER STUDIOS
DVD FORMATS EXPLAINED
96KHZ RECORDING FOCUS
POSTING NIKE SUICIDE

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52

www.prostudio.com/studiosound

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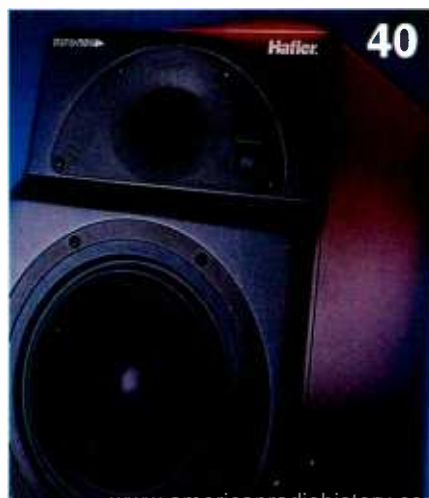
Going to market

HAVE YOU EVER HAD CAUSE to upgrade software or hardware on a system and found the resulting performance disconcertingly different to what your best informed estimations might have expected? You change your computer to a faster one, maybe not the fastest possible, but just enough to not force you to also upgrade every piece of software you have grown to like. Or you change a sound card, even a modem or take advantage of that spell of downtime to install the latest revision of that piece of software that has been sitting in the box with an accompanying manual now swollen to telephone directory proportions.

Almost without exception, the exceptions cover those instances when you have had the luxury to beta test the additions on somebody else's system at great length, you will be surprised, bemused or disappointed. It seems that the number of variables and possible permutations involved is so great that the only sure fire way to see is to assemble and try it. Having 'progressed' from proprietary hardware and software to the democratising influence of the personal computer, the audio industry like every other is now being led by the nose like some castrated domesticated beast by the operating system manufacturers who create the environments in which our applications run.

While I understand fully the logistical and economic reasons why this has come to be, I have to question where it is taking us. I cannot be persuaded that the performance requirements of a professional audio editor figures highly in the plans of those who are salivating at the prospect of power business take-up and every-home-should-have-one multimedia systems where audio is about getting noises over the web or changing the sound of your error bleep.

Some operating systems come closer than others, but if we are to continue pursuing this route, and it seems as if we must, then energies must be devoted to developing perhaps a derivative of an existing operating system that can be developed and grown to serve the specialist requirements of audio in the professional environment. Audio has to make something its own. What exists may be adequate for the time being, but will it get us where we would expect to be in years to come? We may be surprised, bemused, and disappointed.

Zenon Schoepe, executive editor

Wide, fast and bloody

ALTHOUGH IT SEEMED AN ESOTERIC ISSUE to address back in early 1997, 96kHz sampling has taken until now to really kick in. Please don't mistake this for another in the tired tradition of polite trade editorials belatedly acknowledging what an industry has already achieved—we are playing hardball here.

Sitting in on the inaugural 24-bit/96kHz recording session conducted by dCS and Meridian at Cambridge's Queen's College Chapel early last year (and the subsequent 24-192 session), two things seemed pretty evident to me—the potential benefits in terms of audio reproduction and the potential problems in terms of equipment manufacture. At the time, however, the academic approach prevailed and almost all public discussion related to such issues as low-pass filter architectures and spatial information content. Since then much has changed—further sessions have taken place, more engineers and producers have experienced wide and fast recordings first hand, manufacturers have been polarised towards and away from supporting them. Most of all, the politics have kicked in.

Where the bit budget is concerned, the consensus is clear—the details of exactly how you justify 24 bits, everybody who wants digital wants them all. But where almost all convertor manufacturers have signed up for 96kHz or better, other areas of manufacture are less forthcoming. Enquire of your console manufacturer when 96kHz will appear on the spec sheet and you are likely to be fed a main course of processing overheads with a liberal garnish of uncertainty as to its audible benefit. Now, while not all 96kHz sceptics are equipment manufacturers, the way philosophies and commercial activities line up at this point in time is clearly unnatural.

That the issue is a hot one is in no doubt. The response to John Watkinson's critique of 96kHz recording was phenomenal—and it did not simply come from those manufacturers looking to sell flash kit to gullible recordists. The 96kHz features and letters elsewhere in this issue represent a cross-section of that response, but do not fully recognise its breadth.

As I said in opening, this is no polite acknowledgment of developments that are evident to all. *Studio Sound* has tracked the emergence of high sample-rate recording from its inception. Now that the flood gates have opened, there is much to read about it elsewhere, but little of it seems to me to represent the real state of the debate. And the debate is bloody.

Tim Goodyer, editor

ABC
AUDIT BUREAU OF CIRCULATIONS
BUSINESS PRESS

PPA

BPA

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Jan–June 1997
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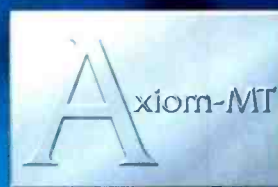*un* **Miller Freeman**
A United News & Media publication

NAB campaign

TO SAY that the Las Vegas NAB Convention has grown too big would be to suggest that the event has in some way risen in a few years from a small regional assembly. It has always been big, yet the proportions of the gathering, like Frankfurt, have become unmanageable for anyone not prepared to plan their campaign in detail and without tracing out their walkabout route on the floor plan before hand. Still a two-centre exhibition, the Sands Hotel section looked as large and busy as the Convention Centre itself this year with a positively staggering level of investment by participants from the broadcast-based industries. Anyone entertaining thoughts of market contraction could not have been there. These are up-beat times and great

times for buyers. Yet still it was possible to waste 90 valuable minutes a day on the whim of attempting a return trip between the two centres.

Audio had it's own hall again, although some manufacturers still chose, or were obliged to choose, space outside it. There is talk of this hall at the Convention Centre being expanded next year with some major building work. Audio highlights included Calrec's digital T-series (see page 42), an entry-level digital desk challenger from Tascam (TMD1000) plus the MMP16 MMR8 variant playback dubber and the first public appearance of the ADSG dubber (p56). We survive NAB only to return again. I love it.

Zenon Schoepe

▲Argentina: Fito Paez' Buenos Aires studio is Argentina's newest and most advanced facility. The work of the Walters-Storyk Design Group, Circo Beat offers SSL SL4056 and Genelec 1039 console and monitors, Pro Tools, 2-inch Otari multitrack, ADAT MDMs and vintage Neve equipment as well as a recording acoustic variable between 0.6s and 1.3s RT time. Walters-Storyk Design Group, US. Tel: +1 914 691 9300.

Reading and reference

Heading this month's crop of reading matter is a new book from the desk of John Watkinson entitled *The Art of Sound Reproduction* (ISBN 0 240 51512 9; £35). Published by Focal Press, its 700-odd pages mix theory and practice in an attempt to provide 'everything the audio engineer needs to know' and covers subjects from audio basics and microphones to loudspeakers and considerations of audio quality and psychoacoustics. SPARS' *Time Code Primer* (ISBN 0 9658309 0 X; $19.95 + $4 P&P) confines its attention to the thorny matter of code under the headings What is Time Code, Handling Time Code and Recommended Practices, making it clear that its 68 pages are a no-nonsense guide for practical application. More specific still is BAL's catalogue of proprietary video and audio interfaces. At a mere eight pages this still manages to claim to be first single catalogue from the manufacturer. Switchcraft's *Molded Cable Assemblies and Patch Cords Guide* is similarly exclusive in its appeal if more extravagant in its 32 pages. File these last two under 'no tyre kickers'. Finally, for those who don't know and who are too ashamed to ask, NTL has produced a simple 3-page Q&A sheet entitled The *NTL Nontechnical Guide to Digital Radio*. Setting aside matters of professional pride, this is the sort of paper too infrequently seen in times of rapid development. Focal Press, UK. Tel: +44 1865 314301. SPARS, US. Tel: +1 561 641 6648. BAL Broadcast, UK. Tel: +44 1203 357827. Switchcraft, US. Tel: +1 773 792 2700 x243. NTL, UK. Tel: +44 1962 822891.
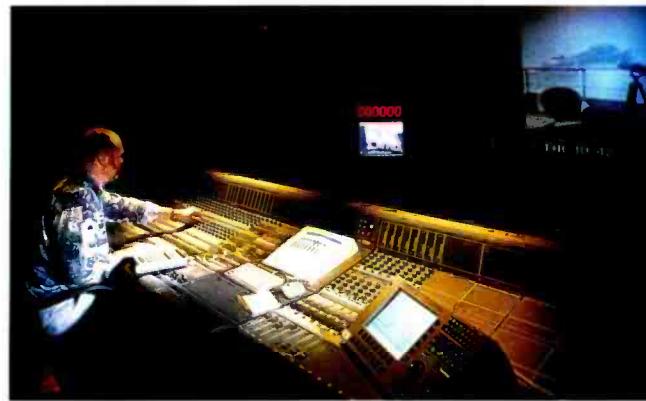
▲UK: Being screened by BAFTA in London to mark the 30th anniversary of international film distribution and production, *Neville's Island* can also claim to be the first film in Europe to be mixed on an AMS Neve Logic DFC console. With tracklaying, Foley and ADR done to AudioFile and posting conducted at dB Post on AudioFile and DASH multitrack, Terry Johnson's film stars Timothy Spall and Martin Clunes, was mixed by Alan Sallabank in Dolby Surround and will be screened on ITV later this year.

Recording awards

US/UK: Brad Guilderman is celebrating a Grammy nomination for using Quantegy 499 2-inch tape to record Babyface's platinum-selling *The Day* album. The Grammy ceremony was recorded by Randy Ezratty's Effanel mobile—on Quantegy tape. BASF's Master Awards, meanwhile have gone to Mike Hedges and David Kahne. Hedges

secured his accolades for No.1s from Texas, The Beautiful South, the Manic Street Preachers and McAlmont & Butler which were all recorded on Studio Master 911 while Kahne, along with engineer John Travis and NRG studios, for Sugar Ray's double platinum album, *Floored.*
Quantegy, US.
Tel: +1 770 486 2800.
BASF, US. Tel: +1 805 295 5551.

▲UK: With audio sourced from a series of one-man shows, *An Evening With Dickie Bird* is a testament to the life of the renowned cricket umpire. When Bird retired from cricket and took to the road, Network Audio's Chris Seymour was charged with the task of capturing and compiling his anecdotes for commercial release. Using only a pair of portable DAT machines, Seymour made two stereo recordings —one of the performance and one of the audience. He then transferred the non-time-coded recordings to a Soundscape DAW where they were shuffled to obtain sync and spun into a final stereo mix. The results were then edited down from seven hours to 90 minutes for release.

▶US: Teddy Riley is pictured with the SSL SL9000j console that has recently been installed in his private Virginia Beach studio, Future Recording. Intriguingly, a key factor in Riley's choice of console was its immunity from obsolescence through advancing standards in digital audio. The console will lose its cherry on a mix of Janet Jackson's 'I Get So Lonely'.
SSL, UK. Tel: +44 1865 842300.

▼ UK: Eight Genesis DPM-101 digital programme meters have been installed in Abbey Road's postproduction and interactive departments, and a further six DPM-101s assigned to 'roving duties' throughout the studio complex. Pictured with Technical Operations Engineer Colin Johnson, the meters have been chosen with accuracy and legibility in mind. Aspen Media, UK. Tel: +44 1442 255405.

▼ US: Burbank's Four Media Company has recently completed the installation of a pair of film consoles: a 3-position 160-input Otari Elite film console and a 144-input Harrison Series Twelve. The Elite (pictured) is the largest yet built and features Otari's PicMix monitoring and busing system, and has three bays and three separate automation systems. The Series Twelve is the third of three ordered and is installed in Studio B which has been redesigned by Tom Holman, acousticians Veneklasen & Associates, and architectural experts Heathcote & Associates. Both consoles have already seen use on a selection of film and television projects. Otari, US. Tel: +1 415 341 5900. Harrison, US. Tel: +1 615 370 9001. 4MC. US. Net: www.4mc.com

High and mighty

UK: Recent recording projects undertaken by British-based classical specialists Green Room Productions have seen a 176.4kHz session take place at Henry Wood Hall where the Leopold Trio performed Beethoven String Trios for Hyperion Records, and a parallel DSD (2.8224MHz) and 96kHz session at Dundee's Caird Hall involving the Scottish Chamber Orchestra performing Schubert's 8th

and 9th Symphonies for Telarc Records. Both sessions were engineered by Green Room's Tony Faulkner whose concern for the longevity of classical masters has pushed him into high sample-rate practices. Presently Green Room is the only UK independent production company offering these facilities which are run in parallel with lower standard recordings at no extra cost to the client.
Green Room Productions, UK.
Tel: +44 1895 822771.

■ America's first Amek DMS has gone to Hollywood's The Bakery postproduction house. The console is the central element in a new all-digital mixing suite which, along with four new editorial suites, constitutes a $2m expansion of the facility. First use of the mixing suite will be a multichannel music production for a Taiwanese theme park. LA's LA Studios, meanwhile, has committed to the dSP Poststation to accompany the opening of its second facility in Santa Monica.
The Bakery, US. Tel: +1 818 508 7800.
LA Studios, US. Tel: +1 213 851 6351.
Amek, UK. Tel: +44 161 834 6747.
dSP, Australia. Tel: +61 2 9557 2292.

■ Sri Lanka's Asia Broadcasting Corporation is to have seven studios and interview studio built by Sonifex which will include Sovereign desks, Sentinel audio loggers and HY02 telephone hybrids as well as other essentials such as power supplies and lighting. The studios will be assembled and tested in the UK before being shipped to Sri Lanka and reassembled for use.
Sonifex, UK. Tel: +44 1933 650 700.

■ London's Decca Record Company has adopted the dCS 972 sample-rate convertor as a house standard. Capable of handling rates from 11.025kHz to 96kHz and a variety of I-O formats, the 972 represents Decca's ongoing investigation of 24-bit/96kHz as a recording and archiving standard. Sain Studios in Wales has opted to make the Genex GX8000 MO recorder the centre of its archiving effort. which involves a large catalogue of sixties and seventies recordings. The transfer from analogue tape to 24-bit disc also facilitates a forthcoming remastering programme which will see the original tapes pass to the National Library of Wales for research and public use. Back in London, mastering house Tape to Tape is claiming to have Europe's first complete Focusrite 'Blue' mastering suite following the construction of a new 4-room facility. Deigned by Recording Architecture. the development includes three mastering suites and a tape transfer room containing a complete set of Focusrite Blue units and Sonic Solutions system.
Decca, UK. Tel: +44 181 747 8787.
Sain, Wales. Tel: +44 1286 831111.
Tape to Tape, UK. Tel: +44 171 371 0978.

■ Croatian Allegro Studio has taken delivery of a 64-input D&R Merlin console for use on its pop music production work. Located in Split, Allegro has clocked up three No.1 hits in 1998. Also in Croatia, 13 BGW Performance Series Model 3 power amplifiers and two M1100 subwoofer systems have been installed in the country's largest concert hall, the Concert and Congress Centre Vatroslav Lisinski. Destined for the smaller of the venue's two halls, BGW is expecting to supply amplification for the larger hall subsequently.
D&R, The Netherlands.
Tel: +31 2940 18 014.
BGW, US. Tel: +1 800 468 2677.

■ LA's A&M recording studios has installed a Euphonix console and Berkeley Hills' Live Oak studio has installed an Otari Elite. A&M's 96-fader CS3000 is sited in its 5.1 surround compatible Studio C which has been renovated by Vincent van Haaf and Blackhole's 48-fader CS3000 is to be found in Studio A. While A&M has hosted the likes of kd lang, Chicago and the Wallflowers since its reopening, Live Oak's Elite represents a timely upgrade for the facility and will run with 32 tracks of ADAT serving a variety of music from rap to opera.
A&M Studios, US. Tel: +1 213 469 5181.
Live Oak, US. Tel: +1 510 540 0177.
Otari, US. Tel: +1 650 341 5900.
Euphonix, US. Tel: +1 415 855 0400.

■ Videolondon's Studio 4 has recently undergone a refurbishment by Munro Associates which has seen a 64-channel SSL Avant film console join its existing Lightworks and Avid systems. The 5-studio complex handles

a variety of TV work and sees the new console as addressing the closing disciplines of film and television. Another London post facility, World Wide Sound, has installed a Fairlight MFX3 workstation at one of its three Soho sites in response to the growing demands made by its load of film and broadcast dubbing work which involves foreign language dubbing for European release.
Videolondon, UK. Tel +44 171 734 4811.
World Wide Sound, UK.
Tel: +44 171 434 1121.
SSL, UK. Tel: +44 1865 842300.
Fairlight, Europe. Tel: +44 171 267 3323.

■ Turkish postproduction house, Sinefekt, has invested in a 16-fader AMS Neve Logic 3 and 24-output AudioFile Spectra. Installed in Studio 1 as part of a refurbishment. the Logic 3 is equipped for surround work and will be used for mixing television commercials.
Sinefekt, Turkey. Tel: +90 212 275 7234.
AMS Neve, UK. Tel: +44 1282 457011.

■ New York's Superdupe has become the first US facility to invest in Soundtracs' DPC-II digital console. The 48-fader desk will take pride of place in a new Walters-Storyk designed 5.1-capable postproduction room and will serve recording and audio post duties for TV, broadcast and advertising. New York's Sound One post facility now has 53 Akai DD8 digital dubbers in use for major feature film work including Martin Scorcese's Kundun.
Superdupe, US. Tel: +1 212 683 6854.
Sound One, US. Tel: +1 212 765 4757.
Soundtracs, UK. Tel: +44 181 388 5000.
Akai, UK. Tel: +44 181 897 6388.

■ Hire companies worldwide continue to increase their stock with Istanbul-based Altincizme purchasing 40-input Midas XL3 console; British-based FX Rentals and Stage Electrics investing in a complete set of TL Audio Ivory processors and 16 channels of beyerdynamic U400 UHF radio mics respectively; the German Westfalensound buying three 48-channel DDA QII desks equipped with DDA's Spatial Imaging System; and California's The Enterprise recording and post facility is sharing three Genex GX8000 machines between archiving and rental duties.
FX Rentals, UK. Tel: +44 181 746 2121.
The Enterprise, US. Tel: +1 818 505 6000.

■ France's Europe 1 radio station has ordered two 24-fader Amek DMS consoles to handle its style of news broadcasting. The first console to be installed in the Paris studio will replace a 17-year old Schlumberger-Digitec desk, the station's oldest, in Studio Coluche while the second will go into the main news studio, Studio Seven. Radio France, meanwhile, has equipped its FM2 mobile with an SSL Aysis Air desk. The new console follows the station's Axiom in Studio 103 and the mobile will be operation by the end of 1998.
Europe 1, France. Tel: +33 1 47 23 12 02.
Radio France. Tel: +33 1 42 30 33 20.
Amek, UK. Tel: +44 161 834 6747.
SSL, UK. Tel: +44 1865 842300.

■ Deutsche Telekom has adopted CCS' CDQ Prima 2Mux-M 2Mb/s codec for an audio programme exchange network allowing inter-studio transmission and direct transfer of programming to the transmitter; the network will also carry ancillairy data. The project will involve the use of some 700 CCS systems and is expected to undergo operational testing at the time of writing.
CCS, Europe. Tel +49 811 5516-0.

■ Maastricht-based regional broadcaster Omroep Limburg has installed a Digidesign Pro Tools 24-bit system complete with DINR and Focusrite plug-ins to handle the restoration and archiving. Operating since 1945, the broadcaster has an estimated 4,000 hours of material which will be digitised and stored on MPEG2 CD ROM. A new OB van has also been built containing a Mac-Audiomedia III system, 24-bit Technica Del Arte convertors and Sound Designer software.
Omroep Limburg, Holland.
Tel: +43 346 7777.

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OverEasy® with VariKnee™ or hard knee thresholds

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

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

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TYPE IV™ output available when equipped with digital output option

Build your own presets using your favourite building blocks

Change programs, parameters, and bypass via midi controllers

XLR and 1/4" balanced ins and outs.

Sidechain functions for advanced filtering applications. Sidechain monitor included

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

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

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

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

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

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

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

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

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

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

It Never Forgets
• The DDP works right out of the box. It comes with 50 factory setups that are guaranteed to knock your socks off. There are presets for every application you can think of, and then some.

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

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

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

dbx PROFESSIONAL PRODUCTS

ARBITER PRO AUDIO • Wilberforce Rd, London NW9 6AX • Tel: 0181 202 1199 • Fax: 0181 202 7076 • web: www.arbitergroup.com

www.americanradiohistory.com

World events

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

18–19
News Technology Conference
Radisson Hotel, Portman Square, London. UK.
Contact: News World Ltd
Tel: +44 171 491 0880.
Fax: +44 171 491 0990.
Net: www.newsworld.co.uk

18–20
Cable & Satellite 98
Earls Court 2. London. UK.
Net: www.cabsat.co.uk

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

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

26-29
Midem Asia 1998
Nusa Dua Beach Resort. Bali
Tel: +331 41 90 46 31.
Net: www.midem.com

29–31
5th Annual Latin-American Pro Audio & Music Expo Miami 98
Miami Convention Centre,
Miami. Florida. US.
Contact: Studio Sound International.
Tel: + 1 914 993 0489.
Fax: + 1 914 328 8819.
Email: chris@ssiexpos.com
Net: www.ssiexpos.com

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

June

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

2–4
Replitech North America
The Moscone Centre, San Francisco, California. US.
Contact: Knowledge Industry Publications.
Tel: +1 914 328 9157.
Fax: +1 914 328 2020.
Email: kipi.event@kipi.com
Net: www.kipinet.com/rep

2–5
5th Broadcast Asia 98 and others
World Trade Centre, Singapore.
Contact: Overseas Exhibition Services.
Tel: +44 171 486 1951.
Fax: +44 171 413 8211.
Email: singex@montnet.com
Net: www.montnet.com

10–13
IRS: International Radio Symposium and Technical Exhibition
Montruex. Switzerland.
Contact: Ms Patricia Savioz.
Tel: +41 21 963 32 20.
Fax: +41 21 963 88 51.
Email: message@symposia.ch
Net:wwwmontreux.ch /symposia

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

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

25–26
2nd Musicom Europe
The Mount Royal Hotel.
London, UK.
Contact: World Research Group.
Tel: +1 212 869 7231.
Fax: +1 212 869 7311.
Email: info@worldrg.com
Net: www.worldrg.com

July

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

July

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

September

26–29
105th AES Convention
Moscone Convention Centre.
San Francisco. California. US.
Tel: +1 415 558 0200.
Fax: +1 415 558 0144.
Email:
105th–chairman@aes.org
Net: www.aes.org

October 12 –November 6
ITU Plenipotentiary Conference
Minneapolis, Minnesota. US.
Tel: +41 22 730 5969.

November

4–5
22nd Sound Broadcasting Equipment Show (SBES)
Hall 7. National Exhibition Centre. Birmingham, UK.
Tel: +44 1491 838575.
Fax: +44 1491 832575.
Email: dmcv@pointproms.co.uk
Net: www.i-way.co.uk/~dmcv/ sbes.htm

4–8
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The 96kHz debate

ON THE SUBJECT of pseudoscience ('Spotlight on 24-96', *Studio Sound*, February 1998) can I point out to Mr Watkinson and any readers he's mislead that the interdependency of time and frequency resolution (p102) has nothing whatever to do with Heisenberg's Uncertainty Principle, grand as they may sound.

For those who have forgotten or never studied basic quantum mechanics, what Heisenberg postulated was that it is not possible simultaneously to determine the position and velocity (or related property) of a material particle with arbitrary accuracy. The relationship between time and frequency resolution, although analogous, is not even a quantum mechanical problem otherwise Fourier would not have been able to state the basic principles in the early 19th Century. It is also inadvisable to refer to Q factor in relation to critical bands and the basilar membrane (Fig.2) since to engineers that immediately suggests resonance. In fact the frequency dependent displacement of the basilar membrane is a travelling wave, not a standing wave, phenomenon.

Readers interested to test Mr Watkinson's contentions about the significance of ultrasonic frequencies to human hearing and the generation of ultrasonic components by musical instruments may care to read the following:

Oohashi *et al.*, 'High-Frequency Sound Above the Audible Range Affects Brain Electric Activity and Sound Perception', Audio Engineering Society Preprint No 3207 (91st Convention, New York).

Lenhardt *et al.*, 'Human Ultrasonic Speech Perception', *Science*, July 1991 p82.

Boyk, J, 'There's Life Above 20 Kilohertz! A Survey of Musical Instrument Spectra to 102.4kHz', obtainable from www.cco.caltech. edu/~boyk.

Keith Howard, London

Dear John

I ALWAYS READ your articles in *Studio Sound* and I most often find myself in agreement with your views. It is still the case after reading your 'Spotlight on 24-96'; however, there are a couple of points that I require comment.

You state, '...a young person might judge 25kHz to be 19kHz, whereas an older person might judge it to be 16kHz. Because of this variability, there is no useful information at these high frequencies'. In my opinion (and Professor Zwicker's, founder of psychoacoustics, and his followers), it is correct to say that there is no useful pitch information, but there is undoubtedly energy and transients which are fundamental to the perception of rhythm, which is an integral part of music, on one hand, and on the other hand to the overall sense of brightness. Your wording may induce some in believing that this information is not relevant and may as well be discarded, which, I understand, is certainly not your opinion.

Also, 'If the human ear really did respond... instruments would have evolved to excite that response.' In fact, these instruments do exist, they are called cymbals, triangle, tambourine,

and so on. Again, they don't produce a specific pitch but rather a large spectrum and it is quite interesting to note that musical instruments have evolved in pure conformity with the findings of psychoacoustics. The pitch discrimination of the human ear is very poor at high and low frequencies; in consequence percussion instruments such as bass drum and cymbals are used, which exhibit both wide frequency spectra. QED.

These are minor criticisms on an article that I have enjoyed reading and on the contents of which I fully agree, in particular when you mention that if it sounds better at 96k, it is because the 48k implementation was not good, it is the conclusion of many DSP engineers, who have decided to process at 96k, because the implementation of algorithms is much easier as the warp effect is significantly furthered, thus reducing the need to 'tweak' processing of high frequencies.

Keep all the good fun and wit.

Jean Luc Moncel, SCV France

I'VE JUST READ John Watkinson's article on the merits of further development of digital audio technology, and must say that this commentary hardly belongs under the heading Technology; though Editorial, Opinion, or Bias (perhaps even 'Open Mic') would all work.

Among the numerous problems I have with this article, let me mention a few. First, the point that 'one hears that the laws of physics are temporarily suspended to allow the latest theory to hold water' is well-made. The author's later conclusions, however, are not based on laws of physics, but rather on 'evolutionary survival advantage'—a 'pseudoscience' if there ever was one. Webster's definition of the scientific method is as follows: 'Principles and procedures for the systematic pursuit of knowledge involving the recognition and formulation of a problem, the collection of data through observation and experiment, and the formulation and testing of hypotheses'. (See www.m-w.com/netdict.htm) As everyone knows, neither evolution nor Noah's Ark may be recreated in the laboratory so this is hardly appropriate to a scientific discussion in your magazine.

Next, the author is careful to point out the physical limitations of the ear as a basis for pointing out the needlessness of any new digital systems. This, also, is not physics but physiology, which while a science is also a little understood one. For all of our physiological knowledge, most of the function of the human body is still a mystery to doctors.

Finally, the author concludes that 'even if a difference can be heard...' Putting physics, pseudoscience, physiology and speculation aside, it can be proven in a laboratory (listening tests) that humans can 'perceive' a difference between 44.1kHz/16-bit audio and 96kHz/24-bit audio (not to mention 192kHz/24-bit audio). I'll even wager that Mr Watkinson can hear the difference—that's my idea of science!

Paul Griffith, Technical University of Gdansk, Sound Engineering Department

10

Alesis XT20

If upgrading the ADAT was a logical move for Alesis, challenging the DA-88's acceptance in postproduction circles was something more. **Dave Foister** looks at longer words and extended features

FORMAT WARS are not supposed to be like this. Alesis *vs* Tascam should have been either a long-drawn-out war of attrition or a bloodbath, but instead the two sides settled quickly and comfortably into their respective territories. Rarely have two such similar systems managed to achieve a peaceful co-existence, and this came about largely because, broadly, DTRS became the choice of the professional facility while ADAT won the battle for the home studio market.

It has to be said, though, that this was not what Alesis originally had in mind. And despite ADAT's undoubted success, it was inevitable that moves would be made to push it further up-market. The Fostex ADAT was a bit too low-key to help, but eventually the ADAT XT appeared, addressing openly the areas where ADAT was perceived to be less of a serious contender than DTRS. Now Alesis has taken the fight to Tascam by giving the format 20-bit capability, making the XT20 the first tape-based MDM to go beyond 16 bits without third-party assistance.

Externally the new machine is identical to the XT apart from a few details of printing, but since the XT itself seems to have passed many people by it is, perhaps, worth looking again at what made it such a significant step forward. It can not be denied that by today's standards, if image counts for anything, then ADAT's presentation needed some attention. The original machine looked and felt like a bit of a lightweight; plasticky and basic, it gave the impression that cosmetics had been sacrificed in the interests of a price point. The XT got the full treatment, with decent styling, better finish and enough extra buttons and displays to make it clear that it could do more by itself. And this was perhaps the key: a stand-alone ADAT had considerably fewer facilities than a stand-alone DA-88, and was more reliant on the addition of a system controller for anything more than basic functions. The Big Remote Control (BRC) in fact adds more operational niceties than Tascam's equivalent, but without it ADAT barely made it as a machine to do a serious session on, largely because of its cludgy transport and its shortage of locators. The original two plus a zero were increased on the XT to 10, enough to cope with a typical song and to provide a couple of extra features. Specific locators are used for looping and punching: 1 and 4 define a section to be looped while 2 and 3 set auto punch-in and punch-out points, for which there is a rehearsal function.

The XT offers a genuine choice of sample rates, rather than the original varispeed workaround, and the rate for a given tape —44.1kHz or 48kHz—must be decided when formatting it, as on the DA-88. A limited amount of varispeed is also available, calibrated in terms of pitch and giving +100 to -300 cents at 48kHz, ±200 cents at 44.1kHz. Delays for individual tracks or whole groups of them are adjustable from the front panel up to 170ms, and a Track Copy function allows up to four tracks at a time to be copied digitally within the machine.

Along with the look of the machine, the feel also responded to user comments and gave the XT some of the solidity and responsiveness its predecessor lacked. ADAT's use of S-VHS tape has certain physical advantages, in that the width of tracks and the space occupied by each bit of data is quite large by today's standards. This provides a certain reassurance, but brings with it a comparatively clumsy operation, with much higher masses to be shifted about when winding and synchronising. This is compounded by the format's use of high running speeds (3x normal video speed) which means there's even more tape to wind to get from one place to another. The resulting relative sluggishness was a key point to be addressed in the XT, and major improvements were made to the transport, giving it a lightness, speed and directness comparable to the competition. This should have improved lockup times in multiple machine setups, another drawback on the originals, but as I only had one machine to deal with I'm not in a position to check.

THE REAR panel ditched the ¼-inch jacks in favour of phonos, a retrograde step in my view in terms of both the quality of the connectors and the compatibility with the earlier machines. At a stroke it ensured that the XT was not a plug-in replacement for the original in a studio using the unbalanced -10dB interfacing, which probably accounts for most of them given ADAT's particular areas of high penetration. Balanced +4dB inputs and outputs remain on a single EDAC, a useful and rugged cost-saver for permanent installation but a nuisance for a reviewer—ever tried

REVIEW

hiring or borrowing an ADAT flail with B-gauge jacks on the end? Strangely, both sets of analogue connections are active all the time, a good idea as far as output is concerned, but, perhaps, not ideal on the input side. Tascams switch the balanced 25-pin D-connectors out when phono plugs are inserted, which, I am told, can give problems when the switches get tired but otherwise prevent any risk of two sources appearing together on a bus never intended for mixing.

The other connections remain the same as on the original, with 9-pin D-connectors for sync in and out, and two ¼-inch jacks for simple remote operation including the use of the little remote keypad, which as before is supplied as standard. This offers basic transport control and access to the four important locators. The 9-pins have the same sync implementation as before, making the XT equally compatible with other ADATs, and the BRC. Besides this there are optical connectors for the ADAT multichannel digital signal, the format which has begun to establish itself as something of a standard because of its convenience and small size.

Like AES-EBU, the ADAT digital format supports word lengths longer than 16 bits, which means it can happily remain on the new machine. As the name implies, the XT20 not only has high-bit convertors, a common enough feature even on 16-bit DAT, but can now record 20 bits on tape, a facility previously only available on MO and hard-disk recorders in this market. Of course, various devices exist to interface with tape multitracks, from companies as diverse as dCS and Drawmer, to put long digital words on Tascam machines, but always by splitting them across tracks. Three or four tracks may be required for a stereo signal, bringing the available number down from the expected eight. This is not a problem for the kind of classical work where the dCS system, for instance, has found favour, as this is generally straight to stereo anyway and it wouldn't matter if all eight tracks got used up for two signals. For conventional multitrack work, on the other hand, this compromise undermines the usefulness of the

machine, particularly if it is to be used on its own and not as part of a multi-machine system.

This then is the breakthrough on the XT20, retaining the full functionality of the ADAT concept but with real 20-bit recording on each track—no data compression or other tricks but a resolution that until recently would have been regarded as too much to expect from this kind of format. Like the sample rate, the word length has to be decided before work starts and is embedded in the formatting of the tape. This means that you can't put 20-bit overdubs on to an existing 16-bit tape, although a complete 20-bit tape could be run in sync with a 16-bit machine for critical overdubs. For new projects, however, there doesn't appear to be any trade-off in setting up to work 20-bit throughout. Of course there must be; since the running time on a given size of tape remains the same, the data must be packed more densely on the tape in order to accommodate the extra bits, but as already mentioned the density of the original format is very low, leaving plenty of room for this kind of expansion. Alesis gives a cogent explanation as to why 20 bits may be regarded as near the real-world limit of the connected analogue gear, but if, despite this, you feel you desperately need 24 bits, the aforementioned type of technique can be used to achieve 24-bit recording using (according to the manual) the Rane PAQRAT or the ubiquitous Yamaha 02R to split the bits across multiple tracks.

Full copying facilities between machines are available as before, with the BRC capable of routeing various combinations of tracks between multiple machines. Since the optical fibre link carries a format capable of transmitting eight 24-bit signals there is, again, plenty of room for the new word length—one up to Alesis for making an affordable convenient format so future-proof when its capabilities could have been pared to the bone on the grounds of economy.

The increasing number of computer cards, convertors and mixers capable of interfacing directly with the ADAT format, notably the various components of the Korg SoundLink

system, makes this new increased resolution very portable, given software and DSP capable of handling it. There may be situations where the destination is limited to 16 bits, and the XT20 takes care of this with a choice of methods of reducing the recorded resolution. On the one hand, it offers dithering down to 16 bits, while on the other it is prepared to simply chop the last four off, on the basis that using truncated words and dithering later in the process may be preferable to dithering here and doing it again later on.

I was very favourably impressed by the XT20 on several counts. I saw the original ADAT very early on, when the whole idea of digital multitrack for less than six figures was big news, and it had the same impact on me as on most; that impact faded a little as we got more blasé about the idea and Alesis had company in the market. The XT revamp makes major strides in making up the lost ground, creating a machine that's comfortable to work with, does much more of what's wanted, and feels more at home in a professional environment. The 20-bit capability of the XT20 puts Alesis' nose back in front in one important area, although it surely won't be for long.

Sonically too, the results are impressive. The sound is smooth and effortless, with the only slight shortcoming being a possible lack of warmth in the lower middle—too subtle to be significant but I felt sure it was there. The low level performance is excellent, with the extra air and space that come with more than 16 bits. A striking example was a moment when I left the machine recording while playing something else back, which was audible only on cans on the floor of the studio. The resulting signal was almost too low to register, but cranking the monitors up showed it to be clear and precise, with any noise sitting below that of the open microphone and the monitoring system.

The XT20 has the potential to introduce MDM to more specialised areas than have currently embraced it. I would like, given time, to take it out on location for classical concert recording, particularly to record SoundField B-Format signals; not only would I feel at ease with it on the road, but it would fulfil the requirements of being an order of magnitude better than the final mixed DAT in a way not previously possible on this kind of machine. In other fields Tascam may be so well entrenched as to make it hard even for the XT20 to make an impression, but it could open things up in an intriguing way. Not only has ADAT come of age; it's moving us on to the next generation. ■

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SSL Axiom-MT

Amsterdam's AES Convention will witness the introduction of SSL's long-awaited digital mixing console. **Zenon Schoepe** goes in for an exclusive preview

HAVING COVERED pretty much all the areas of broadcast and postproduction that could see benefit in digital technology, it was inevitable that SSL would finally have to face up to the challenge of presenting a digital alternative to its market leading analogue music desks. I have seen it, visitors to the Amsterdam AES this month will see it, it will ship in October, and it is called the Axiom-MT.

An outline of what is being offered should set the scene. The MT can run to 96 in-line channels (192 automated mix inputs) plus 12 stereo effects returns to the centre section. It has 48 multitrack, 12 aux, and 12 main mix buses and each strip has access to 4-band fully parametric EQ with high and low filters and a separate compressor-limiter and expander-gate section. Automation is fully dynamic and snapshot with 5.1 capability.

However, most significantly of all, the Axiom-MT looks from even a few paces like a generic SSL analogue desk—bits of the look of a 9000 and a 4000—with bays of 8 channels loaded with in-line long and short motorised fader-equipped strips filled to brimming with pots and switches. It really does not look like the popularly accepted picture of a digital desk. There is predictably some familial resemblance to the Axiom, but the MT pips it substantially on knobs per strip. Dimensionally the MT is identical to the 9000, it actually uses its frame like the Axiom does.

We are looking at yet another manifestation of the original core Axiom technology that started the whole A-Series thing rolling; although it has to be said that this is probably the furthest out from the centre that the technology has been taken as the demands of the music mixing fraternity were seen to be specific and unique.

SSL stresses that the MT does not replace anything and does not signal the rapid abandonment of SSL's analogue desk interests. SSL want us to see the MT as an accompaniment to its analogue desks as the new digital board is described as a mixing product rather than a tracking desk. The later function remains in the lap of the SL9000 which is the company's answer to the requirement for 96kHz capability; although it could in effect be any other high grade analogue board. If readers cast their minds back some years they may remember rumours of SSL ARC (Analogue Recording Console) which while it never materialised was at least conceptualised as being part of a duo for analogue recording and digital mixing. It is also worth recalling that the 9000 and the Axiom were announced simultaneously and very loosely packaged along these lines even though the Axiom never had the blatant music production slant that the MT has.

This time it thinks it really has hit the mark—analogue tracking boards for capturing the source material in wide bandwidth on to the chosen medium. Thereafter the MT fixes it into digital with all the benefits of dynamically automated mixing.

The announcement of a digital music desk from SSL has been something that has been expected, yet the recent concentrated focus on broadcast and post made some believe that such a board would not necessarily emerge, particularly as SSL was keen to play down the appropriateness of such a product. So what has changed? In a word DVD. With the certain remastering of music for the new format(s) the majority of the workload will be at the remixing stage where a digital music console makes a lot of sense and one of the MT's primary features is its ability to pan in 5.1 and other surround formats from large and small faders.

As far as the interface goes you can either go for a physical channel strip per in-line path or you can go for a smaller frame and employ another layer accessed by single buttons in the centre section, per bay or locally on each strip. It is important to remember that 'layering' as a concept was originally decried by SSL in its digital desks, whereas AMS Neve and others adopted it with vigour. SSL then introduced 'banking' to the Axiom, but now freely refers to its desk's use of layers.

The 12U processor rack comes with four Highway ports each of which can split out to a range of I-O, such as 72 mic inputs, an analogue RIO with 48 16 or 20-bit I-O, a digital RIO with 96 I-Os, and an SDIF2 to Highway convertor. Or you can opt for the Hub router and access 24 Highway ports. This arrangement within the MT negates the need for a patchbay as you can control your routeing from the desk once everything is connected up and labelled within the system. DiskTrack will be a future option (the desk I saw was running DiskTrack), but it is clear that the company is keen, in the first instance, to promote the message that this desk can go in anywhere and interface to anything.

The strip itself is neither a 9000 or a 4000 direct lift; although it is close enough for jazz and familiarity with short and long fader paths, an aux section, EQ section, separately splittable filters, a dynamics section with gate and compressor, and gain. It is extremely important to note that this is the processing available per strip not per signal path in a strip and has to be divided between the two, just like on the analogue. I do not have a problem with this.

The desk is status-driven from traditional record, replay, and mix modes. Each bay has a routeing tile at the top for assigning what comes in to the strips as named source groups. Thus where you route is where you would expect to find it, at the top. However, there is a spare assignable routeing tile placed in the centre section that permits some of the more mundane work associated with a session to be performed while sitting. You can also load EQ and dynamics presets from here across the whole desk. Bay swapping, as on the Axiom, can be used to move bays closer to the operator for adjustment. Additionally, you can create bays of stereo channels.

The EQ has high and low-pass fil- >>>>>
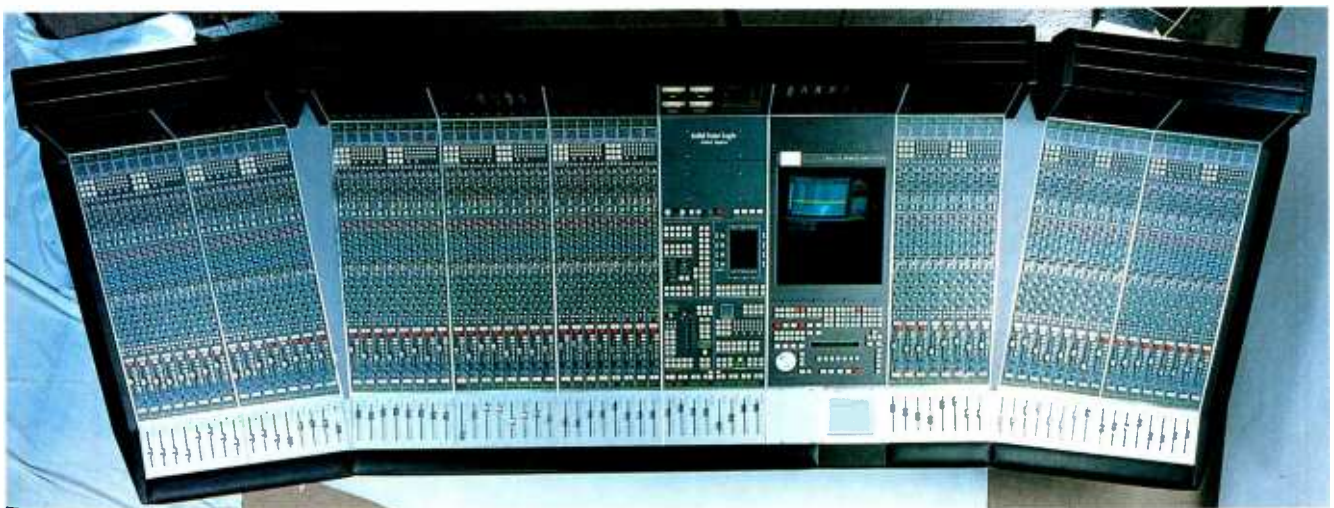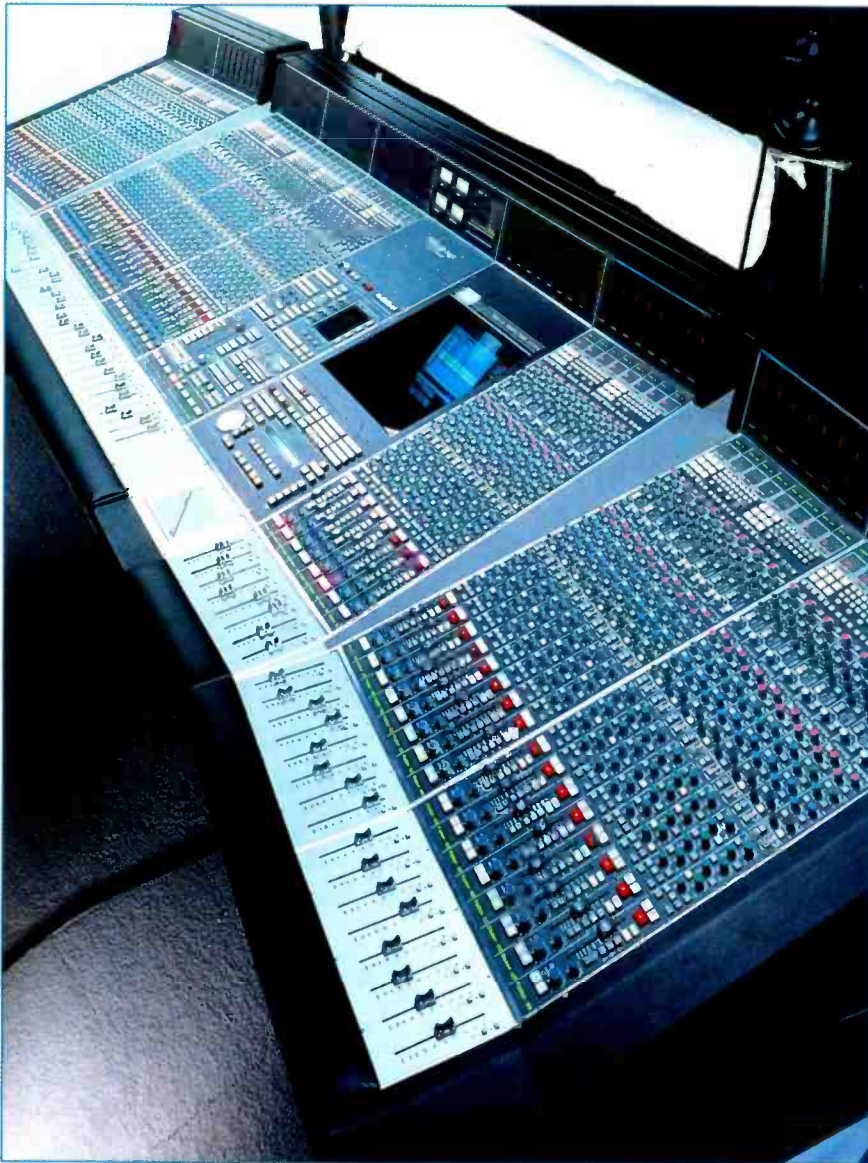
<<<<< ters, accessed from one pot, but usefully indicated by an associated display, while the 4-band fully parametric EQ has two pots per band—Q is assigned to the upper control on the press of a switch. These bands are frequency limited, just like the analogue, and not 20-20 arrangements. MT sports new cards that you will not find in the Axiom. The dynamics offer a compressor-limiter and a noise gate per strip via four controls which are switched between for the separate functions. The desk's dynamics use twice the processing employed in the Axiom.

Each strip has an external switchable insert with the allocation of patching controlled from the monitor.

There are 12 aux sends accessible from the long and short faders, but adjusted from six controls with each switchable for pre-post and groupable for stereo. Additionally, any send can be floated off the aux buses and sent to any of the 48 multitrack buses.

In all instances you get a numerical readout of the control you are turning simultaneously in displays at the top and bottom of the strip. The built-in monitor will also track adjustments made with multiband numerical read-outs superimposed over an EQ curve in the case of the EQ, and a graph in the case of the dynamics. However, the design is such that in reality you will only need to resort to

the screen for the patching, and the finer points of the automation, as the amount of desk visual feedback is enormous.

The continuous rotary controls have display rings around them that mimic the pointer on an ordinary pot cap and the EQ, for example, is even panel legended. You can copy section settings across strips and draw EQs and dynamic settings from a preset library. There are no onboard effects.

Two pan controls alter LR and front-back panning and can be switched between short and long faders with the formats dictated by the selected surround mode in each bay's output assignment routeing tile at the top of the strips.

The centre covers traditional monitoring functions, master compressor (linked 5.1 capability possible), fader, levels, comms and solo. External sources can be 6-channel with selectable large and small surround loudspeaker systems. A small screen with associated rotary controllers and cut and AFL push buttons are switch selected to perform the function of aux masters, effects returns, foldback, and groups. Motion control takes in one parallel port or four serial ports.

The automation is fully dynamic for all desk functions with snapshots and is effectively a new system oriented towards music, but retains terms like snap, autotakeover, autoglide, and match and play. You can >>>>>

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<<<<< choose to write in absolute or trim and the routines of join and revise and all replay are supported which SSL analogue users will be familiar with. The depth of the automation can be specified so if you want to break yourself in with just faders and mutes with the rest manual then you can. One of the great feel-good factor that engineers have about SSLs is that they become proficient in driving the automation computer. MT's package, while not identical to anything that has existed before in the analogue domain, has enough hooks in it to at least sound familiar initially. It is then up to the engineer to learn the extra to tackle full dynamic automation. All the controls can be dropped into write by touch sensing.

However, there are several automation modes that will be more familiar to users of SSL digital post systems. You can, for example, run the desk in a Roll-back mode, which basically writes every single twitch and press on the desk, or a Mix Fill mode. The latter allows EQs, for example, to be tweaked and tuned while listening to the mix, but only remembers the last setting that you arrive at and presumably you are happy with. Again, just like the analogue.

Five mixes can be held in a temporary buffer for immediate direct comparison on-the-fly and for permanent storage the mix plus the routeing, effects returns and everything else is stored in a project.

Dynamic automation is augmented by snapshots that reset the whole desk in less than a second. Cuts are frame accurate.

There is a long and short fader and cut grouping and the ability to group EQ and pan controls plus strip switches for instances where you would want to perform simultaneous multiple aux cuts, for example.

It really is quite a package and a familiar one at that. Getting the interface right on its digital music mixer is probably more important for SSL than anyone else as failing to do so could alienate the generation of engineers who have grown older while sitting behind their analogue ones. The MT manages to achieve the balance of incorporating established working practices and expectations in to a new board.

Anyone with even a passing acquaintance with the philosophy of how an SSL works will grasp the concept of how this digital desk works without too much trouble. It looks largely like an analogue music SSL and to a point it behaves like one. Probably the biggest learning curve resides in mastering the nuances and peculiarities of thought required to get a grip on full dynamic automation. Every engineer has to learn about this at some stage, but thankfully it is not an immediate prerequisite for the MT.

I am most impressed with the fact that you do not actually need to bother with the computer monitor to get the desk going. Things happen on a local level and while there are some dual function controls on the strip, they are obvious ones.

It is ironic how quickly the concept of assignability, the superstrip and a handful of strip controls gained credibility with digital desks when in reality it was introduced originally to allow the same work-surface to be able to attempt to satisfy different applications and, more pertinently, to keep costs down.

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All everybody ever wanted was a digital desk with an analogue feel and a knob and switch on the top for every function. This is still to happen, presumably while the slow process of evolution extends our arm reach to that of an orang-outan, but at this point in time the MT gets closer than the rest.

Yes, you can choose to opt for the smaller work-surface and the two layers, but I have a suspicion that most would probably prefer the visual impact of the fully expanded versions. Shipping will start in October, according to the company, for around the price of an I-O for I-O 9000 J Series. Indeed, if you were to think long and hard about the I-O side of the MT and identify what you could live without and go for the smaller, layered, surface then it would weigh in significantly cheaper than a 9000J.

While the MT is not intended to compete for sales with the 9000J on the grounds that the analogue board outperforms the digital one sonically, and because SSL sees the two as working alongside each other in tracking and mixing duties, I am not sure that their customers will necessarily see it that simply. As a tool for DVD mixing, absolutely, but I suspect that the digital moniker may sway some buyers towards the board with the intention of using it as an all-rounder simply because it will allow them to differentiate themselves from their local competition.

Either way the Axiom-MT is a surprisingly logical extension of the A-Series range and a very promising attempt to transfer analogue operability to digital for music.

The Capricorn and the Oxford have some company. ∎

Digital Audio Labs V8; Big Block; MinneTonka MxTrax

The competition for the nonlinear PC-based recorder-editor market becomes progressively hotter with every new release. **Rob James** assesses an American contender

AFTER AN UNCERTAIN START, there is now an amazing variety of choice in multitrack add-on Digital Audio Workstations (DAWs) for IBM compatible PC computers. A few short years ago, the only real choice was a Mac and Digidesign's Sound Tools or Pro Tools. Over the last couple of years, however, the personal computer market has become almost completely dominated by the Intel-Microsoft axis. The Apple Macintosh percentage share of hardware sales has dropped significantly as the Wintel platform has become more capable of challenging and arguably beating the Mac on its home ground of multimedia. This has resulted in accelerated hardware and software development for the platform as developers go for the biggest potential market. There is now a much larger range of viable choices for pro-audio use available on the PC platform than the Mac.

One of the early champions of the PC platform for audio was Digital Audio Labs (DAL) with its CardD and siblings. The subject of this article, the V8 card, Big Block and MxTrax software from MinneTonka (sic) are the multitrack offering from the company. The CardD family has established a solid reputation as audio interface cards for the PC. The V8 is attempting to capitalise on this success in the multi I-O plus DSP market.

The V8 card, MDM custom interface for ADAT and Deuce Coupe stereo interface arrive in rather garish packaging featuring the kind of automotive cartoon that will be immediately familiar to those who remember *Cozmic Comix*. The manual for the hardware is similarly adorned.

The main V8 card is a full-length ISA type. This is a little unusual these days and may cause some head scratching over installation

in certain machines, but should not cause problems with ATX form factor motherboards. The main reason for the enormous size of the card is the capacity to accept two memory SIMMs (Single In-line Memory Modules) in acutely angled sockets and up to three daughter DSP boards that in turn accept 'piggyback' memory modules (big fleas have little fleas upon their backs to bite 'em and so *ad infinitum*). There is also a 40-pin connector for ISA bus I-O option cards such as the MDM Custom interface. A further 14-pin connector allows connection to a CardD Plus which is another I-O option. The external connection is one 50-pin miniature D 'V8 bus' socket and, despite all these options, the board can happily co-exist with others immediately adjacent.

THE V8 USES two Motorola 56002 DSPs and is capable of sustaining a maximum of 16 tracks of record or replay. The limiting factor is system 'throughput'. This in turn depends on the ISA bus speed, the application software overhead and the disk and controller performance. Equipped with a suitable SCSI interface and SCSI disk with a spindle speed of 7,200rpm or better, the V8 should be able to sustain somewhere between 8 and 16 simultaneous tracks. The V8 supports a maximum of 16 active inputs and outputs at

any one time. A 100MHz or better Pentium processor and SCSI storage for audio are strongly recommended by Digital Audio Labs.

Provided there is sufficient physical space for the card, installation is kept drama-free with only the I-O memory address as a configuration issue. The default value of hex 300 can be simply changed by means of a DIP switch on the board if this address is already in use by another card or whatever. In fact my card arrived set to hex 240 which was already in use. Software installation—the driver(s) for the card is reasonably straightforward and well dealt with in the manual. There is one API driver specific to applications written for the V8 and four others may be installed to allow multiple cards to work with 'wave' applications—other software such as Sonic Foundry's Sound Forge. The catch is, the optional daughter boards, each containing two further 56002 DSPs for EQ and effects processing will only work with applications written specifically for the V8. Any application that DAL considers suitable will carry a 'Gearhead Approved' sticker. Meanwhile the only applications capable of making use of this expensive extra horsepower are MxTrax from MinneTonka and a V8 specific version of the estimable Waves Power Pack plug-ins. For other applications the V8 behaves as a collection of stereo sound I-O cards. All internal digital audio processing is quoted as 24-bit which does not allow a huge amount of headroom when equalising.
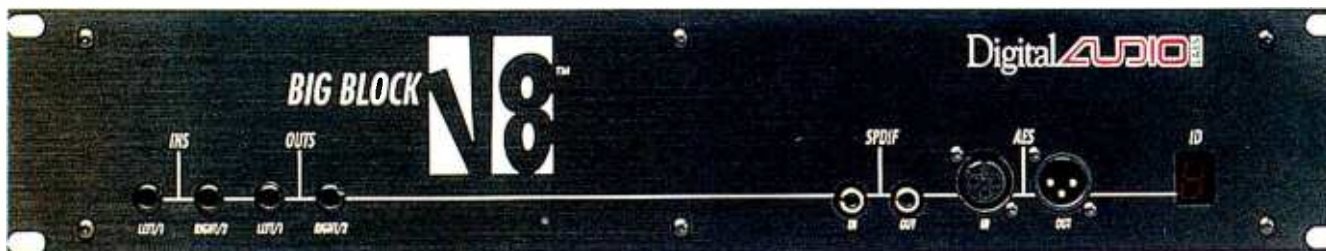
Software supplied includes a Tachometer that gives an indication of resource use and diagnostics to reassure you all is well with the card.

Continuing the automotive metaphor, the Big Block breakout box is a 2U-high, 19-inch

rackmount unit. The rear panel carries two V8 bus connectors, this time of the Centronics type—note though, that the cable used to connect the V8 card to the big box has the same connectors as one flavour of SCSI cable. For all I know, it may even use the same configuration, but I wouldn't like to risk it so, if your SCSI system uses these connectors, label the cables.

Analogue I-O is catered for by two rows of ¼-inch jacks. Digital I-O options are 2-channel; phono sockets for SPDIF; and a pair of TOSlink connectors for optical SPDIF compliment XLRs for AES-EBU I-O. The front panel has XLRs for AES-EBU and phonos for SPDIF. Only one of these five digital interfaces may be used at any one time. The analogue Channels 1 and 2 connections are also brought out to the front panel on jacks. The inputs are arranged such that inserting a jack into the front panel sockets disconnects the rear, but the front and rear outputs may be used simultaneously. A prominent LED alphanumeric display indicates power on with a full stop and the ID number assigned to the Big Block.

The V8 bus connectors allow several Big Blocks to be daisy-chained to enable the creation of large and complex I-O systems. The Big Blocks are fully routeable. Other I-O options may restrict the total number of inputs and outputs available at any one time. The A–D convertors are dual 16-bit sigma-delta 64x oversampling types and the D–As are similar 8x oversampling devices. The quoted signal to noise figures are unusually good for 16-bit devices; although no weighting is given. Levels are selected via the MxTrax software. +4dBu balanced or -10dBu unbalanced with trim of +0.5dB -19dB if the +4dBu option is selected or +6.5dB and -15dB for the -10dB option. Annoyingly, there appears to be no way to make these selections when using non-V8-specific software. It would have been nice to see these options as part of the wave driver software. There are no separate word-clock connections on either the V8 or Big Block. Digital sync must, therefore, be either internal or sourced from digital input.

THE APPLICATION software supplied for review is MxTrax from the implausibly named MinneTonka software. This is a V8 specific 16-track recording, editing and mixing package. The minimum system requirements for this would have been daunting a couple of years ago, but seem relatively mundane now: 100MHz or better processor, SCSI hard-disk subsystem with 7,200rpm spindle speed drive, the DAL V8 card and an I-O option plus 24Mb of RAM and a monitor capable of at least 800 x 600 resolution.

Software installation is straightforward and uses the standard Install Wizard. The first step after software installation is to set up a default monitor output to allow the tutorial files to be heard. These provide a painless introduction to the software and I would strongly advise taking the time to work through them.

The main screen is divided into two main sections—track displays at the top and mixer at the bottom. A splitter bar, that divides the two areas, can be moved with the mouse to reveal more tracks or more mixer according to the current task.

The transport arrangements are fairly conventional with keyboard shortcuts supplementing the on-screen buttons, useful when mixing. The track display is of the moving-cursor type, so the screen redraws when the edge is reached. Recording involves first setting up or loading a previously designed suitable mixer in order to route and monitor input signals. With a blank track display you will need to add a track or tracks to record into. The tracks can be named by selecting them and entering a name. Selecting the REC READY button on the track display arms the track ready for recording and you can then record using the transport controls. The waveform display in individual tracks can be scaled by ±3dB and a single track may be expanded to fill the widow for viewing fine detail. The track display can also be zoomed in time.

Editing is done with either Regions or Areas.
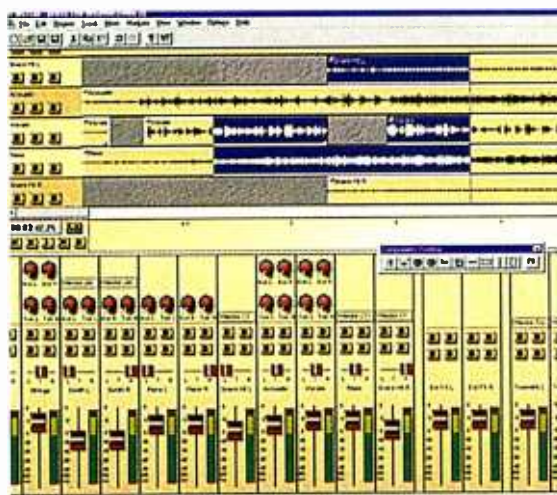
MxTrax from Minnetonka Software

A Region is the complete waveform area on screen that corresponds to one soundfile. An Area is a selected portion of the waveform display in a highlighted track or tracks and can spun across Regions. The editing is pretty rudimentary; cut, copy and paste with no fades or crossfades. I am worried though, about the lack of stereo track functions. It is too easy to move one half of a stereo pair and the only ways of getting it back in sync are either manually or by using Undo which may be highly inconvenient if other edits have been performed after the error.

The mixer section has a Components Toolbox. From here individual building blocks are dragged on to the mixer area to build up complete mixers. These can be saved in their own right or saved with projects. It is suggested channels should have a Channel Name text box at the bottom of the strip followed by a fader, MUTE, SOLO, AUTOMATION and RECORD button block and a PAN control block. Other blocks in the palette are an AUX with 2 aux send and return controls, various spacers to pretty things up and, if you have the optional DSP card(s) 3-band parametric EQ. Input and output assignments are made with pop-up menus attached to the relevant blocks. Pop-up menus also allow channels to be copied, pasted, saved and inserted or deleted. Each fader block is equipped with a meter, the usual graphical representation of a LED ladder. Buses can be freely created and one of the pull-down menu options allows the busing arrangements to be reviewed.

The RECORD button on the channel strip performs the same function as the RECORD READY button on the track when the strip is set up as a track input. The fader automation is welcome and cleanly executed. Global write will probably be the starting point for most users, followed by selective updating. There is no direct provision for saving mix passes, but a work-around is to save the project with a different name.

I also used the hardware with Sonic Foundry's Sound Forge software that behaved perfectly normally using the V8 hardware and wave drivers simply for I-O.

Other hardware options include the MDM Custom Interface For ADAT which uses another ISA slot and adds connectivity for multiple ADAT machines including digital audio and machine control. Since there is only one pair of optical connectors only one machine's audio output can be accepted at any one time. The software allows any connected machine to be selected and controlled and, of course, several machines can be run in sync together in normal ADAT fashion. If there are enough free slots in the PC and your pocket is deep enough, multiple MDM Interface cards may be fitted, daisy-chained together with the V8 via a ribbon cable.

The Deuce Coupe is a stereo analogue and digital board that will provide separate stereo mastering and monitoring. For purchasers already in possession of a DAL CardD this may be integrated in a similar way; although use of any of these options may reduce the total number of inputs and outputs depending on the specific configuration.

In conclusion, the DAL V8 system is solid, with a good sound, if rather unexciting with the MxTrax software. The interface is clear and uncluttered and the learning curve is fairly flat. Anyone familiar with other PC and io packages will quickly find their way around.

The available hardware options are almost bewildering in their profusion and serve to partially disguise the high cost of a fully configured system. A lot of effort has gone into the hardware design, but I feel the lack of software support is a handicap. The system looks expensive in the context of systems released to the market in the last few months, well into 'one box' dedicated hardware territory. I think this downward trend in prices is set to continue for a while yet, and will further disadvantage the V8. Unless there is room for manoeuvre in reducing the prices and DAL can either fully open the platform to a wider range of existing software, or persuade other software authors to support it, the appeal would be seem to be somewhat limited. ■

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Steinberg Cubase VST plug-in effects

Steinberg's Virtual Studio Technology incorporates master and mixer effects into the Cubase multitrack audio environment via a plug-in architecture. **Simon Trask** looks at the standard setup

A S DESCRIBED in last month's article on plug-in audio effects for MIDI+Audio sequencers, the computer-based MIDI sequencing environment has expanded in recent years to incorporate digital audio recording, to the point where sophisticated multitrack audio setups complete with virtual mixer and effects processing capabilities have become standard features of the market-leading professional packages.

Steinberg's Cubase VST MIDI+Audio sequencer is one such package (currently at v3.5), available in versions for the Mac and PC platforms.

The company's Virtual Studio Technology provides a multitrack digital audio recording and mixing environment complete with master (stereo bus) and mixer (channel send-return) effects that can be slotted into graphical 'racks' via an open plug-in architecture. Last month's article provided an overview of the wider MIDI+Audio plug-ins environment as a prelude to more specific coverage of audio plug-ins in what is a growing area of digital audio recording. Steinberg's cross-platform (Mac-PC) VST plug-in format has attracted significant third-party support and established Cubase as the premiere audio-and-effects package for MIDI+Audio users, making it the logical package to look at initially. However, as MIDI+Audio packages may be unfamiliar ground here, before we concentrate on the plug-ins themselves it is worth taking a look at the general audio capabilities and architecture of Cubase, and at how plug-ins can be utilised within the Cubase audio environment.

As discussed last month, Steinberg not only developed their own VST effect plug-in architecture, they also published the spec in order to stimulate a market in third-party VST plug-ins—a decision that has proved well judged. The top-of-the-range Cubase Audio XT package also supports Digidesign TDM plug-ins (as does rival Emagic's new top-of-the-range Logic Audio Platinum package), but these require Digidesign hardware. The advantage of VST plug-ins is that they can run without the need for add-in boards. Cubase VST technology running on a PowerMac can utilise native audio processing to run all its EQing and effecting purely within the computer, while the Mac's built-in audio connections allow CD-quality stereo audio input and output as standard in conjunction with the Mac's Sound Manager software; you can also record direct from CD using the computer's built-in CD-ROM drive and standard AppleCD Audio Player software.

Steinberg's software package also supports a growing number of PCI-based audio I-O cards via its ASIO (Audio Stream Input-Output) open audio interface protocol, enabling more sophisticated audio I-O options for Cubase than the Mac's basic stereo configuration. For instance, Korg's 16-bit 12:12 I-O PCI board (£599 UK) gives you 12 inputs and 12 outputs (in each case eight digital and two analogue—the digital I-O being via the ADAT multichannel optical interface).

The Cubase range consists of Cubase, Cubase Score and Cubase Audio XT (priced at £329, £499 and £619 respectively for Mac and PC versions alike).

All three packages are audio-enabled and utilise Steinberg's VST technology, though the Audio XT package requires that you use only Digidesign hardware and Digidesign's TDM plug-ins, but also includes a VST version as an alternative option. Steinberg also market the entry-level Cubasis AV package (£129), that has a scaled-down version of VST with eight stereo audio tracks. Cubase and Cubase Score can each handle up to 32 channels of Mac-native audio using Steinberg's VST technology, while Cubase Audio XT can handle up to 48 audio channels (Mac) or eight audio channels (PC) using the appropriate Digidesign hardware.

In the Cubase VST environment, MIDI and audio data co-exist in a common graphical multitrack setting, with individual tracks classified as either MIDI or audio tracks. You assign an audio Track to one or more audio channels (discrete audio data streams), with of course two channels being used for recording a stereo audio source. For both MIDI and audio Tracks you can record into any bar range, creating a Part within the Track. You can then move and copy Parts around within a Track or between different Tracks by dragging or Option-dragging the relevant graphical blocks in the Arrange window. Copying does not duplicate the audio data, however; rather, it creates new Audio Events and associated Segments, which essentially are pointers into the audio data. As you record and edit audio Parts, you build up a common pool of Segments for your Song, available via the Audio Pool window. You can then drag Segments into Cubase's Arrange, Audio Editor and Wave Editor windows, automatically creating a new Part complete with Audio Event and Segment information.

An on-screen virtual mixing desk (the Monitor window) provides fader, pan, solo, mute and input channel controls for each channel strip, together with FX and EQ buttons which call up a combined Effects and EQ window for the channel when clicked on with the mouse. Each audio channel has its own input on the mixer, so of course any Track(s) assigned to a channel will automatically be routed through the relevant channel strip. You can enable up to four bands of built-in parametric EQ per channel, with an enable ox-orr button for each band; the actual total number of bands you can use will depend on the processing power of your computer, though.

B Y CLICKING on the WRITE button in the upper left of the Monitor window you can enable a special Audio Mix Track and record an automated mixdown using the fader, pan, mute, solo, FX and EQ settings. Playback of the mix is enabled by clicking on the READ button. You can record into any bar range, loop the range, and overdub as many times as you need to in order to build up a full mix. It's also possible to build up multiple mixer Tracks, one for each channel, so that you can mute and delete mixes for individual channels. All mixer settings can also be edited numerically or graphically per channel in List Edit mode. Multiple audio Tracks, including the audio results of all enabled automated mix settings, can be bounced down onto a new mono or stereo Track in order to free channels for new recordings, to turn a stereo track into a mono track, or simply to 'fix' the automated mix of a track into the audio data.

Cubase VST has two virtual effects racks, one labelled Effects and the other Master Effects. Each rack can hold up to four effects, which you select from a 'pool' of effects plug-ins stored in the VstPlugIns folder within the main Cubase VST folder. Cubase comes with several plug-ins as standard; others are available either free from the Internet

Plugs.eff. Choirus and Choirus 2 are versatile chorus-flanger-phaser processors which can generate some pleasingly smooth, rich-sounding effects. You get 32 program memories, with time (measured in samples), width, LFO frequency, feedback amount, feedback balance, glimmer, and output level parameters. Espacial! is a versatile if not particularly rich or smooth reverb processor that can be used as a Mixer Effect, with a total of 64 program memories and size, width, time, ER start, ER width, ER gain, ER decay, ER output, and output level parameters. The Externalizer is a Master Effect designed to give a more spatial sound for headphone monitoring by creating the 'crosstalk', frequency and delay characteristics that come with listening to external speakers; in practice I found it to be a subtle yet pleasant effect. Fuzz is a little Electro Fuzz 'footpedal' Mixer Effect with 16 Programs and boost, clipback and volume knobs—simplicity itself to use, and great for creating cruddy sounds—while Panner is a simple yet versatile and effective auto pan module, available for the Mixer Effects rack only, with 16 programs and LFO Frequency, Width, Waveform (sine, pulse, sawtooth, triangle) and Out Level parameters. The Stereo Echo is a Mixer Effect that provides 16 Programs and delay, feedback, balance and volume for Left and Right signals, with an option to link the Left and Right settings. Wizard is a subtle stereo enhancer effect, with eight Program memories and just two parameters (width and LR/RL reverse), while Wunderverb3 is a reverb plug-in with a small edit window that provides size, decay and damp parameters on three graphical knobs, plus ten Programs based on different-sized spaces: Hall, Large Hall, Large Room, Medium Room, Small Room, Plate, Gated, Effect One, Echoes, Effect Two. A straightforward yet versatile processor, with a pleasingly smooth, dense reverb quality.

Although they slot into the effect plug-in architecture, Scopion and Tun-a are utilities rather than effect processors. Scopion is a Master Effect that provides an oscilloscope display of the master stereo output. You can select Left or Right channel and adjust the vertical and horizontal axes of the display by rotating a couple of red knobs using the mouse. Tun-a, a Mixer Effect, is a graphical chromatic tuner with a built-in tone generator: you can select from eight tuning scales, set the master tuning in the range 420–460Hz, and select any one of four octaves for playback, then scroll through the chromatic notes using up and down arrows.

Steinberg's VST effects plug-in architecture and user interface provide a well integrated and reasonably versatile extension of the Cubase environment's audio recording and mixing capabilities. The standard effects themselves are of middling quality and offer a reasonable degree of programmability, but they are not about to signal the demise of the dedicated rackmount multi-effects unit, nor does the VST setup match the depth and flexibility of the traditional studio combination of patchbay and effects rack. Still, the whole point of the VST plug-in concept is to provide an open-ended effects environment, and future reviews will look at the growing range of third-party VST effects plug-ins. ■

(go to Steinberg's Web site at www.steinberg.de/english/index.html as a starting point) or as commercial packages from the likes of Prosoniq, Waves, and Steinberg Spectral Design. To add new plug-ins, you simply install or drag them into the VstPlugIns folder. Plug-ins can also have .eff folders, which go in the VstPlugIns folder; these each contain individual effects Programs and a Program Bank which can be loaded into the effects processors via the FILE button on the processor's front panel, giving you a ready-made collection of effects settings.

Within the program, you select the Effects and Master Effects windows from the Audio menu. Each window has four 'slots', with a parameter field initially labelled 'No effect' in each slot. Click on this field within a slot and a drop-down menu presents you with a list of all the effects in the VstPlugIns folder. When you select an effect from the list, a graphical front panel appears in the slot; you then have to turn the effect on by clicking on the graphical Power button, that lights up the front panel's visual display. You can actually assign the same plug-in to multiple rack slots to give you several completely independent processors. For instance, you could assign the Choirus plug-in to all four Effect rack slots, assign a different Program to each processor, and then enable all four effect sends for a particular channel. In this way you can readily create 'composite' effects, though of course at the

expense of being able to use a variety of plug-ins.

Selecting and editing Effect and Master Effect Programs is wonderfully straightforward and intuitive. You can select different Programs and make changes to any parameters live and hear the results in context—and also you can record all changes, including send and effect levels, as part of an audio mix, overdubbing in loop mode as necessary. The immediacy of having all these effects grouped together in front of you on-screen, under your 'mouse hand', is impressive.

There are two types of front panel: Internal and Rack Xpander. Internal effects allow program selection and all editing to be done from the graphical front panel using page and parameter buttons and a value dial.

Rack Xpander effects have a standard interface of Program and Edit buttons.

In this case, clicking on the Edit button calls up a separate window on screen; the design of the editing interface in this window is completely up to the plug-in developer. Once you have edited the selected Program, you simply click in the window's close box in the usual manner to get rid of it.

Cubase comes with the following effects plug-ins as standard: Choirus, Choirus2, Espacial!, Externalizer, Fuzz, Panner, Scopion, Stereo Echo, Tun-a, Wizard, and Wunderverb3. There are also three .eff folders: Choirus.eff, Espacial.eff, and Spectral Design

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Studer D950S

Studer is now on its third generation of digital consoles and its
latest is arguably its most adventurous. **Zenon Schoepe** reports

SOME DIGITAL consoles are far easier
to categorise and pigeonhole than
others simply because their degree of
digitalness is less or more visible. At the
extremes we have consoles that attempt digi-
tal approximations of analogue desks while at
the other end there are creations that have
only fleeting glimpses of analogue analogies.

I will start by saying that the full capabilities
of Studer's D950S are not easy to communi-
cate in the written word because it has a
degree of configurability that makes it difficult
to pin down and offers an answer for most
eventualities. It is easier to demonstrate than it
is to communicate. While undoubtedly an all-
rounder it won't be a desk for everyone sim-
ply because many buyers clearly like
something a little more fixed. Yet it is this flex-
ibility that is one of the D950S' most impres-
sive features and will endear it immediately to
those of a like mind.

What I will touch on here are some of the
ideas and concepts behind the board and
some of the unique and interesting routes the
company has taken to offer a digital desk that
cannot be described as ordinary or plain. Most
particularly I will be looking at the S in D950S.

To understand the lineage of the D950,
which represents the third generation of
large-scale digital desks from the company
and which was launched last June with the
D950S (for surround) version debuting at the
US AES, you have to go back a few years. The
first effort was the D920 some ten years ago
as a pure Studer desk. Following Studer's
acquisition of the Digitec brand and its Virtu-
oso digital console a collaboration ensued

resulting in the D940 (broadcast general pro-
duction desk) and the D941 (on-air) version.
Digitec contributed the DSP and I-O of the
board while Studer did the rest. The D950 has
a redesigned work-surface of the previous
series, but everything else is new. The D950
should not, however, be associated with
Studer's 2000 on-air desks with which it only
shares similar convertors.

It has a completely new platform and
where the last generation was Motorola 56000-
based the newest product is SHARC-based.
The structure is such that it starts with a mass
of processing horse power that can be config-
ured and allocated according to what the user
wants. Studer prides itself on the custom
aspect of its analogue desks and wants to be
seen to be offering the same in the digital
domain. It is intrinsically scalable without ded-
icated cards for specific functions and can thus
create very small or very large systems using
the same boards. The work-surface is also
scalable and the manner in which the avail-
able processing is mapped to the surface is
also flexible.

The difference between the plain D950, or
D950B (for broadcast) to give it its proper title,
and the D950S is the latter has all the functions
of the stereo original desk, but adds surround
panning, surround monitoring and a few other
tricks in terms of machine control. A D950B
can be upgraded to S status which adds about
10% to the cost. They are available in multi-
ples of four fader modules up to a maximum
of 256 faders per desk.

The basic console comes with snapshot
automation, but dynamic automation and an

automation control package are optional. The
surround packages includes a surround moni-
tor panel plus surround source selectors and it
will extend to 7.1. There is also a joystick
option and software for advanced panning
capability. Other options include machine
control as the basic dynamic automation pack-
age comes with a 9-pin serial port. For film
there is the PEC-direct panel option.

With regard to I-O, consoles have combina-
tions of TDM bus satellites into which A–D
cards can be plugged and are then multi-
plexed onto MADI, direct digital connection
and Studer D19 mic A–D pres front ends.

THE D950 uses an off-line approach to the
assignment of DSP power that involves
specifying what you want and then the
software optimises the performance, reduces
propagation delay and creates the desk. This
is done by Studer when a console is spec'ed,
but if you want to be able to reconfigure it
yourself, and power users will, then you need
to buy this Configuration Tools package.

The desk surface connects to the rack via
two optical cables, a multiplexed cable for
screen, keyboard and mouse plus any cables
to feed comms.

Things that can be included on a channel
include 4-band EQ, channel insert, a fader,
input source, high and low-pass filter, phase
reverse, a direct out and the metering. You can
specify multiformat or stereo pan, delay, mas-
ters, auxes, dynamics, solo, PFL, and bus
access. How many buses you have is up to
you within the limitations of the amount of
processing fitted and you can add EQ. >>>>>

<<<<< for example, to any signal path.

A 36-channel desk can live on just two cards, but 20 cards can be housed in the rack for seriously large systems or serious amount of redundancy as the more you fit and the smaller the configuration you choose the more cards are available to sit, scan and kick in to take over from any card that fails. This is an interesting point as a broadcaster could tune redundancy through configuration for particularly important events.

This stuff really does not go down. I witnessed cards being switched out and pulled out of the rack numerous times, with only the smallest of clicks on one occasion. In all other instances the audio was uninterrupted on a 48-track playback. Almost uncanny. In fact if the screen had not thrown up an alert you would not have known anything had happened.

As regards general operation, each channel

can work to ten layers of signal path organised as five banks with two layers per bank. You tweak by assigning functions to strip controllers—there is no centre-section superstrip.

Strips have four identical continuous controller sections each with a 4-character display showing numerical values and parameter name, a switch for performing such things as pre-post switching when dealing with auxes, an LED bar to give a visual display of the parameter value and an associated LED that shows centre for those parameters that have one. You assign functions to these controllers globally from a Master Menu Selector panel that has dedicated buttons for EQ, filters, compressor-limiter, gate-expander and the auxes. Taking the example of the 4-band 20Hz–20kHz fully parametric EQ, you press the EQ button and the controllers show one band of EQ for the strips across the desk. Four PAGE buttons in the same centre section select each of the four bands one at a time again across the desk, but you can also select the processing to be displayed and the pages locally on the strip via buttons. Similar principles apply for the other functions.

Aux levels additionally can be flipped over to the faders for setting as well as copying fader levels directly to the auxes on a single button push. Pan control can also be allocated to the strip rotaries with LR and front-back and additional surround parameters.

The channel strips are fairly busy because you have a wealth of local function select buttons and as such are different to most other digital desk control implementations. In this respect the D950 will not be to everyone's liking.

Automation is straightforward for a dynamically automated board as you only need to write over previously recorded data using the touch sensitivity of the controls. There is global control of read, isolate and write in absolute and relative values. There is also auto takeover and glide.

As far as surround goes, a mono channel can be equipped with a stereo or a selection of surround pans—in multiformat and Studer's Virtual Surround Pan (VSP) modes. The latter is possibly the strongest feature of the D950S and involves a library of software panning functions that allow the placement of sounds within virtual 3D spaces. Grand claims, but read on. Positioning is calculated within the DSP and is optimised for all surround formats. You get the usual controls of LCR pan, front-back pan, LSRS pan, and divergence plus a frequency-dependent pan pot with variable frequency panning filters permitting intensity-based and delay-based panning. An adjustable number of discrete echoes can be produced and routed as noncorrelated, diffuse signals to the surround loudspeakers from any console channel. Echoes are controlled by adjusting ambience, source distance and room size parameters which allow natural reproduction of audio sources from various distances and positions within a virtual room without external effects. It can approximate Doppler effects and sounds disappearing into the diffuse room. All parameters are automatable.

Pan can be performed via joystick or by using strip rotaries. How it works in >>>>>

28

practice is quite convincing as the immediate impression is one of the straight lines of conventional panning being replaced by curves that give the impression of a larger and more circular sound stage. There is certainly more depth to the sound and the feeling that the centre speaker, for example, is further back than it actually is. A neat demonstration is the routine of moving a sound from the edges of the virtual soundfield towards you as this is accompanied by a subtle tonal change that is not dissimilar to real life. The effect is easily heard especially when compared to traditional panning of the same move and I have no doubt that in skilled hands it could be put to quite dramatic effect. You have to hear this for yourself.

It is especially good at positioning mono sources in a surround space by virtue of generating echo signatures for the designed area and, consequently, is a prime contender for DVD remixing work. What Studer has created here is effectively a multichannel ambience processor that is tied into the panning of the console.

There is a multiformat monitoring panel with monitor format selection, pre-post decoder monitoring, meter to monitor switching, and an additive mode selector. The D950S also sports so-called 'dynamic stems' that allow stems to be configured and reconfigured as the need arises.

Not unrelated is the fact that desks can be split for up to four operators, but can also be physically split and run off the same system

as a type of networked desk arrangement with the exchange and sharing of data plus zoned automation.

If you get to see the configuration routine on this desk then you cannot miss the phenomenal potential power of the D950S. It is advanced stuff and a decidedly individual approach to traditional problems. Applicable in one form or another to music recording, production and post environments, it is so obviously attractive to broadcasters because it can be customised for methods of working, but it can also be configured to behave as a completely different desk for completely different applications. It is a lot of flexibility for your money that is also long lived as you will be able to alter and expand the desk fundamentally as your needs evolve.

The work-surface won't be to everybody's taste, particularly the strip pages, and it is different to other manufacturer approaches, but not necessarily unfriendly. I do not doubt that if you really know your configuration inside out and have it set up perfectly for you, then it must be an incredibly fast surface to work on.

More than anything else Studer has a real digital contender in the D950S with all the entry requirements to make it qualify for consideration. It feels like a mature product.

It is reassuring that while the simplification of digital desk's potential seems to be a goal for many manufacturers, Studer can still come out with a console that can be configured to be as simple or as complex as you need. ∎

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VSP

THE SURROUND CAPABILITY of the D950S is extremely well organised and offers own-brand Virtual Surround Panning (VSP). The perception of the distance in a room is largely determined by the time and spatial distribution and the location mechanisms for left-right and front-back are different. To imitate this, in addition to conventional means, delay and filtering is used and the coefficients of these filters vary depending on the selected pan pot angle.

The delay line represents the time difference between the sound arriving at the left and right ear while the principal function of the variable FIR filter is to imitate interaural intensity differences.

Allpass filters can perform a uniform phase rotation of 180° above a certain cut-off frequency to permit sources to be placed outside the speaker base.

Studer says that VSP can be used for surround recordings where main microphones are supported by spot mics and for the creation of synthetic stages during remixing.

Hear it.

The Sony Freedom

Escaping its enduring shroud of secrecy at last, Sony's low-cost Freedom UHF radio microphone system attracts the attention of **Neil Hillman** who asks if it will fly

THE MUCH HERALDED, but somewhat elusive, Freedom series of radio microphones was finally launched by Sony in February at the Frankfurt Musikmesse—a full year after its original launch at the same venue. With a name suggesting an item with wings (in stock at a chemist near you) Sony is keen to enter the budget UHF market, which is at this time the fastest growing microphone sector. Interestingly, the Freedom series is being distributed by Sony Professional and Broadcast in the UK, even though Sony regards the Freedom range as semi-professional, with its intended market being conferences, exhibitions, musicians and entertainers in pubs and clubs.

Sony's stated intention with the Freedom was to break new ground in the price *vs* performance, ease of use and set-up stakes and certainly several features within the range show that they have been single-minded in their pursuit of this; not least of all by their delivery of a unique modular-form 6-channel diversity receiver in a standard 19-inch, 1U-high rack.

The main rivals to Freedom in the 800MHz, UK channel 69 band will be the Trantec S4000-series and the AKG WMPT 300-series, both in price and features, an area addressed by the Freedom's off-the-shelf guitar, hand-held and lavalier-boxed system-packs approach.

The full Freedom range comprises of a hand-held microphone transmitter, WRT 800A, a belt-pack transmitter, WRT 805A, and diversity receivers in the form of either a single channel receiver in a half-width 1U-high, 19-inch rack, WRR 800A, or an expandable receiver supplied with one channel module and accepting up to 6 modules in a standard 19-inch 1U-high rack, WRR 801A.

Both transmitters and receivers use PLL frequency synthesis providing frequency stability, and frequency values are stored on a User Group preprogrammed channel plan allowing for intermodulation-free operation for up to 14 channels simultaneously, as well as access to the individual 64 frequencies available in the UK 8MHz bandwidth.

Sony claims that the Freedom's cost advantage over the existing 800-series has been achieved by providing a limited range that does not have to bear up to the demands of day to day broadcast and location operations, and this certainly becomes apparent when you listen to the cardioid electret condenser hand-held mic which does sound rather average. Channel selection for this device is achieved by unscrewing the bottom half of the microphone body revealing two small rotary switches adjusted by a small screwdriver, thoughtfully provided and clipped into a recess on the inner body. Unscrewing the body also accesses the battery compartment that houses one 1.5V AA battery that gives life

to the 5mW transmitter for typically 12 hours, and both the hand-held and belt pack transmitters share a tiny ¼-wave helical stub antenna that I suspect owes much to the Sony

> Sony's stated intention with the Freedom was to break new ground... and certainly several features within the range show that they have been single-minded in their pursuit of this; not least of all by their delivery of a unique modular-form 6-channel diversity receiver in a standard 19-inch, 1U-high rack

mobile phone factory's parts bin. A slider switch is mounted on the upper part of the microphone body, with a red LED showing when the transmitter is switched on.

The belt-pack transmitter is nicely contoured in plastic to curve with the wearer and the belt clip allows for the pleasantly lightweight unit to be worn either way up; although the battery retaining clip >>>>>

<<<<< gives the distinct impression of being extremely fragile by screaming 'break me!' each time it is operated. This 10mW device is again powered by a single 1.5V AA battery and by simple maths, the battery life is reduced to typically 6 hours.

Frequency response for both transmitters and receivers is said to be 100Hz-15kHz, but to use this optimally, the ECM 77 lavalier microphone would need to be used rather than the cheaper ECM 301, 44 or 121 microphones also available. The microphone connector is either via locking 3.5mm jack or an optional Hirose plug.

The input to the belt-pack transmitter is switchable between -40dBv and -60dBv, a tiny, fiddly plastic slider switch is set on the right-hand side of the unit next to a similar-sized phase reverse switch; a nice touch that could be useful in multichannel setups. Frequency selection and input attenuation of up to 21dB are achieved via 3 recessed buttons—SET and + and - mounted on the front face below an LCD screen showing selected channel, attenuator setting and cumulative operating time. A low battery warning is also sent to the receiver—a real boon for multiple-channel staged events.

The receiver modules are essentially the

Sony claims that there is a complete compatibility across the range of both Freedom and the 800-series, but this does require the removal of the Raycom tweaks—by Sony—at the expense of the audio integrity of existing 800 models. What will present multichannel users do?

same whether you opt for a single or 6-channel diversity receiver with the single receiver having its line-level output adjustable via a knob on the front face through to a balanced ¼-inch jack on the rear of the unit, while on the 6-channel receiver there are 6 individual outputs at 0dBm line-level and a mixed output switchable between Mic level at -38dBm and line level at 0dBm. The receiver units slot into the 6-channel housing allowing a modular build-up of any number between 1 and 6 to be used as needs or finances dictate, with a clever facility for automatic search and setting of available channels. By holding down the SET button on the first position receiver for 3s, the receiver searches for available channels in the selected group and these are then assigned to all tuner units in use. This can then be repeated with a second 6-channel unit, and provided the first 6 channels are powered, the second 6 will be assigned to another 6 frequencies in that group. After this automatic operation it is then possible, should you wish, to manually retune any of the receiver units.

The tuner's LCD panel shows AF level as a moving 'magic eye', RF level as a series of 4 vertical dots with 4 being the optimum antenna reception, Group and Channel selected, and transmitter battery-level shown

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as a 2-segment bar graph in the shape of a battery. There are also LEDs mounted next to the display showing the detection of transmitter battery volts—this flashes with the battery display when volts are low—and the presence of an audio and an RF signal.

The build quality is what we have come to expect from Sony, and in general this is excellent, not withstanding the delicate switches on the belt-pack transmitter and its battery clip, but ultimately this range must be judged on the quality of its audio.

The lineage from its big brothers in the 800-series is easy to see but in shoulder-to-shoulder tests the shortcomings of the Freedom quickly become apparent. In its straight from the box guise, without the optional active antenna system fitted, the usable range of the Freedom falls to about a third of that achieved by an 810-820 transmitter-receiver setup, which is still strong at well over 100m, and there appears to be an 'intelligibility lift' or pre-emphasis in the MF region which gives an overall thinner sound to that of the 800-series. More importantly, when subjected to the HF energy of the 'rattling car keys' test, the compander in the Freedom transmitter decides to exercise its own freedom of choice to go home early and leave an in-tray full of break-up. What is curious about this is that early 800 models of 1994 vintage, at that time almost exclusively used as guitar links, also lamentably suffered from this shortcoming which limited considerably the use to which they could be put. This was cured once and for all by timing and equalisation circuitry designed by radio-mic guru Ray Withers, a modification that meant high-quality speech and vocals could be realised with the system, and hence Sony saw the sales and prestigious use of the 800 radio system rocket. However, for some unknown reason this Raycom circuitry has not been incorporated into the Freedom range and therefore this highlights the dichotomy faced by existing users: Sony claims that there is a complete compatability across the range of both Freedom and the 800-series, but this does require the removal of the Raycom tweaks—by Sony—at the expense of the audio integrity of existing 800 models. What will present multichannel users do? Deliberately down-grade the audio path of their 800-systems for compatability with Freedom, or run the two systems separately? Time will tell, but as mother used to say as she took off her welding goggles—why fix what ain't broke?

So does in fact the Freedom series break new ground in the semi-professional market when applying such criteria as price versus performance, features, ease of use and setup? Well, yes it does, and furthermore at these prices—around a third of the price of the 800-series at about £800 (UK) per channel—I foresee the conference market buying them by the Pantechnicon load. But please, please don't try or expect me to change from my Raycom tweaked 800s. ∎

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Tube Tech MEC1A

The recording, or voice, channel is now an established concept in outboard paraphernalia. **Zenon Schoepe** looks at a variation on the theme from a company that always comes up with something a little different

WELL IT'S BEEN a while but it was inevitable that Tube Tech would turn its attention to the outboard recording channel sector. What it has responded with boasts all the traditional attributes normally associated with the Danish blue brand-quality, sound and quality, and, of course, its valved as well.

Combining a mic-input section with EQ and opto compression in a single channel device satisfies the basic requirements for direct to tape work and it is interesting to note that the box draws from existing models in the company's range. Namely the preamp is from the MP1A, the compressor from the CL1B and the equaliser is based loosely on that found in the more elaborately detailed EQ1A. How the whole lot is patched together is interesting, but it is apparent that fuller implementation of the EQ1A has not been made, nor has the gain reduction control of the newer LCA2B compressor-limiter been employed, presumably because it would have jacked the price up considerably.

You come in through a transformered preamp with 20dB to 70dB of gain on switched coarse and fine controls and encounter switches for phantom, phase reverse, a 20dB pad and a 20Hz–40Hz low-cut filter. This is a slightly different arrangement to the MP1A which provides only one control for the gain setting. Rear-panel connections offer an XLR input and output with two jack sockets accessing the two buses, selected from the front panel, for linking the compressors of multiple units together.

You are then into the 3-band EQ which equates to the high shelving gain band of the EQ1A, a marginally modified frequency selection of its low shelving gain band and a fully parametric mid band that is a sort of amalgam of the frequencies addressed by the EQ1A's three parametric mids.

The whole section is bypassable and employs switch gear to activate boost or cut in all three bands. Given the selection of frequencies and bands offered by the EQ1A-I would say that Tube Tech designer and company founder Jon Peterson has done a pretty good job in distilling these down to just three bands. I think it's important that he's left the low and high ends largely intact as these shelving sections pin down the whole EQ path leaving the mid to roam between the two; although there's a smart degree of overlap if you really are hell bent on unseating those drivers.

I have no reason to believe that the sharpest bandwidth setting in the MEC1A's mid is any more extreme than my recollections of the EQ1A, it is just that the leap between frequencies is greater, and, consequently, the switching effect is more pronounced. The cumulative effect of this section is remarkably pleasing and a good deal more precise and powerful than it would seem on paper. You're not necessarily going to be able to tune out exactly an offending honk, but the available range means you'll get pretty damn close. This is more than countered for by the wider bandwidth settings as broad in Tube Tech parlance is enormous and class-leadingly smooth and delightful.

THE EQ CAN be regarded as performing as two distinct sections—the LF and HF and, then, the fully parametric mid. You want treble 'n' bass, sir? Then I have just the thing for you. This will give you the smiley face curve to end all others with just the beginnings of a smirk in the corners. Phenomenal power is available with the wonderful signature that Tube Tech's manages to massage out of the associated valves. The EQ1A positively glows with good health and the same can

certainly be said here. Of course I would have to point out that this EQ section, while rude with health and vigour, still falls some way short of the EQ1A, which I would consider to be the finest box the company has built to date. Against such a backdrop it is only natural to want more.

The MEC1A compressor is more basic than the aforementioned LCA2B most notably with the absence of the preset constants and the separate limiter, but then it is effectively a CL1B with fixed, or fully variable, manually adjusted time constants. The section can be bypassed independently and has fully variable ATTACK, RELEASE, THRESHOLD and RATIO pots. Its not lightening fast, as you would expect, but when it catches up, then it squeezes firmly yet with consideration, in a manner that is not dissimilar to the way some of those old boxes, that none of us can afford, do their business.

This is almost too good a compressor, more than a mere recording channel deserves. Superb on vocals and mic sources in general, bang up one side of a final mix through the thing and you'll be wondering why the last remaining bastion for mono, aside from TV in certain underdeveloped and so-called developed countries, is the recording channel. They're all the same in this respect and the good ones will always mean that you're probably going to have to buy two.

A 10dB output gain pot is accompanied by gain reduction-output level switching for the vu meter, the obligatory Elvis lookalike signet-ring-style power indicator, and a switch for flipping the order of the EQ and compressor circuits. The last of these is probably one of the smarter little tricks of the MEC1A and it is certainly worth flipping the switch when you are getting close to happy with the sound to see if it can give you something better.

You can do some extremely filthy >>>>>

<<<<< things with the low end by playing around with the order of the processing (a strange word to use in the context of this unit) as it stands to reason that driving EQ into the compressor is going to sound different to processing a nicely compressed sound with EQ.

Mic preamp performance is spot on, not blatantly valvey in character just on the classy side of suggestive. What it also is, is forgiving as it goes pleasantly soft just before you hit the limits of the headroom, meaning that if you are a chancer then you will probably get away with running it a little hotter than you might otherwise dare.

Then, of course, there is the DI that, unusually on a unit of this calibre, is activated by inserting a jack in to the front panel socket rather than by switch selecting it. I have not had cause to use the DI on the MP1A, but the capacity to handle instrument level signals on a recording channel is to be expected. And what a great circuit. Unlike some recording channel boxes with a DI, the MEC1A can take an awfully hot signal without complaining. In fact I'd say it thrives on it and you can push it to a truly acceptable guitar overdrive tone with some sensible decapitation of the top end. It soaks up the level and smoothes it out into sustain with none of that awful fizz you can generate from boxes that do their best work with clean sounds.

I can not pretend that I was not surprised at just how applicable this unit is to instrument level sources. You could track exclusively through the MEC1A with no problems at all. Any suggestion? Well given the level of

EQ Frequencies
Low shelving (±15dB)
20, 30, 50, 70, 100, 160Hz
Mid band frequencies
(±20dB sharp bandwidth/
±12dB broad)
40, 60, 80, 130, 250,
500Hz, 1kHz, 1.6, 2.5, 4.0,
7, 10kHz
High shelving (±15dB)
4.0, 6.0, 8.0, 12, 18, 26kHz
Compressor
Ratio 1.5:1 to 10:1
Attack 1ms to 100ms
Release 70ms to 2.5s

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compression control, and the fact that this box is going to used for, among other things, a fair proportion of straight to tape, and digital tape at that, it would have been a master stroke to have included the altogether wonderful independent limiter from the LCA2B to stand guard over the output. Similarly another mid band would have been even better.

However, you've got to ask where these controls would fit on the front panel. It would have to be bigger and considerably more expensive and that is clearly not to be encouraged. Judged as a whole it performs impeccably. The sound of the constituent parts in isolation is class leading. Put them together and the effect is brilliant. While still not a cheap unit it nevertheless represents a considerable saving over assembling the equivalent from the relevant parts. It's extreme value for money. This really is a killer box.

The good thing about the MEC1A is that it is built from parts of existing proven and popular Tube Tech products. However, it is by no means just a piece of 'best of' outboard as it has its own charm and individuality. Put it this way, if you had a desk full of this stuff you would probably never go out again. Its almost too good for a mono channel and it is not just for rich bedroom-boys as the MEC1A's skills extend into commercial studios and voice-over postproduction.

Highly recommended if you are interested in a characterful high-quality signal path. And I cannot get over the fact that Tube Tech is still one of few brands that make me smile with delight when I take one out of the box. ■

Hafler TRM8

Studio Sound's 'bench test' close-field loudspeaker tests continue with Hafler's TRM8. **Keith Holland** reports

THE HAFLER TRM8 is a 2-way active loudspeaker system, with an in-built amplifier and electronic crossover package. The 200mm polypropylene woofer and 25mm soft dome tweeter are housed in a ported cabinet having external dimensions of 260mm x 392mm x 330mm deep, and an internal volume of 13 litres. The front of the cabinet is shaped such that the tweeter flange is mounted some 20mm behind the woofer and a bass reflex port is located in the rear of the cabinet. The magnet systems are magnetically shielded allowing positioning near to CRT monitors.

The manual, which includes full circuit diagrams and parts lists, recommends that the loudspeakers be listened to with the drivers aligned vertically (portrait).

The electronic crossover is described as being a 24dB/octave Linkwitz-Riley with a crossover frequency of 2.5kHz, and incorporates a 2nd order high-pass filter at 30Hz for subsonic filtering as well as bass and treble shelving controls.

The loudspeaker is finished to a high standard and looks and feels as solid as its 15.88kg net weight suggests.

Discussion of measurements: The on-axis frequency response (Fig.1) lies with ±3dB between 45Hz and 20kHz except for peaks of about 2dB at 700Hz and 4kHz. The response is seen to fall rapidly below 40Hz due to a combination of the port tuning and electronic subsonic filter. Low-frequency harmonic distortion performance is seen to be good, with both 2nd and 3rd harmonics lying below -40dB (1%) relative to the fundamental throughout the entire bandwidth; although the level of 3rd harmonic rises above 100Hz to a maximum of -40dB at 1500Hz. The horizontal off-axis response (Fig.2) is reasonably well controlled except for a broadening of coverage angle for
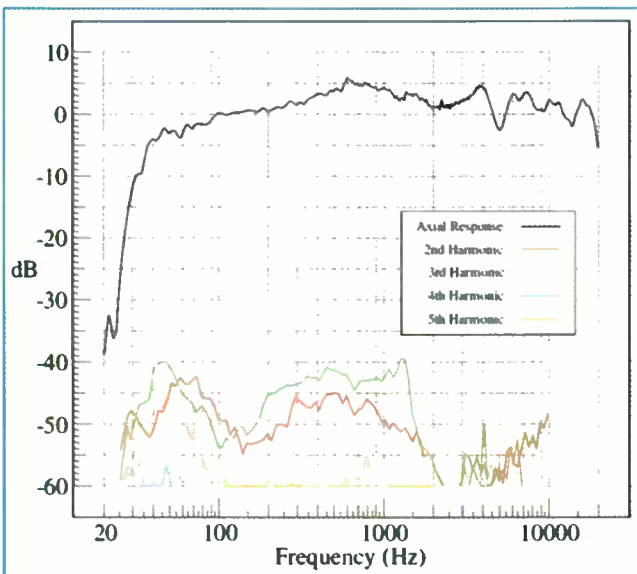
Fig.1: On-axis frequency response and harmonic distortion at 90dB SPL @ 1m
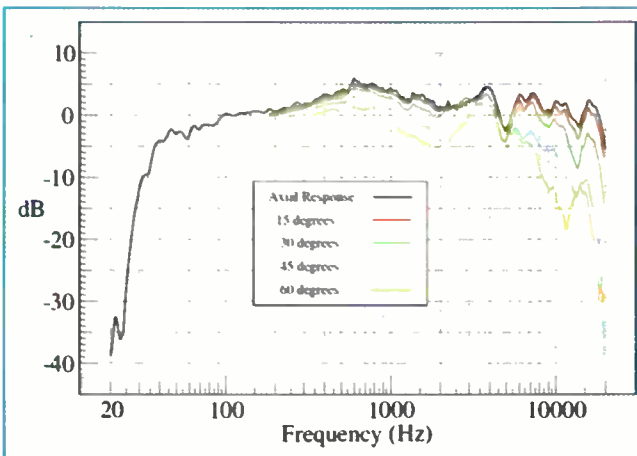
Fig.2: Horizontal off-axis frequency response
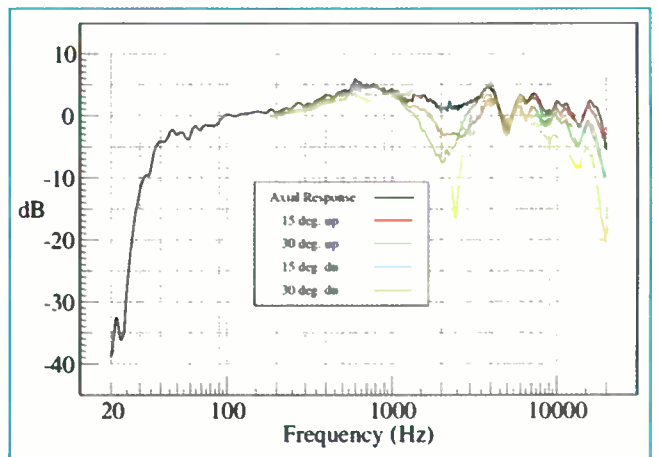
Fig.3: Vertical off-axis frequency response
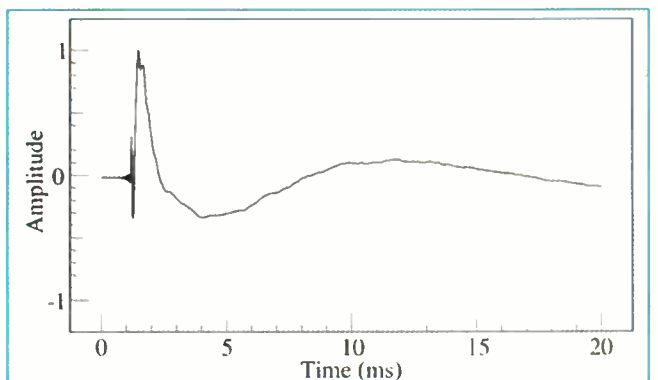
Fig.4: Step response
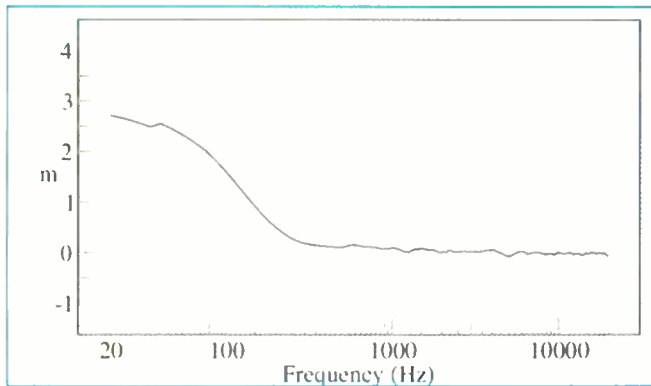
Fig.5: Acoustic centre
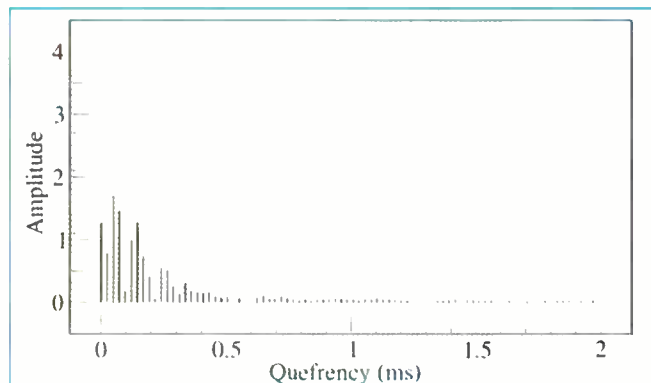
Fig.6: Power cepstrum
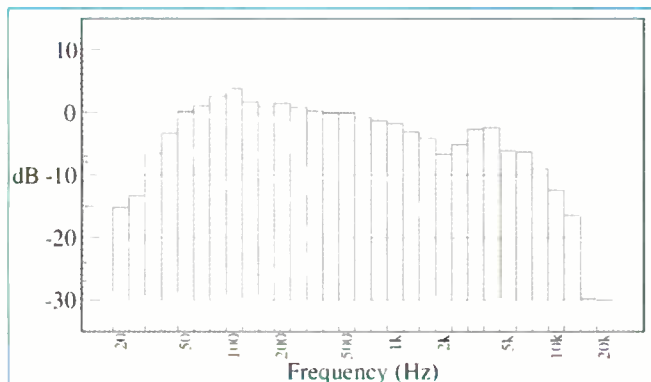
Fig.7: Total power response

about an octave above crossover and some high frequency lobing, evident as peaks and dips in off-axis response between 12kHz to 15kHz. The vertical off-axis response (Fig.3) is dominated by the dip at the cross-over frequency which is characteristic of, and unavoidable in, spaced driver (non-coaxial) loudspeaker systems.

The step response (Fig.4) shows excellent time-alignment and crossover design with a sharp rise and steady decay; although, as confirmed by the acoustic centre result (Fig.5), there is a delay of some 10ms in the low frequency energy, corresponding to a shift in acoustic centre of some 3m behind the loudspeaker at low frequencies. This group delay is probably due to the 6th

order roll-off at low frequencies. The power cepstrum (Fig.6) is well behaved with small spikes evident at about 70ms, 150ms and 260ms. The overall power response (Fig.7) shows a dip at the crossover frequency followed by a peak at slightly higher frequencies due to the broadening of coverage angle shown in the off-axis responses; the lower-mid-range performance is smooth however.

Overall, the Hafler performs well. The design has no obvious flaws and the loudspeaker is particularly well-built. The low-frequency time problem is probably a direct consequence of the adoption of a subsonic filter. The low-frequency harmonic distortion performance is particularly good. ■

www.americanradiohistory.com

Calrec Digital T-series

One year after the unveiling of its first all-digital desk, Calrec has gone digital with its flagship T-series. **Zenon Schoepe** explains

OF COURSE, we always knew it was coming but the revelation of an all-digital, large-scale console from Calrec at NAB was nevertheless something of a surprise, not least because of the form that it has taken.

The writing was on the wall with the unveiling of the company's first digital desk, the radio-orientated X-series previewed at last year's European Munich AES Convention, and the intention was clearly to use this as the test bed for the technology and then to up-scale it for an altogether larger beast. That it should manifest itself as an all-digital version of the established digitally controlled analogue flagship T-series board proves that the company has been looking long and hard at the issue for sometime. Incidentally, it also makes Calrec the first to achieve the all-digital conversion that so many manufacturers steeped in

DCA technology have always been threatening to do.

The Digital T, as it is being referred to, employs the T-series control system. T-series operating system but runs digits instead of analogue circuitry. The system can extend to 96 stereo channel or 192 mono or a mixture thereof, which is as big as any analogue T-series has ever gone.

The design of the T-series started around six years ago and it emerged, in the first instance, as a digitally controlled analogue desk predominantly due to matters of cost. Calrec maintains that it was not that the digital processing was inaccessible but that building in hot-plugability with the requisite amount of redundancy would have made such an all-digital desk prohibitively expensive yet these were requirements that the company wanted to meet. Its digital console had to have the same level of on-air reliability as any of its analogue desks.

It embarked on creating a console that consisted of three parts: the control system based on fast parallel processing, a memory system based on a PC and an audio processing system that was originally analogue but which can be replaced by digital. All parts were designed to be independent of each other and can thus be changed or

updated. For example, the memory system is likely to go to NT next year in response to user requests.

Within this scenario and on the Digital T you can disconnect the control surface and audio continues to pass through the desk and similarly a DSP card can be physically removed and a hot spare kicks in within 4 seconds. The new console will still reset totally within two frames and will boot up from cold within 15 seconds.

As already implied, the X-series was crucial to the development of the company's digital technology as it was deemed better to get it working on a smaller scale while at the same time identifying the unmistakable trend in the radio market towards small uncomplicated digital radio boards. A scalable version of the X-series hardware is employed in the new Digital T-series and the X-series starts shipping this month with the first desk being installed at BBC Radio Nottingham in the UK.

The control surface is likely to remain unchanged for the short term but is likely to be modified in the future as potential users grasp the potential of the digits. Such a remake should also give Calrec a good opportunity to reassess the worksurface components, many of which can now be substituted with cheaper controls than those that were available when the surface was originally conceived. The potential of the all digital processing would also strongly suggest applications in postproduction and should this appeal take off then the control surface would certainly need to be re-evaluated to harness the in-built total dynamic automation capabilities that are currently not supported by the existing control surface.

Internally, the Digital T-series uses 40-bit floating point, 24-bit available on the mic inputs and the potential to run at a higher sampling frequencies than 48kHz should it be required.

The Digital T will be available for shipping in January 1999 and Calrec is confident of this timescale because it states that so much of the console has already been well proven. The objective is to weigh in within 15% of the price of an analogue T-series depending on configuration.

The Digital T is undoubtedly an important step for the small UK company but one that still adheres to its trademark of sensible and waste-not engineering. It is a credit to Calrec that its new flagship digital desk looks, for all intents and purposes, very much like an analogue console. After all, is not that what everybody is trying to achieve? ∎

NEW TECHNOLOGIES

Cheap digi desk

Tascam unveiled an entry level digital console at NAB, called the TMD1000, which is expected to ship in July for around $1199 US. The 16-channel mixer has four mix inputs with phantom, 8 channels of TDIF and two AES-EBU and SPDIF outputs. It can be expanded through two optional cards which are expected to ask in the region of $150 US. The IFTD1000 adds 8 channels of TDIF and 4 channels of AES-EBU or SPDIF

while the FX1000 effects boards doubles the on-board processing of four channels of dynamics and a stereo pair of effects. The desk boasts 128 snapshots and MMC, but no moving faders.

The news coincided with the arrival of the MMP16 derivative of the MMR8 dubber which offers 16-channel of playback, 20-bit D–A and 24-bit digital output capability and can load and play Pro Tools session files, Waveframe projects and OMF compositions directly.

Version 1.25 software for the MMR8 and MMP16 includes OMF export, 9-pin, PC-AT keyboard connection with the MMRC remote and Pro Tools 3.2 and higher session files compatibility. For DTRS machines, the RC898 remote control incorporates all the functions of its predecessor but adds individual input monitor switches for each of the six machines, confidence mode access on the DA-98, dedicated function keys for 20 macros, a large LCD, illuminated transport keys and the ability to access track copy functions on the DA-98 and DA-38. Tascam also introduced the 4-machine RC828 remote.

Tascam, US. Tel: +1 213 726 0303.

AudioFile 98 and networking

AudioFile 98 is based on AMS Neve's 24-bit platform with networking capability and Version 3.0 software. Features include 24-bit recording with 16-bit and 24-bit editable together, waveform display and >>>>>

Frontier Design Zulu

Taking its lead from the ADAT, Frontier's Zulu convertor is
raising computer audio standards, writes **Dave Foister**

ONE OF THE SMARTEST THINGS Alesis did when designing the ADAT system was putting all eight digital signals on a fibre-optic link. Not only did this give the most cost-effective and space-saving multichannel digital interface in the field, but it invited others to exploit those very advantages, giving the format an even wider appeal than it might otherwise have had. The usefulness of the interface in such a space-intensive application as a computer sound card is obvious, and, indeed, several have adopted the idea.

One such card is the WaveCenter from Frontier Design, that has ten digital inputs and outputs, two on SPDIF, and the rest on ADAT optical connections. For those with access to ADAT tape machines this is all that is needed, but in the absence of the recorders the problem remains of how to get analogue signals in and out of the computer. The expense of suitable convertors would once have negated the benefits of the card, but the price keeps falling; keen to maximise the appeal of their own system, Frontier has produced Zulu, a simple set of 20-bit convertors dedicated to the ADAT digital format.

Zulu is very much a real-world practical box, making sure to offer only what is needed in order to keep costs down. The first sign of this is the fact that; although it has the full complement of eight analogue outputs, it only has four analogue inputs. This assumes, probably fairly, that few of the kind of jobs it will be doing require more than four at once. The four channels are supposed to be duplicated across the digital outputs, appearing on 5–8 as well as 1–4; although this did not appear to be the case on the review sample; even so, most destination systems, including tape machines, should be capable of routing the signals as required themselves.

Four tricolour LEDs show analogue input levels, changing from green to yellow as level increases and turning red at digital maximum. There is no level control on the box, and the inputs are designed for unbalanced -10dBV signals, which means that +4dBV sources must be used with great care—there is not a lot of headroom here. The translation between incoming levels and digital indication on the destination machine seems to behave as expected, with 0vu giving -15dBFS on an ADAT meter, and Zulu's red lights coming on exactly as the ADAT

maximum bar lights up.

The eight analogue outputs are also unbalanced -10dBV feeds, appearing like the inputs on ¼-inch jacks. Again, no level control is necessary, neither is there any metering—any damage will have been done by now. The small size of the box means that the back panel is a forest of ¼-inch jacks for the 12 analogue signals, the two optical connectors hiding in the corner next to the power connector. This is fed from a wall-wart, and the one supplied to me was a European 2-pin version necessitating a little dodgy brutality with a rubber 13A boot to make it work. I trust Zulus sold in the UK will have the correct power supplies. I was also surprised that no optical cables were included.

The other main cost saving comes from the fact that Zulu's convertors do not have their own sample-rate clock on board. Safe in the knowledge that almost everything the box is likely to be connected to will have a clock, Frontier has decided to eliminate the duplication and use the external clock at all times. For this reason it has to have a digital input connected even when it is being used purely as an A–C, so two optical cables are required. A green LED shows the status of the incoming digital signal, going off in the absence of anything, glowing steadily when receiving ADAT format signals, and flashing to show the presence of an inappropriate format, Toslink SPDIF uses the same connectors but means nothing to ADAT.

This arrangement has the advantage that it avoids the dangers of clocks chasing each other that used to plague some systems. Its only slight drawback is that it cannot easily be used with the original ADAT machines, because if they are set to digital input they can only slave to incoming clock, not act as a master. If the possible extra resolution of the Zulu convertors is needed another external clock has to be used.

Straight comparison of Zulu with the convertors in an XT20 showed them to be very similar in terms of sound, giving them the potential to bring very good quality indeed to a computer card and maintain 20-bit compatibility with the new ADAT if required. One difference worth noting is that the processing delays are significantly different from those in the machine itself, which could lead to cancellation if a careless mish-mash of sources is used.

Zulu does its job very well, with quality and simplicity its priorities. Getting decent audio in and out of a computer without spending a fortune seems to be a common quest at the moment, and this little box is just what the doctor ordered. ∎

<<<<< threshold recording. Strip silence can be used to tidy up an EDL while multi-levels of Undo and Redo are now available. The machine supports 4.3Gb and 9.1Gb drives, OMF2.0 import and a MADI I-O option. StarNet networking provides multiple AudioFiles with access to multiple drives connected via fibre-optic cables. The network is a star configuration running 400Mb/s optical connections. When an AudioFile is connected to StarNet the direct connection between the worksurface and its storage devices is replaced by connection to any storage device on the network with all disks housed in removable chassis.

A system-wide directory of network disks can be viewed at any time and the transfer of individual sound files is available to all StarNet users while the off-line backup and restoration of projects is also facilitated by the Media Toolbox that manages this and the import and export of files from other systems, such as OMF2.0 and .WAV files. Media Toolbox is available as a stand-alone device for use with a single machine or as a networked device.
AMS Neve, UK. Tel: +44 1282 457011.

Waveframe 408 Plus

Waveframe's 408 Plus workstation plays back 8 tracks from a single SCSI bus which is identical to Tascam's MMR8 dubber. Version 6.2 software adds OMF2.01 file compatibility, support for optional multichannel digital I-O and import-export filters for .WAV, AIF and SDII formats. Other features include 64-bit graphics, Sony P2 and ES bus machine control, ISA and PCI expansion slots and integrated Ultra SCSI controller for OMF and multimedia applications. The 408 Plus shares the same underlying architecture as its predecessors the 401, DCS and DAW80 with prices starting at $10,995 US for an 8-track, 8 analogue I-O configuration.
Advanced Systems Group, US.
Tel: +1 510 654 8300.

Cream Scope

Creamware has announced the Scope (Scalable Object Processing Environment) modular DSP platform for Mac and PC, which combines the technologies of sampling, synthesis, effects and mixing. Based on a multi-DSP PCI board that can use 12 SHARC processors, it includes a large library of components from DSP modules to complete devices with their graphical >>>>>

LA Audio SPX2

Finding yourself short of desk inputs is as inevitable as it is inconvenient. **Jim Betteridge** may have the answer

CAN A DESK ever have enough inputs? Of course it can not. Whatever configuration makes sense at the start of your installation, you always end up making some compromise or other, having less-used sources available on the patchbay rather than ready to go on a fader, it is no big problem, but I think we would all agree the less time spent crouching by your client's lap fiddling with patch leads, the better.

Some of the main contenders for relegation to the patchbay are the stereo sources: the cassette, the ¼-inch machine, the CD player, the VHS, possibly the Beta and even the DAT machines. Except for good old-fashioned radio productions involving the live spinning-in of unnumbered carts and ¼-inch machines, it is seldom that more than one of these sources is required at a time. With this in mind, a painless and inexpensive compromise between the profligate assignment of precious inputs and the inconvenience of the patchbay is the SPX2 Stereo Source Selector from LA Audio.

This 1U-high rackmount box takes six stereo line sources, provides screwdriver-operated trim pots wide enough to accommodate -10dB and +4dB standards and then, via 6 front-panel buttons, allows any one or more to be selected and sent, via its mix bus, to its stereo outputs. You can select any number of sources at once and they are mixed together according to their trim settings. To ensure there are no clicks and thumps on the monitor speakers, the SPX2 uses silent CMOS switching on the input select, and soft, relay controlled switching on the main outputs.

Each button has a status LED to show the currently selected source(s). There are 3 stereo outputs on the rear panel, all attached to the same stereo bus. The two main outputs are balanced on TRS ¼-inch jacks. Each has a front panel on-off button with status LED plus a TRIM knob. The third, called RECORD OUT, has no level control, is unbalanced on phonos and remains unaltered by any of the front-panel controls except for the input trims. In addition, there's a stereo headphone output on the front panel with its own level control. Also on the front panel, a single BALANCE knob plus a DIM and a MONO button relate equally to the two main outputs and the headphone output.

Effort has been made to offer a variety of connection possibilities: input one is on balanced XLRs, Input 2 and Input 3 are on balanced jacks while Inputs 4-6 are unbalanced on phonos. Input one has a parallel pair of break jacks on the front panel to accept the occasional visiting stereo source and associated with it is a STEREO TRIM knob that replaces the screwdriver trim provided on Inputs 2-6.

It seems to me that all balanced inputs, even if on TRS jacks, would have been far better. A lot of potential users will have almost exclusively balanced analogue sources and will be very reluctant to unbalance them here. I put the point to LA's head of R&D, Paul Bird, who said it would not actually be that much more expensive and he'd definitely consider it for v2.

So I strapped the SPX2 into the rack and connected my Beta SP and the two DAT machines to balanced inputs and the VHS machine, cassette deck and CD player to unbalanced inputs, being careful to keep cable runs to a few feet. I then set a stereo channel on the 02R to expect +4dB, set the trim knob on the SPX2's Output 1 trim to mark 7 (actually 3 O'clock) and adjusted its input trims to achieve a good nominal level. No Problems. All as expected.

One of the limitations of the stereo inputs on the 02R is that they have no balance control. It's sometimes necessary to have individual control over the left and right channels (when conforming audio for video when you may only want one of a split pair). The 02R has pan controls for the left and right channels but no balance or individual faders. For this reason it is sometimes necessary to take up a pair of precious mono inputs. The provision by the SPX2 of a balance control, therefore, is a boon.

I then connected Output 2 of the SPX to the analogue 2-track return on the 02R. This allows the monitoring of a stereo source, whose inputs are also normalled to the stereo output of the desk, without fear of a feedback loop. VHS machines are particularly nasty in this regard as they switch to line-in (E-E) on stop. Very handy.

The components used in the SPX2 are the same as those used in many other professional analogue equipment and its sonic performance is certainly good quoting around 90dB S-N with 0.015% THD and noise. It sounded okay to me.

Once in the rack I became irreversibly attached to SPX2 and at this price it was an easy decision to whack a cheque in the post and throw away the boxes. My main criticism is that all the inputs aren't balanced; and if we're looking at the possibility of a slightly more pro version for the future. 6 dual-coloured LEDs to show signal present and clipping for each of the inputs would be very useful. It's still a very handy little box as it stands, though. ∎

NEW TECHNOLOGIES

<<<<< user-surface and presets. Scope will be able to operate several devices at the same time to emulate a complex studio set up.

The system will be open to third-party developers and integrates into the Triple-DAT family of workstations. Many of the Pentium-native DSP effect modules of TripleDAT will be able to be used within Scope and all Creamware's present hardware can be integrated into the system to increase the number of physical I-Os.
Creamware, Germany.
Tel: +49 2241 59580.

Digital valve mic

Each side of the CAD VX2 valve mic capsule has its own independent valve head amp and output amplifier including separate custom output transformers. This, says the company, allows the head amp valve to be optimised for low noise while the output valve is optimised for driving the transformers and cables. Polar pattern switching is performed at the output of the mic rather than at the capsule. Cardioid, figure-of-eight and omnidirectional patterns are available. Noise floor is claimed to be lower than many FET designs and the valve circuit has been optimised for flat phase response and has a claimed bandwidth that extends beyond 100kHz. The capsule is 1¼-inches in diameter.

A 24-bit, 32kHz to 96kHz sampling rate output module with a claimed 120dB dynamic range is also available
CAD, US. Tel: +1 440 593 1111.

Marantz CD-R

Marantz has what it describes as an entry level priced CD-R machine in the CDR630 which can play CD, CD-R and CD-RW discs and record CD-R and CD-RW. Features include balanced analogue inputs, coax and

optical digital inputs, automatic indexing, sampling-rate convertor bypass, CD sync for auto start recording from digital sources, rackmounting casing, and remote control. Shipping is expected in June for around £750 (UK).
Marantz, The Netherlands.
Tel: +31 40 273 7978.

Digi radio desk

Klotz Digital has targeted the radio market with the Paradigm digital console, it has sample rate convertors on every input channel, all mic inputs with voice processing, an AES reference clock for master synchronisation and an integrated timer, cue speaker and clock. It boasts 24 stereo input sources, 8 input channel faders, and 4 output buses in a compact and attractive package. The announcer's mic feeds the talkback bus, there are 16 GPI and >>>>>

SD Systems LDM94; LCM89

Viewed with suspicion by purist players, brass bugs are still the best solutions to many problems, **Dave Foister** comes over all saxy

SOME MICROPHONES are born specialists, others have specialisms thrust upon them. Many of our favourite one-job microphones were intended as all-rounders, but the discovery of a particular characteristic linked them irrevocably with a specific application. Less common are the microphones designed from the ground up to perform best in a one-niche application. One give-away is a fancy mount for a particular instrument, so no prizes for guessing which pigeon-hole these two models from SD Systems belong in.

Sax players are a breed apart. On stage they can strut their stuff almost as much as guitarists, with an instrument almost as phallic. They can do this because they clip little microphones on to their bells, freeing them from the constraints of standing still in front of a microphone stand. Trouble is the little microphones usually aren't up to studio use, yet the players resent having to stand rooted to the spot in front of something that might actually record their sound properly. It's the same with double bass players; the bug that's good enough night after night on stage sounds crap in the studio, yet the difficulties of miking them up don't interest them.

Several solutions exist for the saxophone, usually in the form of a grown-up clip-on microphone like the DPA miniatures on their little goosenecks. While many remain all-rounders with a mount that happens to work with saxes, these SD Systems microphones commit themselves 100% to the sax with big wire spiders that clip over the bell and could hardly fit anything else. There are two models, dynamic and condenser, distinguished largely by their colour and the presence of a powering box on the end of the condenser's lead.

The mount is common to both and resembles a small tripod, with a soft plastic covering round the ends of the legs. These are gently curved to clamp snugly on to the bell of either a tenor or an alto (soprano's too small and baritone too big) without scratching the lacquer. The legs are surprisingly long, and place the microphone capsule itself considerably further out of the mouth of the bell than some such devices, which can only be a good thing in my opinion. Regardless of the quality of the microphones themselves, many of these arrangements fall down because they are simply too close, putting spill rejection and inconspicuous mounting before sound quality. I've always wondered why they bother, as

most saxes are loud enough that spill isn't a problem and most players seem only too happy to have so much stuff strapped to their instruments that they look like a weapon out of *Star Wars*.

The mount, then, holds the microphone firmly in a fairly suitable place, and has at its top a rubber suspension (spares provided) that supports the microphone element itself. The simpler of the two is the LDM94 dynamic, a remarkably small black capsule connected by a thin wire direct to an XLR. Properly balanced, it can handle up to 155dB SPL (just as well really) with a reasonably respectable frequency response. For me this one remains more suited to the stage, as it puts brightness before depth to give quite a pretailored sound, with plenty of edge to cut through. It also seemed susceptible to LF bumping effects which could have been either the mechanical shocks of the keys being moved or a bit of wind from the bell itself—perhaps less likely as the capsule has a foam shield to prevent this.

More interesting is the LCM89 condenser. This is coloured brassy gold to match the instrument (tough if you've got one of the flash black lacquered ones) and its lead ends in a small plastic box with a belt clip. This can take a 9V battery (with a claimed life of 1-2 years) but also accepts 48V phantom power via its output XLR. It even has two controls on it, allowing the player to adjust both volume and tone (funny but I always thought they did that with their mouths). There is also a MUTE switch, which is either a boon for the player or a pain for the engineer according to your point of view.

The big plus point is that its sound is altogether cleaner and smoother and more appropriate for use in the studio. It's still on the bright side compared with a more conventional studio microphone in a more conventional position (a 414 in this comparison) but much more complete than its dynamic stablemate at the lower end. The overall curve gives a full spectrum with a useful presence lift, and this can be smoothed out with the TONE control on the player's box if desired.

Sometimes a player comes into the studio saying 'it is all right mate, you can take a feed off this' and your heart sinks at the prospect of the mix of diplomacy and EQ it is going to take to make it work. If it is a sax player with an LCM89 you can relax—it will work and it will carry on working when he starts swinging that bell around. ∎
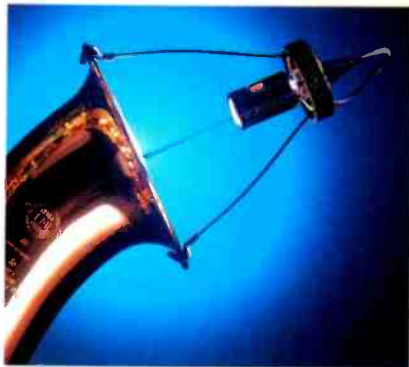
<<<<< GPOs, an intercom port with automated input sensor, console set ups, LCD channel labelling and a mix-minus system for phone lines.

Also, the desk is expandable to 48 input sources and 16 faders and can add a profanity delay and redundant power supply. All this for around $6,000 US, in August.
Klotz Digital, Germany.
Tel: +49 89 4623380.

tc electronic 02R card

tc electronic has released details of a plug-in effects card for the Yamaha 02R mixer called the Unity. Offering M2000 performance, the card plugs into one of the four 02R I-O expansion slots and can be ordered with or without an 8-channel AES-EBU I-O connector. The card is loaded with M2000 effects and two effects can be run at the same time with all settings remembered and automated in the 02R's automation.

The two effects can be run from the desk's aux, group or main buses and external signals can be mixed with the Unity returns on the card.
tc electronic, Denmark.
Tel: +45 8621 7599.

GPS DTV console

Graham-Patten used NAB to air its plans for a DTV-capable digital audio mixer.

Hardware platform for the scalable processing is a 333MHz Pentium II using Compact PCI architecture to provide dual plug-in power supplies and hot-swappable modules. The DSP core is modular and can be configured with up to 144 SHARC processors with a full-blown desk able to cope with 512 channels.

I-O for the system is modular and available as a combination of analogue and digital and connects to the engine using IEEE-1394 networking technology. Mixing engine control in large systems can be through dual 100-base T network connections while smaller systems will use RS422 serial control ports.

GPS has introduced Comparisonics sound matching software that can be used to catalogue and search sound effects libraries to find the occurrence of specific >>>>>

Soundcraft B400

The broadcast quarter's use of live desks prompted the design of the B800. **Paul Shure** meets its smaller sister

A COUPLE OF YEARS AGO, Soundcraft introduced a broadcast mixer in response to the observation that many broadcasters were ordering modified live sound consoles for broadcast production use. The reasoning was that live consoles provided the large number of aux sends broadcasters needed to create clean feeds but lacked the more specific broadcast features. Consoles providing this feature set may be available but generally not at a pricing level that fitted average budgets. The B800 is intended to address these requirements, hit the required pricing range yet still remain flexible in specification.

The B400 is the result of some potential customers of the 800 considering it over-specified for radio work. It draws on the structure of the B800 but is far more radio production-orientated and can perform as an on-air mixer if needed. If you directly compare the two consoles, you find a common approach but a different feature set—both share the concept of providing a range of frames from 24 to 56 modules within which can be specified a choice of modules. In the case of the B400

these are chosen from Mono, Stereo and Stereo Telco for the inputs; the 8 groups can be addressed with Mono or Stereo group modules; leaving the essential 3 modules —Stereo Master, Comms and Monitor. In total this creates a bus structure with 8 audio groups, master stereo output, separate mono output, 3 mono auxs and 1 stereo.

A second layer of user specification is added with one of the most comprehensive internal jumper-link option-sets you'll find. As an example, the mono input module has at least 15 options covering EQ functions and signal chain positioning of key features, plus logic options that mainly concern the operation of the fader cueing. This flexibility continues throughout mixer and makes detailed explanation redundant as, if you don't like what you see, there is a strong likelihood that it is a jumper option. To this must also be added basic transformer I-O and fader type choices.

The major differences between the 400 and 800 are to be found in the input modules. It

would be wrong to give the impression that the 400 modules are slimmed-down versions of the 800—they have been designed with differing priorities, and a quite different appearance—the order of the EQ and aux sections is reversed.

The Mono Input module has both mic and line inputs with the expected ancillary controls, routing to the 8 groups in LR pairs and stereo bus. 4 aux sends (1 stereo) which are individually switchable pre-post, and a 3-band EQ with sweepable HF and MF, but fixed LF plus a variable high-pass filter. The module also has a direct output whose level can be controlled by reallocating Aux 1's LEVEL control.

The ON-CUE buttons are particularly 'radio orientated' in their functions with the ON button going to half illumination when the fader is fully lowered. The Cue modes differ according to how quickly the button is pushed and the internal jumpers; although it would be most common for it to route to the Cue buses. It can also be used, in the Line mode, to set up triggers for remote start-stop of external sources. An LED bar graph running parallel to the fader can be jumper selected to read different points in the channel.

The Stereo input is similar, but with no mic inputs and can select from both or either sides of the L&R inputs. The EQ is simplified as reflects the likely type of line sources in radio—CD, tape, hard disk and so on. The Stereo Telco module differs mainly in its provision of a telephone input from an external hybrid, a dedicated direct output and a TALK-BACK button that takes the desk talkback to that direct output.

The Group modules add stereo and tape returns and 4 aux sends. The stereo group adds the ability to control the stereo image width from mono to wide stereo.

The Stereo Master module brings aux masters, main stereo bus with optional limiting, and the separate mono output, to hand. The Comms and Monitor modules add the monitor functions and select—both external and from 6 internal sources, phones, cue, talkback and oscillator functions.

Meter bridges come in two styles—a DIN standard type that accepts any DIN dimensioned meters, and a more basic vu bridge with lower sightlines over the console. The talkback mic and small speaker are also housed in the meter bridge as are LEDs to warn the user that any of the numerous internal VCAs have been selected for external control.

The B400 has pulled off the same trick as the successful B800—it is compact, easy to use and remains flexible at the control-surface level, plus has the ability to reset internal jumpers on a per-channel basis. If this is added to the ability to mix and match modules when specifying, you can end up with a 'custom' console at off-the-shelf prices. ∎

<<<<< sounds in soundtracks. The software contains proprietary algorithms which reduce complex sounds to compact data signatures and databases can be searched rapidly to find a sound using a 'prototype' of the desired sound.
GPS, US. Tel: +1 530 273 8412.

8-channel convertors

Benchmark has an 8-channel convertor system made up of the AD2008 8-channel A-D, DAC2008 8-channel D-A plus the ADA2008 4-channel A-D/D-A combo unit. All have 44.1 and 48kHz capability plus varispeed with prices starting at $3950 US.

Claimed specifications include THD+N of -108dBFS (0.0008%) and CCIF intermodulation distortion that is -128dBFS (0.00004%).
Benchmark, US. Tel: +1 315 437 6300.

Smaller Logitek digital

Logitek now has a 5-fader version of the ROC10 digital radio mixer called the ROC5 that runs with the company's Audio Engine processor.

ROC5 prices for a 16-channel configuration start at $8,000 US and the system can be scaled up to 64 inputs. Target markets are newsrooms, newsbooths, dubbing studios and automated stations.
Logitek, US. Tel: +1 713 782 4592.

TV80 desk

The TV80 broadcast console draws on the TV1000 desk, but is presented in a more economical package with all electronic switching. Available in a range of frame sizes it includes computer recall of preselected audio sources, alphanumeric source displays, a house router interface, an eight mix-minus system and master confidence IFB panel. Talkback is available to submasters, auxes, and mix-minus while standard configuration includes eight mono submaster and two stereo master outputs, eight user-configurable aux sends, four mute groups and direct outputs from every input.
Wheatstone, US. Tel: +1 919 638 7000.

DAB processor

Orban's Optimod-DAB 6200 is designed to meet the demands of digital audio radio,
Two-channel TV broadcasting and webcasting. Features include Protection >>>>>

TL Audio Ivory C-5021

Extending TL Audio's range of valve outboard range is the
Ivory compressor. **Dave Foister** tracks its progress

THE IVORY TRADE is flourishing, so to speak. The Indigos have gone, replaced in TL Audio's colourful range by a new and striking series of processors, again valve based, and again pushing the boundaries of what can be done at a price. The expected TL line-up of preamps, EQ and compressors is included, and here we have the C-5021 dual-channel compressor.

Each channel of the unit includes a multi-input preamp, a simply featured compressor and a basic gate, laid out with TL's usual logic and clarity. An obvious omission compared with previous models is a microphone input, that seems to have been a standard component of every TL processor for years, but is the sensible thing to leave out if the design is to be rationalised. Retained, however, is the

front-panel jack for a high-impedance instrument input, that according to the block diagram appears to be mixed with the rear-panel inputs rather than overriding them. This block diagram is a regular and very useful feature in TL manuals, showing clearly how the signal path is laid out and where, in this case, the valves are doing their jobs.

The 5021 has two valve stages along the way, with the rest of the gain blocks being solid state in line with TL's familiar hybrid philosophy. The first is the preamp stage following the combining of the inputs, and this comes after the input gain control and the rear-panel switches for level matching. These last introduce a feature I have never seen before; while the unbalanced jack input can be matched to -10dBm or +4dBm, the balanced XLR is switched between +4dBm and +18dBm. This is a very sensible reaction to the high levels often produced by DAT and other digital recorders, bringing the operating parameters of the compressor and other controls much more in line with familiar analogue setups.

The valve preamp feeds straight into the valve compressor, in parallel with the DRIVE LED which glows brighter as the level climbs higher above nominal 0dB. Compressor functions are kept as simple as ever, with fully variable THRESHOLD and RATIO controls but push button FAST-SLOW switches for ATTACK and RELEASE. As before, all is not quite that simple as there is a certain amount of programme dependence in the release time, which also interacts with the attack setting, resulting in a surprisingly flexible and tolerant range of

possibilities. The ratio can reach 30:1 for near-limiting, with the characteristic TL soft knee, and a small round illuminated vu meter can be switched to show output level or gain reduction. There is a wide-ranging gain make-up control, a side-chain insert on a TRS jack, a bypass switch, and the two compressor channels can be linked for stereo use. Unusually, this only buses together the control voltages in order to prevent the image shifting; the two sets of controls both remain operative, and must be set up the same for proper stereo working.

The compressor is followed by the simplest gate you'll ever see, with a single control for its Threshold. Although its label suggests expansion as well, that is stretching it a bit; basically it turns the signal on and off with fixed attack and release times and a light to tell you it is doing it. For corrective cleaning up it is fine but don't expect too much more—it is a little bonus rather than a major selling point.

In a way this is uncharacteristic, because with these boxes what you get is generally more than what you see. This is certainly as true of this compressor as any of TL's others, leaving one wondering, yet again, how they can do it for the money. There is something about the signal paths in these TL Audio devices even before you start using the processes that lifts them out of the league suggested by the price.

An illuminating moment with the 5021 came when I plugged it up to a vocal track, and before I'd even brought the threshold down to make it do something, one of the band asked 'what have you just done to the vocal? It sounds great.' For whatever reason, and whatever you think about valves, the voice sounded bigger and more lifelike, and I was then able to add compression to as gentle or as brutal a degree as I wanted. The apparently limited set of controls are no handicap at all.

The 5021 is just as happy on overall stereo programme compression, and here I found its high-level matching useful. The fact that DAT will produce +22dBm at digital full scale can be a worry with some equipment—I have a mixer whose tape returns are overloaded by such a signal—and it is good to see this acknowledged by a circuit that can avoid being driven perilously close to its limits. TL points out that the matching does nothing more than manage the internal headroom—setting the outputs to +18dBm does not increase the unit's dynamic range.

Once again TL has produced a compressor that, despite its price, could find itself among the first choices in a rack rather than the last resort workhorse. Recommended. ∎

The Leading Edge in Digital: now with Surround

With the D950B Digital Mixing System, Studer has introduced a product that sets new frontiers in the realm of digital audio. The D950 uses state-of-art technology and highly flexible DSP power balancing to satisfy the needs of the audio professional. The console can easily be reconfigured to match the specific needs of various applications.

And now, **the new revolutionary D950S Surround Version** is available, comfortably supporting all Surround monitoring formats and featuring the unique Virtual Surround Panning™ (VSP) software. The D950S easily takes care of all the aspects of Surround production and postproduction in a modular and advanced fashion! ■

AES Amsterdam, Booth No. F34

STUDER 50 ears
professional audio equipment

H A Harman International Company

Purple Audio MC76

When Urei's limiters became obsolete, their usefulness and value were assured. **George Shilling** weighs up their successor

I HAVE YET TO MEET a recording engineer who dislikes or actively avoids using Urei 1176 limiters. Every commercial music recording studio worth its salt usually has two or more of these mono units in the rack as standard. Useful on vocals, drums, guitars, basses, wind instruments—in fact, almost anything—there is something pleasing to most ears about the way they control dynamics. The meter shows gain reduction clearly and its needle ballistics are almost as visually pleasing as the audible effects. They possess wonderfully 'warming' properties with not a single valve in sight—instead, these units make use of FET technology.

Production ceased following the takeover of the company by JBL (Parent company: Harman US) who, no doubt, wished to concentrate on the much larger sales of their other lines. Prices of secondhand units have crept up recently, which has undoubtedly led to Andrew Roberts and John Klett stepping into the breach to develop the Purple replica. Their MC76 (Mono

Compressor or Roman numerals for 1100?) model has a front panel virtually identical to the Urei model, except, obviously, the colour.

Although the manual studiously avoids direct reference to the original, even the Purple logo borrows heavily from the Urei one. Urei models were black or silver, the early Universal Audio silver-blue 1176 making way to the black 1176LN (low-noise) and later the silver 1176LN. There were steady upgrades of the model during its life designated 'A' to 'F'; the MC76 most closely resembles an 'E'. The MC76's knobs look the same but the pots are a little out of whack—the OUTPUT pot stops 3dB before the legending reaches zero; although it starts in the right place. The RELEASE knob pointer is also misaligned, and the ATTACK knob has a strange feel to it, like a rheostat or a pot with lots of very small steps. It retains the click-off bypass operation of the 1176. The Purple features the same sets of push buttons as the 1176: ratios of 4:1, 8:1, 12:1 and 20:1, and (power) Off, +8dBM level, +4 dBM level or GR (Gain Reduction) metering and a similarly illuminated vu meter.

Most 1176s have rear-panel connections on solder tags-screw clamps. Purple sensibly uses XLR connectors instead. If you want to link two 1176s you need the 1176SA (stereo adaptor) to make use of the phono socket on the back of

most units. The MC76 vastly improves on this situation with two jack sockets for stereo link operation, one marked DIRECT and the other OFFSET with a locknut trim-pot for offset adjustment and a phase toggle, all neatly grouped on the back panel. Obviously I was unable to assess stereo operation with only one unit, but such use is apparently facilitated by an internal 1.5V AA battery that I am told will give way to a tap from the mains power transformer on units made from June. This case-mounted toroidal transformer is the most obvious internal difference from the 1176.

It is also worth noting that the main circuit board is far more rigidly located than the Urei: Purple has worked hard on the build quality and physical durability of these units. Another sensible move is the provision of an IEC mains socket with fuseholder and voltage selector. I only received the manual by fax at a late stage, which despite being an early draft appears well written. It promises purchasers spare fuses, mains leads and mounting bolts; although mine lacked these.

My initial impressions of the unit compared to a late-model, silver-faced, 1176LN were that the MC76 was a tad quicker in its characteristics. However, some of this may have been more in the bouncy meter ballistics than the sound. After a quick comparison recording a DI'd bass guitar, I thought the Purple lacked some of the depth of tone of the Urei. Later, though, I was able to make a direct comparison with a mid-period, black face, 1176LN and an unusual, light-grey, Haeco-badged, mastering unit (with the Urei name on the back and obviously similar construction). With the benefit of the excellent monitoring at Dave Gilmour's Astoria studio I was able to determine subtle differences between the units. The Urei was the warmest and richest sounding, with the Haeco not far off. In comparison the Purple seemed slightly brighter, which I assume may be due to the 'newness' of the components. The compression characteristics were similar, however the Purple was very slightly faster than the others, and seemed to distort more easily when driven hard. This was especially noticeable when I tried the (top-secret) all-four-ratios-simultaneously trick. I preferred the warmth and gentle overdrive of the Urei to the Purple's slightly harsh fuzz, but in most situations the Purple acquitted itself perfectly well. It did the job perfectly when gently compressing vocals or acoustic guitar on a 4:1 ratio. With the benefit of easy stereo linking, unless you can find a black-faced 1176LN for the same price then this unit would make an excellent purchase. Choose carefully though: the manual states that there may be some colour variation between units! ∎

(Thanks to Nick Ryan, Mark Thompson and Fletcher for background information for this review.)

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ADSG Digital Audio Disk Recorder

Talk of a digital dubber in Sony livery has been rife for the last few months. **Zenon Schoepe** reports on its first true public appearance

RUMOURS CONCERNING the existence of a digital dubber bearing the Sony brand name have been circulating for some months; although hard facts on the precise nature of the product have been scant. Indeed, the very existence of the product has been questioned by some observers, with sightings either side of the Atlantic not being verifiable and early direct questioning of Sony yielding no enlightenment. A marketing push to raise profile was clearly at work with presentations of the dubber reportedly made to the European and American film communities both of which were claimed to be on the verge of signing up for the new and still unseen format *en masse*. ADSG, as the entity began being referred to, even had a presence at the dubber's AES meeting held just prior to NAMM (*Studio Sound*, March 1998).

Perhaps I am being unkind but the word hype springs to mind as the methods employed to introduce this machine have been quite contrary to the more usual routes exploited by manufacturers. However, the machine was seen for the first time in public at last month's NAB Convention in Las Vegas, but not on the Sony stand, on the Iomega stand. Technically it is not a Sony product, it is an ADSG (Advanced Design Systems Group) product and the tie up with Iomega centres around this memorably entitled Digital Audio Disk Recorder's use of a 'Professional' variant of Iomega's new 2Gb Jaz media that is soak-pretested to, in ADSG's words. create drives and media that are virtually bulletproof and capable of the demands that the ADSG box has of the media—24-bit, 16-channel and seamless punch in-out.

ADSG is a wholly owned subsidiary of Sony Pictures Entertainment which is itself a wholly owned subsidiary of Sony. The idea for the box came through Sony Pictures Entertainment which claims that the existing dubber products were not giving them what they wanted. The box was developed originally for SPE's own use, but ADSG says there was always the intention to bring it to the world market. Announcements are still to be made about who will distribute and sell the product.

The dubber contains a number of SHARC processors, certainly more than are needed for a plain old dubber, and their presence is attributed to the potential for future full onboard editing capabilities. This, says ADSG, is the sort of capability that is now wanted on the stage and the dubber is said to providing a solution that will be fully capable as player, recorder or editor and that can be logged on to via a networking system.

Even so, the available processing does still seem to be substantially more than is necessary, and ADSG claims that it has built a 'total solution' box and that it will be announcing other capabilities in the future. Rumours that this processing could be harnessed by a third party for digital mixing would seem to be unfounded.

A curious point is that the PCM uncompressed file format is currently unique to the ADSG machine; although the company claims that compatibility issues are being addressed through discussions with other (rival) manufacturers.

To outline the specifications of this 16-channel 16-bit, 20-bit, or 24-bit dubber, it claims to offer 16 channels of individually selectable record and play, continuous output in forward or reverse play from stop to 1.5x speed, seamless punch in-out monitoring, audio input switching modes, 16-channel headphones monitor, cut and paste editing on individual tracks, units or across an unlimited number of machines, programmable levels of undo and individual track, unit or machine slipping. Transport follows biphase, LTC, VTC, P2 and position in feet/frames, minutes/seconds and millimetres/samples. I-Os are on AES-EBU with optional 24-bit A–D/D–A convertors. 96kHz capability is planned as a software upgrade.

Each 4U-high rackmount contains dual Jaz drives and an external SCSI port, 16 LED meters, alphanumeric displays, an LCD display, jog-shuttle wheel plus mode and transport keys. There is said to be no Sony Oxford technology in this box.

Price for the 16-channel unit is $15,000 US with shipping claimed to be immediate from a Sony manufacturing plant in San Diego starting with 50 a month and with the capability to produce 200 a month. SPE is planning to install 250 units at its studios.

Anyone looking at the dubber market with a perfectly unbiased eye could be forgiven for thinking that there is no shortage of viable contenders with Akai and Tascam's TimeLine units doing sterling work bolstered by the strong progress of Fairlight's DaD package.

However, ADSG believes that only 10% of the potential market is currently being addressed and this is clearly what it intends to capitalise on saying that it expects major decisions to be made in the next few months.

The promise looks good, but you have to question the mock-secretive manner in which this product has been broached and the effect this might have on the purchasing decisions of existing and established digital dubber technology. Even so, clearly one to watch closely. ∎

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<<<< convertor, the 8 ins and outs which can be configured as 24-bit convertors up to 48kHz or 96kHz as an option. It connects optically to Pyramix.

DUA has an onboard MTM interface that allows it to be interfaced to the Kefren and Keops audio cards. It occupies a single ISA Type Bus PC slot and contains all A–D/D–A electronics and AES-EBU I-O circuitry while a breakout box provides analogue connections and headphones monitoring. DUA has 4 analogue inputs, 6 analogue outputs, 2 AES-EBU 24-bit inputs and outputs, internal crystal lock, external AES-EBU sync and external wordclock sync.
Merging, Europe. Tel: +44 1763 849356.
Merging, US. Tel: +1 847 272 0500.

Audio codec

The Barco RE8960 audio codec encodes according to ITU-R rec724 and ITU-T J.41-J.42 and MPEG Layer II and is claimed to be a solution for studio-to-studio links, FM transmitter feeds, remote pickups, OB and remote coverage.

Bidirectional 1.5 and 2Mbit/s encoding and decoding send audio and data signals on the network and it supports digital and analogue audio formats with bandwidths from 7kHz to 20kHz. It can interface with six types of audio and data format including dual channel analogue, AES-EBU and RS232-422.

The device is also surprisingly compact at only 1U high with all settings, controls and monitoring software controllable via the ROSA graphical interface.
Barco, Denmark. Tel: +45 39 170000.

Lucid convertor

Lucid Technology has a series of multichannel A–D and D–A computer audio interfaces planned with the first, the ADA8824-Sonic, interfacing directly to Macs equipped with Sonic Solution's 16.24 digital I-O card. The device has 8 channels of 24-bit A–D and D–A conversion as well as 8 channels of AES-3 I-O and 2 channels of SPDIF I-O.

It has digitally controlled analogue input and output attenuation while 15 LEDs cover metering. Internally controlled sample rates are 44.1 or 48kHz while external sample rates can be controlled by AES-3 sync or wordclock.
Lucid Technology, US.
Tel: +1 425 787 3222.

www.americanradiohistory.com

Mr Smith
A blast from the past

Norman Smith's days as The Beatles' recording engineer and his work with the fledgling Pink Floyd earned him a succession of nicknames. **Richard Buskin** charts the successes that saw Normal become a Hurricane

THEY WOULD CALL HIM 'Normal' and '2dBs', and they used to rely on him for their sound, yet Norman Smith was never credited for any of the landmark recordings that he engineered for The Beatles from 1962 through until the end of 1965. No engineers were credited in those days. Only in 1968 did Smith's successor, Geoff Emerick, have his name printed on the liner notes to The Beatles' *White Album*, and even then it

was misspelled, 'Jeff'.

'During those years, we were limited as to how much bass we could put on a record without causing too much difficulty for the disc cutter.' Smith explains of his early experiences. 'Paul McCartney would often ask me for more bass frequencies, and I'd say, "All I can give you is 2dBs". After that I was labelled as 2dBs Smith, not to mention 'Normal'.'

As a jazz brass player who could also find his way around the drums and piano, Norman

Smith had been working in studios for several years before he embarked on a career behind the board at EMI Studios in Abbey Road. That was in 1959, his first child was on the way and, in need of a steady job, he lied about his age and secured the position of recording assistant as one of about 200 applicants.

'The age ceiling at that time was 28,' he recalls. 'I was 34, but I lied and fortunately was one of the three "young" men to be taken on.'

After a short spell serving as >>>>>

More than a saucerful of secrets—Smith's days with Pink Floyd

<<<<< a general gofer Norman Smith took his place at the mixing console, and among his early assignments were the recording tests that potential artists were then subjected to. 'That's how we started as engineers, because we couldn't really cock anything up,' he explains. 'Meanwhile each of the producers at EMI had their own assistants, and they would be the ones to keep an eye on the potential talent that was coming in. One day, of course, this group with funny haircuts came in...'

Not John, Paul, George and Ringo, but John, Paul, George and Pete—as in Pete Best, the handsome but professionally inept drummer who The Beatles would soon be ousting in favour of Ringo Starr. The group had already been turned down by every major British record label, including EMI's own Columbia and HMV, so there was a lot at stake when they undertook a recording test for that company's small Parlophone label on Wednesday, 6th June 1962. Nevertheless, their performances of 'Besame Mucho', 'Love Me Do', 'PS I Love You' and 'Ask Me Why' hardly made a startling impression on Parlophone's Head of A&R, George Martin, his production assistant Ron Richards or engineer Norman Smith.

'Visually, they made quite an impression, but musically we didn't really hear their potential,' recalls Smith. 'They had tiny little Vox amplifiers and speakers which didn't create much of a sound at source. Every sound engineer wants some kind of sound at source which he can then embellish—but I got nothing out of The Beatles' equipment except for noise, hum and goodness knows what. Paul's was about the worst; in those days we had echo chambers to add on the reverberation, and I had to raid the Studio 2 echo chamber in order to fix him up with a sound so that we could get something down on tape.'

The solution was to fetch a large Tannoy speaker from said echo chamber and solder a jack socket onto the input of a Leak TL12 amp. Only at that point could the recording test properly get underway. 'Afterwards we brought them all up into the Studio 2 control room,' Smith continues, 'and for about half an hour we laid into them about their equipment

and the fact that we needed to get some decent sounds. They didn't respond at all, and so George Martin then asked them if they had anything to say to us. Perhaps there was something they didn't like. George Harrison looked George Martin up and down and said, "Yeah. I don't like your tie". With that I just creased up. I always admired the Liverpudlian accent for its potential humour, and that just about confirmed it for me. After that the rest of them joined in with the wisecracks, and that was far more impressive than the musical side of things. When they left George Martin said to me, "Well, what do you think of that lot?", and my response was to say that I'd never seen anything like them. They were so different.

'At that time there was an enormous number of groups coming in for tests and none of them really showed any potential or anything visually different, but of course these guys did. I said, "For that alone we should sign them. Just because of their humour and the way they present themselves they are different." So George said, "Well, I'll think about it," and shortly afterwards he signed them.'

GEORGE MARTIN, in fact, didn't even take charge of the aforementioned recording test. That task was assigned to his assistant, Ron Richards, as Richards was really the pop producer of choice. (Martin had forged something of a reputation for producing comedy and variety records by the likes of Peter Sellers, Beyond the Fringe, Matt Monro and The Temperance Seven.) Nevertheless, George Martin became involved in the session after Norman Smith had alerted him to the commercial sound of 'Love Me Do', and he subsequently produced the first recording session proper, which took place on 4th September 1962, with Ringo Starr on the drummer's stool.

Not for long, though—less than impressed with Ringo's debut performance on 'Love Me Do', the Parlophone men had The Beatles return for another session a week later, and this time, with Ron Richards temporarily back at the helm, Ringo was relegated to playing the maracas on 'PS I Love You' and tambourine on 'Love Me Do' while >>>>>

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<<<<< session man, Andy White, took over on the skins.

'Two sessions may not sound like a very long time to get 'Love Me Do' completed,' Smith opines, 'but it does when you consider that we did The Beatles' first album [*Please Please Me*] in one day. Ringo's drumming, I thought, was okay. He never was a great drummer, even at the height of their fame, but he was adequate for The Beatles and the main thing about him was his personality. On the other hand I didn't see any reason for Andy White being brought in, because a little boy could have played the drum part on 'Love Me Do'.

Several years ago, George Martin asserted that, while Ringo may not be the most technically proficient of drummers, he does have a distinctive sound, and he was also very innovative in his use of percussion instruments in the studio. As a former drummer himself (and the son of a drummer), Norman Smith challenges this.

'The truth of the matter is that Paul had a great hand in all of the songs that we did, and, to the best of my memory, Ringo would generally ask him what he'd like him to do.

Hurricane days: the haircut that fooled *Top of the Pops*

After all, Paul was no mean drummer himself. Anyway, I can't understand the "innovative" remark at all. Perhaps Ringo came up with the odd thing or two...'

'Love Me Do' eventually climbed to No. 17 in the UK charts—'better than we expected,' according to Smith—and with confirmation that Parlophone had a new hit act on its hands George Martin took over full time as The Beatles' record producer. 'I think that was largely because of [manager Brian] Epstein's influence,' Smith comments. 'He and George had a pretty good rapport, and initially he wanted George to come in and give his okay to 'Love Me Do'. Then, after that was a minor hit, we really started to hear John and Paul's writing ability, and so the boss of Parlophone took over from there on in.'

Lennon and McCartney's compositional talents became evident with the songs that they concocted for their follow-up singles, 'Please

Please Me' and 'From Me to You', not to mention their first album which took just under 10 hours for 10 new tracks to be added to four that had been recorded previously. Furthermore, The Beatles were no slouchers when it came to covering other artists' material, as was evidenced when, with nine of the 10 numbers in the can, they performed a breathtaking rendition of The Isley Brother's 'Twist and Shout' on the first take.

'After a whole day of singing John's voice very nearly didn't hold out,' recalls Smith. 'He was always under a bit of strain given the timbre of his voice, but the band had a large jar of cough lozenges—Zubes—as well as a carton of cigarettes, and he just went for it. Due to the state of his voice, we knew we had to get that song in one take, and fortunately we did. There again, if you listen to it, perhaps the sore throat and the hoarseness improved the performance. It's a terrific performance.'

THEREAFTER, hit followed hit, and with each successive album came a new and distinctive sound: double-tracking on *With The Beatles*, the trademark jangling of George Harrison's Rickenbacker 12-string on *A Hard Day's Night*, and on and on. This progression was all the more remarkable when taking into account the fact that, during the years 1963 to 1965, the group was releasing two albums a year comprising largely Lennon-McCartney compositions, and that the musicians' and technicians' ideas were often racing ahead of the available technology.

'EMI always gave us—the engineers—the chance to experiment on our own in the control room if we had a certain kind of sound that we were after,' Smith explains. 'We could go in and try different ideas, and out of that did come certain things, such as tape delays, ADT and other things of a semi-technical nature. Actually, back then the sound engineers were judged—quite unfairly, in my view—by the number of hits that they'd worked on. Consequently, once The Beatles broke through I was also walking on water at Abbey Road and I could do no wrong, so I could more or less do exactly as I wanted. Before then, Ron Richards and myself had been struggling to get hits with people like Shane Fenton—who later became Alvin Stardust—and Paul Raven—who would become Gary Glitter—and so The Beatles made it for me, and from then on I was the number one engineer.

'Up until the time when I became a sound engineer the other engineers would always use screens. Everything was screened off so that the separation was good on each mic, but I didn't like that idea for The Beatles once it had been decided that I was going to record them. I wanted to set them up the way that they looked, in line with their attitude and how they approached things, and it seemed to me that they would be far happier >>>>>

<<<<< if they were set up in the studio as though they were playing a live gig. I therefore threw all of the screens away, and the Abbey Road management warned me that I was taking a little bit of a chance, but The Beatles performed as they did on stage and although the separation on each mic wasn't terribly good it did contribute to the overall sound. We also got a bit of splashback from the walls and the ambience of the actual studio, and in my view that helped create what the press dubbed the 'Mersey Sound'. I'd receive phone calls and letters from America asking how I managed to get all of that sound on tape.

'As far as my side of it was concerned—and all of the other engineers, for that matter—we were sort of restricted. The material had to be transferred from tape on to acetate, and therefore certain frequencies were very difficult for the cutter to get on to disc. I mean, if we did, for instance, slam on a lot of bass it would only be a problem when it got up to the cutting room, but at the same time we were all a little bit frustrated that we couldn't get certain kinds of sounds that we would have liked.

The golden wedding party: '...still can't remember the title of thet song...'

Instruments like the sitar were terribly, terribly difficult to record due to the range of sound frequencies—the meter would be bashing over into the red, and so you didn't get any value for money.

'Of course, there was more equipment becoming available as time went on, and then there was the contribution of all six of us—or five of us, not counting Ringo—so all of that contributed to the progression of sounds, as did the kinds of songs that were coming across. Don't forget that their writing was changing too and this demanded a different sort of approach. By the time of *Rubber Soul*, for example, we were dealing with completely different types of songs, and to be honest with you the attitude towards not only the recording but also each other was beginning to show not too nicely.'

This is an interesting disclosure. Previously The Beatles and their associates have indi-

cated that everything was hunky dory in the studio until the *White Album* sessions in 1968. Smith's recollections, however, point to the beginnings of dissent creeping in a full three years earlier when Beatlemania was still at its height.

'Up to that time we had a kind of family setup between the four guys. George and myself,' he continues. 'It was superb, and you can probably imagine what a great time it was to be part of all that. Also—and I don't want to take anything away from anyone—production of The Beatles was very simple, because it was ready-made. Paul was a very great influence in terms of the production, especially in terms of George Harrison's guitar solos and Ringo's drumming. It was almost like we had one producer up in the control room and another producer down in the studio, and John Lennon also knew what sounds he wanted...'

This is another interesting remark concerning the inner workings of the group, given the tensions that would later arise out of John and especially George's problems with Paul trying to run matters both in and out of the studio.

'When *Rubber Soul* came around it was taking a lot longer to record each title,' says Smith. 'I could see the friction building up and I didn't like it at all. I thought, "To hell with this", and I told George Martin, "I don't like what I see and I want to get off this train". He said, "They're going to be very upset about this." But I said, "Well, that's the way it is." A few days later, what should appear by special delivery but a solid gold travel clock from Aspreys, inscribed with thanks from The Beatles. That was a sweetener to stay on and initially I agreed, but it was nevertheless very difficult and I still didn't like what I could see coming up, particularly between Paul and John. I could see them drifting apart and I did not like that one little bit.'

Work on *Rubber Soul* ended in November 1965. Three months earlier, George Martin had struck out on his own along with Ron Richards and John Burgess to form Associated Independent Recordings (London) Limited—or AIR—and so there was room for Norman Smith to move up within the EMI organisation and become an in-house producer. This is what he opted to do in February of 1966, and so he bid *adieu* to the Fab Four and farewell to the faders, and within a year he was producing an up and coming band by the name of Pink Floyd.

'I left engineering completely behind,' Smith says. 'I could still put in my four penn'th, but as a producer with Pink Floyd I had too many other things to worry about, such as Syd Barrett.'

Having seen a Floyd live performance Smith was sufficiently impressed by their stage presentation to sign them to EMI. 'I can't in all honesty say that the music meant anything at all to me,' he admits. 'In fact, I could barely call it music, given my background as a jazz musician and the musical experience that I'd had with The Beatles. After all, with The Beatles we're talking about some- >>>>>

interview

<<<< thing really melodic, whereas with Pink Floyd, bless them, I can't really say the same thing for the majority of their material. A mood creation through sound is the best way that I could describe Floyd.

'There was something about Syd Barrett's songs which was indescribable—nondescript—but obviously had that Barrett magic for an awful lot of people. His songs were interesting as far as I was concerned, but more for the question, "Well, why the hell did he write that?". You know, "What inspired him?" Nevertheless, we got along as well as anybody could with Syd Barrett. He really was in control. He was the only one doing any writing, he was the only one who I, as a producer, had to convince if I had any ideas, but the trouble with Syd was that he would agree with almost everything I said and then go back in and do exactly the same bloody thing again. I was really getting nowhere.'

Add mind-altering chemicals to what Nor-man Smith describes as Barrett's 'peculiar personality' and it isn't hard to see why matters quickly began to get out of hand.

'I never actually saw him taking drugs in the studio,' says Smith, 'but he had the kind of character that, even if he hadn't taken any, you'd think he was on drugs. You couldn't hold a sensible conversation with him for longer than 30 seconds. Roger Waters had the best rapport with Syd but even he found it difficult. I remember them being on *Top of the Pops* with 'See Emily Play', and beforehand I took them into Number 1 studio purely to rehearse what they would do on television for the first time. I was almost choreographing them—silly little old me, thinking they would actually stick with my choreography. Of course, that all went out of the window thanks to Syd. He wasn't happy about doing *Top of the Pops*, he didn't like singles—he only liked doing albums—and it was while waiting around for an appearance on [BBC

Radio's] *Saturday Club* that he walked out of the door and went missing. That really was the first sign of his complete mental breakdown, and he never did come back into the studio after that—meaning that I had a hell of a hard time with the recordings.

'As musicians, the Floyd were capable enough, but again Nick Mason would be the first to agree that he was no kind of technical drummer. In fact I remember recording a number—I can't now recall which one—and there had to be a drum roll, and he didn't have a clue what to do. So, I had to do that. Nick was no threat to Buddy Rich. Roger Waters, on the other hand, was an adequate bass player, but to be honest he used to make more interesting noises with his mouth. He had a ridiculous repertoire of mouth noises, and we used that on one or two things; you know, a bit of Rolf Harris.

'So, they were capable, but I had a great struggle with them after Syd Barrett went funny and left. They tried very hard to write material—I remember them writing a song about apples and oranges, which I dressed up and released as a single, and that sold about six copies. It was around the time of the *Saucerful of Secrets* album, and I really did think that it was all over. We still had to keep the singles coming for the audience that they had established, but the songs that were being composed by the other three guys were, to say the least, lacking in commerciality. Their recording career was going down the drain fast, and then along came Dave Gilmour and things started to pick up again. They were able to get back to what Pink Floyd was all about and re-establish themselves.'

After producing *Atom Heart Mother*, Norman Smith was the executive producer on 1971's *Meddle* album.

'These guys knew what they wanted, and so it was silly for me to contribute anymore,' he says. 'I thought I'd done my bit with them and encouraged them to produce themselves—they were producing tape loops at home and bringing them into me—as well as to be resourceful in the studio. Having done that it was time for me to retire gracefully and offer to help them if they needed my advice at any time. However, they didn't, because Roger—a bit like Paul McCartney—had the makings of being a good producer, and Dave Gilmour showed this ability as well.'

BY THIS TIME, Hurricane was standing in the wings. Having been songwriting since he was 10 years old, Norman Smith had been dabbling on the piano in Studio 2 during a dinner break from a Floyd session. Out of that came the germ of an idea that would evolve into 'Don't Let It Die', a number that he thought would suit John Lennon and which he thus demo'd in his best pseudo-Lennon voice. This he subsequently played to that well-known hit-maker, Mickey Most, who, unaware of the identity of the mystery vocalist, identified the song as a Top 3 hit. On learning who the singer was, he insisted that the recording should be released as is. The composer-producer-performer wanted to add strings. Most told him to go ahead, although not before pointing out that he wished to pick up the publishing rights. An astute move, and he ended up getting his

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way—EMI initially turned the song down, only to do an about-face when Smith informed them that Most would do so instead. Suddenly 'Don't Let It Die' was hot property, and Smith managed to cut a deal whereby the company saved face by releasing the product of its own employee while allowing Most to secure the publishing.

'My wife and I were on holiday in Corfu at the time of the song's release,' Hurricane recalls, 'and while we were there I received a telegram to say that it had entered the charts at No.18. Soon after that it jumped to No.2, but as I was still away I couldn't do *Top of the Pops*. At that time I had long hair, and so Mickey had the brilliant idea of finding someone who had hair like me and having the back of his head photographed. That's all that people saw when they did the run-down on *Top of the Pops*.'

Eventually the man himself was available to promote his own song, as well as the subsequent 'Oh, Babe, What Would You Say?' which topped the charts in America, leading to an appearance on *The Tonight Show* with Johnny Carson and a full-fledged concert tour.

As for the 'Hurricane' moniker: 'I wanted to retain the name Smith, and after trying different ideas I saw the old movie *Hurricane Smith* being advertised in the *TV Times*. As it happens a group already had that name, and halfway through the labels being printed for 'Don't Let It Die' their lawyer contacted EMI and said we couldn't use it. In the end they thrashed out a deal whereby whoever got a hit first would retain the name... I got the hit.'

After Hurricane hung up his frilly shirts and flares, Norman Smith undertook a pair of independent productions with Denny Laine and Malcolm Roberts. Then he decided to call it a day. Nevertheless, it should be pointed out that the Hurricane had very nearly enjoyed huge success as a songwriter several years before. The time was June of 1965 and The Beatles were nearing the completion of the *Help!* album. Smith takes up the story which he describes as being 'like something out of a Hollywood movie'.

'It was a Friday evening, I shall never forget it as long as I live. They'd recorded 13 songs for the album, but we always aimed for 14, and so they said, "Well, what are we going to do now for this last one?". They were running through different ideas and discarding them, and I was sitting up in the control room with George Martin and [their music publisher] Dick James, so I turned to George and said, "I happen to have written a song for John and it's in my inside pocket." He said, "Well, tell them over the talkback." I said, "No, I can't do that you tell them. But don't get them all to come up. Only Paul". So George Martin asked Paul to come up and he told him that I'd written a song. He said, "Really , Norm?" I said, "Yeah, I have actually." "Well, let's hear it then. Come down..." I said, "No, no, no. I'm so nervous about it. Let me go across to Number 3 with you and I'll play it to you.

'So that's what I did and he really did think it was good. He said, "That's terrific, Norm! It sounds good for John." I said, "Well, no offence, Paul, but I've written it for John." "Oh, right. Let's get John in." So John came into Number 3. I performed it again

and he said the same. "Yeah, smashing. We'll do that."

'I couldn't believe it.

'When we returned to the control room both Paul and John asked me to record a little demo over the weekend in time for the session on Monday, and that's what I did, but in the meantime Dick James offered me £15,000 to buy the song outright. Back in the mid-sixties that was a terrific amount of money, and to me it was staggering, being that I had about four pence in the bank. I was about to say, "I'll take it", but George Martin was sitting behind Dick James and he was shaking his head as if to say I should ask for more, so I went, "Well, I'll tell you what, Dick; I'll think about it and I'll let you know on Monday". When Monday came, I was sitting with my demo at the mixer a couple of hours early, but when the guys came in they said "hello" and walked straight down into the studio. I thought, 'That doesn't look too good',

and then Paul spoke over an open mic and asked me to come down. I went down and Paul said, "Look, Norm, don't be upset about this, but do you realise that on this album we haven't done a song for Ringo?" Ringo always had a song, so they now had one for him to record, and Paul said, "We promise you that we'll do your one first on the next album." Well, the next album was Rubber Soul, by then their attitudes had changed, and so that was the closest that I ever got to having one of my songs recorded by The Beatles.'

And the name of said song? 'I thought you'd ask that, and it's produced another ache in my heart, because not only could I never ever find that damned manuscript or the little demo that I gave them, but I can't even remember the blasted title either. It was a good song with a solid beat, almost like the solid beat of 'Twist and Shout' but with a bit of romance too... I try not to think about it too much.' ■

Few failed singers go on to claim responsibility for three generations' of American hit records. **Dan Daley** believes that Brooks Arthur is the only one

Arthurian Legend

H E WANTED TO BE a singer. But instead, Brooks Arthur wound up as an engineer and producer with a discography that few of his contemporaries could equal, covering many of the great records of the 1960s, from 'My Boyfriend's Back' (The Angels), 'The Locomotion' (Little Eva), 'Hang On Sloopy' (The McCoys), 'Chapel of Love' (The Dixie Cups), 'Leader of the Pack' (The Shangri-Las), on through virtually every major hit that Neil Diamond ever recorded, as well as the Janis Ian classic, 'At Seventeen', to best-selling Broadway shows and comedy records for Robin William, Jackie Mason and, most recently, a multiplatinum for film and television comedian Adam Sandler.

And he still takes singing lessons.

'What can I say? I wanted to be one of the crooners,' laughs Arthur, who has penned a stage show by that name now playing regionally in the US and heading for a London opening later this year. 'That's been a big part of the secret of my success: I could relate to great singers because I know how singers think and what they need in the studio. Three of the greatest I ever worked with were Neil Diamond, Van Morrison and Gene Pitney. But on all the records, the vocals were the most important thing.'

Arthur knew he wanted to sing early on, and thats when the voice lessons first started, as he was growing up in the very Italian-American neighbourhood of Bensonhurst, in Brooklyn, New York. But it was his part-time after-school jobs in the mail rooms of Decca Records and Kapp Records in Manhattan that actually gave him the foot in the studio door. Literally.

'I was delivering the mail all over Decca Records and the guys at Decca's studios [one in Decca's midtown offices on West 57th Street and a larger one downtown known as the Pythian Temple] and the studio guys liked me and let me hang around while they were making records,' he recalls. 'One A&R guy, Milt Gabler, who later turns out to be Billy Crystal's uncle, let me watch my first recording session. It was Ella Fitzgerald and Gordon Jenkins together. And Bob Thiele had Coral Records in the same building and I remember watching him do a session with The McGuire Sisters. It was incredible. The guys in the mail room kept wondering whatever happened to me. I was never around. It's cause I was always hanging around the studios.'
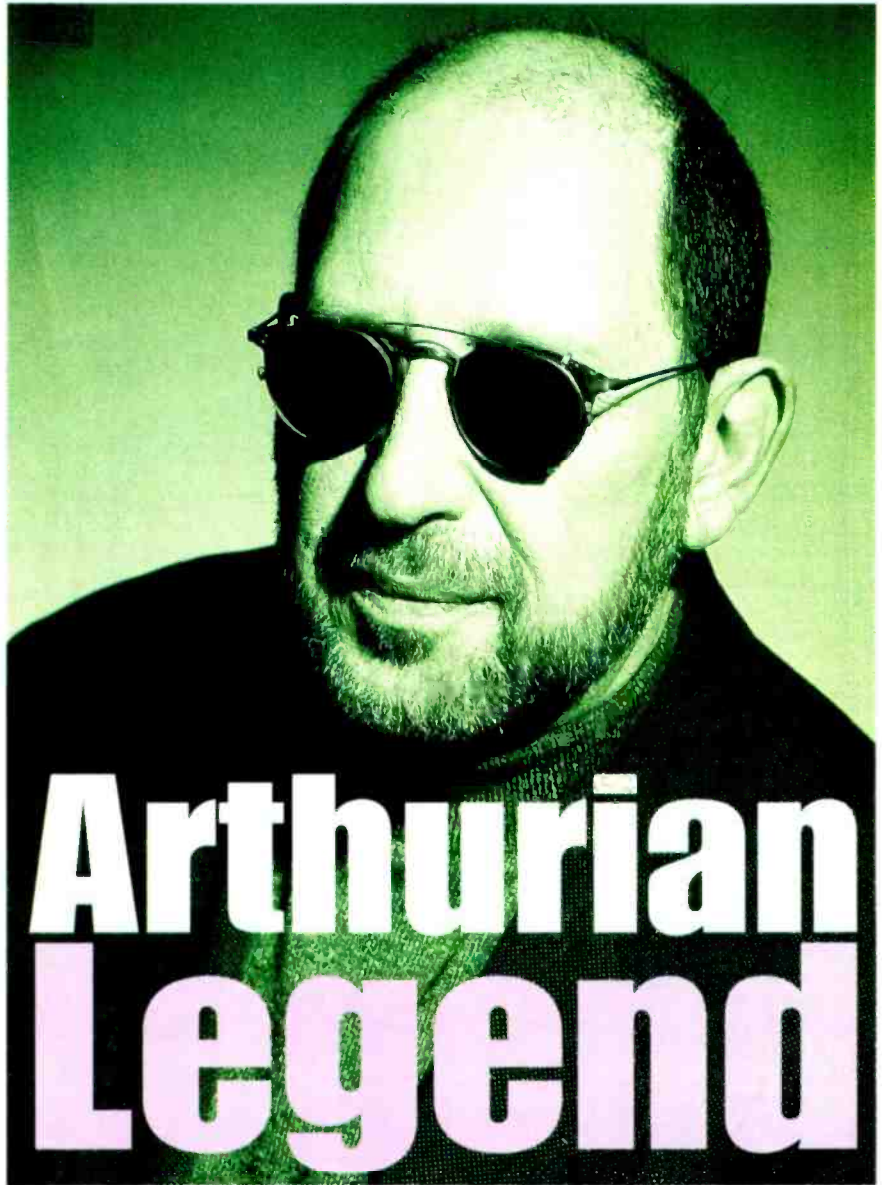
Arthur found that the staff engineers at the studios in Manhattan at the end of the 1950s were an affable bunch—the crankier ones were soon to start opening their own independent studios in the next decade—and they were more than happy to answer his innumerable questions. Meanwhile, he was also pursuing songwriting, and it was at wunderkind publisher Don Kirschner's office that the young Arthur met Barry Mann, Cynthia Weill, Neil Sedaka, Carole Bayer Sager and other up-and-coming composers working out of Kirschner's offices on Broadway. He sang, he composed, he hung out at studios. All of that came together in 1961 when he suggested to several writers in the Kirschner stable that he record their demos at a local independent studio, Dick Charles Recording, with its custom console fitted with Langevin faders and huge Daven rotary pots, was happy to see Arthur, and didn't question his engineering credentials when informed that the kid from Brooklyn was bringing in his own paying clients.

'I was flying by the seat of my pants,' says Arthur of the time. 'But I was happy to be in the studio. All those gems of microphones —RCA 77 and 44s, Neumanns and Telefunkens. Back in those days, having them around was normal.

'I was recording through the custom console at Dick Charles, which had stepped EQs. But it coloured the sound and loaded it too much, so instead of using them, I ran the signal right through a rack of Pultecs and sent straight to the mono Ampex tape machine. That machine had huge electronics and a giant vu meter—you couldn't miss it. We were using two Ampex mono machines and bouncing between them. I would record the rhythm track onto one machine, then play that back to the singers through the phones and rerecord the track while recording the singers all onto the second mono machine. Just that process created the sound. You can really hear it on 'My Boyfriend's Back' >>>>>

Cashing in on celebrity status with The Angels

Arthur with Dusty Springfield

<<<<< by The Angels, and on 'The Kind of Boy I Can't Forget' by The Raindrops (two of whose members were the composers, Ellie Greenwich and and Jeff Barry).

'I was using the Universal limiters and compressing the shit out of it, and after four or five generations with overdubs and more vocals, it made for an incredible sound. I'd describe it as somewhere between gating and breathing. Especially the drums—I limited the hell out of the drums, the kick and snare. It was totally adding noise to the tracks each time, but that was part of the vibe. We didn't look at it as noise; to us it was part of the vibe, part of the sound. And the 45s and 78s that you played them back on had surface noise that rivalled the signal anyway.'

As a result of the demo work, Arthur began making a name for himself as an engineer in New York. He was sought out by the production team of Bobby Feldman, Jerry Goldstein and Richard Gotterher—known collectively as FGG—for records with groups of the day like The Angels and The McCoys. Arthur moved from Dick Charles. where he'd been pulling down $65 a week since the owner put him on staff, to the princely sum of $120 per week at Associated Recording Studios. Arthur was developing recording methods of his own by now. 'My Boyfriend's Back', for instance, was recorded at Associated with Arthur using an EV 666 on the kick drum, an old—even then—Altec 'Saltshaker' mic on top of the snare and an Altec 'Lipstick' mic beneath, an RCA 77 under the toms, a Neumann 67 for the acoustic bass, a pair of Altec mics on the piano…

'Even though we were recording mono, I still needed two mics to get the width of the piano,' he explains. 'I used a 67 a lot on vocals because it gave the girls great body. To their voices, I mean. A 47, too. We ran the track down a few times with the girls singing it for the band in the room, then the band laid down a track with no guide vocals. Then we started bouncing between

machines. I did the overdubs on 'The Loco-motion' that way, too, and the records I did with The Shirelles. It worked because even though the recording setups were pretty much the same, the groups all had very different personalities that would come through.'

By the time Arthur and Neil Diamond met, 3-track and 4-track machines had become the norm; although Arthur says that the 3-track format came and went so quickly he hardly used it. Bert Berns, owner of Bang Records and cowriter of, among other classic songs, 'Piece of My Heart' and 'Hang On Sloopy', brought Diamond to Arthur to engineer his first record.

'Neil was awesome, a true star,' recalls Arthur. 'You knew it from the moment he walked into the room. He also knew what he wanted on the record. Me and [producers] Jeff Barry and Ellie Greenwich called it "the Neil feel". That sobriquet eventually became the title of his 1966 debut album, which produced the hits 'Cherry Cherry' and 'I Got The Feelin'.'

Arthur marshalled his Ampex 4-tracks at A&R Studios fairly conventionally: Track 1 was bass and drums ('That's a great way to go,' he exclaims. 'It means you have to commit'), Track 2 was guitars, Track 3 was percussion and the fourth track held Diamond's guide vocals. Then the first three tracks were bounced over to two tracks of a second Ampex or Scully 4-track. Horns, background vocals and other parts were added in a similar manner until it was time to do Diamond's vocals. It was here that Arthur's touch seemed most magical.

'We actually had some keeper pilot vocals from Neil—'Shiloh' and 'Kentucky Woman' were two like that,' he recalls. 'But he was such an amazing singer that the hard thing to do, but the right thing, was to get him do his stuff.'

Other groups needed more coaching. Arthur engineered 'Come a Little Bit Closer' for Jay & The American in 1964 at Mira Sound in Manhattan, produced by Artie Ripp. The group was made up of typical Brooklyn kids who did most of their singing on the street under lamp posts and in reverberant high school stairwells. The studio was something else, and a check on the vocal pitch of many of the records of the period on more modern equipment reveals that many of the groups were far from the studio veterans of whom Arthur was now a cohort.

'We were very concerned about >>>>>

Arthur in his Century Sound studio in Manhattan, opened in the mid-70s

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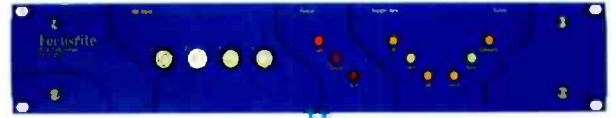
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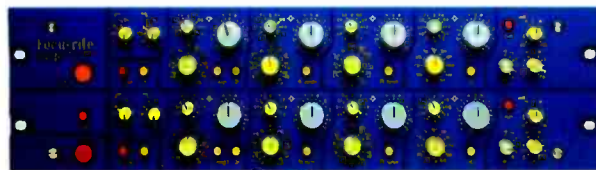
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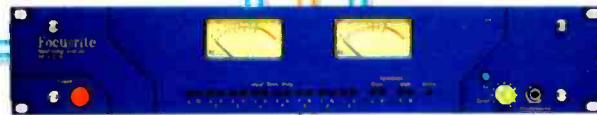
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Directing The Chiffons

<<<<< intonation,' says Arthur. 'Artie was going for the big Tony Bennett-Mario Lanza vocal performance on these records, so the vocals were very out front and had to be good. The pitch was often atrocious on some of those old records. You should have heard what didn't make it onto the records. It was menacing. Monitoring sucked at the time, and these kids came in off the street or they had been singing on stages in clubs. I taught them about taking one side of the headphones off. In other cases, I used speakers in the room while they were singing. I would wheel a couple of JBL or Altec 604 monitors into the studio and position them to the dark side of the microphone, which I had set on a hyper-cardioid pattern. Then I would reverse the phase on the speakers so whatever did leak would disappear. I did that a few times with Jay and even with Neil, and with the girl groups. It was a more familiar way for them to hear their own voices while they sang instead of on headphones.'

Speaking of girl groups, Arthur engineered two classic records for The Shangri-Las —'Leader of the Pack' and 'Walking in the Sand'. Both were done at Mira sound and with George 'Shadow' Morton producing 'Leader'; Greenwich, Barry and Leiber & Stoller produced the other track. Morton's nickname derived from his propensity to unexpectedly disappear from the scene for extended periods and then just as mysteriously show up again. The pattern applied to the studio, as well.

'He was a very mysterious person and producer,' remembers Arthur. 'He was like Quentin Tarantino. He was making *Pulp Fiction* records—vignettes, very cinematic and dramatic, and they had to have a narrative, a story to them. 'Leader's' multitrack came in from a studio called Ultrasound out on Long Island, cut by my friend named Rod McBrien. We were doing mostly the vocals and some overdubs, but the most famous overdub was the motorcycle sound. My assistant engineer was Joe Venneri, also a sometimes member of The Tokens. He used to ride his motorcycle to work and park it out in front of the studio. So we ran a mic cable out the front door and set up an EV microphone on a boom outside and ran it directly into a mono tape machine in the studio. And then Joe cranked away on his bike.'

Just for the record, the motorcycle on the record was a Harley Davidson.

Arthur loved to double and triple background voices, and pretty much stayed with Neumann 67s and 47s for the process. The 67 was the one he used on most of the girl groups, and what Jeff and Ellie and Shadow snapped their fingers to in the hallway at Mira Sound as percussion for The Shangri-Las' 'Walking in the Sand'.

Arthur was itching to expand as a producer, though. Shadow Morton had produced—and Arthur engineered—'Society's Child', Janis Ian's controversial 1967 hit single, but took one of his famed long walks after that. A few years later, Arthur again met Ian and began an intense professional relationship that in 1975 produced *Between the Lines*, a platinum album that contained the hit 'At Seventeen', which garnered her a Grammy for Best Female Vocal of the Year, and was an auspicious start to Arthur's career as a producer.

'It was more or less my first step into production,' he says. 'She was definitely precocious. I remembered her from 1966 in the studio and the way she was, she was not intimidated by the studio—the studio was intimidated by her. She knew exactly what she wanted to hear. And this was a 15-year-old girl at the time. 'Society's Child' we did on a couple of 4-track machines Self-Synced together. When we did *Between the Lines* in 1975, by then I had a studio of my own in Blauvelt, in upstate New York but not too far from the city. A&R Studios was my partner in the studio, but mostly it was for my own projects. It had Scully 16-track, 4-track and 2-track decks and a Suburban Sound (SSI) console built by Neil Monsey, who also partnered with A&R, which is how I got that console.

'Janis and I were working together up there on the record. She was living in a rented bungalow not far from my house, and during the day we worked and at night we ate >>>>>

Brooks Arthur at the console in New York City's Mira Sound Studios

Rob Schneider, David Spode, Brooks Arthur and Adam Sandler

<<<<< pasta. Making that album became a way of life for us. And I remember [Rolling Stones executive coproducer] Alan Steckler dropped by one day and heard 'At Seventeen' and said, You have a hit on your hands there. That gave us a lot of encouragement. While I was mixing the record, I was listening to Carole King's *Tapestry* album, which had recently come out. I always wanted to do a record with her because I remembered her from the songwriting days on Broadway, but Lou Adler beat me to the punch. I set up a Thorens turntable and a Macintosh integrated amplifier in my office and I set up a direct patch from the studio to my office in the same building and constantly compared my mixes of *Between the Lines* to *Tapestry*. I was in love with that record, and even though I have a mind of my own, I really wanted *Between the Lines* to compete with *Tapestry*, which was the record I had always wanted to make.'

If making so many classic American artists of the sixties and seventies was not enough, Arthur brought an overseas classic to the fore. Bang Records' Bert Berns had been developing UK artists, most notably the Belfast club band Them, the protean unit assembled by its lead singer Van Morrison, to back him in his club at the Hotel Maritime. Van Morrison disbanded Them in 1966 after a frustratingly short career, but Berns sent Morrison a plane ticket to the US and in 1967 he was in the studio with Arthur recording *Blowing Your Mind*, from which 'Brown Eyed Girl' reached No.10 on the Billboard charts that year.

'We did that record at A&R Studios,' says Arthur. 'We did the record in Studio A, and I used a 6⁻ tube mic on his vocals, which were always incredibly strong. The double stops on the guitar on 'Brown Eyed Girl' were played by [New York session player] Al Gorgoni and by Van. Eric Gale was known as a guitar player, but he played bass on the sessions. Herbie LaValle played drums, Artie Butler played organ and Hugh McCracken played acoustic guitar. Someone recently found my take sheets from those sessions, so I can tell you that the master of 'Brown Eyed Girl' was take 22, done on 28th March, 1967. The first song we cut on the record was 'Row, Row Rosie', and we used take four.

'I had never worked with any more emotional singer than Van Morrison,' Arthur continues. 'He could be volatile at times, but that wasn't it, really. He was just putting his marrow, his soul, into every take. He just coiled up and gave it. He was a soul singer. He was not singing—he was baring his soul. When he was doing his pilot track vocals for 'Brown Eyed Girl', it would take him a good three minutes to come out of the trance he put himself into to sing. It was something to behold.'

Astral Weeks followed in 1968, a jazzy, meandering record that attracted the initial core of the Van Morrison cult. Arthur asserts he was the engineer of that record, though Morrison's management 'forgot' to mention this fact on the album sleeve and, he says further, reneged on a promise to correct that on the record's subsequent pressings. 'But it was the album that showed me I was growing up as a producer,' he says.

Arthur also engineered the reformed Blues Project eponymously named MCA recording in 1973, and has produced a number of records for jazz greats like John Pizzarelli, many in Century Sound, the studio he opened in Manhattan in the mid-seventies on West 52nd Street, unofficially dubbed Swing Street by generations of New York players. But he never lost his affection for the crooners, nor they for him. Gene Pitney apparently forgave Arthur after Pitney flew in to New York from Texas for a 50-piece session that Arthur had set up at Mira Studios.

'Gene wouldn't work with anyone else but me at that time,' Arthur recalls. 'I think it was, again, because I could have real empathy with singers about their vocals. But before he flew in for that date, I had been working incredible hours, starting at 9am and going till 4am. I used to get a room and sleep at the America Hotel, a fleabag that was in the same building as the studio. Nipsey Russell lived there and he used to stop into the studio all the time. But I had caught this virus because I was so run down and it affected my inner ear. Not good for an engineer. I got so sick I went to the hospital and I was in for three days. So Gene shows up at the studio, but no Brooks. So the producers come to the hospital and beg me, saying, "Can we prop you up in the chair for the session?". I said, "Yeah, okay", and after I got to the studio, I proceeded to throw up all over them. Sometimes you have to know when to say "No".'

Arthur remains incredibly busy as a producer these days, with his most recent success in comedian Adam Sandler's three records, and a forthcoming one from another *Saturday Night Live* alumnus, Norm MacDonald, a connection made through Sandler. And he approaches comedy records the same way he did music.

'It's the same as producing singers', he says. 'It's all about performance, and knowing how to capture them.' ■

L OOK THROUGH any film encyclopedia and you will find numerous actors, editors, directors and cinematographers. Less numerous are sound designers. Fewer still are sound designers who have also excelled in picture editing, directed their own movie and even done stints as camera operator and actor. In a field that, even today, gets overlooked in favour of special visual effects and other technical disciplines, Walter Murch is something of a hero, a fact underlined by the number of people queuing up to talk to him or at least shake his hand when he gave a lecture at the 'School of Sound' seminar held in London during April.

These sessions were accompanied by a season of movies with sophisticated or notable soundtracks at the National Film Theatre. Among the titles were four of Murch's best: *American Graffiti* (1973); *The Conversation* (1974), which he introduced at the NFT, and for which he received his first Oscar nomination; *Apocalypse Now* (1979), his first Oscar winner; and *The English Patient* (1996), for which he picked up two statuettes, one for the sound, one for picture editing.

Born in New York, Murch was part of the first generation of movie lovers who went to film school. It was at the Cinema School of the University of Southern California that he met the writer-director-producer George Lucas, who would go on to become one of the most successful commercial movie makers of all time. Murch's early career is entwined with that of Lucas and both became involved with another huge figure (in both senses) when Lucas won a scholarship to observe Francis Ford Coppola making the out-of-character *Finian's Rainbow* for Warner Bros in 1968.

On Coppola's next movie, 1969's *The Rain People*, Lucas took the role of production associate, while Murch got his first sound credit. In his work, Murch has always acknowledged the correlation between what is seen and what is heard, something that was not necessarily the product of film school lectures. 'I think that, to a degree, it was already obvious to me,' he says. 'As I progressed in my career, the depth of the relationship between the two elements became more and more obvious. There is an opportunity to take advantage of this complex relationship but you are not free to do that completely because you are always an observer of the film as well as someone working on it.'

M URCH IS NOT an admirer of the soundtrack for its own sake style of working, preferring to recognise that it has to work with everything else in the production. 'Sound has a great power but it is a conditional power,' he observes. 'It places the image in a physical and emotional context, helping us to decide how to take the image and how it integrates itself into everything else. This goes back to the silent movies, which were never completely silent because there was music and even sound effects. The only thing that was missing was the spoken word. Silence can be a useful tool, although it isn't used that often. One memorable scene that I used it on was the end of *Godfather II* [1974], where Michael Corleone [Al Pacino] is sitting by the lake. I dropped the soundtrack down to the atmos and then shut it down completely, which transmitted the interior emotions being felt by the character.'

After *The Rain People*, Murch and Lucas were among the army of camera operators employed for *Gimme Shelter* (1970). >>>>>

Walter Murch
The sound film man

Immortalised by his soundtrack for *The Conversation*, Walter Murch has a rare understanding of the elements of cinema. **Kevin Hilton** listens carefully

The English Patient

Apocalypse Now and the Conversation are Murch career highlights

<<<<< the concert film of the Rolling Stones' now notorious Altamont gig. 'Everybody automatically assumes that I must have done the sound on that film.' he says. 'but I was just a cameraman. The makers came to Zoetrope Studios to hire cameras. including a 1000mm lens. which I operated. The final shot of the movie—a man taking a piss. silhouetted against the sunrise. which I shot from a great distance so he had no idea—was mine.'

Next came *THX 1132*. a dystopian fantasy developed from a prize-winning short Lucas made at university. As well as designing the sound, Murch received a co-writing credit, cementing his creative partnership with the director. The two went on to make the nostalgic *American Graffiti* but the sound designer was not involved in Lucas' next project, the movie that made his name and established Dolby Stereo as a cinematic tool—*Star Wars* (1977). Part of this is due to being involved in the sprawl that was the production of *Apocalypse Now* (1979) and also two other movies. *The Black Stallion* (eventually released in 1980) and *Julia*.

After *American Graffiti*. Murch worked more with his mentor than his peer. becoming involved in *The Conversation* (1974). which is regarded by many as much as the sound designer's film as Coppola's. The plot revolves around a surveillance expert (Gene Hackman) who, using sophisticated microphones. may or may not have recorded two people discussing a murder plot. Nominated for the Academy Award for Best Sound. Murch has distinct memories of this movie: 'For me it had all the uncertainty of a first film because it was the first one where I both edited and designed the sound. Perhaps because of this there is an emotional edge to it that I can recall to this day. A difficulty was that it is both a character study and a murder mystery and it required a knife edge balance between the two. which are almost contradictory. If you have a murder mystery. the characters are normally subservient to the plot. something that Hitch-

cock was a master of. Ultimately *The Conversation* had to be both and there was struggle in the sound and the editing to find the edge and perch on it.'

MURCH'S NEXT MOVIE for Coppola was the near-hallucinatory *Apocalypse Now*, for which he was Oscar nominated for film editing and won the Award for Best Sound. Many stories are told about this production. some of which are documented in both Eleanor Coppola's diary and the making-of film *Hearts of Darkness* (1991). 'I remember it in terms of being a first.' Murch recalls. 'It

Walter Murch filmography

Touch of Evil (1998, restoration sound and film editing)
Dumbarton Bridge (1998, consulting editor)
The English Patient (1996, sound, film editor)
First Knight (1995, rerecording mixer, film editor)
I Love Trouble (1994, film editor)
Crumb (1994, rerecording mixer)
House of Cards (1993, film editor)
Romeo Is Bleeding (1993, sound rerecordist, film editor)
The Godfather: Part III (1990, film editor)
Ghost (1990, rerecording mixer, film editor)
The Film School Generation (1994, actor) (TV)
Call from Space (1989, film editor)
The Unbearable Lightness of Being (1988, film editor)
Return to Oz (1985, director, cowriter)
Dragonslayer (1981, rerecording mixer)
Apocalypse Now (1979. sound montage, film editor)
Julia (1977, film editor)
The Conversation (1974, sound montage, sound rerecordist, co-film editor)
The Godfather: Part II (1974, sound)
American Graffiti (1973, sound)
The Godfather (1972, production consultant)
THX 1138 (1970, sound montage, cowriter)
Gimme Shelter (1970, cameraman)
The Rain People (1969, sound)

was the first multitrack film I had worked on and it was new territory because it was a multichannel soundtrack with low frequency enhancement. At the time I looked at the way the film was shot and thought to myself, 'Does he [Coppola] really need to do this?' because there was so much else going on. But when I looked at it later. with the big Panavision visuals, I realised that the sound-track we did was the thing to do.'

From the stories of *Apocalypse Now* and the evidence of *Hearts of Darkness*, an image emerges of Coppola as a single-minded dictator. Murch says this is not the case. 'Francis hires heads of departments and collaborators and allows them to experiment. He always gives us plenty of rope, sometimes enough to hang ourselves with. *Hearts of Darkness* emphasised some aspects of his character and the way he works but it was meant to be cinematic as well as just a record of what happened during the making of *Apocalypse Now*.'

Another collaborative director Murch has worked with is Anthony Mingella, whose *The English Patient* was the *Titanic* of the 1997 Oscars, winning Murch his second Best Sound Award and his first Best Film Editing statuette. 'He's one of those very collaborative directors.' he says of Mingella, 'evolving ideas with everyone and not necessarily sticking with his own concepts. His goal seems to be not to dictate but to harness the ideas and talent of others. That's the role of a director, to see that all the different points of view of the craft departments are pointing in the right direction. He has to set in motion a very complex machine and create a unity. The advantage of many points of view is that the work will be multifaceted, rather than monophonic, which is the danger with more dictatorial directors.'

In Murch's most recent project, he indirectly worked with one of the great mavericks of modern cinema. Orson Welles. In restoring *Touch of Evil* (1958). one of the greatest film noir thrillers and yet another Welles film that suffered from studio interference (to the point where Welles was barred from the lot and the movie was finished by another hand), Murch again combined sound and vision. 'I got a phone call out of the blue from the producer,' he recalls. 'They had found the memos Welles had written during production, before he was fired from the movie. These notes are half

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about the sound and what he wanted to do; the other half is about the pictures. I suppose the restorers considered that they needed someone who could both work on the sound and the pictures and that Walter Murch was the man.'

WELLES IS RECOGNISED for his precision but even this reputation did not prepare Murch for what his notes held. 'I was shocked by the detail of the memos and Welles' articulation of what he wanted,' he says. 'Every film should have a document like this as a kind of touchstone but rarely does a director have the time, the ability or the articulation to do it. Welles had all three. Because of this we were loathe to stray from what he had said. If he had been there we could have talked about it but unfortunately he's dead so we held ourselves strictly to what was in the memos, which are not only articulate but also specific. However, the memos did allow a certain degree of interpretation so it was a bit like collaborating with Welles on the work.'

Touch of Evil was loaded reel by reel into an Avid nonlinear editing workstation, where Murch cleaned up the pictures and remixed the audio as an 8-track digital soundtrack. It is generally held that film making has moved away from the old mag recorder methods but Murch says that it is still relatively new for him. 'Last year I remastered the soundtracks of *The Godfather* trilogy using it and I've made a few music videos that way but the first film I worked on using it from start to finish was *The English Patient*. Surprisingly it is also the first film to have been edited electronically and win an Oscar. I would have assumed that this barrier would have been passed before that but it's not the case.'

In terms of the differences between the old and new technologies, Murch says that the primary one is flexibility, with computers bringing more functionality. 'For the sound on *The English Patient* we used the Sonic Solutions system. They have in place something that other manufacturers are only now addressing—a network function that enables people working on the different elements (footsteps, atmos) to access the edit decision list and see what is being done. In this way the two can be played together to see how it will work, without interfering with anyone else. It's very different to how it was before, when you would have to stop people from working to try things out. You also have the ability to have the computer on the stage itself and fix anything that goes wrong there and then.'

Walter Murch's career in movies has been both diverse—in terms of the people and material he has worked with and the different roles he has taken—and highly successful, despite his one excursion into directing, the overly bleak *Wizard of Oz* sequel *Return to Oz*, being a commercial and critical failure. In his work he has shown how audio can be used to enhance and drive along a film and that technology is just a means towards creativity. The main surprise in meeting him is discovering that a man so associated with The Doors song 'The End' was not a fan of Jim Morrison and his chums. Still, that's the movies—nothing is straightforward. Which is as it should be. ■

The investment of over £2m in taking BBC Pebble Mill over to digital working has helped define the Corporation's approach to both technology and its provincial centres. **Kevin Hilton** pays a visit

EVERY NATIONALITY has its version of capital-centricism. where those who live in the first city are dismissive of everything that happens outside its boundaries. In the US, there is a fear of the boonies; in the UK this translates as 'the sticks', or, more politely, the regions. So accepted is this word as a euphemism for 'somewhere we don't know much about, but we know that it's not London' that it is heard in news reports and seen in serious newspapers. But with growing decentralisation, combined with a new-found pride in not coming from London, many areas of life are now proclaiming their independence.

In broadcasting, this has been furthered by the growth in independent production, with many new companies being formed away from the media Mafia haunts of Soho. Regional production centres are becoming seen as more than places to deal with London overflow. Which is probably why BBC Pebble Mill, the Corporation's centre in England's second city, Birmingham, is heavily publicising the opening of its refurbished Studio A, which, after a £2.2m upgrade, is being hailed as the first permanently installed digital widescreen television studio outside the capital.

To the BBC, the outlay displays a commitment to the regions, as underlined by Alan Jones, regional head of BBC Resources: 'This investment represents a major commitment by the BBC to Birmingham and the Midlands as a whole, confirming it as the most significant production base outside London. By employing the very latest technologies, we're ensuring Studio A will be in business well into the next millennium.'

Digital widescreen is the prime concern for many broadcasters, with digital terrestrial television (DTT) due to start in the UK towards the end of this year. This start-point was projected several years ago; accordingly any refurbishment has incorporated the installation of new digital cameras or switchable facilities to cope with both the 4:3 pictures that everyone is accustomed to and the 16:9 aspect ratio of the wider picture.

In this way, a number of studios at Television Centre in West London have already been upgraded to digital widescreen, as have the leading commercial studio complexes in and around the capital, including Teddington, Pearson's Stephen Street operation and the Capital Group. The studio business is equally competitive out-of-town and while Pebble Mill's competitors offer digital equipment, it is hired in, making Studio A the only permanent installation of its type.

Bob Stacey, manager of studio and location resources at Pebble Mill, explains that the coming of DTT would have meant an upgrade at some point, but that the new Studio A is also the result of other >>>>>

Pebble Mill

<<<<< considerations. There was a need to refurbish anyway,' he says. 'we would not have just gone widescreen for widescreen's sake. It's something tied in with the coming of digital broadcasting. There is a delay in the start of digital terrestrial television at the moment, the roll-out has slowed down, otherwise I would have expected us to be producing more widescreen programming than we are at this time.'

Another driving factor was the installation of a digital control system—the BBC's own BNCS (BBC Network Control System)—and a Trilogy Commander digital talkback and intercom facility. Stacey says of BNCS, 'It's a cheap, fully functioned system that can be easily expanded. It's just a case of installing another monitor wherever it's needed.' This ability, which may appear peripheral to other, larger technological questions, was a high priority in the planning of the new production areas. 'There were intense discussions about

where people sit,' says Stacey. 'At the front console in the production gallery there is the vision mixer, with a second mixer if necessary, and then still store operators and sometimes the producer, because some do not like to be stuck at the back of the room. All this is something we had to bear in mind when designing the desks and placing talkback microphones, and so on.'

This flexibility extends to the technical supervisor being able to be positioned either in the production gallery or the adjoining lighting gallery. This is made possible by a portable pod containing all the necessary monitoring, talkback and phone systems. The production control room is centred around a 24-channel Sony 7250 2.5 ME digital vision mixer, with an integrated DME 7000 digital video effects panel. Up to 11,600 stills are stored in a Pinnacle Lightening, with everything monitored on a mixture of Sony and Ikegami monitors.

Stacey comments that in this transition period between the familiar rectangular screen and widescreen, some compromises have to be made. While all the 4:3 monitors are standard professional cathode ray tubes, the 16:9 units are customised domestic sets as manufacturers are not prepared to commit production runs to small numbers of product and are waiting for the coming of new screen technologies.

As some of the monitors used are switchable between the two ratios, there was discussion as how far away from the lighting gallery's operating desk the screens should be placed. The worry was that, while it would be comfortable to see material on the 16:9 units at the usual distance, the smaller 4:3s would be more of a strain. To this end the monitors in this room have been placed in little trucks, specially built by ATG Spectrum (which was been awarded the technical refurbishment contract), that can be trundled backwards and forwards accordingly. Lighting control is through a venerable Rank Stand console, with power over 420 dimmers (plus DMX).

Adjacent to lighting control is the VTR machine room, which houses five DigiBeta machines, with five VHS recorders as backup 'shadows'. The bulk of programmes recorded here are on DigiBeta, but other formats can be brought if required, as is the case with the show Can't Cook, Won't Cook, which is laid back on a mixture of Beta SP and D5. The move to a self-contained VT room enabled Resources to free up space in the post-production area for other equipment, which, comments Stacey, has proved lucrative for that department.

IN THE INCREASING move to digital vision, one area that has retained an analogue character is audio. The refurbishment at Pebble Mill does not change this trend. The new sound control room (SCR) was built in the space previously occupied by a visitors' viewing gallery, a fact, combined with the emphasis on the vision areas, that restricted the size of this area. 'The shape of the sound control room was dictated by the size and shape of the adjoining rooms,' explains Stacey. 'It may be smaller, but it is better than what we had before. We've made the area slightly smaller, which was caused by extending the production gallery. As a consequence the lighting and operating-operations gallery have been moved along.'

The SCR features a voice-over booth, enabling it to be used for a wider range of work than just being a studio recording area. Stacey explains. 'We want to do digital multi-track work here, which is why we've installed the necessary disk drives and built in a huge patch panel where we can wheel in digital multitracks. The sort of work we're looking at includes music programmes, which we have done here. The desk we have gives us the ability to perform a multitrack mix: a big

problem with TV studios is that during turn-around, when the set is being struck, the kit is sitting there doing nothing. This way we can use the SCR as a self-contained facility.'

Dominating the SCR is a 60-channel Calrec Q-series, with 36 mono and 24 stereo channels, a common and enduring site in many TV audio suites. 'It was decided to stick with the existing analogue desk,' Stacey explains. 'While things were moved around and the vision side was replaced, the sound department had only been changed about four to five years ago. We looked at digital desks, but we weren't entirely convinced that digital is good enough at the moment, at least not for the cost of it. It doesn't feel like the time to jump for digital at the moment; that time is probably a year or so down the road. Our opinion is that digital desks are not that much better than a really good analogue at present.'

Another reason for staying with an analogue desk was that the bulk of work passing through Studio A is daytime TV material, either live or recorded as live. In this way an analogue audio desk is seen as the best option. 'Immediacy is very important,' says Stacey. 'Once we abandoned the idea of an assignable console, any thoughts of installing a new desk disappeared.' Despite this, digital has not been completely overlooked in the SRC. A major change in the SCR was the replacement of the traditional BBC grams cabinet with various digital play-in equipment: an Akai DL1000 with its own sub-mixer, CD player, two DAT machines and good old cassette. Radio mics have been built into a wall rack, as opposed to being on a trolley. The acoustics were designed by Tony Woolf of the BBC's in-house engineering department. The wall coverings are mixture of absorbent and reflective material, the latter being perforated metal panels.

In the studio itself, the only major change has been the installation of four Sony BVP-500P heavyweight digital cameras (switchable between 4:3 and 16:9) and two lightweight BVP-550Ps, which were due to be delivered on 8 May. Stacey comments that the bulk of the £2.2m refurbishment budget went on the new equipment, with only a small proportion spent on rewiring. The whole refit took nine weeks and was supervised by BBC Project Management Services, with building and furniture design by ATG Spectrum. In the new commercially aware, profit-based BBC, there is no longer an internal building and furniture design department, meaning that such projects as Studio A are tendered out. As there is only a small number of specialist studio building companies across Europe, competition is tight. Founded four years ago, ATG Spectrum has established a relationship with the BBC, having also worked on the refurbishment of Studio 5 at TV Centre for BBC Sport and Studio A at the Centre for Broadcast Skills Training.

At Pebble Mill, ATG Spectrum designed and installed the new production and lighting galleries, while also upgrading the SCR. All technical furniture was built at the company's factory in Letchworth. With the tight turn-around time, the contractors worked over Christmas and New Year, but Stacey says that this could be justified because the studio kept working during the rebuild. 'Although the galleries were being worked on, we didn't take the studio out of commission,' he says. 'We

Studio Sound May 1998

hired a Type 8 scanner from BBC OBs at Kendal Avenue in London and kept producing programmes. It was economic to pay the builder over the holiday period because the cost of working out of hours was less than that of the scanner.'

This meant that the studio's usual roster of programmes could continue uninterrupted, an advantage given that Pebble Mill largely concentrates on daytime television, which is a voracious devourer of material. Pebble Mill has a long history of day-time TV production, going back to the days before it was called

that and evolved into what we know today. At a time when there were only three terrestrial channels in the UK and two spent much of the day-time off the air, the only concession to people who did not go out to work was the Pebble Mill at One magazine programme.

Today there is a continual sequence of infotainment shows to keep people amused, entertained and informed during daylight hours and BBC Birmingham produces a fair proportion of this programming for the network, including *Can't Cook Won't Cook*, *Call My Bluff*, *Style Challenge*, *Telly Addicts*, *Going for a Song*, *The Entertainment Game* and occasional forays into more serious matters with *Question Time*. The majority of these programmes are produced by the BBC itself, but some are independent productions. It may be a while before viewers see a daytime cookery show in full widescreen, but, as the new Studio A promises, it cannot be that far away. ■

THE SETTING is the high school gym and the coach is putting the girls' basketball team through its paces. As the pretty young things run up and down the court Mr Sadistic is barking orders. These gruelling workouts are known in America as 'suicide drills', and, in a series of US TV commercials, we see the girls progress from the preseason team trials to the practice sessions, the 'suicide drills', the games and, yes, the championship. Still, never mind all of the sweat and skill. Just put it down to that trusty Nike footwear.

Jeff Payne is a sound designer and mixer at post facility POP in Santa Monica, where he works on an average of three commercial spots everyday, sometimes as many as five or six, and it was he who got the assignment relating to the Nike *Suicides* advertisement. Just for the record, the ad agency in this case was Goodby, Silverstein & Partners, the agency's creative director was Jeff Goodby, the production company was Propaganda, Jan O'Malley was the producer, Antony Hoffman the director, with Emily Dennis and Michael Elliot taking care of the editing.

'Initially I met with the people from Nike and the people from Mad River Post who, as the editorial house, are my clients.' Payne explains. 'They came to me with the dialogue from the shoot, and they also brought me music which had been scored by Michael Boyd Music in San Francisco together with some sound effects; mainly crowd noises.

Basically, what I then did was to go back and take some of the effects supplied to me by Mad River Post as well as some of those from Michael Boyd, added a lot of my own stuff, manipulated theirs, and pretty much created our own sound design for it.

'The dialogue that had been recorded while they shot the film mainly consisted of the girls yelling as they went through their drills. So, I took those voices, placed them accordingly and manipulated them by slowing them down, pitch-shifting them and adding all kinds of strange delays, flanges, reverbs and other weird stuff. There were also some wild lines from their coach, and I threw those in at strategic spots. Then I took all of those elements along with the music and ran them through different kinds of processing; the Lexicon, the Eventide and so on. I turned a lot of sounds around, played things in reverse, slowed them down, flipped them around, sampled them, and basically just created a lot of interesting, more mood-provoking noises. Even with the music, I slowed that down and played it in reverse in one section, and added all kinds of weird processing to it. They really wanted a surreal presence to the sound in the middle of the spot. For the most part, however, I was working with the actual dialogue from the shoot and manipulating it.'

Payne asserts that Emily Dennis and Mike Elliot are among the most creative editors that he's worked with, and they were around nearly all of the time when he was doing his own job. 'With a lot of the spots in this Nike campaign we didn't know that we were going to be sound designing them or changing things until we actually got into them,' he says. 'So, we'd end up saying, "Well, wouldn't it be cool if we tried this here or tried this here," and it was a much more collaborative effort than normal, where I just get the effects and mix them. This was a unique spot and I was a lot more involved in the sound design portion of it. That happens, but it's rare.'

The 30-second *Suicides* spot—probably five or six months in production from conception to selling the idea to the client, producing and shooting it—ended up taking Payne between four or five hours to sound design and mix. 'This one took a little bit longer and that's always the case when you're creating effects,' he says. 'That's as opposed to me just receiving them and mixing them, EQing them, panning them and putting some reverbs on or whatever I do. In this case we were doing a lot of sampling and we were finding sounds; real sounds. There were some basketball sounds and foot squeaks that I had to find off the DATs, as well as all kinds of weird things that I had to put in to make it work. I took some elements from sound effects libraries, but I would say that 90% of these sounds came from the actual shoot... The finished spot was a pretty intense 30 seconds.'

The Nike ads aside, among the more complicated spots that Jeff Payne has worked on recently were a couple for Lexus, one called *Capture* and the other titled *Escape*. High production values and surreal effects amounted to plenty of sound design, and Payne was more than up to the task.

'In one of the spots they're chasing the car through some weird circus and there are all of
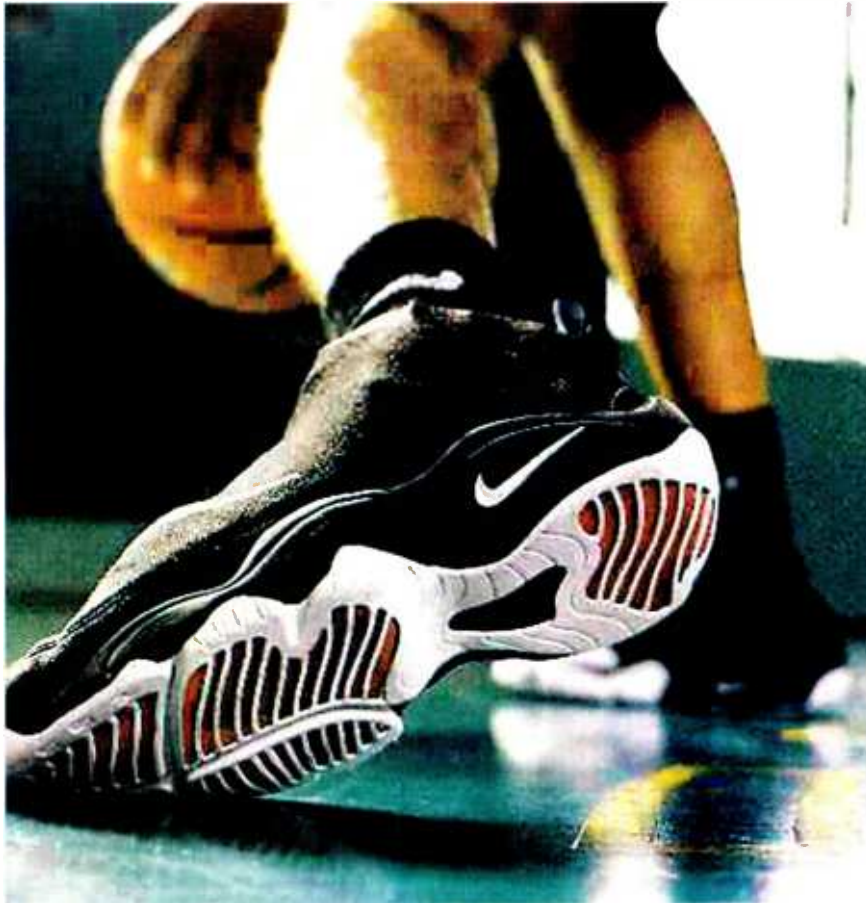
Soft shoe shuffle

With global advertising campaigns come similarly sized budgets and headaches. **Richard Buskin** discusses suicide with sound designer Jeff Payne

these guys throwing knives at it,' he says. 'That was a lot of fun. I got the effects from a company called Primal Scream and the music was done there as well, and I had all of the elements split out. In other words, all of the effects are split out having been prebuilt and prelaid down for me, and with all of the music being split out I've got control over the oboes, the woodwind, the percussion and all that kind of stuff. So, that was a pretty complicated mix, because there were a lot of elements going on, a lot of dialogue that ran through it which needed to be understood, and very dynamic music and sound design. It was a challenge.'

HELPING PAYNE to meet that challenge was an AMS Audiofile together with a 96-input Logic II console, and he is quick to state that he is more than a little appreciative of the state-of-the-art gear that is available to him at POP.

'As far as audio goes, ad people fortunately have always been the first ones to jump on the train,' he says. 'They were the first to really use workstations—way before film and television got into them —and so I was able to get in on that boom ten years ago. Furthermore, the production values have gone up and up, they're spending so much more money on commercials now—it's not uncommon for a spot to cost a million dollars plus—and the great thing is that we get to work with people who are the tops in their field, whether they're composers or sound designers. They're real together. They've been doing this for a while and they really know what's going on, so that's the nice part about doing commercials.

'The Logic is plenty large and now that I've been using it for five years it's almost like playing an instrument. I mean, you learn the subtleties of the console, you learn how to do things quickly, and it's just a really nice working surface. You see, a commercial's got so many quick cuts that in 30s you've got track after track after track of effects, and that's another benefit of having automation. I don't have time to lay them out like a film mixer might, laying out all of these split tracks onto different channels. I'm not dragging three 24-tracks with me. I may have 15 to 20 frames between the sound effects that the editors have supplied me with, and because I'm using automation I can go in and completely change my EQ, my aux sends and everything else on one channel 15 different times. That makes a big difference.

'The technology has changed so much over the years and that's given people more options. As a result they don't necessarily finish their sessions any quicker, but they do tend to try to do more during those sessions. Likewise, the clients' expectations have become greater, quite a few people have systems where they can listen in Dolby surround at home, we do some spots in surround, and so they're more sophisticated and used to hearing better quality stuff.'

But while a few people may have surround systems at home the majority still don't, and Payne therefore has to bear that in mind when it comes to monitoring the commercials that he's working on.

'I only really use my large monitors for my EQs and for getting my basic levels,' he says. 'Almost all of my mixing is done with the Auratones and then final approval is always on a TV set. I run everything back through a tiny little speaker on a little Sony monitor, and it's kind of sad and it's unfortunate, but those are the limitations of what I do. I sometimes get to mix stuff for theatrical release, but in the main it's got to sound great on a really shitty TV. Ultimately that'll change and I'm sure that the ad people will roll with the punches, but in the meantime the advantage is that, listening to this crappy TV at low levels, I'm still going to have my hearing in ten years! The other mixers will all be deaf!'

Being that Jeff Payne only sound designs about 20% of the time—normally he mixes things that have already been presound designed—he usually pulls material out of effects libraries. At the same time, having learned the ins and outs of his profession over the past dozen years, he is more than familiar with what does and doesn't work, and so he will often revert to the tried and the trusted.

'I'll know that if I manipulate something in a certain way, play it in reverse or change the pitch and slow it down it will give me this effect,' he says. 'That just comes from experience.'

However, what with all of that experience, derived from working on so very many different commercial spots, isn't it difficult to retain interest and objectivity on the job?

'No, actually I think it's almost the reverse,' Payne replies. 'The nice thing about it is that the turnaround is so quick. You work on something and then it's gone, and in the afternoon you're working on something else, and the challenge is to be able to focus and to give the clients what they're looking for while also trying to take it in the direction that you feel it needs to go in. That's the challenging part and that's the part I like about it. It's fast paced, it moves, and in my end of the business people tend to come very prepared.

'Like, for instance, this Nike job. It wasn't that these people weren't prepared when they came in to the session, but they just wanted to be creative in the session. A lot of times they come in and they basically have laid everything out, and you're set to go and ready to mix, and your job is to give them what >>>>>

<<<< they're hearing in their heads. Don't forget, by the time they get to me they've probably spent a week with the sound designer and two weeks with the composer, and they've really had a lot of time with this. On the other hand, I'm seeing it for the first time and then I'm mixing it as I'm looking at it. So, I really thrive on the pace of things: I've got to learn it fast, I've got to quickly get a feel for what they want and then deliver it before the session's done.

'What makes the whole thing a little scary is the fact that, by the time it gets to me, it's the end of the line. This is the last process: dub 'n' ship time. There are many, many times when I'm finishing and the spot's going out on a satellite to New York at 8 o'clock tonight. That seems to happen more often on the higher-end projects, and I can't tell you how many times I'm trying to get it over to FedEx by 8 o'clock so that they can get it on the air. They therefore don't have the time to make any changes, and so it's got to be right by the time it comes to me. I remember, when I first got into the business that was the part that had me thinking, "Whoa! Hold on! I'm not that bright!"'

Why work to that kind of deadline with the devil?

'Well, I've asked my clients that, and I think the reason's simply down to economics. They're giving them shorter deadlines now and they have less time to post. Whereas normally they would have four weeks to do all of their post work they're squeezing that down to two weeks and they're squeezing their shoots down to less days. That way they can sometimes spend less money, and I'm really, really noticing a difference in that. Another thing that I've been noticing more and more during probably the past year and a half is that I'm finishing spots before the picture is finished. So concurrently, while I'm doing the final mix, they're bringing me the work picture, and I may or may not lay back to the final picture at the end of the session. It may not even get to me by the time I've finished my mix, so they may lay it back somewhere else at a later date or they'll get me the final picture right as I'm finishing and I'm seeing that when I lay it back. It's crazy!' ■

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Mike Hedges

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Bauer power

Maintaining a place among Germany's top studios is a demanding task, requiring an understanding of many dynamic markets and technologies. **Tim Goodyer** visits a pace setter

'AT A TIME when many people speak about returning to core business, we are extending our core business. That means knowing what the core business is—we know we're audio people, we will stay audio people, but we will extend this core to all the needs of the people.' The speaker is Reiner Opelland, one of the two managing directors of Stuttgart's Bauer studios which is today celebrating the opening of its revamped main studio. The event sees its formally embrace 7.1-channel surround formats with new acoustics, desk, and monitors. The move sits comfortably with Opelland's claim, as do the studio's Foley-equipped second room, which serves smaller sessions, post, and radio plays and commercials; its three Sonic Solutions mastering rooms; its four record labels and a new venture addressing the emergent world of multimedia. Towards the end of the year, Bauer Studios will have another celebration to host: its 50th anniversary.

The story of Tonstudio Bauer—as it was previously known—begins with the grandfather of Opelland's wife and comanaging director, who opened the radio shop that inspired her father to begin rebuilding tape machines for South German radio after the second world war.

'My father started with a colleague after 1945,' Eva Bauer-Opelland confirms. They had to reconstruct types of tape machines and after three or four years he said, "No, that's not my job; I will work with these machines". So he started a studio in our living room. Then he moved to two rooms in town, and then we moved again. And in Autumn 1960 we moved to this building which used to be an old cinema.'

Opelland's involvement in the studio began when the couple met at a Tonmeistertagung in 1972—although the exact sequence of succeeding events is light-heartedly contested between the two. Regardless, he began a formal technical education in Berlin and followed it with freelance work in Munich (including sessions with Eberhard Schöener which saw him making the first recordings of Sting as a vocalist), while she studied sound engineering in Dusseldorf, making both partners equally capable of dealing with the sharp end of the studio's activities.

'Even when I was a child I worked at the studio,' Bauer-Opelland recounts, 'so it was certain that one day I would come back here and take it over. It was such interesting work —I started as a kid to cut tape and to set up microphones. It was always an interesting job and an interesting business.'

'I used to say I need one recording session per year to maintain my membership status in the Tonmeister association,' Opelland continues. 'There are some kinds of recording which we do. We don't have any problem to set up a good microphone set and have the musicians

play. This is one thing we still like to do.'

The design of the new control room began with proposals from Bauer's house staff which were taken up by Berlin-based studio designer Karlheinz Stegmaier. The result is a larger, irregularly-shaped modern room capable of accommodating both the AMS Neve VXS console and the Genelec monitoring system, the new console deposing a Cadac desk that had endeared itself to its users and will leave a number of EQs behind when it is sold.

'Film sound mixing was why we planned to develop the control room and to buy the new console,' states Bauer-Opelland. 'That was also one of the main reasons for having the "S" option—to go into scoring—and also, because of our three Sonic mastering suites, we know the plans for DVD and digital radio, and so we also think that for good sound mixes in surround it's necessary to have such a console.'

'Deciding the specification of the new console was easy in a way,' Opelland explains. 'We

knew that 60 channels would accommodate all the situations we could imagine for the future. The Cadac console, in comparison, had 36 channels equipped with six spares not equipped. In combination with the 48-track Studer DASH machines, we found that 48 channels would be a minimum, so at first we thought of the 48-60 specification and we though we should have 60 channel.. I think this is enough; I know that the larger film studios are using larger versions of the VXS and the digital counterpart, that would raise the question of whether a digital console works better.

'It was possible to do surround sessions with the Cadac but it was a matter of how much time and work you wanted to invest. The Cadac console, on the other hand, was a great console which made it necessary to find the "right" console to replace it. Sonically, the Cadac is still on top, maybe. I can't make a direct comparison. Obviously, the sonic quality of the Cadac, including Clive Green's philosophy of having as few audio stages as possible and on-phase linearity.

These are some of the things that made his consoles so successful. Some people think that Cadac is exotic—maybe they are right—but this meant that we would have to find another in this way exotic, good-sounding console.'

'And there was something one of our producers said,' adds Bauer-Opelland. 'A good analogue console won't be old-fashioned in ten years, but what about a digital console?'.

'We had a deep look into the market,' Opelland continues, 'and we found that up until last year, all the digital consoles have not been installed in Studio A, but in Studio B. Nobody that far had dared to have a digital console in Studio A. We would have liked to have, for instance, the Sony console, but what would have happened after five years?

ALL THE MONITORS are Genelecs—but contrary to current thinking, they are not all identical—the LR pair is 1039As, the centre channel is a 1038A, and the four surrounds are 1030As.

'They are all from the same family and I think that is important,' Opelland contends. 'Normally the Dolby rule is that all the loudspeakers should be the same model, but we talked to the German Dolby representative and agreed to modify it so that it's the same manufacturer and it's adjusted to sound equal enough.

'Choosing monitoring is always a very difficult part. We did long tests and it was a hard decision because the general approach to testing speakers only makes audible the change from one speaker system to the next; it doesn't make audible the real difference. It would need at least one month of experience with one system and then rebuild the studio with another system for one more month. So one fine evening we had to sit down and order some speakers... For the kind of music we are recording, we still need natural sounding speakers—to reproduce chamber music, a big band and also pop music. It's a very broad range.

'Also, if you think about the people who come in, sometimes you have to explain why you have chosen the speakers, and so we thought it would be easier to have the Genelec speakers because they are a kind of world standard at the moment. There are good things and bad things about any loudspeaker, of course. We feel that Genelec is a good choice at the moment because they represent these kinds of music very well. We had a fine tuning last week after we had enough experience of them, and I think everybody is content now.'

Bauer Studios' history is as intriguing as it is unique. During its formative days it helped define, as well as record, the sound of traditional German brass music. Balancing this expertise with that for jazz and classical music, the studio's forte became acoustic recording in general—a fact supported by the wealth of photographs adorning its walls. But all of this required careful management when electronic music came to the fore.

'You don't need a big studio with trained tonmeisters for a MIDI production,' Bauer-Opelland concedes, 'so our main work was and is still to record acoustic music. The musicians know what they want to play, so they come into the studio and typically we can finish a pro-

duction within one, two, four days...

A recent stage production of *Beauty and the Beast* became the latest in a series of high-profile 'soundtrack' recordings undertaken in association with a nearby theatre. On this occasion, with the entire cast occupying both studio rooms, it saw Bruce Botnik seated behind the new desk. But the expertise built up in acoustic recording should not detract from Bauer's interest in ongoing developments like the imminent fusion of the South and South-West German state radio stations and the take-up of digital radio, and the maturing of multimedia—for which a new venture, Gadget, has been set up.

'We realised that you need specialists,' Bauer-Opelland begins, 'and our sound engineers are specialists in their field but they are not interested in computer animation. Other guys are really interested in graphics and start projects with the video and then they say, "Sound—we can do that too". And that's not good. When you have a combination of specialists you have much better quality in both parts.'

'If you consider audio to be one piece of a pie chart,' Opelland elaborates, 'you end up with the question: "What do you want to offer in the future—one piece, or all the pieces?". We decided that we would like to have the complete thing so we started a multimedia production in conjunction with a brass music association who wanted to use the project for their 75th anniversary. These guys said, "We want to found our own company," and I said, "Okay, let's do it together".'

'We also hope that we can give our part to the radio scene,' says Bauer-Opelland. 'At the moment, all over the world, there is less money so they must source out, so we hope that the fusion of the two big radio stations in South Germany will give a push to the major scene —in Stuttgart we have government sponsorship to get more film and major work into Southern Germany.'

Part of the studio's future planning lies with the past, however, and while most new recording projects use a 48-track Studer DASH machine and Pro Tools system, or a 20-bit DA-88 Prism Sound setup used for location work, older formats are also available.

'We feel that we are specialists in all parts of audio,' Bauer-Opelland asserts. 'We can still reproduce shellac records, we've recorded mono, stereo, we had one the first 3M digital multitracks, we were one of the first people in Germany recording with PCM, and we're still very busy in mastering and restoring using NoNoise, and doing location recording We have a Mitsubishi MX850 and an Otari DTR900, so we can still handle all the main digital formats. It quite often happens that we get tapes to copy and other enquiries about what standards we have.'

Although the facility itself is worthy of its growing reputation, coming away from Bauer Studios, you are left with the feeling that its physical presence is quite outweighed by the thinking behind it. The astute management of old achievements and new opportunities overshadows the admittedly impressive new control room—an object lesson for a time when it is easy to believe that the route to success lies simply in following technical developments. ■

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Penthouse Suite

The first client through the doors of the Penthouse Studio in 1998 discovered that it had been completely rebuilt and represents the latest turn in Abbey Road's recording philosophy. **Tim Goodyer** accepted an exclusive invitation to find out more

WHEN ABBEY ROAD technical operations manager, Neil Aldridge, talks about his studio's 'duty' to help develop the audio business, you know that it is no sales pitch. His sincerity is evident from his account of the downside of being the world's most famous recording studio—that everything you do is judged in terms of its taking place in 'the Beatles' recording studio'. His frustration is implicit in Kanji graffiti on the front wall declaring, as it ever did, the loss of John and the love of Paul. So as Abbey Road opens its refurbished and surround-equipped Penthouse Studio for business, it is worth remembering that, while Studio 1 maintains the orchestral tradition set by Sir Edward Elgar on its opening, Studio 3 continues to build on its involvement with Pink Floyd's *Dark Side of the Moon*, and Studio 2 retains the mystique of the Fab Four, it is now just over two years since Abbey Road opened the world's first multimedia suite.

The new-look Penthouse studio hosted its first session on 18th February, having been closed for its refurbishment just three months earlier. Apart from looking more inviting than it had previously, it has enjoyed an acoustic rework by Sam Toyashima's Acoustic Design Consultancy, and now offers a scoring panel on its Capricorn console, discrete B&W-Chord 5.1 surround monitoring, and greatly improved patching facilities. All these aspects played their part in prompting the refurbishment. The day-light that streams in through windows and sky-lights converted from ventilation ducts can be blocked for those still enamoured of the traditional 'studio tan', while the removal of the sof-fit-mounted Quested monitors has seen the front wall recede by several feet. In place of the Questeds stand three distinctive new B&W 801s, with a further pair toward the rear of the room. More discrete are the comprehensive patching panels that appear throughout.

'We took it right down to the floating floor,' explains Abbey Road senior technical engineer Colin Johnson of the extent of the project. 'One of the problems with the old room was that a busy session quickly became a mess with wiring all over the place.' Devised by Johnson, the room now offers comprehensive patching for ana-logue and digital audio signals, time code, MIDI, and has ties linking it to the rest of building including the multimedia suite. Wiring has been provided for a projector and screen system (up to, and including, selection of the hardware), and the installation of a Sonic Solutions Medi-aNet system is also under consideration.

'The trouble was that we had to build the stu-dio physically somewhere else because there wasn't enough floor area to do it here,' explains Johnson. 'Then we brought it in and reassem-bled it. It was quite strange really, being able to walk into this room and say "oh yeah, this goes here..." and be somewhere else.'

Once on site, the technical installation also went well: 'We just worked to Colin's spread-sheet,' confirms Kelsey's Michael Whiteside, the man responsible for the wiring job. 'It all went very smoothly.'

The only elements of the old studio to make it into the new one without challenge were the outboard and the Capricorn—which took a return trip to Burnley for an overhaul and the fitting of the scoring panel. 'We took it back to the factory and gave it a new coat of paint and some coloured knobs,' confirms AMS Neve's Paul Berry. 'We also went through all the racks with a fine-tooth comb and checked it was within specification, replaced anything that was-n't. Since this desk was installed, there have been various modifications both in hardware and software, so everything's been brought up to date with the developments of the last five years.

'This is the first one we've had back to the factory,' he continues. 'Normally we do our modifications on site. It wasn't a case of "let's get it back and patch it up", but it gave us the chance to go through the whole desk and accep-tance again and make sure everything was up to spec. We replaced a few mic-line cards, but we didn't find anything we didn't expect to.'

But if the physical aspects of the work were straightforward, the logistics were not.

'We were expecting a quiet period over Christmas,' Johnson recalls, 'but we got a big film job book into Studio 1 and that was it. The noise from here just went everywhere.'

Abbey Road's location in a residential area means that it is considerably restricted in terms of noise that can be heard outside the build-ing—as could the diamond drill cutting neces-sary to complete the Penthouse. The situation has, however, recently eased somewhat.

'She moved,' says Johnson referring to one res-ident who had regularly complained at the stu-dio's activities for a period of years. 'The thing is, this was a recording studio before she moved in, but it was still our fault. But it's a lot easier now.'

Apparently, the owners of a London restau-rant are now the subject of her complaints.

THE PENTHOUSE'S new B&W 801 moni-tors are powered by new Chord 1424 monoblock power amps and are signifi-cant for a number of reasons. Still the subject of an embargo at the time of writing, they delayed the studio's official opening party until after their launch on 1st May. For the studio, however, they represent a serious venture into discrete surround monitoring.

'It is a move by us to be in at the beginning and to some degree trying to force the issue along,' Aldridge confirms. 'We'll do a presenta-tion to our producers and potential client to show them what can be done with surround sound and show them the way it's going to go. Even if they can't release on a mainstream car-rier right now, I believe there will be a market in maybe five years or so time.'

But while Aldridge is in no doubt about the future of surround, the present presents serious problems.

'I spent a good part of my time when I was at LA and New York AES Conventions talking to everyone I could find who might have an opin-

Capricorn Scoring Panel

THE NEW SCORING PANEL on Abbey Road's Capricorn is the first installation of the new facil-ity, and represents a logical extension of the console's capability. Conceived as recently as autumn 1987, the panel follows AMS Neve policy of using proven technology as opposed to developing new technology where established technology exists. In this case, although it is the first 8-way panel the company has produced, it is based on the 6-way panel already found on the V-series console and displaces an earlier scoring panel that replaced the desk's machine control buttons.

The new panel offers full 8-way assignation complete with PEC-Direct switching. Operationally it is quite comprehensive, including various monitoring options, inputs for play-back and dialogue, and numerous inserts including those necessary to accommodate a Dolby encoder.

ion to see if our way of thinking was the same as theirs,' he continues. 'And the reassuring thing was that nobody else seemed to know either.'

The issue is largely one of speaker placement, but is complicated by the traditions of stereo music recording and soffit-mounted speakers.

'In the early days we were a bit hesitant about the speakers,' Aldridge admits. 'We really didn't know if we should have some built in to cater for the pop side, but once we'd had a look at the 801 prototypes in Studio 1 at the end of November we were fairly confident that we could use these for everything. And I'm very keen to do that because my impression is that there is no agreed location for surround speakers.

'We've done surround sound work for quite a number of years now, but we thought long and hard about this and we couldn't quite get to grips with where to put the speakers. We were going with a design that had built-in front monitors, but the longer I thought about it, the more there didn't seem to be one right answer.

'Quite a lot of clients—whether it's the producer or the composer—have a strong opinion on where the speakers should be, but so many of them are so different from each other that

we thought the best thing to do would be to have a free-standing system—and the new 801s turned up. We tried them downstairs and as soon as we heard them it was "great, this is going to work".

'Everyone who has heard them has liked them instantly,' Johnson agrees.

'That's why they're standing on a moveable plinth,' Aldridge continues. 'This way we can have the standard 60° stereo spacing at the front with an additional centre speaker or, as the DTS guys suggested, we can have them at 90°—in a square. We can put them there, no problem. The rear speakers can be 60° to the rear, 90° to the rear, or at 210°, almost to the side. We can accommodate all of that. Many of our major film recordings have been done with the rear speakers to the side, so I think that's the one that will catch on as anything like a standard, especially for DVD-style work because it's akin to what goes on in the cinema and in most homes.

'Basically we've drawn a circle around the engineer to time align the speakers and the

room has been designed to accommodate any formation without looking wrong. And for anyone doing a stereo job up there we can just take the additional speakers out and the plinths become coffee tables. So a fair amount of thought has gone into how to accommodate surround.'

A similar system has been adopted in Studio 1, but the Penthouse will benefit from the use of carpet markings of some sort to assist placement and repeatability. 'It might be triangles set into the carpet.' Aldridge explains, 'or it can be integrated into floor design. It could even be brass screws. We just use a Chinagraph in Studio 1...'

The embargo on the monitors may have delayed the official opening of the revised Penthouse, but it hasn't prevented it from receiving its first clients—regardless of a gentle hike in the room rate. In fact, the rate wasn't even discussed when the first booking was made. And—Beatles or no Beatles—that fact alone sets Abbey Road aside from many studios today. ■

Microphone principles: 2

Microphones are often taken for granted, but knowing their limitations is an important step towards using them properly. **John Watkinson** tries to explain

LAST MONTH we discussed how the most important characteristic of a microphone is its directivity pattern as this has the greatest bearing on what it sounds like in practice. In my view, differences due to directivity are more marked than differences due to operating principle. In other words there will be more difference between a cardioid and an omni than, say, between a moving coil and a condenser. This is why the operating principle will be treated in a later piece.

The ideal design directivity of a microphone is only achieved at frequencies where the dimensions of the diaphragm do not form a significant part of the wavelength. In practice, at high frequencies this will not be the case and the actual polar diagram will change as diffraction becoming significant. Fig.1a shows the result of a high frequency sound arriving off-axis at a large diaphragm. It will be clear that at different parts of the diaphragm the sound has a different phase and that in an extreme case cancellation will occur, reducing the output significantly. Essentially this is an aperture effect. The diaphragm is not sampling the sound wave at one point, but over a finite distance.

When the sound is even further off axis, shading will occur. Consequently at high frequency the polar diagram of a nominally omni microphone may look something like that shown in Fig.1b. The HF polar diagram of an eight may resemble Fig.1c. Note the narrowing of the response such that proper reproduction of high frequencies is only achieved when the source is close to the axis. For sound sources which are off-axis, a drooping frequency response will be obtained.

Strange frequency responses may be chosen for creative purposes, but this should be done in an equaliser. The frequency response of a microphone should ideally be flat. This is often tested on-axis in anechoic conditions. However, in practical use the surroundings will often be reverberant and this will change the response at high frequencies

because the directivity is not independent of frequency. Consequently, a microphone that is flat on axis but that has a directivity pattern that narrows with frequency will sound dull in use. Conversely, a microphone that has been equalised flat in reverberant surroundings may appear too bright to an on-axis source. Pressure microphones being omnidirectional are the worst offenders in this department because shading makes it impossible to maintain the omnidirectional response at high frequencies. Clearly an omni based on two opposing cardioids will be better at high frequencies than a single pressure capsule.

The magnitude of the aperture effect can be reduced by making the microphone diaphragm smaller but this results in smaller signals. The penalty may be that the signal-to-noise ratio becomes worse. However, developments in low-noise circuitry will allow diaphragm size beneficially to be reduced. It should be noted that in reverberant conditions pressure and pressure-gradient microphones can give widely differing results. For example where standing waves are encountered, a pressure microphone positioned at a pressure node would give an increased output whereas a pressure gradient microphone in the same place would give a reduced output. The effect plays havoc with the polar diagram of a cardioid microphone.

The proximity effect should always be considered when placing microphones. As explained in an earlier piece, proximity effect causes an emphasis of low frequencies when a PG microphone is too close to a source. A true PG microphone—such as an eight—will suffer the most, whereas a cardioid will have 6db less proximity effect because half of the signal comes from an omni response. In the case of the shotgun microphone, the tube will become acoustically small at low frequencies and it will become ineffective causing the polar diagram to widen. If high directivity is required down to low frequencies the assembly must be made extremely long. The parabolic reflector microphone has the

same characteristics, needing something like Jodrell Bank to be directional over the full frequency range. Fortunately, directional microphones are not normally used for full frequency range sources; the goal is normally dialogue capture or birdsong. In these cases, the addition of a high-pass filter is beneficial as it removes low frequencies without affecting the quality of, for example, speech.

THE ELECTRICAL OUTPUT from a microphone can suffer from distortion with very loud signals or from noise with very quiet signals. In passive microphones distortion will be due to the linear travel of the diaphragm being exceeded, and things have to be pretty dire before this happens. In active microphones there is the additional possibility of the circuitry being unable to handle large amplitude signals. Generally a

maximum SPL will be quoted at which a microphone produces more than 0.5% THD.

Achieving low noise in practice requires many issues to be addressed. Electrical noise may be due to thermal effects in the transducer itself and in the circuitry. Microphone noise is generally quoted as the SPL which would produce the same level as the noise. The figure is usually derived for the noise after A weighting. The difference between the 0.5% distortion SPL and the self-noise SPL is the dynamic range of the microphone. A figure of 110dB is considered good, but some units reach an exceptional 130dB.

In addition to thermal noise, microphones may also pick up unwanted signals and hum fields from video monitors, lighting dimmers and radio transmissions. Considering the low signal levels involved, microphones have to be well designed to reject this >>>>>

a) Aperture effect due to off-axis sound causes H.F. loss.

b) OMNI at LF / OMNI at HF

c) 8 at LF / 8 at HF
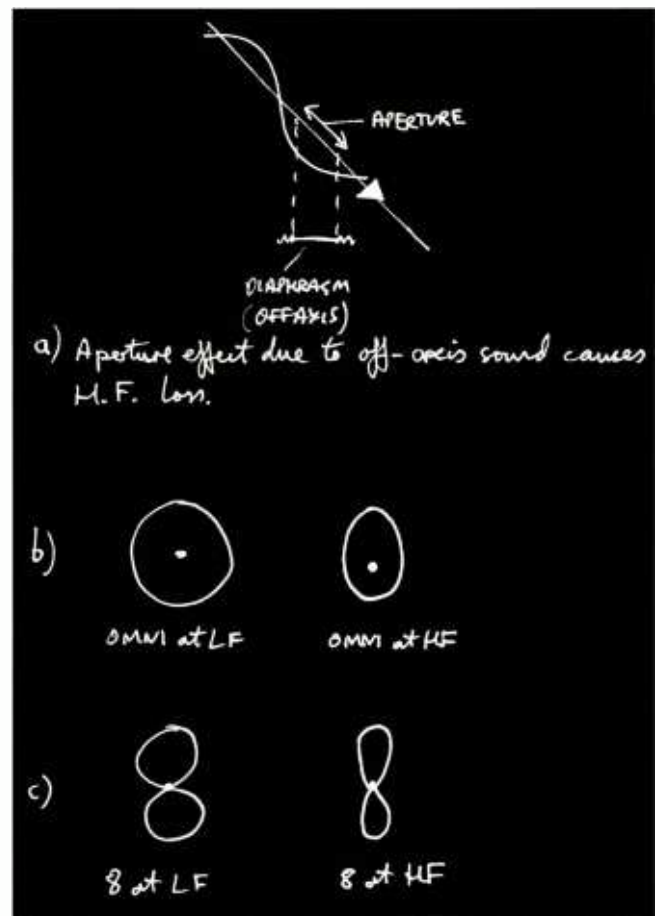
Fig.1:
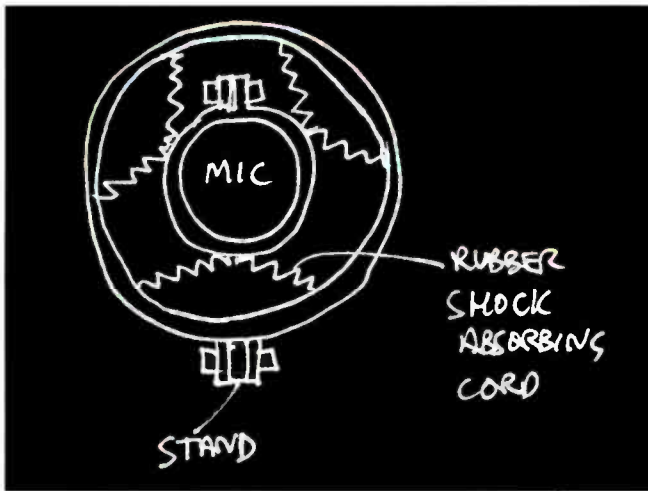
Fig.2: Handling noise is effectively reduced by rubber cord mounting. Looks bad but very effective

<<<<< kind of interference. The use of metal bodies and grilles is common to provide good RF screening.

If any external vibrations reach a microphone they can cause relative movement between the diaphragm and the capsule body which will result in spurious output signals known as 'handling noise'. Goosenecks do not absorb vibration and the rubber jaws of a normal clamp serve only to protect the finish. Almost all microphones contain some flexible elements between the capsule and the body, but there is generally not much space for them and they are of limited effectiveness. For high quality work, many manufacturers supply an external flexible mount such as that shown in Fig.2. The microphone is clamped near its centre of mass to the inner part of the mount, and the outer part is fixed to whatever stand or support is being used. The two parts are joined by rubber cords that allow effective decoupling. It is important that the vibration isolation of the mount is not bypassed by a taut cable. The cable should be fixed to the stand and an isolation loop should be formed between the stand and the microphone.

Noise can also be due to wind and plosives from close speaking. Pressure gradient microphones are more sensitive than pressure microphones because pressure gradients can be caused when the body of the microphone obstructs the airflow. Wind generally causes turbulence as it flows around the microphone and the amplitude can be extremely high. There is a distinct possibility of overloading the associated amplifiers. This is called 'blasting' and the sound is actually interrupted as the amplifier recovers from driving its output voltage to the power rail. Turbulence is usually restricted to low frequencies and a high-pass filter is useful but such a filter will not prevent overloading of any amplifier stages that precede it.

The only practical solution is to enclose the microphone in a specially made windshield. Turbulence will still take place, but it will be around the windshield instead. The windshield is generally cylindrical and consists of a metal or plastic grid supporting a porous material such as foam which damps the low-frequency noise due to turbulence but allows through the lower amplitudes in the audio spectrum. In extreme conditions the turbulence may be reduced by fitting the shield with an outside layer of long piled textile material. These windshields are often called Dougals because of the resemblance to the character in the British children's television programme *Magic Roundabout*.

Windshields always cause some loss of high frequencies. The effect of plosives can be reduced with less drastic measures. Hand-held microphones can be fitted with a spherical foam ball or a light metal frame carrying a layer of silk or similar material can be placed in front of the microphone. In extreme cases the vocalist can be given a microphone to eat but the recording is actually made with a second microphone further away.

The output voltage for a given SPL is called the sensitivity. The specification of sensitivity is subject to as much variation as the mounting screw thread. Some data sheets quote output voltage for 1 pascal, some for 0.1 pascal. Sometimes the output level is quoted in dBV, sometimes dBu. The outcome is that in practice preamplifier manufacturers provide a phenomenal range of gain on microphone inputs and the user simply turns up the gain to get a reasonable level. ∎

US: Three cheers for the middle class

Neither dead nor forgotten, but certainly mislayed for a while the American middle class may yet be the saviour of audio excellence writes **Dan Daley**

THE DAY OF TAX RECKONING for Americans is here. And on the 15th of June, as on the Ides of September and again the following January, the Internal Revenue Service expects those of us running businesses and being putatively self-employed to send them more money that we think we should and less than they would like to get, with the perversely humane notion that paying taxes quarterly takes the sting out of the process, and also makes sure you haven't blown your wad by the end of the year before they can get to it.

Tax reform has been in the air here since congressional hearings on how the IRS handles its charges were held last year, which elicited apologies from the agency to the families of those whom it hounded, literally to death in some instances, and some vague promises of reform that were quickly forgotten once the President of the United States dropped his pants for the media again.

But tax time is a time for politicians to wheel out their rhetorical cannons and aim them squarely at what passes for the middle class in the States. No-one quite knows who constitutes the middle class here anymore, since families making $35,000 a year are applying for government-subsidised food stamps and millionaires are being made on an hourly basis on Wall Street. No one knows who its constituents are, but the term 'middle class' has a nice ring to it.

But things are looking up for the studio community here. I was recently talking with a UK engineer, an expatriate now living and loving it in LA (something about beaches), and one of the things he waxed on about was the plethora of mid-sized, affordable, decently equipped, run and maintained studios in the US. He's right—once you stop slathering over the palaces and fretting over the projects, it becomes apparent that quite a number of studios that fall into neither category have cropped up hereabouts in recent years. These are not converted garages with some plug-in gear from Sam Ash's pro-audio sales division. They have real consoles and tape machines and the odd digital audio workstation or two, some decent outboard gear (including the one or two prized possessions, such as a vintage tube microphone

that is broken out separately on the rate card), and—something in increasingly short supply in the studio business in general these days—a small but intense core of optimism dressed with an almost boundless layer of enthusiasm.

This is the generation that started out with personal recording equipment a decade or so ago. They found themselves in the studio business by design or not—more than you might think woke up one day and found that they had bought so much gear that they had to start letting it out to cover the loans, and that leads to building a lounge, and once you've got a lounge, you're a regular studio. But either way, they're here *en masse*.

What happened? The scenario is, I think, similar to the one that prompted the proliferation of good, affordable Italian restaurants all over Manhattan in the last couple of years. I can't walk out of my flat there and swing a dead power supply without hitting a very decent *Frutta di Mare con fusilli* priced at less than six quid. Even within the seemingly boundless money pits that are New York in the Wall Street testosterone rush of the late nineties, a few people realised the huge,

Europe: DSP on delivery

Next generation delivery formats are set to complicate issues of quality with consumer DSP writes **Barry Fox**

NEW FORMATS, new opportunities. Or not, as the case may be. The DVD Forum has come up with an interesting idea to make DVD-Audio more widely attractive. Many consumer hi-fi systems already use digital signal processing to add delays, echoes and phase shifts to make a small living room sound like a large hall. But DSP doctoring has so far been a hit-and-miss affair because different manufacturers use different circuitry, and different listeners pick different settings from a menu of generic acoustic. DVD-Audio will use Smart Content. Here, the recording engineer can analyse the natural acoustic of the recording venue, and store it as coded data on the disc along with the music. When the disc is played the data automatically sets the DSP circuitry to an exact match. The system should also be able to calibrate the room to subtract the existing acoustic from the acoustic of the recording venue.

The DVD Forum in Japan will publish the final standard for DVD-Audio and Smart Content in June. Six companies—JVC, Panasonic, Pioneer, Samsung, Toshiba and Warner—are so keen on the idea that they want to launch players in 1999. Meanwhile digital television is being sold on the promise of 'better pic-

tures and sound'. Although many satellite services are already delivering digital programming, the UK looks likely to lead Europe, and the world, with digital terrestrial TV. The BBC and commercial broadcasters are all due to start broadcasting in the autumn, using the European DVB standard for MPEG2 video and audio.

Although the DVB standard is broad enough for HDTV and multichannel surround, this is not what viewers will get. The British government licensed digital terrestrial transmission on six 'multiplexes'. Each is in the UHF band, with the same bandwidth (8MHz) currently used to carry a single analogue TV channel with Nicam digital stereo sound (often with an embedded Dolby surround matrix). The multiplexes are on so-called 'taboo' channels—UHF frequencies that cannot be used for analogue broadcasting because they would interfere with other analogue transmissions, elsewhere in Europe

Lack of international clearance for the taboos means that only 55% of UK households will be able to receive terrestrial TV from terrestrial transmitters. The rest will have to use satellite dishes to pick up terrestrial programmes simulcast from satellite. Coastal areas, especially in the South of England, are

least likely to get terrestrial TV. Each multiplex has a data capacity of around 24Mbits/second. With current compression technology, this is enough to carry five or six TV programmes, depending on their content (movie material, talking heads or fast-action sports).

Commercial broadcaster BDB, British Digital Broadcasting, will start broadcasting with 15 channels, spread over its three 24Mbits/s multiplexes. The BBC may use more bits per programme, but because the prime object of the broadcasters is to make money, there is little likelihood of anyone introducing the HDTV option—with one programme occupying a full multiplex (20Mb/s for high quality

Although many satellite services are already delivering digital programming, the UK looks likely to lead with digital terrestrial TV. The BBC and commercial broadcasters are all due to start broadcasting in the autumn

progressive scan pictures and multichannel surround for HD sets and 6-channel sound systems, and 4Mb/s standard quality simulcast for conventional sets). Much more likely they will cram more channels into the available space.

BDB is already promising that advances in

pent-up demand for some good Italian—not necessarily great but above average—and relatively cheap. What that observation has done for gastronomy in New York, it is doing for the middle class of recording studios in America.

The rush of personal recording raised consciousness and empowerment of musicians, turning them into engineers, producers and recording artists without having to deal with the usual, dead-end routes of major record labels. That gave rise to the independent record label phenomenon (still the third largest collective distribution entity here) and produced a competitive playing field that exists almost wholly in its own universe, regional cottage businesses that, taken as a whole, add up to a fairly massive industry.

The money still is not there in most cases to use the upper-tier facilities, that are often as intimidating to this group as they are unaffordable. But they are beyond the training wheels of the home studio now. They need real, acoustically correct recording spaces, reliable signal paths, exposure to gear they did not or could not purchase themselves, and expertise they never acquired. And, of course, a lounge.

A small but growing portion of that generation has recognised that need and is filling it, with small, affordable studios. The middle class. Something we weren't sure we'd ever see again. Combined with a decent puttanesca sauce and pasta al dente, who could ask for anything more?

compression will allow 60 channels, which means a data rate of 1.2Mb/s, for pictures and sound. The UK's Digital Television Group acts as a communications bridge between the digital broadcasters, set-makers and retailers. The DTG has good technical expertise and is worried about the squeeze on audio. Digital viewers will get only MPEG2 stereo, with embedded Dolby surround. The data rates are not yet decided, and will very likely be flexible.

The broadcasters need data space for teletext information and to carry authorisation codes which set the receivers to receive only those programmes the owner has paid for. The more subscribers there are, the more frequent the authorisation updates must be, so the less space there is spare to improve pictures and sound. Hence the DTG has been warning against the use of marketing adverts which claim 'CD quality' sound. The meaningless, and thus legally safe, phrase 'digital quality' will more likely be used.

In a confusing responsibility split, the UK's Digital Network group (formerly called the DeMux Group) represents many of the same companies as the DTG, but seems largely unaware of the DTG's work. The Digital Network group seemed genuinely surprised when asked about multichannel sound. It confirms that the receivers sold to the public when the services start will not be able to handle multichannel audio, even if any of the broadcasters spare the data space for it.

So while DVD capitalises on the promise of AC-3, and DVD-Audio promises superfi surround with Smart Content, Digital Television is stuck in the mid-fi matrix mud.

Is the sky falling?

The countdown to the new millennium is underway and panic over year 2000 compliance is rising, writes **Kevin Hilton**

CARE STORIES are great; they are a journalist staple, with a grand tradition tracing all the way back to the Bible: Bad Weather Predicted—Boat Building The Answer Says Noah. As time moved on, the stories became less apocalyptic but no less hysterical. It is something that has affected all areas of reporting and, in recent years, the technology press has been as reckless as any of its counterparts in this. Remember the 'CDs will spontaneously combust after six years' hilarity from the mid-eighties Sometimes they are more serious in their implications. At the moment there are two circulating and both are much more worrying than the idea of someone's entire Pink Floyd collection going *phoom*. One concerns mobile telephones and the increasing opinion that their prolonged use can cause cancer. The other is the Millennium Bug, the enormity of which is based on the premise that, on the 1st January 2000, computerbased systems around the world will fall over because their clocks will think that it is the year 1900, not 2000.

It is undoubtedly a problem but just how big or otherwise it truly is has been lost in recent hysteria. Like most technological kinks, the Millennium Bug was identified a while ago but it is only now that people are starting to run around in little circles shouting 'The sky is falling! The sky is falling!" Fittingly that source of all considered information, the Net, is packed to bursting with Web sites highlighting the calamity; although, in keeping with the nature of society these days, there are probably not as many pages about the Bug as there about Led Zeppelin, *Star Trek* or body piercing. These sites are a mixture of official government information group pages or efforts by independent pressure groups and consultancy. One of the latter calls the Millennium Bug 'the worst crisis since World War II", a typically lurid, alarmist quote that would have been more convincing if not for the artwork chosen to accompany the message. Sure, we're talking an insidious, unpleasant creature here but somebody got it very wrong to choose a graphic of a Ladybird (or Ladybug for people who eat jelly instead of jam), a benign insect that is considered an omen of good luck and is much loved by gardeners because it kills those nasty aphids.

Much scarier is the black wriggly thing on the Department of Trade and Industry (DTI) site, the UK government's signal that it knows what the Millennium Bug is and it is doing something about it. To back this up, on 30th March, the British Prime Minister, Tony Blair used his best 'trust me, I'm your friend, I want to help you' voice and smile to explain to both big business and the proletarian masses the havoc that the Millennium Bug could cause for individuals, industry, the country and the world in general.

To emphasise the immediacy of the situation. Mr Blair said, 'So the problem starts now, but we will gather pace up until the year 2000. And it is serious because there are few, if any, areas of modern life that are not touched by IT. If we don't tackle this problem, the economy will slow as many companies divert resources to cope with computer failures and some even go bust. If we do, we can avert major problems and enhance our reputation as a safe country to invest in." The government has increased the budget of the Action 2000 task force and will use £40m to establish 'centres of excellence' in IT training. This, Mr Blair hopes, will not only deal with the problem but help to alleviate unemployment by creating new jobs for people as Bug hunters.

Observers have been unmoved by Mr Blair's sincerity. Many believe that it is much too late to start a major campaign now, given that there are only one year and something like 200 days left before the ball drops. The retraining scheme has been derided and compared to UK terrestrial TV station Channel 5 sending out teams of newly 'qualified' technicians to retune viewers' video to avoid interference. It took longer than expected and there were many complaints about the work done. In that instance the most annoying thing that happened was the contrast and colour were not perfect; translating to computing could mean data loss beyond comfortable imagination.

Maybe. The threat could be overstated. Prime among people's concerns are PCs at home and in the office but the high turnover in new systems probably means that these are the least at risk. The real worry is over older systems, mainframes, networks and software of a dubious lineage. Many in the financial world are getting twitchy but some broadcasters are admitting that some of their scheduling and automation systems are not Year 2000 compliant. Who or what is safe? Is this really a threat to life as we know it? How much do you really know about the Millennium Bug and what it might do to your business?

> *Like most technological kinks, the Millennium Bug was identified a while ago,[a] but it is only now that people are starting to run around in little circles shouting 'The sky is falling! The sky is falling!"*

DVD: Generations

Forget VHS and Beta, forget CD and vinyl—the next media evolution is the big one. After last month's DVD Audio primer, **Tim Frost** charts the course of the next generation of media carriers

ONE WAY OR ANOTHER, DVD is going to make its mark on the audio, video and computer world. For a start, its advantages in the computer market make it an obvious and logical successor to CD: DVD-ROM drives have the distinction of being considered essentially future-proofed CD-ROM drive. This neatly sidesteps the problem that, although drives are beginning to sell in quantity, DVD-ROM software won't really start flowing until the end of this year. With manufacturers already predicting that CD-ROM production will entirely give way to DVD-ROM within a matter of a few years, DVD is guaranteed at least one place in the unravelling history of storage formats.

If and when DVD breaks into the consumer market as a replacement for sell-through and rental VHS, or for audio CDs, is less certain. The Jeremiahs say it is dead in the water already; as a non-recordable format it can't usurp VHS and the mass market will not go out and spend £500 on a play-only gizmo. At the other end of the spectrum, DVD's supporter's say it has been a runaway success, penetrating the market faster than CD or VHS. Somewhere in the middle, business analysts and independent commentators believe that DVD-Video will ultimately replace VHS as a distribution medium for prerecorded video; although it will take between five and ten years to get established.

But what is DVD? Is it simply a disc with seven times the storage capacity of CD, delivering just better quality video than the Video-CD and better audio than audio CDs, or is it more than that? What are the audio implications of the format—in fact, what is the format?

Those happy few who have been following DVD's development over the years, will have been able to splice together many of DVD's details from various sources. However, since the DVD Forum, which sets the standards, likes to keep the details close to its chest (unless you want to pay the $5,000 to become a Grade Two member) the full facts are often hard to come by. Here, then, is the story for you mortals.

The most obvious feature of DVD is the amount of user data that can be fitted onto the disc. Sticking with the 12cm silver disc (8cm is an option that no-one has shown the remotest bit of interest in) DVD has managed its seven-fold move up in capacity to 4.7Gigabyte (Gb) without any magic tricks. Mostly its been done by tightening up the tolerances throughout the system. The pitch of the tracks is smaller, so more tracks can be placed on the disc. The pits are smaller, which puts more pits on each track; although this requires a laser with a smaller wavelength to read the smaller spot.

The structure of the data put on the disc is also more efficient. DVD has a more efficient and effective error correction, so freeing more space for real data. (See Tables 1 and 2.)

Another major physical difference is how the data layer is placed within the disc. In a CD, the reflective data layer is close to the top, printed, surface of the disc. In DVD it is dead in the centre, specifically so that DVD discs can be made double sided by placing data layers back-to-back, doubling the amount of information that can be stored on each disc.

A further method of increasing the storage capacity of a DVD disc is to take the dual-layer approach, where a second semitransparent data-layer is sandwiched between the main reflective layer and the surface of the disc. The player's laser can either focus on the semitransparent layer and read the information off that, or beam through it and read the data on the main reflective layer behind it. However the complexity of creating this semitransparent layer, restricts its capacity to a mere 3.84Gb. But in theory at least, a combination of dual layer double sides could push a single disc's capacity up to 17Gb.

Manufacturing process for single and double-sided DVDs is a cross between CD and Laserdisc. The two halves of the disc are pressed in polycarbonate, the surface sputtered with aluminium much like the CD process and then finally the two halves are 'glued' together. Current DVD manufacturing technology can cope with single-sided (DVD5) and double-sided (DVD10) discs relatively easily; although the smaller pit sizes and the need for absolute flatness presents a raft of manufacturing difficulties. Making the dual-layer (DVD9) discs is somewhat more problematical, with few pressing plants even attempting it yet. (See Table 3.)

The two established forms of pressed DVD are DVD-ROM and DVD-Video, with DVD-Audio still being discussed and DVD-Karaoke as a variant of DVD-Video. DVD-ROM can be looked on as an extension of CD-ROM, maintaining and building on the ISO9660 and UDF formats that existing computer operating systems can understand. But it is the work done to create the DVD-Video format that holds the key to DVD's potential as a universal carrier.

The DVD-Video specification extends the logical layout of DVD so that the one disc can be read on every type of DVD drive, whether its in a PC, a video or a hi-fi system. DVD-Video also sets out the formats for video, audio and data content and includes its own

DVD compared with CD	DVD	CD
Disc diameter/thickness	120cm/1.2mm	120cm/1.2mm
Track pitch	0.74 micron	1.6 micron
Pit length	0.4 micron	0.834 micron
Laser wavelength	635-650nm	780-790nm
Speed	3.49 m/s	1.2 m/s
Max data rate	11.08 Mbit/s	1.536 Mbits/s

Table 1: Comparison of CD and DVD

WHERE DVD GETS ITS EXTRA CAPACITY	Capacity increase compared to CD
Closer track pitch	2.2 x
Smaller pits	2.1 x
Improved modulation efficiency	1.07 x
Improved format efficiency	1.4 x
TOTAL	6.8 x

Table 2: The capacity account

TYPE	12cm discs		8cm discs	
	Name	Capacity	Name	Capacity
Single side-single layer	DVD5	4.7Gb	-	1.46Gb
Single side-dual layer	DVD9	8.54Gb	-	2.66Gb
Double side-single layer	DVD10	9.4Gb	-	2.92Gb
Double side-dual layer	DVD18	17.08Gb	-	5.32Gb

Table 3: DVD options

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DVD Writable Formats

ALTHOUGH DVD-RAM and DVD-R are the only official DVD re-writable formats, there are others around the corner—each with different strengths and weaknesses.

DVD-R: 3.9Gb Record Once; Uses the same basic principle as CD-R, where the laser burns the data pits into a special dye layer, DVD-R was developed by Pioneer as a tool to author and test DVD software.

DVD-RAM: 2.6Gb Re-writable: The Forum's only agreed re-writable standard is DVD-RAM, which using a mix of phase-change and magneto-optical technology. As standard, the disc comes in a outer cartridge—like the shell around a 3.5-inch floppy disc—although the disc can be removed and read on a suitable DVD-ROM drive out of its cartridge. The cartridge gives DVD-RAM ruggedness, but moves it away from the bare disc concept of CD and DVD.

DVD+RW: 3Gb Re-writable: Developed by HP-Philips-Sony-Ricoh, this 'unofficial' DVD format uses phase-change technology like CD-RW. DVD+RW is a lot closer to existing CD and DVD technology and comes as a bare disc.

DVD-R/W: 3.95Gb Re-writable: DVD-R/W is designed by Pioneer as a re-writable format for authoring and other applications where the discs contains files that are rarely changed. Like DVD+RW, it is based on phase-change technology and should also be playable on most if not all, existing DVD players.

standardised navigation information and programming, so that all players can find their way around the disc.

Nearly all of these control structures and file standards will also apply to DVD-Audio. So understanding the thinking behind the published DVD-Video standard gives a good idea about most of the possibilities that will be offered by DVD-Audio.

At the heart of DVD-Video is its ability to deliver near-broadcast-level video using MPEG2 video encoding. MPEG2 can reduce CCIR-601 component video down to a data stream running at a mere 3Mbit/S or 4Mbit/s so that a 2½-hour movie can fit on one side of a DVD disc. MPEG2 stores the video in component form, (colour signals stored separately) with the signal merged into composite NTSC or PAL by the DVD-Video player.

Video can be stored in 4:3 format or as true wide-screen images, anamorphically squashed on the disc and stretched back to shape again by the player with minimal wastage of resolution by letterboxing the image with black lines top and bottom. Coming down-market, the DVD-Video standard also embraces the lower quality MPEG1 video standard, which might be relevant for low-cost productions. However, with the rapidly dropping price of MPEG2 encoding systems, within a year or two there will be little or no cost advantage for opting for encoding video using MPEG1.

Since DVD-Video builds on DVD-ROM formats, PC or Mac readable files can be easily included onto DVD-Video discs in a data area. These files can be anything from games to Web links. Conversely, DVD-ROM discs can contain DVD-Video and DVD-Audio files that will be recognised by the relevant consumer players. As the icing on the cake, DVD has a 'Burst Cutting Area' which can be individually written to, immediately after the disc is pressed. This allows each disc to be individualised with a unique serial number.

THE LAST BATTLE for DVD-Video, and what brought the format to the attention of many audio professionals, was the much publicised spat between multichannel MPEG2 audio and Dolby Digital (the format formally known as AC-3) which was finally settled in Dolby's favour. Technologically, this battle has relatively small impact on the overall market for DVD and certainly did nothing to help explain the full range of audio options that are part and parcel of the DVD-Video standard.

Where CD offers only one audio standard, 16-bit/44.1kHz, DVD offers well over 60; audio for DVD is not so much a fixed standard, more a range of allowable options.

The audio formats fall into two primary groups: linear PCM and compressed-audio streams (compressed-audio in this case means audio that has been encoded using Dolby Digital, MPEG or other similar data stream reducing algorithms).

The first significant point about LPCM audio on DVD-Video is that it follows the video world's adoption of 48kHz as the core sampling rate. 44.1kHz doesn't get a look in.

Word lengths start with CD's 16 bits, but DVD also has options for 20-bit and 24-bit working, and this can be matched with a 96kHz sampling rate. This already shows that DVD-Video can carry off the job as the audiophile carrier if the demand is there— >>>>>

<<<<< producers, artists and consumers do not have to wait for DVD-Audio.

LPCM audio on DVD is also not restricted to stereo. The disc can carry a range of multichannel formats to a maximum of eight discrete channels at 16-bit/48kHz or five discrete channels at 24-bit/48kHz. The limitation is the storage capacity of the disc—some space has to be left for the video after all, and the rate at which audio data can be extracted from the disc in replay. This is the more practical limitation as the standard allows for maximum data rate of 6.144kbit/s for the combination of all the linear PCM audio tracks. But within these limitations, engineers and producers can have their own choice of bit and sample rates and number of channels. (See Table 4.)

The problem recognised by Hollywood from the outset is that linear PCM audio is simply too disc-hungry; use multichannel PCM and there is not be enough space left for the video content of an average length movie. So adopting compressed audio formats—stereo Dolby Digital in the US and MPEG1 stereo in Europe was an obvious move—allowing more space for video and the option to include several separate audio streams for different languages.

The DVD Forum has settled on four specific compressed-audio formats for DVD-Video—Dolby Digital, MPEG Audio, DTS and SDDS; although the door is left open for others should the need arise. (See Table 5.)

After several skirmishes between Dolby and Philips (representing MPEG; although not owning it) the final decision about

DVD-Video Linear PCM audio options		
SAMPLING FREQUENCY	QUANTISATION	MAXIMUM NUMBER OF CHANNELS
48kHz	16 bit	8
48kHz	20 bit	6
48kHz	24 bit	5
96kHz	16 bit	4
96kHz	20 bit	3
96kHz	24 bit	2

Table 4: Audio on DVD-Video—uncompressed

DVD-Video Compressed-audio options	
Format	Channel options
Dolby Digital	Mono to 5.1 channel
MPEG1 Audio	Mono and stereo
MPEG2 Audio	3 to 7.1 channel
SDDS	5.1 and 7.1 channel
DTS	5.1 channel

Table 5: Audio on DVD-Video—compressed

what is mandatory for DVD-Video has now been settled.

The ruling essentially says that where audio is present on a PAL region disc, it must have one stream at least, in LPCM, or MPEG1 or Dolby Digital. On an NTSC disc, it has to be in either LPCM or Dolby Digital. All this relates

only to mono or stereo audio—the use of multichannel audio is purely optional.

Confused? You're not the only one. But if you turn the problem on its head and look at what has to be on the players, it becomes a little clearer as to what audio formats should appear on the disc. The cheapest, most >>>>>

At TDK, we believe that repetition puts an end to all art. That's why we know you can only accept the best for recording, editing, mastering, presenting and archiving your performance. For this reason, we developed TDK STUDIO products, a range of digital audio media providing flawless recording characteristics and uncompromising reliability. From now on, there will be no repetition due to technical failure of recording media, so you can concentrate on your art first time, every time.

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<<<<< minimalist NTSC region player will have stereo LPCM and stereo Dolby Digital capability. The equivalent PAL region players will have stereo LPCM, MPEG1 Audio and stereo Dolby Digital capability. So to absolutely guarantee a disc will play its audio then it needs to have the basic stereo (or Pro Logic) mix in one of these formats. What else goes on the disc after that is entirely up to the producers. It can be a combination of LPCM, MPEG and Dolby Digital, mixed in with multichannel SDDS and DTS to perversely cover every possible base—or a disc could just contain linear PCM. So after the mandatory requirement has been met, these additional formats are used as a value-added features on the disc.

In practice, multichannel Dolby Digital, MPEG2 audio can be used instead of the stereo streams, since DVD-Video players can extract the stereo signal from either of them.

So much for the rules. In reality, the movie

industries in Europe and the US have already standardised on Dolby Digital as the compressed-audio stereo-multichannel carrier for their release, some with the addition of LPCM stereo, carrying a Dolby Pro Logic version of the soundtrack.

The general idea is that each disc will contain two or three different language soundtracks, so that one master can be used for several different countries. Users can then select which language soundtrack they want to listen to.

Censorship and other legal issues, as well as the practicalities of gathering all the different soundtracks together and putting them on a single disc, has resulted in a release schedule that zones Europe into at least three separate DVD-Video masters. Most major movies will be released in a UK version, a German-speaking territory version and a release to cover the rest. Each release will have the movie's original language soundtrack

> In reality, the movie industries in Europe and the US have already standardised on Dolby Digital as the compressed-audio stereo-multichannel carrier for their release, some with the addition of LPCM stereo, carrying a Dolby Pro Logic version of the soundtrack

plus additional soundtracks and subtitles for each local territory.

As far as the players are concerned, the first generations have mostly offered stereo analogue outputs, with a digital connector to send the multichannel digital audio stream out to an external Dolby Digital decoder. However, the cost of multichannel audio decoding is dropping rapidly and there are already several low-cost single chip decoders. So it is likely within a year or two all but the lowest entry-level players will have discrete multichannel capability built into them. The same applies to high quality PCM. Decoding of 96kHz material is arriving on a number of players and this too will become standard in the not-too distant future (even if the analogue sections on most players will not be able to do the signal justice).

Without a karaoke option, no Japanese-inspired audio-video format would be complete. DVD-Video Karaoke disc displays background video pictures with a text overlay of the lyrics and breaks the LPCM >>>>>

DVD: A Potted History

PHILIPS-SONY started the ball rolling by announcing its HDCD format in September 1994. They proposed a new disc format, based on CD's size and shape that could store significantly more data and could be used for audio, data and for high quality video, using the newly established MPEG2 compression technology.

However, unwilling to hand over control and royalties to Philips-Sony in they same way they did with CD, the rest of Japan Inc, headed by Toshiba and its strong connections with Hollywood, went for its own option, which was announced the following January. Where HDCD would still require a movie to be put on two discs, the Toshiba/Time-Warner format was double sided so that a single disc could hold a complete movie, working the same way as Laserdisc.

A format war was in the offing until the computer industry, headed by IBM, turned around and made it clear that it wanted a single solution and would not support either of the opposing formats. This concentrated the two side's concentration wonderfully and by the middle of 1995 parts of the two formats were integrated into a single solution—the family of DVD that we know and will learn to love.

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<<<<< audio up into five separate audio channels. Two are for the stereo backing track, one for a melody guide track and two for additional vocal tracks. Karaoke players have controls for the user to select and mix tracks as they sing along—great.

One of CD's selling points over LP and cassette is the random access to individual tracks. For Laserdisc users this is nothing new, but for the vast majority who have only experienced VHS, DVD takes this feature for the first time into the video world. DVD-Video achieves navigation control by having its own standardised program control system. Video can't just be placed in streams one after another as with VHS, the disc has to be programmed with a full set of instructions that tells the DVD-Video player what video and audio streams to access where to start from and where to finish. Without these instructions, a disc will simply not play. When the disc is loaded into the

player, the control system is used to select the appropriate language soundtrack and subtitles and whether the movie is to be shown in wide-screen or 4:3. In operation, DVD follows the model of Laserdisc, splitting the video up into chapters that can be accessed either directly from a remote control or from on-screen menus.

But the programming can go much further than simple menu-driven navigation. The programming language includes logical arguments so that it can operate as a simplified interactive disc. DVD programming can be used, for example, to create one of those annoyingly addictive video trivia discs. This will play out video clips followed by a menu to select the answers. It will then branch off in different directions depending on the choices while also keeping tally of the score. If this does not seem that adventurous, remember DVD-Video discs will do this on every ordinary DVD-Video

DVD-Video Karaoke disc displays background video pictures with a text overlay of the lyrics and breaks the LPCM audio up into five separate audio channels. Two are for the stereo backing track, one for a melody guide track and two for additional vocal tracks

player—no additional computing power required.

CD is open to any and every form of copying and Hollywood wasn't about to make the same mistake with DVD. Offering almost master-tape quality video, Hollywood was paranoid about committing its material to DVD-Video without some form of copy protection and came up with two separate systems. The Content Scrambling System (CSS) was developed to stop direct digital to digital copying. Frames of video are encrypted with a key that cannot be copied, so bit-for-bit copies of the entire disc or single video files will not function. Each DVD-Video player also includes a Macrovision >>>>>

Regional Codes	Territory
1	US-Canada,
2	W Europe-Japan-MiddleEast
3	SE Asia
4	Australia-Pacific Islands-Central & South America
5	E Europe, Africa, Indian sub continent
6	China

Table 6: Territorial coding

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<<<<< processor to make it more difficult to make simple analogue VHS copies and this is activated using an instruction on the disc.

Region Coding enables distributors to maintain some control over the territories that discs can be used in to avoid extensive exporting of discs from the US or from some of the piracy prone areas such as China. Each player (including PC DVD-Video systems) has a set area code for the territory it is sold into. Discs for those markets are similarly coded and will only play on matching players. Much has been said about Regional Coding being used intentionally to protect markets.

Regional Coding has been introduced specifically to create a barrier between territories, since most distributors hold the rights to sell a title only in specific territories, and not worldwide. Although it must be pointed out, that the system does nothing to stop movement between different territories in each region.

The copy protection issues are being rethought with the possible addition of some variety of watermarking. But despite all the pain and suffering that hammering out the copy-protection standards has caused —including delaying the launch of DVD by a year—the movie industry is resigned to a more pragmatic view on the value of copy protection. None of it is bullet-proof as CSS, Macrovision and Regional Coding are all defeatable, the view is that these act as technical barriers to limit piracy, especially in the home, rather than eliminate it. (See Table 6.)

Since DVD-Audio has already been discussed at length in *Studio Sound* (April 1998). I will skirt around this up-and-coming format—important though it may be in the end. DVD-Audio will follow in the footsteps of DVD-Video, sharing its logical format and some of the file standards to make it universally compatible with the other DVD forms. The range and quality of audio formats include what is on DVD-Video, but go beyond

> The copy protection issues are being rethought with the possible addition of some variety of watermarking. But despite all the pain and suffering that hammering out the copy-protection standards has caused, the movie industry is resigned to a more pragmatic view on the value of copy protection

that into even higher-fi variants and there should be some ability to create a CD-DVD disc. Following DVD-Video's basic format, DVD-Audio will also support both video and data content, varying from DVD-Video more in nuance than in essentials.

AFTER ITS ADDITIONAL storage capacity, the most significant function that DVD offers over CD is its universality. When CD was developed it was purely as an audio distribution format. Turning the CD format into a carrier for data, then coping with recordable variants, pictures, video and finally a mix of all of them together has proved a nightmare. Each format is different and there is little standardisation of the types of audio, video and control files that go on the disc. Enhanced CD promised much in bringing the worlds of CD-ROM, video and audio together, but has delivered little of lasting impact.

Recognising that this is a key issue, DVD resolves enough of the underlying compatibility problems to make it possible to put video, audio and data on a disc so that the relevant parts are available to computer drives, DVD-Video, and future DVD-Audio players. The way the data is laid out on the disc (the logical format) is standardised. So all of the DVD disc types will be recognised by all the DVD player types, even a disc primarily designed as a DVD-ROM will be recognised as a valid disc by a DVD-Video or DVD-Audio player. And since the audio and video files obey a common set of standards then, for example, the video content of a DVD-ROM can be viewed on a consumer DVD-Video and the hi-fi music content played on any type of DVD player.

DVD-ROMs are already being developed to function as fully interactive games and edutainment titles for the PC, but will also play their video content on a DVD-Video player. Since their audio files can also be stored in

DVD or Digital Video Disc?

THE ORIGINAL ACRONYM DVD came from Digital Video Disc, reflecting its origin as a video carrier. After the realisation that computer applications would be just as important to the new format as video, some wit redubbed DVD as Digital Versatile Disc. Now it is sort-of accepted that 'DVD' is the name and no longer stands for anything.

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Writing to DVD

DVD FOLLOWS the footsteps of CD into writable and rewritable formats, but in a different order. The industry believes that to be successful the medium must be rewritable, rather than write-once, so it has put all its development and manufacturing effort behind rewritable DVD. Consequently DVD is in the unusual position of having the rewritable DVD-RAM drives available at $500, when the more limited DVD-R write once drives are $10,000. Currently DVD-RAM and DVD-R cannot store the full 4.7Gb that can be squeezed onto a single layer of a pressed DVD. The aim is to introduce second generation drives and media next year that will bring all forms of recordable DVD up the 4.7Gb level.

one of the DVD audio formats, it is not only a vast improvement on the generally poor sound found on CD-ROMs, the audio can also play on DVD-Video and DVD-Audio players.

It is important to register this fact, because it describes a level of convergence between audio, video and data content that makes DVD a powerful, but dangerous carrier.

Powerful because it really doesn't care what the content is—it will happily store and deliver it on the right hardware. Dangerous, because it permanently blurs the distinction between audio, video and data discs.

The most significant function that DVD offers over CD is its universality. When CD was developed it was purely as an audio distribution format. Turning the CD format into a carrier for data, then coping with recordable variants, pictures, video and finally a mix of all of them together has proved a nightmare

It raises key questions about the nature of content. Assume for a moment that most producers will want to add video or graphics content to an audio album, especially if it doesn't cost much to do and doesn't stop the disc being universally playable. In that case where is the dividing line between a music video that happens to contain a full version of the album or an album that also contains additional video material?

This is where audio-only world needs to start taking notice of DVD-Video. The disc may be called DVD-Video but the video content does not have to be its main driving factor. Using DVD to release 'enhanced CD' type albums with additional video content, and audio in stereo, multichannel or 24-bit/96kHz super audio, is a quick-fix solution to upgrading the quality and functionality of an album release.

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Unlike DVD-Audio, the standards are fixed already and the user-base of players is on the rise. In the short and medium terms, DVD-Video players—both stand-alone and in PCs—are going to vastly out-number DVD-Audio players, making the choice to go to DVD-Video for an album release more logical, and profitable, than DVD-Audio. For those artists and producers who want to go beyond CD's 'limitations' (with apologies to John Watkinson) then DVD-Video offers a ready-made distribution format. The only problem is that consumers will have equipment potentially with play-back capabilities beyond the recording technology regularly used in most studios today.

This also has a knock-on effect on the recording formats. If material needs to cross the boundaries of DVD-ROM, DVD-Video and DVD-Audio, then it makes sense to originate at 48kHz sampling rates and DVD-Video also breaks out of the stereo world. Most movies

are in surround, an increasing number of live performance music videos are in surround, even video games are in surround, so why should audio-only albums restrict themselves to stereo? The next generation of record-buyers may have little allegiance to stereo and will want to know that surround content is there on the disc, even if they are going to access it on fairly rudimentary surround sound systems.

DVD, as personified by DVD-Video, goes far beyond being simply a larger data carrier. Unlike its predecessors, it moves published media comfortably into the multimedia world. It offers better picture, better audio, bigger data storage and the possibilities of seamlessly crossing the divide between the fixed standards required by consumer hardware and the flexibility needed for computer applications.

So that's it. A tour of the more essential elements of DVD and you can raise your head up from that empty beer glass—'cause it's your round and mine's a pint. ∎

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The high life

With the spotlight shining brightly on 24-bit 96kHz audio, **Mike Story** of DCS offers possible explanations for some of the audible advantages of working with higher sample rates
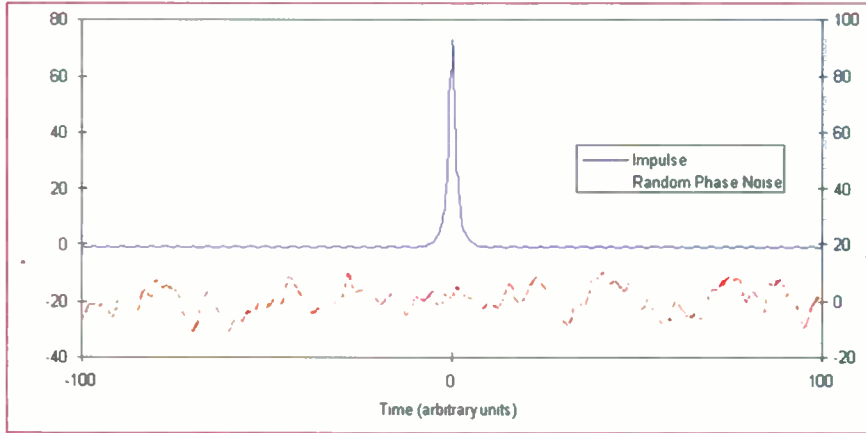
Fig.1. Contrasting waveforms and audio share common power spectra

THE BULK OF THE WORK on explaining how we hear has been concerned with frequency response. Frequency response of the ear has been extensively investigated, and the frequency response of a recording or reproduction chain is by far the dominant fac-

tor in determining how realistic or intelligible the resulting signal is. Although there is some argument about exact numbers, about 20kHz is generally accepted as the maximum frequency that humans can hear. We may not be able to tell one frequency from another above about

12kHz—they all appear in the 'max frequency' bin of our discrimination mechanism—but we can tell they are there.

Digital audio systems have historically made use of this 20kHz limit to set sampling rates. When CD formats were first established, the problem of storing the large amount of data needed for about one hour stereo playing time was substantial, so sample rates were set as low as reasonably possible, consistent with maintaining a 20kHz bandwidth. 44.1kS/s gave and still gives an unambiguous frequency range of 22.05kHz (Nyquist principle).

In principle, if frequency response were the only issue, there would be no advantage in moving to formats with higher sampling rates. However, the evidence is otherwise. Direct comparisons of the same source material, recorded and reproduced at 44.1kS/s, 96kS/s and 192kS/s show that there is an advantage in going to the higher rates—it sounds better. The descriptions of those used to making such comparisons tend to involve such terms as 'less cluttered', 'more >>>>>

The author, centre, with the team that made the first periphonic 24-96 recording in 1997
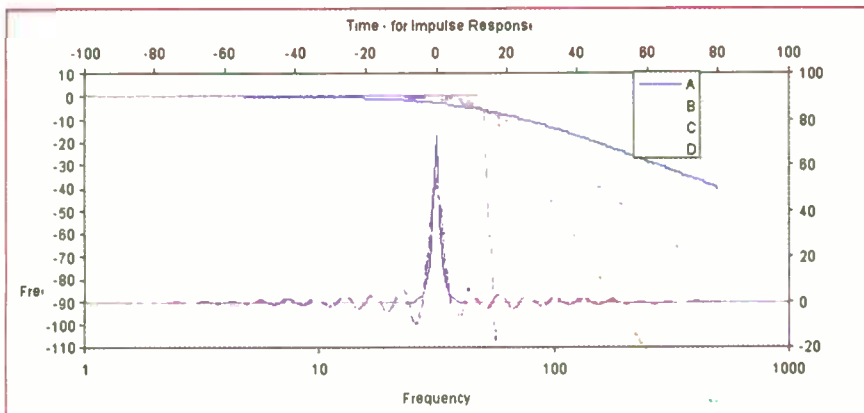
Fig.2. Roll-off characteristics for various filters
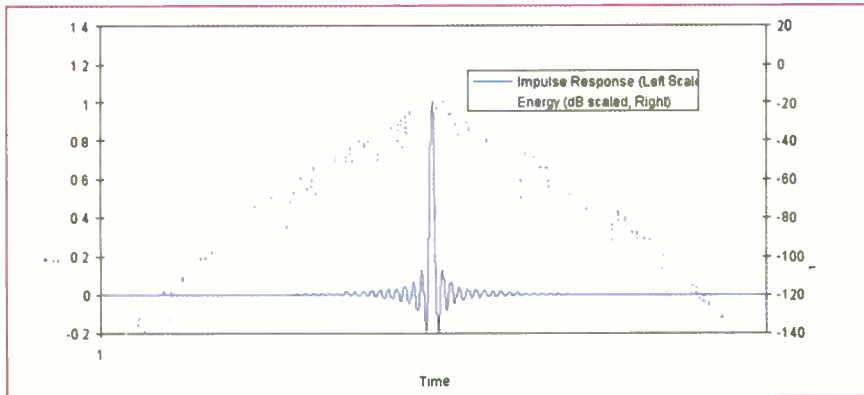
Fig.3. Anti-aliasing filter energy distribution

<<<<< air', 'better HF detail' and in particular 'better spatial resolution'. We are left wondering—what mechanism can be at work? It seems unlikely that we have all suddenly developed ultrasonic hearing capabilities.

Actually, a little thought also suggests that frequency response cannot be the only factor at work in our hearing apparatus. Fig.1 shows two waveforms that have identical (power) spectra, and yet sound very different—a bandlimited impulse (a click) and a type of white noise. Other waveforms can easily be generated that have the same amplitude response, but sound (substantially) different still. Something else must be going on.

To generate the waveforms, we diddle with the phase of the individual frequency components. Mathematically, phase information is the only other variable needed to convert from a spectrum to a waveform—but phase is really a mathematicians plaything, that is not a natural in explaining real physical effects.

Sampling Effects: At this stage it is worth looking at some of the baggage that sampling a signal carries with it. One of the processes that needs to happen is anti-alias' filtering. On the recording side, anti-aliasing filtering must be carried out to prevent signals present above 20kHz being aliased back on to the recording within the audio band. On the playback side, anti-alias filtering is usually carried out to prevent large amounts of high frequency energy being presented to subsequent items in the playback chain—such as amplifiers and speakers. In general, anti-aliasing filtering needs to be quite vigorous—typically from some fraction of 1dB down at 20kHz to -100dB or so at half sampling frequency (22.05kHz for CDs).

Sharp filtering inevitably causes a ringing transient response. Filter designers and mathematicians are familiar with the problem—the effect is referred to as the Gibbs phenomenon. Fig.2 shows responses for four filters with increasingly sharp cutoffs, and the associated transient responses. It is worth noticing that the ringing sets in at quite a gradual slope (filter B)—it is not the extreme severity of the sharper filters that causes ringing, but the move away from a very gentle filter function (A).

Anti-aliasing filtering causes this type of ringing transient response. The effect is well known and unavoidable, and tends to be dismissed as a mathematical irritation with no audible effect—because the frequency response is flat. The dismissal gains credibility, because if attempts are made to roll frequency response off a little earlier, recording engineers complain. Certainly, it seems that frequency response flat to 20kHz to 1dB or so is important.

Energy Dispersion: The ringing contains energy, and we can plot energy against time. For anti-aliasing filters we get the sort of shape shown in Fig.3. This shows that; although the energy in the input transient is concentrated at one time, the energy from the anti-alias filter is spread over a much longer time—the audio picture is 'defocused'. We might be tempted to argue that the energy is ultrasonic, but this is certainly not the case at 44.1kS/s or 48kS/s—our bandwidth constraints mean that to get good anti-aliasing, we must filter as fast as we can, and only pass the audio bandwidth. Ergo—any energy in the output signal is in the audio band. At sample rates above the >>>>>

<<<<< standard, the energy in the ring still has the full bandwidth of the pass-band—maths tells us so. We can also note that the energy in the ringing is large—for a sharp filter it can be 12% (-9dB) of the energy in the main lobe.

Fig.4 shows the frequency responses of some filters we might consider using, for audio purposes, at different sample rates. The digital filters do not represent any particular hardware, and are all designed to give the following performance:

-0.1dB at 20kHz
-120dB at half sampling frequency
no in-band ripple
24-bit coefficients

An analogue frequency response with a Gaussian filter set to -1dB at 20kHz is shown for comparison. Gaussian filters have a non-ringing transient response—if a filter rolls off faster than a Gaussian, it starts to ring. The ringing may be acceptable, of course. In practice, it is improbable one would use this filter, because -1dB at 20kHz is unlikely to be acceptable for professional purposes. A filter that was flatter to 20kHz, gentle to say 50kHz and fast after that, with some ringing, might be more likely. The purpose of the comparison is to show just how much extra bandwidth is needed to contain ringing (energy defocusing), and how much our ears may have taken for granted as the processing of the data they produce has evolved.

Fig.5 shows the transient responses of the different filters, operating at their different sample rates. >>>>>
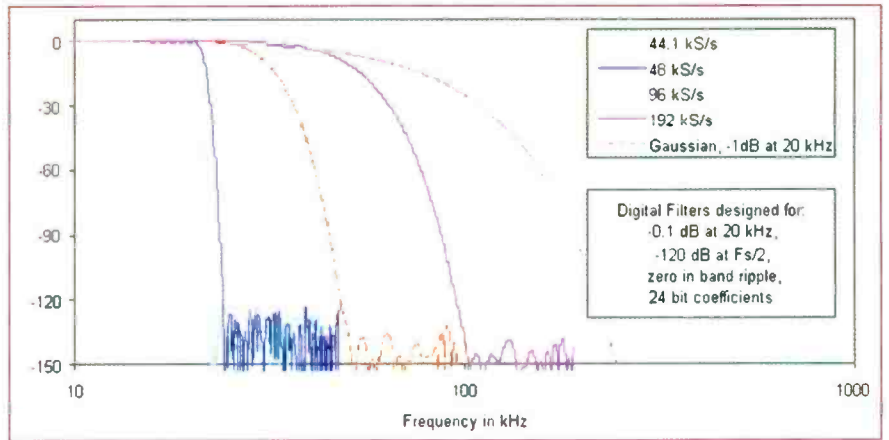
Fig.4. Filter frequency responses
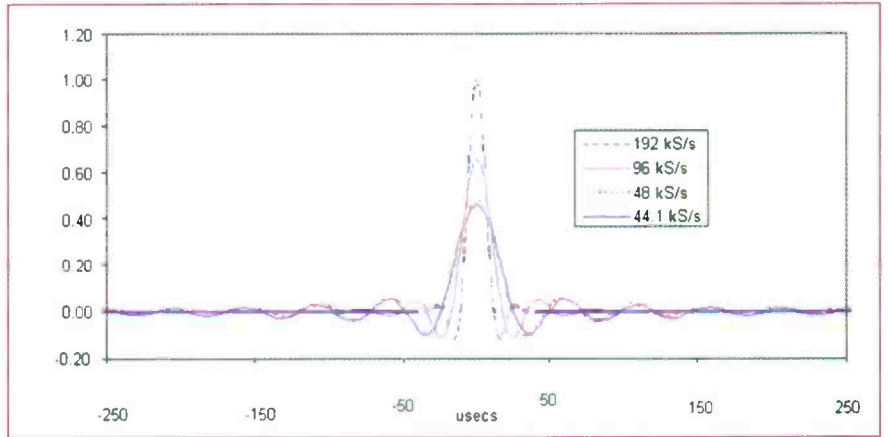
Fig.5. Transient filter responses

Energy Dispersion at Different Sample Rates: Fig.6 shows the energy associated with the transient responses. 44.1kS s and 48kS s filters spread audible energy over 1ms or more. The 96kS s filter is much better, keeping the vast bulk of the energy within 100 msecs. The 192kS s filter can be very good indeed, keeping the energy within 50µsecs. The analogue Gaussian filter is just a little better still; although the improvement is almost certainly academic because of energy dispersion from today's speakers and mics.

Taking into account the speed of sound, we can convert energy defocusing in the time domain to 'smear' in distance estimation by the ears. Energy spread over ±500s is the same as a distance smear of ≃15cms. 96kS s keeps almost all the energy within about ≃50 secs, or ≃1.5cms. One of the observations people make about 96kS s material is that the spatial localisation of everything is very much better than 44.1kS s. 192kS s is better than this;
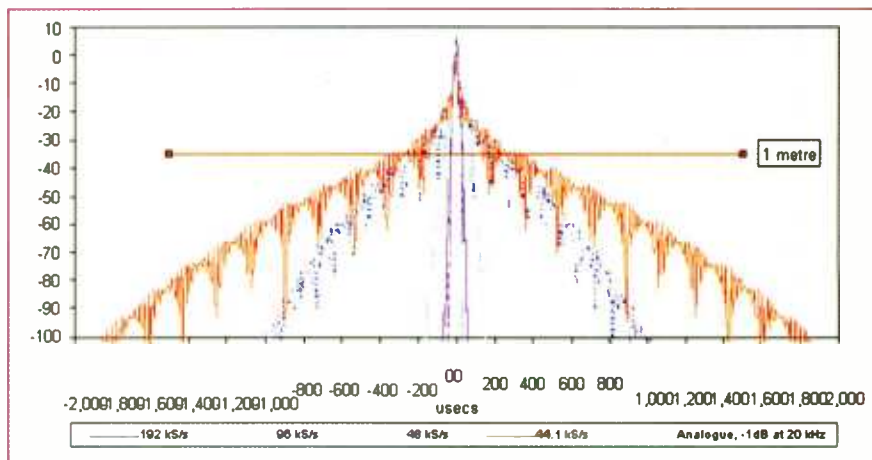
Fig.6. Transient response energy distribution

although very dependent on amp and speaker performance to demonstrate it.

You can get yourself into a bit of a twist thinking about the energy in the ringing. After all, if it is in the audio band, allowing extra energy at higher frequencies through the system surely cannot cancel out some that is in the audio band? It does, though—so although we may not be able to hear energy above 20kHz, its presence is mathematically necessary to localise the energy in signals below 20kHz, and it is possible (and our contention) that we can hear its absence in signals with substantial high frequency content. A high sample rate system allows it through (fact) —and allows the high frequency signals to sound more natural (contention) but allowing better spatial energy localisation (fact).

It is our suggestion that some of the audible differences between conventional 44.1kS s and higher rates (88.2, 96, 176.4, 192kS/s) may be related to this 'energy smear' or defocusing caused by anti-alias filtering, and that the ear is sensitive to energy as well as spectrum. This is further backed up by our two original 'same spectrum, sound different' signals (Fig. 1.). In the impulse, all the energy is concentrated at one time, whereas for the white noise the energy is uniformly spread over time. There is a precedent to this suggestion that the ear is sensitive to both spectrum and energy—the eye is as well. For sensitive vision or vision off the main beam, we use energy (luminance, or black and white information), whereas for detailed identification when we are looking at something, we use spectrum information (chrominance, or colour). In fact, most sensing processes are sensitive to energy. If the ear is sensitive to energy, it would almost certainly use the information for spatial localisation.

Multichannel: In conclusion, it is worth noting that if this suggestion is correct, then it would be sensible for any multichannel audio formats to use one of the higher sampling rates. The purpose of multichannel is for better spatial localisation of sound sources—so it needs a sampling rate that can support this. ■

Footnotes

1. I refer to a recording or reproduction chain as having a frequency response, and I refer to a signal as having a frequency power spectrum —or sometimes, because frequency is so overwhelmingly important, just a spectrum.
2. See, for example, *Digital Signal Processing*, Alan V.Oppenheim & Ronald W.Shafer, pub. Prentice-Hall 1975.
3. Even people with grey hair, who must have significant high frequency hearing loss.

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High jinks

In response to John Watkinson's challenge over the value and use of high sample-rate conversion systems, independent broadcast engineer **Tony Drummond-Murray** raises issues of perception and fidelity

IN THE CHALLENGING article 'Spotlight on 24-96' (*Studio Sound*, February 1998) John Watkinson is mistaken in his assertion that to stop development when a product-system has achieved 'good enough' status through development, (at the invisible point B) on his Fig.1, is a matter of some intellectual pride? While few would disagree with his analysis of the factors affecting human hearing, in the real world the signals are processed by electronic systems responding to different criteria, long before they are released at some distant monitoring point, for the delectation of the ear. Who is to say what compromises will be regretted later, if system development is arrested prematurely? Currently it appears that in the parallel field of video the quest for digital resolution is far from over as TV cameras and telecines strive for ever finer quantising steps, in the quest for higher originating quality linked with digital postproduction flexibility.

It seems unlikely, as John points out, that Shannon's Sampling Theorem has suddenly been proven wrong by those with so-called golden ears. The answer is more likely to reside within the envelope of the Theorem, which asserts that a band-limited waveform can be fully described by sampling at twice the highest modulation frequency. The actual frequency of the signal to be sampled may not be relevant, as it is possible to sample a 1MHz signal with 5Hz modulation, with a 10Hz sample pulse, and still extract the data.

The performance of the low-pass anti-aliasing filter on the input to the digitising electronics is critical to the whole system, since it rejects (the out-of-band) frequencies above the first Nyquist zone—that is, those that will be reflected back into the base band. The engineering of this filter becomes impossible when the highest modulation frequency is half that of the sample pulse, becoming increasingly practical as the highest modulation frequency recedes below this point. Phase linear, brickwall filters are the stuff of rocket science. The highest modulation frequency that a system will satisfactorily handle is way below the fs/2, by a factor directly related to the filter's in-band performance and required attenuation out of band. The **>>>>>**

Fig.1: 16kHz sine wave

Fig.2: D–A conversion on 8kHz sine wave

<<<<< apparent theoretical ability of a system to respond to frequencies up to fs/2 must represent the absolute limit and should not, surely, be advocated as a practical condition. The suggestion that anyone would wish to digitise a full amplitude 40kHz 'audio' signal that had originated in nature, with a 96kHz sample frequency, is just too far fetched; as is perhaps the assertion that all power amplifiers suddenly run out of loop gain at 20kHz, causing massive distortion.

Consider the situation where microphones capture sound coming from a number of sources simultaneously; there is no way in which the electrical waveform generated by these microphones is anything other than strictly analogue. At no instant do they naturally assume some convenient time (or amplitude) quantising steps prior to the A–D convertor. these are strictly imposed upon the input signal by an impartial central arbitrator. Once these steps have been imposed on the analogue signal by the A–D process, they become frozen in time until the next sample. This 'packet' concept is quite alien to nature, but must be true for all channels since they share the same sample clock. Although originally the channels may individually contain 'audio' frequencies up to perhaps 20kHz, there would be a rich blend of phase relationships between them, but this would have been crudely fixed by the sample clock, with great precision.

Consider the case of a 16kHz sine wave sampled at 48kHz, as in the AES standard: there are only three samples per cycle. Fig.1 top, shows the output signal (shortly after emerging from the D–A), the lower trace shows the effect of the interpolation filter, and represents the system output. The only way in which the output wave shape could be improved, to more closely resemble the original sine shape, would be to reduce the bandwidth of the interpolation filter centralising it on 16kHz—increase the filter Q at 16kHz. This would of course be complete nonsense, since the system could only then respond to 16kHz stimuli, an almost complete proof of the fact that a sine wave itself conveys virtually no useful information. .

This relatively coarse time-sampling must

Perhaps the ability of the 96kHz system to more accurately resolve phase differences due to the faster clock is, in some degree, responsible for the perceived difference, rather than the performance of the interpolation filter alone

destroy the subtle analogue phase relationships present in the original material, since the interpolation filter responds to the stimuli it receives after the step change initiated by the previous clock pulse. It can not 'know' where the next level will (should) be, and perform some look-ahead prediction. If the case of a single half-cycle of a 16kHz sine wave from silence is considered, it is impossible to >>>>>

Fig.3: 3kHz sine wave after 48kHz and 96kHz D–A

<<<<< predict with what polarity, phase and amplitude it will occur; but it is certainly possible to compute an envelope outside which it could not exist, when the filtering parameters are known.

Fig.2 shows the effect of the D–A conversion process on an 8kHz sine wave, while this is still a relatively high audio frequency, it's certainly audible to most humans, and is well within the range most people would consider vital to any form of musical reproduction. Surely, the move to 96kHz sampling is a natural progression based upon the perceived difference in quality, apparent to so many, in tests described within the pages of this Journal, for example. Perhaps the ability of the 96kHz sys-

tem to more accurately resolve phase differences due to the faster clock is, in some degree, responsible for the perceived difference, rather than the performance of the interpolation filter alone. Certainly one does not wish to grossly over engineer, even if the accountants were to sanction such a luxury, but the 10% lower 44.1kHz sample frequency of CDs et al would only make the sample wave forms more extreme, and a higher clock increasingly attractive.

Fig.3 shows the difference, before the low-pass interpolation filter, of the output of a D–A with a 3kHz signal, and 48kHz and 96kHz equivalent sample clocks. I think I know which one I would rather listen to.

Finally, a paradox for consideration. Many engineers will remember fondly the early Broadcast VTRs made by Ampex and RCA. Such VTRs used to reduce the 20 or so octaves of the input video signal to a single octave by frequency modulating the input video signal on an RF carrier for recording on tape. The VTR's playback systems had an RF HF equaliser, which was adjusted to give a flat frequency response of the demodulated video signal. The

apparent paradox was this: How could adjusting the HF equalisation of the off-tape FM signal, before 60dB–80dB of dynamic and static limiting, affect the frequency response of the demodulated video after limiting? The solution is simple: the equaliser, which was carefully designed to be phase linear, affected the level of the second harmonic that was recovered off tape, thereby changing the position of the RF

> The solution is simple: the equaliser, which was carefully designed to be phase linear, affected the level of the second harmonic that was recovered off tape, thereby changing the position of the RF signal's zero-axis crossings

signal's zero-axis crossings. No amount of limiting thereafter could ever alter the timing of these crossings.

Is it possible that the effect of increasing the sampling frequency to 96kHz is perhaps more subtle than at first appreciated? Sampling the audio signal, with the lowest frequency clock that the Standards Designers thought that they could get away with, represents considerable (data) compression. Whether this compression is lossless, or not, is clearly open to debate. ∎

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PUTTING IT ALL TOGETHER

As the AES celebrates its 50th Anniversary, we continue to expand in these pages our tribute to the people and ideas central to half a century of service to audio in its manifold forms. Our initial instalment accompanied the 103rd Convention in New York; the last instalment will coincide with the 105th in San Francisco later this year. Our intention is to examine the Society's origins, its evolution, and its potential as the global forum for professional audio.

As an important part of this overview we need to recognise the gifted individuals who have put us in touch with new frontiers. Here, in brief interviews and career profiles, they offer us their insights into important trends in the industry, or share with us the most significant events in their experience of audio. As we come to realise the real value and significance of their work, both they and their contributions are honoured.

Finally, there are two timelines. One marks many of the important events in audio in the last century, the other charts the parallel progress of the Audio Engineering Society. Ours is a history full of remarkable people, inventions and innovations. We are involved in a never-ending quest to push the envelope of what is possible in audio, asking "how?" rather than "why?" Our past accomplishments contribute to what we are today, and signpost the future. If the medium is the message, then let us make certain, whether in service to music or the spoken word, that the medium is and remains clearly audible.

The Committee for the 50th

SPRING 1998

THE AES – YESTERDAY,

JOHN ANDREWS
Marketing Director, Solid State Logic

SIGNIFICANT EVENTS? I've seen many big changes in a career that started on the operational side and then moved into manufacturing. I could list them but they're probably much the same as most other people.

However, when I was a BBC Studio Manager ('Beeb' terminology for sound engineer), the changeover from rotary faders to in-line faders was very significant.

When you're mixing on a console with rotary faders you can only control two at once - or maybe three if you spread your fingers spread out across the knobs! Compare this to when you have flat in-line faders and you can put a finger on each.

You may not think that this was a problem but the later mixers we used had up to 26 rotary faders, although admittedly four were subgroups that you didn't touch very much.

When developments made it possible to build the console electronics actually inside the console itself, there was a change, a move to quadrant faders, and then on to flat faders. It was then possible to get 20 faders in-line in a manageable sized desk whereas in the same sized desk you would previously have managed only about eight or 10 rotary faders. Working with large numbers of live musicians suddenly became a lot easier.

It is unlikely that the group of resolute people gathered in the RCA Victor Studios in Manhattan, fifty years ago, spent too much time thinking about a long term future for what they were about to form. The effort of just trying to make the next meeting happen was probably about as much as most could consider. However they must have had a vision for what the organisation could become because it set this Society off on a path that has brought us through 50 successful years to where we are today.

Let's look precisely at where we came from in that half decade. From that meeting of 150 people in New York, the Society now has nearly 12,000 members. From a single section there are now 110 sections, including 39 student sections, in eight regions throughout the world. The first annual convention took place in October 1949 in a New York hotel with 22 technical presentations and 45 exhibitors. The last convention, the 103rd, in New York attracted over 20,500 visitors and 370 exhibitors, had over 150 technical papers, numerous workshops covering the more practical aspects of the business, and numerous special events.

While the scale of operation has changed dramatically there still remains a great deal in common - just look at these two conventions.

Firstly the background to the events was similar, both being against a time of rapid technological change. In 1949 the technical papers presented clearly represented a realistic cross section of the leading edge technologies of the day but many of the topics would not be out of place now - automated signal compression systems for broadcast, contact printing tape duplication, audio techniques for TV, sound reinforcement developments, and standards.

The exhibition floor, much as today, presented displays of the very latest technology, although few of the names from that event still exist in the same form

today. Preempting the sizable proportion of non-US companies now exhibiting at US Conventions, there was a single exhibitor from the UK. There were also important technical demonstrations - reports of the Convention list milestones such as some of the first working demonstrations of audio tape, binaural recordings and new record manufacturing techniques. For an attendee at that first convention it must have been as exciting an event as any modern convention. It set a pattern for conventions that we haven't strayed too far from even today.

Away from the large showcase events, I've always believed that the real strengths of the AES are the diversity of its activities and the way it meets its aims of disseminating information in the field of audio. Much of this happens at roots level, within the local sections - not everyone has the ability to attend the conventions. It is these monthly local activities that spread knowledge and give the opportunity to meet colleagues in the same area. The Society owes much to those that run these local sections. The advent of the regional convention has helped support the more distant sections - six in Australia and seven in Japan so far: plus the 14 International Conferences and the numerous section conferences have brought up to the minute discussion to a wider audience than could be addressed just by the Conventions. Regional activities become ever more important as the membership diversifies - expansion into Eastern Europe and Latin America is recent and it may take a few years before we see large numbers of attendees at the Conventions from these areas.

Of course the other means of keeping in touch with the activities of the Society has been through the many publications, foremost of those being the Journal. It is worth considering how that has also grown, starting in 1953 as a quarterly and now is the foremost peer-reviewed audio publication in the world, carrying the reviewed papers, conference and

AN AES TIMELINE

1947: C.J. LeBel, John D. Colvin, C. G.McProud, Norman C. Pickering, Chester A. Rackey, Ted Lindenberg and others lay groundwork for the "Audio Engineering Society."

1948: Group creates charter for the new Society. On March 11th, 150 people gather at RCA Victor studios in New York City to see what this new group is all about, and to listen to a formal lecture by Dr. Harry F. Olson on loudspeaker

design. Two months later other Sections are organised in Denver and Los Angeles.

1949: Under guidance of Louis S. Goodfriend, AES organises first of four lecture series groups, entitled "Elements and Practices of Audio Engineering." The first convention (40 exhibitors, 3,500 attendees) held at Hotel New Yorker. Program features papers and staging of a live-versus-recorded

demonstration.

1953: The first International Section is formed in Japan. The first AES Journal published, pioneered by Louis Goodfriend and Vincent Salmon.

1954: AES publishes its first standard, TSA-1-1954: "Standard Playback Characteristic for Lateral Disk Recording."

1955: The AES is incorporated

with a "not-for-profit" status.

1957: Westrex demonstrates a 45/45 stereo cutting system at the AES Convention.

1965: The New York group becomes official Section. Facilities Committee constructs sound reinforcement/recording console for use at Conventions.

1968: Johan L Ooms organised an 'Audio Abroad' session at the New York Convention Centre attracting 18 papers from Europe. This led to discussions about forming AES Sections in Europe.

1970: The Board of Governors meets with John C.G.Gilbert, John Maunder and Percy Wilson: approves formation of British Section. Central European Section, comprising Austria, Belgium,

AN AUDIO TIMELINE

A selection of significant events, inventions, products and their purveyors, from cylinder to DVD.

1877: Thomas Alva Edison, working in his lab, succeeds in recovering Mary's Little Lamb from a strip of tinfoil wrapped around a spinning cylinder. He demonstrates his invention in the offices of *Scientific American*, and the phonograph is born.

1878: The first music is put on record: cornetist Jules Levy plays "Yankee Doodle."

1881: Clement Adler, using carbon microphones and armature

headphones, accidentally produces a stereo effect when listeners outside the hall monitor adjacent telephone lines linked to stage mikes at the Paris Opera.

1887: Emile Berliner is granted a patent on a flat-disc gramophone, making the production of multiple copies practical.

1888: Edison introduces an electric motor-driven phonograph.

1895: Marconi achieves wire-

less radio transmission from Italy to America.

1898: Valdemar Poulsen patents his "Telegraphon," recording magnetically on steel wire.

1900: Poulsen unveils his invention to the public at the Paris Exposition. Austria's Emperor Franz Josef records his congratulations. Boston's Symphony Hall opens with the benefit of Wallace Clement Sabine's acoustical

advice.

1901: The Victor Talking Machine Company is founded by Emile Berliner and Eldridge Johnson. Experimental optical recordings are made on motion picture film.

1906: Lee DeForest invents the triode vacuum tube, the first electronic signal amplifier.

1910: Enrico Caruso is heard in the first live broadcast from the Metropolitan Opera. NYC.

Robert Lieben of Vienna develops an amplifier using vacuum tubes enabling playback through a loudspeaker rather than headphones.

1912: Major Edwin F. Armstrong is issued a patent for a regenerative circuit, making radio reception practical.

1913: The first "talking movie" is demonstrated by Edison using his Kinetophone process, a cylinder player mechanically synchronized to a film projector.

TODAY & TOMMORROW

convention reports plus news of AES activities. For many it is the only contact that they have with the Society on a regular basis and is still given as the most common reason for joining the AES.

It would be easy to say that we're fifty, all is well, and we're looking forward to another fifty years just the same. But the next fifty aren't going to be the same, nor are the next ten. We have to develop.

One important change is the way that the AES is beginning to take advantage of new technology where its use is appropriate to our needs. In 1997 we made the Preprint and Journal indices available on CD-ROM. This is a tremendous asset to researchers looking for particular items amongst the vast store of technical knowledge held by the AES. There is a strong likelihood that the near future will see Convention preprints on CD-ROM supporting the paper versions. There are obvious other possibilities for such a medium but there has to be due consideration of the suitability of CD-ROM for all printed information.

A major influence on the way that members, and non-members, connect with the Society has been the introduction of www.aes.org, the AES website. Faster news and large amounts of information, checking up on the progress of AES standards - these are the attractions but following the introduction of a secure server to enable safe credit card transactions, far more is now possible. In the first six months of operation

Roger K. Furness
AES Executive Director

over a quarter of all transactions were processed via the internet - including registering for Conventions, paying membership dues and purchasing AES publications. This has proven far more popular and more quickly that we thought possible.

E-mail is playing a greater role within the AES and there may be further ways that we can use that to further disseminate information.

As technology changes, so do the subjects that we need to discuss. It also brings business areas and people who were previously not concerned with audio within the interest areas of the AES. We have to learn to absorb the interests of these new-to-audio people and expose them to the breadth of opportunity that the AES can offer while learning from the new entrants about possible future directions. In this way the AES will remain relevant and vital for anyone with an interest in audio, in all its multitude of areas.

So 50 years is a time to reflect on what has been achieved, what we have, and where we might be going. Ultimately the role of the AES is to provide a platform for discussion of all areas of audio. Precisely how that is maintained and kept at the cutting edge is for those guiding the Society to judge.

The long term future direction of the AES however lies in the hands of its members. They'll be the ones who'll be deciding where we are at 100.

MALCOLM TOFT
Managing Director, MTA

NO DEVELOPMENTS IN recent years have been so dramatic as the move to multitrack recording. It changed the entire face of the recording industry and we could be far more creative.

The critical period was just four years at the end of the 60s and the beginning of the 70s. The industry had been 4-track since the early 60s and in late 1968 I was an engineer at Trident Studios which just had the first 8-track in the UK. Because of the shortage of tracks all recording had to be planned - and it wasn't possible to bounce on adjacent tracks. Mixing sessions used to follow the recording - about an hour per title - tacked on the end of the session!

Going to 16-track increased the possibilities while moving to 24-track really opened up the doors - you had so much freedom. It changed recording techniques and introduced the 'mixing session' as a major part of the process. There came the idea of using different desks for the mixing which led to the idea of tracking and mixing rooms. Bigger control rooms were needed for desks capable of handling the extra tracks as well as the large amounts of outboard gear now needed. It was no longer certain that the engineer who did the track laying would also do the mix.

I stopped engineering in 1970 to develop Trident consoles. After the arrival of the 24-track recorder it was a totally different business. The move from 4 to 24-track was evolutionary but the effect was revolutionary.

Germany, Luxembourg, The Netherlands and Switzerland, formed in Frankfurt, Germany.

1971: First European Convention held in Cologne, Germany, with 250 registrants, 29 papers, 12 exhibitors. First Workshop ("A Workshop on Studio Tape Recorders") held at the 41st Convention in New York.

1973: At the fall Convention in

N.Y., the New York Section presents: "Look What They've Done To My Song, Ma," tracing the effect of technology on recorded music. Programme narrated by Michael Tapes, features John (Jack) Mullin, who illustrates the history of recording through demonstrations. The programme concludes with an electronic music composition written for the event by Walter Sear, reproduced on a 16-track tape

machine wired directly to 16 power amplifiers and loudspeakers placed around the room.
Belgian Section formed.

1974: Netherlands Section formed.

1975: New York Section, at the 52nd Convention held at the Waldorf-Astoria Hotel, presents "Those Magnificent Men and Their Music Machines," recapping the his-

tory of electronic music, narrated by Michael Tapes and Robert Moog. Electronic music is composed and performed by Suzanne Ciani.

1976: AES Gold Medal awarded to Georg Neumann and a Silver Medal to Dr Willi Studer, at the 53rd Convention in Zurich

1977: Centennial Issue "The Phonograph and Sound Recording After 100 Years," JAES volume 25,

number 10/11, October/November. Exhibition of historic audio equipment at 56th Convention. Paris, to celebrate centenary.

1978: Board of Governors approves the formation of the Technical Council and the Technical Committees, and sets new standards procedures. First Anthologies are published: "Loudspeakers, Vol. 1" and "Sound Reinforcement."

1979: "Microphones" Anthology is published.

1980: The Los Angeles Section inaugurates a series of interviews with historical figures, entitled "An Afternoon With..." First guest of honour is John G. (Doc) Frayne. Standards Committee (AESSC) reorganises under the chairmanship of Geoff Langdon, with Daniel Queen as secretary. American

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1916: A patent for the superheterodyne circuit is issued to Armstrong. The Society of Motion Picture Engineers (SMPE) is formed.
Edison does live-versus-recorded demonstrations in Carnegie Hall, NYC.

1917: The Scully disk recording lathe is introduced.
E. C. Wente of Bell Telephone Laboratories publishes a paper in *Physical Review* describing a "uni-

formly sensitive instrument for the absolute measurement of sound intensity" the condenser microphone.

1919: The Radio Corporation of America (RCA) is founded. It is owned in part by United Fruit.

1921: The first commercial AM radio broadcast is made by KDKA, Pittsburgh PA.

1923: Reisz microphone developed by Eugen Reisz and Georg

Neumann using carbon granules to modulate applied DC voltage.

1925: Bell Labs develops a moving armature lateral cutting system for electrical recording on disk. Concurrently it introduces the Victor Orthophonic Victrola, "Credenza" model.
This all-acoustic player with no electronics is considered a leap forward in phonograph design
The first electrically recorded 78 rpm disks appear.
RCA works on the development

of ribbon microphones.

1926: O'Neill patents iron oxide-coated paper tape.

1927: "The Jazz Singer" is released as the first commercial talking picture, using Vitaphone sound on disks synchronized with film.
The Columbia Broadcasting System (CBS) is formed.
The Japan Victor Corporation (JVC) is formed as a subsidiary of the Victor Talking Machine Co.

1928: Dr. Harold Black at Bell Labs applies for a patent on the principle of negative feedback. It is granted nine years later.
Dr. Georg Neumann founds a company in Germany to manufacture his condenser microphones. Its first product is the Model CMV 3.

1929: Harry Nyquist publishes the mathematical foundation for the sampling theorem basic to all digital audio processing, the "Nyquist Theorem."

The "Blattnerphone" is developed for use as a magnetic recorder using steel tape.

1931: Alan Blumlein, working for Electrical and Musical Industries (EMI) in London, in effect patents stereo. His seminal patent discusses the theory of stereo, both describing and picturing in the course of its 70-odd individual claims a coincident crossed- eights miking arrangement and a "45-45" cutting system for stereo disks.

EARLY BEGINNINGS

RON GODWYN
Then Head of Technical Dept. Philips Records Now Editor of IBS Journal, Line Up

ASKED FOR A SIGNIFICANT event in audio I'd say, almost without thinking, transistors, closely followed by the arrival of Rupert Neve.

In those pioneering days of the early '60s, fledgling recording studios were vying with each other for an 'edge' to attract their clients - the sound was the thing. How could you make your sound better than the next guy's? Our studio had a tube console with channel correction amounting to +/-6dB at 60Hz and 10kHz . That was it. Then along comes Rupert offering a transistorised box with a selection of switched frequencies for us to play with. A revelation! We marvelled at the sounds that could be produced.

We ordered a free standing case containing a set of six equaliser modules with six chosen HF and LF turnover frequencies and hi-pass filters. The improvement in general sound was astounding as we could now reach musical areas that had been previously denied to us. It was so successful that we ordered a much needed transistorised transportable console from Rupert for outside recording work - a 16 channel, modular set up, in two units.

The low noise, low distortion and general transparency, attracted much attention. Rupert was then commissioned to make a replacement console for the studio in about 1964, the very first of his hundreds of studio consoles. The rest, as they say is history!

The British Section was among the first AES Sections to be formed outside of North America and today it is proud to be one of the largest in terms of activities and individual membership. The section's rapid growth and development owed much to the groundwork that had been laid down decades earlier for a totally different organisation.

The 1930s was a period of experimentation and innovation. The British Broadcasting Corporation, the largest employer of audio engineers at the time, undertook extensive research programmes and set engineering standards. This was also the period when now legendary audio companies such as Tannoy, Wharfedale, Celestion, Radford, Rogers and Quad appeared, supplying audio equipment to enthusiasts.

The British record industry was also well established. The newly created EMI, formed from the merging of The Gramophone Company and Columbia, had opened its recording studios at No 3 Abbey Road in 1931 and furthered a reputation for top artists, producers and technicians.

In 1936 a group of dedicated audio professionals and amateurs formed the British Sound Recording Association. Actually more concerned with sound reproduction than recording, the BSRA benefited from the large number of audio pioneers that had emerged in this exciting period.

In retrospect it was unfortunate that the BSRA chose to merge with the British Kinematograph Society in 1964. With audio being only one of the new organisation's interests, the total coverage of the subject was reduced. A number of members chose to join the AES
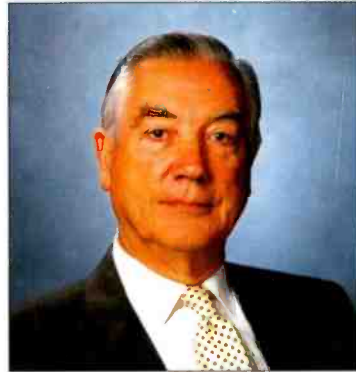
John Borwick
Fellow, AES

but to gain access to the same insights on audio topics as the BSRA had provided meant crossing the Atlantic for Conventions. Towards the end of the 1960s, three UK AES members - Percy Wilson, John Gilbert and John Borwick started laying plans for a possible British Section at the offices of John Maunder, UK distributor for Shure Bros. In early 1969 a meeting of prospective members was held to approve the idea and the details were concluded at the AES New York headquarters by Wilson and Maunder. The British section formally came into being in September 1970.

Today British Section membership stands at 700 but it began with just 50. Recruitment of Sustaining Members began early on and this now stands at 57 which helps to finance the regular meeting programme. The section has also been very keen to provide officers for the AES at all levels, including three AES Presidents.

Although London has been the location for three AES Conventions since 1975, the capital lacks suitable facilities to host European Conventions. British engineers have, however, been keen contributors to all Conventions, while British companies also form a significant percentage of the Convention exhibitors, with many of the well known names having been present for decades. A regular annual AES Conference has been successfully held in London for the last seven years drawing international attendees.

The British Section provides a regular meeting place for its members while also acting as the base for extensive international input and participation in AES activities, an attitude held positively for the past 28 years.

AES TIMELINE

National Standards Institute (ANSI) gives AESSC jurisdiction over its S4 Committee on Audio Engineering. Anthology "Disk Recording Vol.1: Groove Geometry and The Recording Process" is published.

1981: Anthology "Disk Recording Vol.2: Disk Playback and Testing" published.

1982: First AES Conference. "The New World of Digital Audio." held

in Rye, New York. Standard AES6-1982 r1992 (ANSI S4.3-1982) issued: "Method for measurement of weighted peak flutter of sound recording and reproducing equipment."
AES Gold Medal awarded to Dr Willi Studer at 71st Convention, Munich.

1984: Board of Governors gives AESSC full jurisdiction over AES standards development. Anthology:

"Loudspeakers Vol.2" is published. AESSC issues new standards: AES2-1984 r1992 (ANSI S4.26-1984) AES Recommended Practice: "Specification of loudspeaker components used in professional audio and sound reinforcement.
"AESS sets the digital sampling frequency standard. AES6 and AES7 sets standards for tape recorder flux and flutter.
At Montreux Convention, AES

encourages Philips and Sony to agree common digital standards for Compact Disc.

1985: AESSC releases AES3, digital interface standard. Fifty years of Magnetic Recording celebrated at 77th Convention in Hamburg with session dedicated to eary tape history, and a display of historic equipment

1986: Board of Governors

approves reorganisation of Technical Council with Technical Committees based on disciplines. Anthology on "Stereophonic Techniques" is published.

1988: The 85th AES Convention in Los Angeles features John T. (Jack) Mullin's historic audio equipment collection. Exhibit visited by more than 2,500 attendees. Anthology on "Time Delay Spectrometry" is published.

An "Afternoon With Jack Mullin" session is preserved on the first videotape produced for the AES Publications Department.

1991: Russian Section formed. St. Louis Section formed.
The Standard AES10-1991 (ANSI S4.43-1991) "AES Recommended Practice for Digital Audio Engineering— Serial Multichannel Audio Digital Interface (MADI)" is issued.

AUDIO TIMELINE

Arthur Keller and associates at Bell Labs in New York experiment with a vertical-lateral stereo disk cutter.

1932: The first cardioid ribbon microphone is patented by Dr. Harry F. Olson of RCA, using a field coil instead of a permanent magnet.

1933: Magnetic recording on steel wire is developed commercially.
Snow, Fletcher, and Steinberg at

Bell Labs transmit the first inter-city stereo audio programme. Eduard Schuller of AEG, Berlin develops magnetic heads with torroldal cores, a key component in the realisation of the Magnetephon.

1935: AEG (Germany) exhibits its "Magnetophon" Model K-1 at the Berlin Radio Exposition. Dr Wilheim Gaus of IG Farben (later BASF) prepares the first plastic-based magnetic tapes.

1936: BASF makes the first tape recording of a symphony concert during a visit by the touring London Philharmonic Orchestra. Sir Thomas Beecham conducts Mozart.
Von Braunmuhl and Weber apply for a patent on the cardioid condenser microphone.

1938: Benjamin B. Bauer of Shure Bros. engineers a single microphone element to produce a cardiold pickup pattern, called the Unidyne, Model S5. This

later becomes the basis for the well known SM57 and SM58 microphones.
Under the direction of Dr. Harry Olson, Leslie J. Anderson designs the 44B ribbon bidirectional microphone and the 77B ribbon unidirectional for RCA.
RCA develops the first column loudspeaker array.
Two track magnetic tape head developed for sterephonic recording by Eduard Schuller of AEG, Berlin.

1939: Independently, engineers in Germany, Japan and the U.S. discover and develop practical AC biasing for magnetic recording. Western Electric designs the first motional feedback, vertical- cut disk recording head.
Major Armstrong, the inventor of FM radio, makes the first experimental FM broadcast.
The first of many attempts is made to define a standard for the VU meter.

1940: Walt Disney's "Fantasia"

is released, with eight-track stereophonic sound.

1941: Commercial FM broadcasting begins in the U.S. Arthur Haddy of English Decca devises the first motional feedback, lateral-cut disk recording head, later used to cut their "ffrr" high-fidelity recordings.

1942: The RCA LC-1 loudspeaker is developed as a reference-standard control room monitor.

CHALLENGES FOR THE FUTURE

The AES needs to maintain its membership. As our membership grows older it is essential that we attract new members or the AES will decline. So how do we get new members? We can't look for them in the industries that existed ten years ago as those are becoming less important, as we get closer to internet broadcasting, 'video on-demand', even 'music on-demand'. There is a change coming and there are people working in these new industries that are going to make it happen. They are sound engineers, they work in audio but in fields that we are not very aware of. It is our job to make the AES known to them and to give them the opportunity of becoming members because the AES can offer them something that no other organisation can. In return the AES membership will gain valuable insight into these new audio technologies, to the benefit of every one.

Subir K. Pramanik
President AES

There is enormous interest from Eastern Europe in the Conventions. There always has been but now, with the opportunity to attend, the AES is being asked to support visa applications and make it easier for them to come. While the sections have very lively activities in their own countries, it's important that we encourage our Eastern members to become part of the AES 'family' - otherwise they are just subscribers to the Journal.

Han Tendeloo
104th Convention Chairman

There are two distinct events that were important in creating the European AES. The first was a special technical session at a US Convention in 1968, organised by Johan Ooms of Philips Phonographic Industries, going under the name of 'Audio Abroad'. It helped to raise the awareness within the AES of what was happening in Europe. The second was the first Convention in Cologne, 1971. This was the first stop in gaining the AES entrance to continental Europe. That first Convention was almost a 'family' affair and from that beginning it has developed into a large industry-wide show which closely parallels the way that the US Conventions developed. We must expand that family and ensure it remains welcoming.

Peter K. Burkowitz
Fellow, AES

HARRY THUMAN
Studio owner / engineer

MY SIGNIFICANT MOMENT was the arrival of the Moog synthesiser. It ability to make a synthetic sound that one could manipulate had a powerful affect. I was just a kid and had been learning to play drums but it changed my life completely because from that day on I knew that was what I wanted to do. I imported the first ARP 2600 synthesiser in Germany, and then Moogs. It all started there.

I had realised that sound was no longer just about reproducing audio that has happened somewhere; sound was something that we could modify and manipulate, and out of that would come incredible new things.

I didn't stop liking real instruments. I still had a career as an engineer and producer where the Moog was something that I'd use only if I needed it. But the idea of control voltages had entered my head, and their more modern equivalents are still there. The idea of integrating synthesisers with the studio - mixers, processors and tape machines - to be more creative was important.

The tools to create music are now readily available - years ago that APR 2600 cost me DM 17,000 and you still needed a 2-track costing only a little less. Now you get these kind of facilities for a fraction of that. It was Moog who kicked that off.

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he AES-11d-1991 AES Information Document is published: "Plane wave tubes: design and practice." The AES11-1991 AES Recommended Practice for Digital Audio Engineering is published: Synchronisation of digital audio equipment in studio operations." The AES15-1991 (ANSI S4.49-1991) is published "AES Recommended Practice for Sound-reinforcement systems — Communications

Interface (PA-422)."

1992: Long-awaited AES14-1992 (ANSI S4.48-1992) issued: "AES Standard for Professional Audio Equipment —Application of connectors, part 1, XLR-type polarity and gender." (Pin 1 shield, Pin 2 high, Pin 3 low). The AESSC establishes a secretariat-level liaison with International Electrotechnical Commission (IEC) Technical Committees.

1993: Czech, Slovenia and Croatian Sections formed. Berlin Convention celebrates 50 years since first stereo recording by awarding honorary membership to Helmut Kruger, and issues CD of original recordings

1995: AES24-1-1995 standard protocol for sound system control issued. AESSC is awarded an Emmy for the AES3 Digital Interface Standard.

1996: The 100th Convention held in Copenhagen. Donald Plunket's address singles out pioneering work of Joe Ooms of Philips that sparked AES activities in Europe, work carried on by Herman Wilms and Titia Bakker. Three anthologies published: "Sound Reinforcement 2," "Loudspeakers Vol.3: Systems Crossover Networks" and "Loudspeakers Vol. 4: Transducers, Measurement and Evaluation."

1997: AES/SC establishes official Category-D liaisons with IEC TC 100 subcommittees, raising possibility of joint IEC-AES standards. The 14th International Conference, internetaudio. aes.org held in Seattle. For the first time. sessions streamed live on the Internet. The 103rd Convention: "AES Goes Gold, Celebrating 50 years as the Global Forum for Professional Audio," held in New York City. Beginning a year of special events that will trace its history and anticipate an exciting future for the Society.

1998: Golden Anniversary celebration held in New York on March 11, the exact date of the first AES meeting in 1948, with ten of the original members present. ∎

Dr. Olson patents a single-ribbon cardioid microphone (later developed as the RCA 77D and 77DX), and a "phased-array" directional microphone.

1943: Altec develops its Model 604 coaxial loudspeaker. The first stereo tape recordings are made by Helmut Kruger at German Radio in Berlin.

1944: Alexander M. Poniatoff forms Ampex Corporation to make electric motors for the military.

1945: Tape decks are sent back to the US in pieces in multiple mailbags by Army Signal Corps Sergeant John T. (Jack) Mullin.

1946: Webster-Chicago manufactures wire recorders for the home market. Brush Development Corp. builds a semi-professional tape recorder as its Model BK401 Soundmirror. 3M introduces Scotch No. 100, a black oxide paper tape.

Jack Mullin demonstrates "hi-fi" tape recording with his reconstructed Magnetophon at an IRE (Institute of Radio Engineers) meeting in San Francisco.

1947: Colonel Richard Ranger begins to manufacture his version of a Magnetophon. Bing Crosby and his technical director, Murdo McKenzie, agree to audition tape recorders brought in by Jack Mullin and Richard Ranger. Mullin's is preferred, and he is brought back to

record Crosby's Philco radio show. Ampex produces its first tape recorder, the Model 200. Major improvements are made in disk-cutting technology: the Presto 1D, Fairchild 542, and Cook feedback cutters. The Williamson high-fidelity power amplifier circuit is published. The first issue of *Audio Engineering* is published; its name is later shortened to *Audio*.

1948: The Audio Engineering Society (AES) is formed in New York City. The microgroove 33-1/3 rpm long-play vinyl record (LP) is introduced by Columbia Records. Scotch types 111 and 112 acetate-base tapes are introduced. Magnecord introduces its PT-6, the first tape recorder in portable cases.

1949: RCA introduces the microgroove 45 rpm, large-hole,

7-inch record and record changer/adaptor. Ampex introduces its Model 300 professional studio recorder. Magnecord produces the first U.S.-made stereo tape recorder, employing half-track staggered-head assemblies. A novel amplifier design is described by Macintosh and Gow.

1950: Guitarist Les Paul modifies his Ampex 300 with an extra preview head for "Sound-on-

SETTING DIGITAL STANDARDS

KEN TOWNSEND
*General Manager
Abbey Road Studios
(retired)*

THE ARRIVAL OF THE MULTITRACK
tape machine was the biggest influence on the way that the studio operated. The move from stereo machines to 4-track changed the whole pattern of recording. Before that everything was virtually live, both pop and classical. The balance engineers were 'true' balance engineers who got the mix right at the time. Multitrack meant that engineers who weren't all that good at mixing could actually lay down the tracks and, if necessary, it could be fixed later.

As larger machines with more tracks followed the situation changed even further.

What multitrack did was introduce 'inefficiency' into the recording process. Mono to stereo had been a straightforward change: you just had two tracks rather than one, but it was still live. Four-track changed all that because overdubs could be done at different times. For example, on a mono or 2-track pop session you expected to record four titles in a 3-hour session. With 4-track you'd be lucky if you recorded those in four days! From there onwards it went downhill rapidly. Of course studios would sell more studio time and in turn, there was a need for more studios because the extra time requirements of multitrack meant that there wasn't enough capacity.

Work on digital standards was prompted by the increasing number of demonstrations of digital audio equipment from the mid-70s onwards, all using different values of linear quantisation and sampling frequency. To address many of the issues created by this situation the AES Digital Audio Standards Committee was formed with its first meeting in 1977. This initiative and the successful implementation of several well supported meetings resulted in the AES being given responsibility for generating digital audio standards by all the other US organisations who had interests in the subject.

Work on digital standards continued despite a problem with the American Anti Trust laws that resulted in major changes to the organisational structure and working practices of the AES technical committees. This structure did function as well as had been expected and the present organisation was introduced in 1984 to give a clearer focus.

It wasn't until 1981 that the controversial topic of sampling frequency was agreed on by a compromise, recommending a choice of frequencies for differing applications - 48kHz for professional applications but recognising 44.1kHz for consumer use and 32kHz as a widely used broadcast standard. This became AES standard AES5-1984, and still remains controversial.

Work on the digital audio I/O interface was published in 1985 as AES3. It was the first AES standard to be developed in conjunction with the EBU and became known as the 'AES/EBU interface'. The uptake of the standard by both manufacturers and users was

J. P. Nunn
*Chairman of
AES Standards Committee*

very quick, demonstrating the real need for this standard.

The AES/EBU interface has proven very resilient although there have been several revisions and amendments published, adding clarification and extensions. AES3 has been in use for over thirteen years with no sign of it being replaced, in fact further development is more likely. The importance of the AES/EBU interface was celebrated in 1995 by the awarding of a Technical Emmy to the AES and EBU by the National Academy of Television Arts & Sciences.

The AES is not generally involved in standards relating to consumer digital audio products except where the resulting products may have applications in the professional field, or where there may be a resultant effect upon the production process. It therefore monitored but was not involved in the standardisation of a consumer digital audio interface which was handled by the IEC. AES3 was later incorporated into the same IEC standard to reduce the confusion being generated by the two similar but different standards.

The 1990s have seen the development of other digital standards - the AES10 serial multichannel interface (MADI), AES11 concerning synchronisation of digital equipment, and AES17, the digital audio measurement standard.

While digital audio standards may have been more visible, it would be wrong to forget other AES standards work such as that on peak programme meters, instrumentation, disc recording, audio polarity, architectural audio, sound reinforcement, transducers, forensic audio, listening test standards and sound levels - all making the industry function more easily.

INNOVATOR
Georg Neumann

There are few names that have survived intact from the very earliest days of radio and sound recording equipment innovation to the present quite like that of Georg Neumann. Although Neumann himself died over twenty years ago his influence is just as strong in modern microphone design - both in the products that bear his name and in many microphones from other manufacturers.

Neumann first made his mark working for Eugen Reisz, the principle result of their collaboration being the Reisz Microphone. This was an arrangement of carbon granules set in a block of marble which modulated an applied external 12v DC voltage. The whole arrangement was suspended on springs within a large steel ring and found favour with Berlin-based German national radio from its opening in 1923. Although it boasted a linear frequency response between 50Hz and 1kHz it suffered from a gradually rising HF response to approximately +15dB at 10kHz. but it is recognised as an historic piece of engineering.

Reisz and Neumann were soon to part company as Neumann had differing ideas about how to develop the technology and went on to found the company which bore his name in 1928. Neumann wanted to mass produce condenser microphones but there was considerable scepticism about his chances of success as they had previously only been made in laboratory conditions.

Perhaps the real reason for his split from Reisz was his growing interest in the record making process, and the receipt of a commission from contacts in England to build a machine for cutting discs. It is not realised that Neumann was equally active in several fields from the very start of his new company.

His first mass produced condenser microphone, the CMV3 was launched in 1928 and was quickly nicknamed the 'Neumann bottle' due to its unusual shape. This was far superior to the Reisz mic and became the German standard for studio and broadcast use, hardly changing until the end of WWII.

Development on the disc cutting machines continued and by 1930 Neumann had already made the transition from the belt drive of the early designs to direct drive. Further diversification of research generated a range of products including electro-acoustic measurement equipment with specialist microphones, and cinema gongs. The development of the Pistonphone allowed very accurate calibration of all types of microphone using optical techniques.

The war disrupted Neumann's activities, bombing causing them to move production from Berlin, not returning for several years. 1947 was a very prolific year for Neumann - one of the less known sides of his research made the Nickel Cadmium battery a practical device. He developed a design for NiCds that didn't produce large amounts of oxygen and therefore could be

sealed. To many, this is Neumann's most important achievement.

1947 was also the year of introduction of the U47 condenser microphone. It was unique because is used a double diaphragm capsule where each diaphragm could be polarised independently with respect to the central plate. The result was the first switchable pattern condenser microphone offering omni and cardioid characteristics.

In 1949 Neumann obtained a patent for the first remotely controllable pattern microphone with the M49, while the M50 introduced a year later used a pressure capsule in a acrylic sphere to create a very even omni pattern.

1953 saw the first move away from the large diaphragm designs with smaller microphones for TV applications, while 1956 saw the SM2, an early stereo microphone that was unique for many years.

Georg Neumann died in 1976 - he would have been 100 this year if he'd lived. The company that bears his name was acquired by Sennheiser in 1992 but even a casual glance at current Neumann designs, and those of numerous competitors reveals a continuing influence that has retained a popularity with users for 70 years.

INNOVATOR
Mark Crabtree

"Most people wanted to use the AudioFile to replace existing technology. It was an uphill struggle to explain that it could do much more than that."

The AudioFile, the first hard disk-based editing system, is perhaps the product that Mark Crabtree is best known for. However, as a digital audio designer, his track record is impressive - the first full bandwidth digital delay line, first full bandwidth digital reverb, the first sampler and one of the earliest practical digital consoles.

An early interest in music and electronics led to an engineering degree at the University of Cambridge. The industrial experience components of the course took him to a division of Lucas Aerospace making test equipment for the military aircraft, where "they gave me some Intel microprocessors to play with and let me loose on their mini computer."

Following an MSc in digital electronics, Crabtree joined Lucas full time, where he designed a modular automatic test equipment suite of cards using parallel techniques to replace the mini computer system in use. He was keen to see what was ground breaking technology in 1974 - colour graphics, digital routing matrices etc, find an external market but Lucas declined.

His thoughts returned to musical interests and the possibility of developing a solid state replacement for the Watkins Copicat tape-loop echo unit beloved of musicians in the 60s and 70s. Noise restrictions with the BBD halted that direction but the research did generate ideas for tape phase simulation using the right combination of analogue and digital electronics. Prestigious sales to Abbey Road and AIR Studios of the DM2-20, the first product under the AMS name led to a new career direction. *Continued*

COLIN SANDERS
Founder, Solid State Logic

ALTHOUGH THERE IS MUCH today to celebrate for the AES it is also sad to reflect that the industry has recently lost one of its audio pioneers in a helicopter crash earlier this year. Colin Sander's concept of expanding the accepted range of processing facilities found on a studio mixing console, making it the central controller for the complete studio, and then adding a computer to handle an increasing amount of the control and automation, seems a totally logical approach these days.

In 1969, he formed Solid State Logic to handle this control system of which there are now tens of thousands in use.

The company's first console was shown at the 1977 Paris AES Convention, gaining much praise and four orders. Development, research and feedback continued leading to the unveiling of the 4000E Master Studio System in 1979, triggering the start of a major change in the recording studio industry. It was very different from other consoles but its functionality had a natural appeal to engineers and producers.

Colin Sanders was a quiet man with a very sharp mind and a need to retain a hands-on approach to his company and their products. With its rapid growth this became less possible and from the late 1980s he took less interest in the business, severing his connections with SSL in 1991. However having radically changed one industry, he threw himself into a host of other non-audio technologies, applying the same creative enthusiasm and drive to these as he had to audio.

1970: The first digital delay line, the Lexicon Delta-T 101, is introduced and is widely used in sound reinforcement installations. Ampex introduces 406 mastering tape.

1971: Denon demonstrates 18-bit PCM stereo recording using a helical-scan video recorder. RMS and VCA circuit modules introduced by David Blackmer of dbx.

1972: Electro-Voice and CBS are licensed by Peter Scheiber to produce quadraphonic decoders using his patented matrixes.

1974: D. B. Keele pioneers the design of "constant-directivity" high-frequency horns. The Grateful Dead produce the "Wall of Sound" at the San Francisco Cow Palace, incorporating separate systems for

vocals, each of the guitars, piano and drums. 3M introduces Scotch 250 mastering tape with an increase in output level of over 10 dB compared to Scotch 111. DuPont introduces chromium dioxide (CrO2) cassette tape.

1975: Digital tape recording begins to take hold in professional audio studios. Michael Gerzon conceives of and Calrec (England) builds the "Soundfield Microphone," a coincident 4-capsule cluster with matrixed "B-format" outputs and decoded steerable 2 and 4- channel discrete outputs. EMT produces the first digital reverberation unit as its Model 250. Ampex introduces 456 high-output mastering tape.

1976: Dr. Stockham of Soundstream makes the first 16-bit digital recording in the U.S.

at the Santa Fe Opera.

1978: The first EIAJ standard for the use of 14-bit PCM adaptors with VCR decks is embodied in Sony's PCM-1 consumer VCR adaptor. A patent is issued to Blackmer for an adaptive filter (the basis of dbx Types I and II noise reduction). 3M introduces metal-particle cassette tape.

1980: 3M, Mitsubishi, Sony and Studer each introduce a multitrack digital recorder. EMT introduces its Model 450 hard-disk digital recorder. Sony introduces a palm-sized stereo cassette tape player called a "Walkman."

1981: Philips demonstrates the Compact Disc (CD). MIDI is standardised as the universal synthesiser interface. IBM introduces a 16-bit personal

computer.

1982: Sony introduces the PCM-F1, intended for the consumer market, the first 14-/16-bit digital adaptor for VCRs. It is eagerly snapped up by professionals, sparking the digital revolution in recording equipment. Sony releases the first CD player, the Model CDP-101.

1983: Fiber-optic cable is used for long-distance digital audio transmission, linking New York and Washington, D.C.

1984: The Apple Corporation markets the Macintosh computer.

1985: Dolby introduces the "SR" Spectral Recording system.

1986: The first digital consoles appear. R-DAT recorders are introduced in Japan.

Dr. Gunther Theile describes a novel stereo "sphere microphone."

1987: Digidesign markets "Sound Tools," a Macintosh-based digital workstation using DAT as its source and storage medium.

1990: ISDN telephone links are offered for high-end studio use. Dolby proposes a 5-channel surround-sound scheme for home theater systems. The write-once CD-R becomes a commercial reality. 3M introduces 996 mastering tape, a 13 dB improvement over Scotch 111.

1991: Wolfgang Ahnert presents, in a binaural simulation, the first digitally enhanced modelling of an acoustic space. Alesis unveils the ADAT, the first "affordable" digital multitrack

recorder. Apple debuts the "QuickTime" multimedia format. Ampex introduces 499 mastering tape.

1992: The Philips DCC and Sony's MiniDisc, using digital audio data- reduction, are offered to consumers as record/play hardware and software. The Nagra D is introduced as a self-contained battery-operated field recorder using Nagra's own 4-channel 24-bit open-reel format.

1993: In the first extensive use of "distance recording" via ISDN, producer Phil Ramone records the "Duets" album with Frank Sinatra. Mackie unveils the first "affordable" 8-bus analogue console.

1994: Yamaha unveils the ProMix 01, the first "affordable" digital multitrack console.

GEOFF EMERICK

Engineer, Producer

I'M NOT SURE THAT there have been any significant technologies that have really changed the way I work. I'm basically using the same techniques that I used on 'Pepper' (Sgt) and 'Revolver'. The arrival of more tracks changed things, but was it for the better? I guess it was.

Previous to that you had to do the mixing process at the time of recording. When we were laying the drums, bass and maybe a guitar on one track you had to set all the echoes and all the finished EQs. So in a way it was even harder especially when you were dealing with huge orchestras, you had to get all the sounds and mix it to stereo.

Our efforts were aimed at capturing the musical moment, whereas so often we are now not capturing that moment and then have to spend a year trying to recapture it. And often we miss it altogether.

"We kept being asked for digital delay lines but found that the available 12-bit converters were not good enough and we had to come up with a way of expanding their dynamic range to effectively give us a 90dB dynamic range. It never occurred to us to restrict any of the performance parameters"

Following his modular systems work at Lucas, Crabtree designed the DDL architecture around an expandable microprocessor controlled bus. Using the first generation of memory cards up to a second could be achieved, increasing as larger chips became available, culminating in a special capable of 26 seconds of stereo delay, for editing sections of songs.

The versatility of the concept allowed the development of digital reverb and pitch changers around the basic design. But is was almost by accident that another feature became important. On the front panel was a Lock-in switch that stopped audio playing into the RAM and put it into an infinite loop.

"It was just a nice effect - sort of usable but the ends clicked and it wasn't controllable. I remember a late night call from Frank Zappa who was determined to use this as a performance instrument. People wanted to be able to trigger it, and later, trigger it with audio." So was created the first sampling unit.

Awareness of the possibilities of electronic editing and the volatility of RAM meant a search for a suitable storage medium. The first AudioFile hard disk system was delivered in 1985, but the concept had been created on paper for some years just waiting for the right drive to slot into the design and create a practical product.

"Right from the start this technology allowed you to place audio wherever you wanted it. When the AudioFile was first shown, the industry was naive about hard disk systems and early showings were educational rather than sales orientated." At the same time they were learning how it could be used to create new opportunities in audio tasks from mastering to audio post production.

There was also the need to create the icons and terminology of hard disk recording - many of the things now taken for granted in our familiarity with such systems.

"A lot of what in retrospect is obvious, like the moving tape graphic representation and the imaginary replay head - we had to carve all those things out of solid rock."

I N N O V A T O R

Willi Studer

"When I look back on my life, it really has been a series of lucky breaks..... I can't don't even know whether I really deserved them." The words of Dr Willi Studer taken from an interview published in *Studio Sound* in 1991, shortly after retiring from active involvement in the company he'd founded 43 years earlier.

They were fairly typical of the understated but confident manner of Willi Studer - an approach that he ensured ran throughout his company. Visitors would see products being built and assembled with maximum precision and attention to technical detail. Typically Swiss you might think, and you'd be correct, but it went further as Willi Studer kept a very tight control on all aspects on the company so that it reflected his priorities, everywhere.

As with most technically creative people, his interest in engineering began young. At 12 years old he built a simple telephone link but was ordered to take it down by the Swiss Post Office as it didn't meet official regulations! Aged 15, he took a two year apprenticeship which before starting to work in several shops servicing and installing domestic radios. There was little test equipment and all repair work was by what Studer called "methodical troubleshooting".

This grounding saw Studer, just 19, designing and building radios for other companies. This continued with varying degrees of commercial success until 1948 when he found a company willing to distribute an oscilloscope he'd designed.

These were to be the first products of the Willi Studer company.

A year later he was approached by an importer of American tape recorders to convert and test them for European use. This inspired Studer to develop his own design of domestic tape recorder. By the year's end, the first models were on sale under the name of Dynavox.

Work then started on a professional recorder but there needed to be new arrangements for manufacture as recorder heads were still baked in the oven at home. Trials of the transportable Studer 27 Professional Recorder began in 1951 with Swiss Radio. In the same year Studer acquired the rights to use the name 'Revox' and a range of domestic radio/tape recorders was designed.

The Studer 27 proved successful and the professional side quickly developed through the A37 to the C37 by 1960, which had far sightedly been mechanically designed to form the basis of future multitrack recorders. The first of these was the 4-track J37, one of which was supplied to Abbey Road Studios in 1967 to meet the demands of the Beatles recording 'Sgt Pepper'.

1954 saw the introduction of the G36 domestic recorders which were already equipped with a 3-motor drive. The availability of stable silicon transistors triggered the development of the 77 series in 1967 which found a home in virtually any situation where a small recorder was required.

The professional tape recorder products developed through several generations and found homes in studios worldwide. Today Studer are one of the few remaining manufacturers still committed to analogue tape recorders.

Dr Studer's contributions to the development and refinement of the tape recorder were recognised in 1982 when he was awarded the Gold Medal of the AES during the Montreux Convention. This coincided with the early showings of Studer's first digital multitrack recorder, the product of an ambitious digital R&D programme, directed by Studer, which resulted in it being the only European company capable of building a digital tape recorder of its own design.

Willi Studer died in 1996 aged 83. His interests in the technology and development of his company never faltered even to the extent of still walking several miles to unlock the Studer building very early in the morning well into his 70s, something that few others would emulate.

AUDIO TIMELINE

1995: The first "solid-state" audio recorder, the Nagra ARES-C, is introduced. It is a battery-operated field unit recording on PCMCIA cards using MPEG-2 audio compression.
Iomega debuts high-capacity "Jaz" and "Zip" drives, useful as removable storage media for hard-disk recording.

1996: Record labels begin to add multimedia files to new releases, calling them "enhanced CDs."
Experimental digital recordings are made at 24 bits and 96 kHz.

1997: DVD videodiscs and players are introduced. An audio version with 6-channel surround sound is expected to eventually supplant the CD as the chosen playback medium in the home.

1998: The Winter Olympics opens with a performance of Beethoven's "Ode to Joy," played and sung by synchronising live audio feeds from five continents with an orchestra and conductor at the Olympic stadium in Nagano, Japan, using satellite and ISDN technology. ∎

On your way, Jack

Modern interconnection schemes are in confusion, and the continued use of out-dated standards is an unnecessary complication. **Philip Newell** calls for the death penalty for the jack plug

SURELY, THE TIME has now arrived for mono jacks, and RCA jacks—or phono plugs, to be British about it—to be abandoned as audio connectors for equipment intended for studio use. In a simple, serially connected hi-fi system, where only a left and right signal path is the norm, and where interconnect cables are kept short, unbalanced interfaces are usually perfectly satisfactory. However, even in the simplest of domestic studio setups, the existence of multiple and parallel signal paths (for reverbs and delays, for example) can make the optimisation of grounding systems a nightmare.

To ensure that a system is free of hums, buzzes and radio frequency interference which can all compromise the transparency of a sound, even at low levels, a well-conceived grounding (earthing) regime is virtually obligatory. The fact that some people 'get lucky' and end up with a hit and miss hook-up that is trouble free does not detract from the general requirements of good engineering practice.

Where jacks must be used for flexibility or convenience, the 3-pole 'stereo' variety can still be used by mono jack plugs if the input and output circuitry design has given due consideration to this need. Only by allowing separate access to chassis and signal grounds can complex hook-ups be optimally interfaced. Two-pole connectors cannot provide this very necessary access, and hence have no place on any equipment to be considered a serious choice for studio use.

The use of 2-pole connectors leads the inexperienced into all manner of nonsenses, especially when wiring 3-pole jackfields (or patchbays, in US jargon). When a 3-pole, balanced input or output is being connected to a 3-pole jackfield, the connection convention is more or less universally understood by all people with even a smattering of knowledge of recording: hot to tip, cold to ring, and ground to sleeve. However, in an alarming number of cases, I came across people installing equipment in studios who seem to think that the 2-pole, unbalanced outputs should simply omit the cold (ring) connection. That is, they connect the tip of the mono jack to the tip of the 3-pole jack, and the screen of the mono jack to the screen of the 3-pole jack. This is absolutely incorrect practice.

Some people seem to get away with the above nonsense by virtue of the equally nonsensical connection of other pieces of equipment, although their noise floors may be compromised. The error of their ways will be abruptly pointed out to them if they try to patch in a piece of equipment with transformer balanced, floated (ground-free) inputs and-or outputs. Only silence will be forthcoming, as the open circuit ring connection will provide no return path for the non-ground-referenced signal.

When wiring a 2-pole in or out to a 3-pole jackfield, even if single screened (2-conductor) cable is used, the signal return must go to the ring and not the sleeve. This may look rather odd with the screen (shield) of the cable attached to the ring connection, but that is what is necessary. If, however, 3-pole connectors were used on all equipment, balanced or unbalanced, then even to the uninitiated, the wiring arrangement to the jackfield would be much more obvious.

Manufacturers of studio equipment are all too ready to promote the 'professional' nature of their equipment (though much of it clearly does not deserve such a description) yet in their search for every cent of profit, they do terrible disservices to their customers, who they frequently do not treat professionally. Perhaps they should begin worrying a little more about this situation, because to promote something for studio use while failing to ensure that the equipment interfaces comfortably into a typical studio system, could leave them open to legal action for selling equipment which is unsuitable for the purpose for which it is being advertised. Such laws exist in many countries around the world, so the battles could extend far and wide. In fact, some enterprising lawyer could probably have a field day, and make a fortune, just from pursuing this type of case. The recording industry is awash with violators of these laws.

The time has come for many manufacturers to stop merely bench-testing their equipment, in the isolation of a test room, and to witness—as I do all too often—the frustrated attempts, of studio owners and operators, to wire a noise-free system. Calls to the appropriate manufacturers or dealers for help, almost invariably brings the response that it must be a problem with the other piece of equipment, of different manufacture, to which it is connected. Sometimes, noise-free systems cannot be achieved, yet countless hours are wasted by people, driving themselves to distraction, before discovering this fact. When I tell them that the fault lies not with them, but with the manufacturers, they often refuse to believe me, because to them, the manufacturers are revered like gods. It is time that many manufacturers began to treat their customers with the respect that they deserve in return for their loyal support, and not to treat them like idiots and fools. A good starting point would be a generally better attempt at standardising interface protocols, and the abandonment of the use of 2-pole jacks. ■

> The fact that some people 'get lucky' and end up with a hit and miss hook-up that is trouble free does not detract from the general requirements of good engineering practice