Studio sound

INSERT
DYN
AUTO
PEAK
SAFE (SIP)
FDR CALL
GROUP CNTRL
AUTO
SELECT
METER READS
INPUT
TAPE
GROUP

L/R   CENTER   SRND   SW   L2/R2   RTNG   JOY1   JOY2

CAL   SAFE(SIP)   GROUP SETUP
UPPER

LCR   ST-SRND   SAFE(SIP)   GROUP SETUP
LOWER   PAN

INPUT   TAPE/PB   GROUP/DIR   PEAK/VU
METER
READS

MODULE SELECT

OCTAGON CONTROL
MODULE

EXCLUSIVES

D&R Octagon
CEDAR for Pro Tools
Digitech Studio 400
AD MIDA System-4
BSS Opal DPR422
Sony CDP-D500
dbx Blue 160S
Roland VS880
Nagra ARES-C
EL Distressor
Symetrix 606

SOUND GENERATION
Posting DS9 and Voyager

The
KEITH
OLSEN
Interview

IN THE MIRE AT GLASTONBURY
POLISH RADIO EXPANDS
REVERB: THE GUIDE
192KHZ IS HERE

08

9 770144 594024

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Polskie Radio S.A. **81**

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25

INFINITY

AL timix

Solid State Logic

Give them time

HAVE YOU EVER WONDERED would have become of Jimi Hendrix had he lived through to middle age and beyond? Would he have continued to develop along the same lines, would the draw of film scores have proved too strong or would he have been relegated to the occasional walk-on jam session with the guitar-greats circus? Similarly, what would Keith Moon and John Bonham have made of their lives and what of Jimmy Miller, Alex Sadkin, Chas Chandler, Joe Meek or the late and great Frank Zappa?

While they had all achieved much, none were anywhere near the point of throwing in the creative towel, leaving us all to wonder what might have been. The tragedy of their deaths is equalled by the fact that their experience, character, ingenuity and brilliance has exited.

Equally tragic is the waste and demotivation of the young who are entering our industry. Those that have something to prove are as valuable as those that have already demonstrated a contribution. As if the road to studio, live, broadcast and post employment is not narrow and fraught enough with obstacles, newcomers frequently have to contend with suspicion and patronising derision.

For a business that is still so young at heart this is peculiar. Do you not feel immediately taken aback by the innocence of the type of questions you can be asked by a student? Do you dismiss them politely for fear of being locked into what could be an interminable conversation that ultimately leads to no personal gain? Or do you rise to the occasion spurred by the enthusiasm of your audience?

Each of us has the enormous responsibility of presenting pro-audio as an exciting and welcoming community. Exciting is easy—it's an exciting business and you're preaching to the freshly converted—it's the welcoming that has to be worked on. We owe them the time. Remember the dead but praise the living, they are the future.

Zenon Schoepe, executive editor

The art of agreement

A NUMBER OF RECENT ISSUES covered in *Studio Sound* have conspired to throw up a consistent theme. In part, this theme has less to do with technical considerations or pieces of equipment than it has to do with people and business—although it has its roots in technical progress. However you categorise it, should it go unaddressed, I feel certain that it will hinder us as surely as any technical problem.

In this issue of *Studio Sound* you will find several of the pieces of evidence: features on surround sound monitoring, high-sample-rate recording and an 'Open Mic' column on open file interchange. In recent issues you'll find more evidence concerning file interchange, miking techniques for surround recording, and high sample rates. The common thread is that of agreement—standards, compatibility, call it what you will—or the lack of it.

Without discussing the details of each case, it is enough to say that all carry contentions of varying nature and importance. Some can be seen as matters of personal or artistic choice while others are more fundamental. Some will take a while to resolve, while others are already quite urgent. All are important.

The thread that links the placement of microphones for surround recordings with DAW file compatibility and the DVD specification with studio control room design, is that we all rely on other people and other systems for our work to be useful. And while it might seem attractive to a particular manufacturer to corner some part of a market, or for some studio to build a 'superior' monitoring system, history tells us that the real gains are to be made through cooperation and consensus. And while collaborations necessarily mean compromise (file exchange) they are not necessarily preclusive (high sample rates for DVD).

The bottom line here is that if we want to move forward, we're going to have to get used to making decisions—together.

Tim Goodyer, editor

Miller Freeman
A United News & Media publication

www.americanradiohistory.com

Decca closes Recording Centre

UK: The Decca Record Company has announced the closure of its postproduction facility in Kilburn, North London. The Centre has been one of London's leading recording studios for decades and in recent years it has been a unique editing and mastering centre for the classical label's prestigious projects.

In a statement from the record company, the move is described as part of a number of 'restructuring' initiatives to reduce costs. The statement reads, 'Decca is the last classical record company in the UK to run such a facility for its sole use. For it to work effectively and efficiently in the future, it would require a disproportionate amount of investment relative to the needs of the classical market.'

The statement then emphasises that Decca's location recording activities will continue at its regular venues including the Konzerthaus in Vienna, Walthamstow Hall in London and the Concertgebouw in Amsterdam, and that it will continue to employ a team of technical producers based in London.

Six employees are to be offered transfers to Decca's Chiswick, West London, offices, while the remaining 30 employees from Kilburn are to receive severance and freelance packages. It has not yet been revealed how the equipment in the Recording Centre will be relocated or dispersed. **Phil Ward**

Harman buys out Amek

INTERNATIONAL: The Harman Pro Group has bought the remaining outstanding shares in Amek Systems and Controls Limited. Philip Hart, president of the Harman Pro Group, described the move as 'exciting' as it allowed the combination of designers, Graham Langley from Amek, and Graham Blyth from Soundcraft, plus a continued association with Rupert Neve who designed Amek's 9098 analogue superconsole, and the 9098 series of outboard processors. However, former Amek chairman Nick Franks, who cofounded the company with designer Graham Langley, has left the company.

'Amek has an outstanding reputation in the marketplace and will provide Harman with a range of high quality performance consoles that will complement the product offerings from Soundcraft, Studer and Allen & Heath,' says Hart.

Console manufacturer Amek sold a 30% share of the company to AKG at the beginning of the decade and this was inherited by Harman when it bought AKG in 1993.

Cine Expo

AMSTERDAM: Cine Expo is the annual meeting place for the European cinema exhibition industry. Here, between 29th June and 2nd July, cinema owners and personnel were able to get up to date with what is new, and see previews of forthcoming films.

For the event, the main auditorium at the RAI convention centre was transformed into a large cinema—allowing the audience to hear the new JBL 5000-series 3-way cinema loudspeaker system which

▲ **US: A new 3-studio recording complex has recently opened in Salt Lake City, Utah. Conceived by partners Gianni Skolnick and Blair Sutherland, and constructed by LA-based design team studio bau:ton, Counterpoint Studios caters for differing scales of recording project from the 750-ft² recording area of Studio A to the MIDI composing room in Studio C—with post and multimedia services also on offer. studio bau:ton, US. Tel: +1 213 251 9791.**

was installed in the 5-channel screen system. All the Expo's screenings were presented with digital soundtracks, and all three main formats were represented, although the majority of films were screened in Dolby SR-D. However, it was evident that more and more films are being released in multiple formats in order to cater for varying cinema installation requirements.

On the equipment side, and in addition to JBL's 5000-series speakers, there was a compact range of screen and subwoofer systems on show from Electrovoice and some new systems —large and small—from EAW. Stage Accompany also created a lot of interest with its cinema system which uses a ribbon HF transducer in line with its sound-reinforcement systems.

The digital age is certainly making inroads into cinema sound in all areas, and the three manufacturers of digital soundtrack formats are now offering complete systems from decoder to amplifier inputs. The latest processor from Dolby, the CP500 has already been out for some time and is now joined by a decoder-processor from Sony SDDS, with preproduction version of new 6AD from DTS also being shown. All systems offer crossover functions, extensive EQ facilities and so on, with the SDDS unit also having the possibility of networking to auditoriums as required.

Also worthy of mention is the Smart Panastereo, a digitally controlled analogue cinema processor with digital interfaces for both Dolby SR-D and DTS and offering full audio processing facilities.

From the level of activity, it's safe to conclude that the cinema business is definitely not in the doldrums and expansion is taking place rapidly for Middle and Eastern Europe as well as other 'emerging territories'.

The advent of digital sound as a standard for most films is bringing its attendant wake of problems (such as over-loud soundtracks), but hopefully these will be sorted out once the producers understand both the advantages and limits imposed by the new technologies.

Dolby DVD disc

US: Dolby Laboratories, together with an independent classical music label from California, Delos International, has announced the release of the first audio-only DVD disc. Entitled *DVD Spectacular*, it features musical selections in Dolby Digital 5.1-channel sound, including performances of Tchaikovsky by the Dallas Symphony Orchestra, and Carol
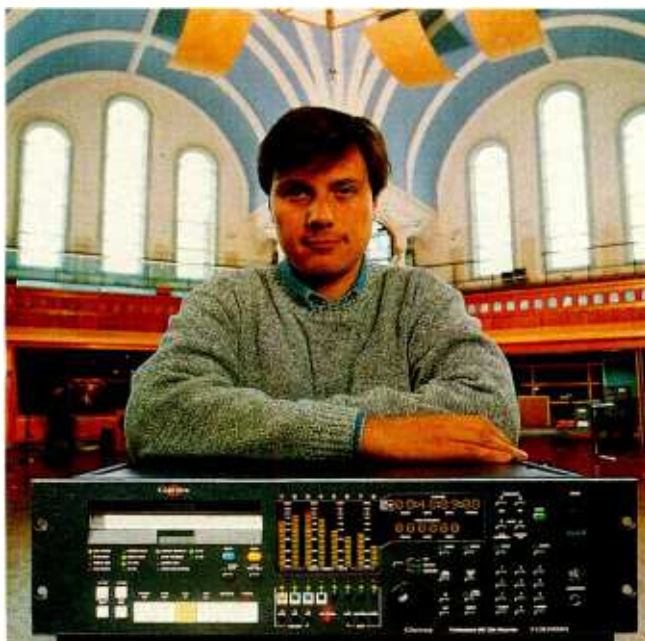
◀ **Genex at Air**

UK: Accompanied by the comment 'We can relax in the knowledge that the sound quality of the Genex is vastly superior to that of all the tape-based digital 8-track recorders' from chief engineer Geoff Foster comes news of Air Studios' purchase of a Genex GX8000 MO recorder. The sale was secured by HHB who has recently relocated it US operation to Santa Monica under the control of David Beesley. Air has also recently taken delivery of a 16-track Pro Tools system with 888 I-O and a selection of TDM plug-ins.
Air Studios, London.
Tel: +44 171 794 0660.
HHB, US. Tel: +1 310 319 1111.
Fax: +1 310 319 1311.
Digidesign, UK.
Tel: +44 1453 653322.

'LIVE'
at
MUSIC ASIA
held in conjunction with Pro Audio & Light Asia '97

▲ SINGAPORE: Now nine years old and able to claim attendance figures comparable to those of a European AES Convention, PALA can claim to have come of age. Whereas just a few years ago the event was dominated by the sights and sounds of the disco business, the 1997 show demonstrated most of the attributes of a mature Western pro-audio event than ever before. And at a time when the Asia-Pacific region is on its feet and looking to the West for information and inspiration, it couldn't be more timely.

PALA isn't the place to look out the latest in product development. Rather, it's an opportunity for local distributors to parade Western technology to the 'locals'. To this end, the likes of Team 108, Electro-Systems and Daxco were predictably busy and enjoying greater support from Western manufacturers than in previous years, and also from some of those in Japan. This support was evident not only in the size of this year's event, but in the organiser's figures that claimed an increase of over 30% both in terms of number of exhibitors and floor space and 7,933 visitors from 35 countries. The show was also significant from the point of view of the launch of a new regional title, *Pro Sound News Asia*. Sister to the American and European editions, *Pro Sound News Asia* is intended to serve Southeast Asia in a similar way—with local news and stories offering an insight into the region's business. From most exhibitors, the reports of numbers and quality of visitors were generally healthy. The only

serious exception came from the DTS-supported British section which seemed to suffer from being tucked down an uninspiring alley near the entrance, but not on any logical touring route. However you else you choose to measure it, PALA made sense. And it's not too hard to see why—SE Asia largely lacks the sophistication of the Western in terms of experience and expertise, but not necessarily in terms of appetite. Whether you're talking films or cinemas, record production or karaoke, radio production or shopping malls, all areas of sound and video are having to measure up to standards established elsewhere in the world. That the challenge is being taken up is indicated not just by the presence of Synchrosound and Measat, but by word that all eight of Taiwan's music studios have installed mixing consoles this year.

On the business front, import access to much of the Asia-Pacific region is dogged by punishing import taxes, and in spite of the propaganda, there remain questions about Hong Kong after its handover to the Chinese. As the busiest port in the world, Singapore is free of the tax handicaps of Taiwan and India, and it is eager to challenge Hong Kong for the upper hand. With these things in mind, the PALA show remains a hospitable and rewarding contact point between East and West for pro-audio. Next year's show will continue the alliance with Music Asia established this year and is scheduled for 17th–19th July at the World Trade Centre.

Tim Goodyer

Rosenberger playing a composition by Richard Rodney Bennett on an Imperial Concert Grand Piano. DVD Spectacular will be available in the US from 22nd July, 1997 (US$24.98). It offers a range of media options and musical selections, plus audio and video test signals developed by Dolby, which are suitable for

use with home-cinema systems. According to Delos' president, Amelia Haygood, 'We wanted to keep the disc closely focused on the sound, to demonstrate the holding power of spectacular music and sound without moving pictures'.
Dolby Labs US.
Tel:+1 415 588 0200.

Studio Sound August 1997

■ New York's BMG Studios has joined the 24-bit, 96kHz recording movement with the upgrade of six Daniel Weiss 102 digital processing systems to 24-96 capability.
BMG Studios, US.
Tel: +1 212 930 4055.
G Prime, US. Tel: +1 212 765 3415.
■ Kuala Lumpur's All Asia Broadcast Centre has purchased Quested monitoring systems for its seven recording studios and four audio post studios. The installation will include HQ410, HQ210 and H108 surround setups for the two large recording rooms; HQ210 and H108 surround setups for the smaller rooms and H108-H210 LCR setups for the post rooms; and HQ210s for ADR rooms. As well as being the world's largest digital broadcast and production complex, ABC is home to Astro, Malaysia's new satellite broadcasting company.

Malaysia's newest broadcaster, NTV, has ordered a AMS Neve Libra and Logic 3 digital consoles, and a 55-series analogue console for its Kuala Lumpur facility. The 48-channel Libra will be used for live broadcast, while the Logic 3 will be used with a 16-output AudioFile for dubbing and sweetening.
Quested Monitoring, UK.
Tel: +44 181 566 2488.
AMS Neve, UK.
Tel: +44 1282 417282.
■ London's PolyGram Digital Mastering has taken delivery of a pair of Spendor SA30 close-field monitors. The new speakers will see service in preproduction work in the facility's copying suite.
PolyGram, UK.
Tel: +44 171 493 8800.
Spendor, UK.
Tel: +44181 388 5000.
■ LA's Complete Sound has is to open its 3-room SSL Axiom-based facility complete with 60 Hafler power amps to meet its surround-sound monitoring requirements. A new division of Complete Post, the new facility will use EV TS992 and TL880 speakers driven by 9303 amplifiers in its THX-certified rooms. Complete Sound is set to open in September to handle DVD and DTV programming.
SSL, US. Tel: +1 212 315 1111;
+1 213 463 4444.
Hafler, US. Tel: +1 888 423 5371.
■ Paris-based Mikros studios has commissioned Belgian-based Fundamental Audio Research to design and build its new mixing room. The design will incorporate FAR DBW80 and CR10a monitors, and FAR's new Multi Acoustic Panels.
FAR, Belgium. Tel: +32 4 367 65 10.
■ Las Vegas' newest recording facility, Voyage, has just opened for business. The main recording room, Studio A, is built around an Otari RADAR nonlinear recorder and Status console pairing while Studio C is a post room using a Dyaxis IIi workstation. Initial business is centred

on music for Voyage's established record label and audio for movies, TV and advertising but is expected to move into more album projects.
Otari, US. Tel: 415 341 5900.
■ Swedish CD mastering house The Cutting Room has chosen Merging Technologies' Pyramix as the heart of its audio network. The system serves four studios, each running identical 166MHz Pentiums, Windows NT with the Pyramix Kefren DSP board serving multichannel ADAT-AES I-O.
Merging Technologies, Switzerland.
Tel: +41 21 946 0444.
■ British television and post facilities Central TV, Maidstone Studios and Films at 59 have all enlisted Munro Associates in recent expansion-refurbishment work. Birmingham-based Central is nearing completion of its new £15m HQ which will accommodate staff and technical facilities. The independent Maidstone Studios has revised its Munro-designed rooms, including equipping the main studio for discrete surround use with a Logic 2 console and custom DynaudioAcoustics M3 monitors. Bristol's Films at 59 has opened a third surround room using C3 monitors and two ABES sub units.
Munro Associates, UK.
Tel: +44 171 403 3808.
■ Taiwan's recent flurry of activity has included Life Recording Studios adding a new 56-input SSL 4000 G+ studio and a second AMS Neve VR Legend with Flying Faders; the Thai army's Channel 5 TV station adding a 36-input SSL 8000 GB-series console, and new recording studio Megaforce readying for business with SSL 9000j-series and 4000SE consoles. Megaforce also represents Munro Associates' biggest ever overseas order with three DynaudioAcoustics M4 and two M3 monitoring systems.
■ India's first Logic 2 installation will go to new Mumbai film facility, Gaurav Digital. Along with a stand-alone AudioFile, the Logic 2 will serve audio posting of Indian language films and the dubbing of foreign films. Gaurav Digital honours its name in supporting DTS, Dolby Digital and SDDS sound formats.
AMS Neve, UK. Tel: 1282 457011.
■ The Lebanese American University is to install two DDA consoles for use on its Theatre Audio Techniques course. The 12-input and 24-input CS3 desks were chosen for their LCR panning facilities to support SIS (the Spatial Imaging System).
EVI Pro Audio Group, UK.
Tel: +44 1562 741515.
■ The Arizona Conservatory of Recording Arts & Sciences has added 16 modules to its Neotek Elite console. Originally installed in 1995, the console serves students on the Conservatory's Recording Program.
The Conservatory, Arizona.
Tel: +1 602 858 9400.
Martinsound, US.
Tel: +1 818 281 3555.

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September

7-10
PLASA 97
Earls Court 1, London, UK.
Contact: Marcus Bernie
Tel: +44 171 370 8231
E-mail: marcus.bernle@eco.co.uk
Net: www.harvard.co.uk

8-11
**MIDEM Latin America &
Caribbean Music Market**
Miami Beach, Florida
Contact: Jane Garton
Tel: +1 33 1 41 90 44 39
Email: jane_garton@midem-
paris.ccmail.compuserve.com
US tel: +1 305 573 06 58
Net: www.mldemcannes.com

12-16
IBC 97
RAI International Exhibition and
Congress Centre, Amsterdam
Tel: +44 171 240 3839
Email: show@ibc.org.uk
Net: www.ibc.org.uk/ibc/

16-18
Infocomm Asia
Singapore International Convention
& Exhlbition Centre
Tel: +1 703 273 7200
Email: icia@cia.org
Net: www.usa.net/cia

17-19
**19th Annual Satellite
Communications Expo
& Conference (SCEC 97)**
Washington Convention Centre,
Washington DC, US.

Tel: +1 800 288 8606.
Fax: +1 913 967 1900.

23-26
TVLink Brazil 97
International Trade Mart,
San Paolo.
Tel: +44 181 910 7841.
Fax: +44 181 910 7813.
Email: zoe.cano@reedexpo.co.uk

26-29
AES 103rd Convention
Jacob K Javits Convention Centre,
New York, US.
Tel: +1 212 661 2355.
Fax: +1 212 682 0477.
Net: www.aes.org

October

2-5
18th Nordic Sound Symposium
Bolkesjø, Norway.
Contact: Seem Audio:
Tel: +47 66 98 27 00.
Fax: +47 66 84 55 40.
Email: soundsymp@nrk.no
Net: www.nrk.no/soundsymp/

9-11
LLB
Stockholm, Sweden
Tel: +46 8 24 07 00.
Fax: +46 8 21 84 96.
Email: llb@branschkansliet.se

16-19
7th Intermedia 97 Music Expo
Hala Ludowa, Wystawowa Street 1,
Wrocław, Poland.
Tel: +4871 481821.
Fax: +4871 481451.

16-20
**10th International Audio, Video,
Broadcasting, Motion Picture
and Telecommunications Show
(IBTS 97)**
Milan Fair, Porta Metropolitana
Pavilions 9/1, 9/11 10, Milan, Italy.
Tel: +39 2 481 5541.
Fax: +39 2 49 80330.
Email: assoexpo@assoexpo.com

20-22
**Asia Cable, Satellite &
Broadcast 97 (ACSB 97)**
Putra World Trade Centre, Kuala
Lumpur, Malaysia.
Tel: +6 (03) 264 5663.
Fax: +6 (03) 264 5660.
Email: acsb@mfsb.po.my

23-26
Reproduced Sound 13
Hydro Hotel, Windermere, UK.
Tel: +44 1727 848 195.
Fax: +44 1727 85 0 553.
Email: Acoustics@clus1.ulcc.ac.uk

26
**The National Vintage
Communications Fair**
Hall 11, NEC, Birmingham, UK.
Tel: +44 1392 411565.

28-29
**Broadcast India 97
Technical Symposium**
Chavan Centre, Mumbai, India.
Contact: Kavita Meer
Tel: +91 22 215 1396.
Fax: +91 22 215 1269.
E-mall: saicom@bom2.vsnl.net.in

28-29
**9th ITA Magnetic and Optical
Media Seminar, MOMS**
The Palm Springs Marquis Hotel,
Palm Springs, California, US.
Contact: Charles Van Horn
Tel: +1 609 279 1700.

29-30
**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

30-1 November
Broadcast India 97
World Trade Centre, Mumbai
(Bombay), India.
Contact: Kavita Meer
Tel: +91 22 215 1396.
Fax: +91 22 215 1269.
E-mail: saicom@bom2.vsnl.net.in

November

4-6
Vision & Audio 97
Earls Court 2, London, UK.
Contact: Michelle Calder
Tel: +44 181 948 5522.
Email: michelle@smexpo.demon.co.uk
Net: www.aprs.co.uk

5-8
Apple Expo 97
Grand Hall, Olympla, London.
Contact: Lisa Parrett
Tel: +44 171 208 5020.

CD-R alterations

I have read with interest your article on the HHB CDR800 published in the most recent issue of your magazine (*Studio Sound*, July 1997, pp19). I have recently had extensive talks with HHB about this unit, which have resulted from discussions with an experienced classical recording engineer who asked for my advice. The engineer concerned has been responsible for some fine recordings, at least one of which has been honoured with a Grand Prix du Disque, but he has had little involvement in the burning of one-time CDs, whereas I have been burning one-time audio CDs of high quality for three years.

The majority of the new ranges of CD-Rs being introduced to the market contain built-in sample-rate conversion. It is now standard practice to apply redithering to the converted result, even where the source material has a nominal sampling rate of 44.1kHz. This redithering frequently corrupts any super noise shaping which has been carried out on the source material, in order to convert it from 20-bit or 24-bit formats to a 16-bit format. The result is to burn data onto the CD which is not a replica of the original and which can contain audible differences. You will find that there are very few CD-Rs now available which allow for 16-bit digital source material to be burnt to CD without digital data alteration.

For your information, I make use of a pair of Marantz professional 610 CD burners, driven through an HHB Bit-Box, which is equipped with a special E-Prom to bypass the DSP if a sample-rate of 44.1kHz is detected on the input. The HHB Bit-Box processes the DAT subcode, in order to put the index markers into the correct place in the data block to be written to the CD burner and also applies adjustable digital data delay, in order to allow for the index-marking run-up on the CD burner. Since the DSP is bypassed no corruption of the digital source data takes place, but it is necessary

for the clock frequency in the sending DAT deck to be relatively accurate and jitter-free.

I believe that it is important that at some stage you draw the attention of your readers to this problem, which is now general in the industry and which is affecting the use of CD-Rs in the burning of one-off CDs for replay by artists and management, who need to be assured that what they are hearing is in fact a replica of what is obtained from the mix-down.
John D Barnes, Barnes Consultants, UK.

The Alberts

Just read your interview with Ron and Howie Albert in the July issue (p74). Hmmm.

I engineered and-or mixed six LPs for Frank Zappa (including *Zoot Allures* and *Bongo Fury*) mostly at the Record Plant, LA. Along with about 200 other albums in 20 years in LA. I started and worked on *Thoroughfare Gap* with Stephen Stills and during that time, R&A Albert worked on the same album. So I have a

little bit of a connection with the brothers. (They had a nickname at the RPLA that reflected their particular difficulty with getting that guitar sound they mentioned in the interview. But that's another letter.) I also had the distinct pleasure to do some minor engineering work on the Eagles' 'One of These Nights', with Bill Szymczyk; also working on several live tracks on that follow-up tour with the famous Record Plant truck (now owned by Design FX in LA, I believe). So I have a connection with Bill S too.

The comments that the 'A' Brothers make in their last paragraph about being the ones responsible for the cutting of 'Rocky Mountain Way' while Bill 'wasn't in the studio or (even) in the same state' is typical of a fact of life for almost any engineer worth a dam in the industry. Big deal boys. To read their comments about Szymczyk points out exactly why they can say 'Bill's a great guy' and why they are... well, the Alberts. As producer's, they can't carry Bill's track sheets.
Michael Braunstein, US. ∎

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www.americanradiohistory.com

Nagra Kudelski Ares-C

The wait for a prestigious manufacturer to deliver a practical solid-state digital location recorder-editor is over. **Neil Hillman** reports

IT'S NO MEAN FEAT to be first in the market with something that revolutionises to the extent that conventional practices are rendered redundant. The first sea-change for digital audio came with the workstation —perhaps AMS would care to raise their hand for the AudioFile. Then came desk-top editing with Avid, Lightworks and SADiE. Now Nagra Kudelski brings us duvet editing: the ability to record digital audio on location, retire to your hotel bedroom where you edit, and then send the material down a conventional phone-line or over an ISDN link. Couple this with a price in the UK of around £35,500 and it all sounds just too good to be true, which in fact it is —but this is an intriguing machine that, but for some puzzling design decisions, could actually generate more custom for the Cheseaux factory than that of the rope shop at the foot of Mont Blanc.

Unlike the motoring similes of Volkswagen with their sadly named 'Sharon' or the amusing translation of 'Pajero' into Spanish, the Ares is named boldly after a Greek god, rather than the colloquial English 'harris', so presumably Nagra has high expectations for this device. The 'C' signifies that this version uses PCMCIA flash cards as its recording medium. In fact it uses two. conforming to Series 2 or 2+, with up to 64Mb of storage currently available per cartridge. That's enough to give the Ares a total recording capacity of well over two hours in stereo, or double that in mono configuration.

The original ¼-inch open reel design was soon rejected, as indeed were DAT and Mini-Disc. as storage mediums. In fact the only moving parts associated with this machine are the needle of the idiosyncratic modulometer, the jog-shuttle wheel of the editor and the fumbling fingers and mumbling lips of the operator attempting to manipulate the machine's setup.

The Ares-C is designed to fulfil three functions that. up until now, have never been incorporated into one portable package: a solid-state digital audio recorder, an onboard non-destructive digital editor and an in-built ISDN codec. Certainly, it is immediately recognisable as a Nagra; it has its microphone inputs on the left-hand side and a pair of small, slotted switches between the female XLR input sockets for selection of microphone type (dynamic, +48V/+12V phantom, or 12V Tonader powered microphones). A traditional analogue level meter is to the left of the input level pots on the machine's front face, and the large play-record rotary control switch is in its customary right-hand position. What is noticeably new on the front face is the single row LCD screen that displays track number and duration, and five transport buttons that operate Track Forward. Rewind, Play Forward, Play Rewind and Stop. But the warm glow of familiarity ebbs abruptly when you start to examine the controls and their associated labelling. The left and right microphone inputs for instance; marked as 'sensitivity' and therefore quite correctly expressed as 'mV/hPa'. Look, I'm sorry, but millivolts-per-hectapascal means nothing to me, and I suspect not much more to other sound recordists

either. What was wrong with the previous incarnation of Nagra recorder that told us the industry standard unit of microphone amplifier gain in dBs?; and for a machine marketed on its 'familiar Nagra interface' it seems strange to have dispensed with the Nagra IV-S headphone mode selector that enabled the user to select left or right inputs, mono or stereo outputs, or indeed headphones 'off' and replace it with a single feed; and why no M-S decoding matrix for a professional location recording machine clearly designed to be so up-to-date?

But the main weakness of the Ares-C must lie in the man-and-machine interface of the audio recorder itself, which requires configuring via a complicated software driven menu-tree accessed by the SHIFT key and the five transport keys beneath the LCD screen on the front panel. These keys alter not only the occasional changes, like auxiliary inputs or system beeps. but those of a more basic nature that a sound recordist may make at a moment's notice—like switching between recording in mono or in stereo. So, if you decide to record in stereo, but the machine is set for mono, into the menu you must go. And next time I go there. I shall be armed with a bell. a book and a candle. The machine needs to be told that both of the microphone inputs should be active, as the surplus input is switched off to reduce noise in its single microphone, mono configuration. Further down the tree to select MPEG as the compression standard required, rather than the G722, G711-A or G711—standards otherwise on offer and then comes the selection of J-Stereo and the sampling rate of either 16kHz. 24kHz, 32kHz or 48kHz. This is not in itself a difficult task, but a total of some two dozen key strokes are needed—providing the instruction manual displaying the menu tree is to hand and that you haven't attempted to navigate the menu intuitively—all simply to

Too good to be true? Nagra's Ares-C

change from mono to stereo recording. It's at this point that we should breathe a collective sigh of relief that Nagra has not as yet turned its attention to the lucrative domestic white-goods market—what a washing machine that would be.

Plainly then, the Ares-C is not designed to fill the gap left in the flight cases of sound recordists who moved from the time-code Nagra IV-S to DAT. Let me put it this way, no sound recordist I know is going to want to download to the postproduction house his rushes via ISDN at the end of a day's filming; and what are the economies of passing over a flash-card costing several hundred pounds with that day's rushes?

So just who is this device aimed at? Nagra is very clear where the Ares-C will fit in to the market, and I believe they have been very perceptive.

Rotate the machine's concept in your head and view it as an audio journalistic tool. Think 'digital Uher' perhaps, but do not think of it purely as an acquisition device.

While not quite transforming itself from an ugly duckling to a beautiful swan, this gangly, awkwardly talented machine takes on a whole new aura and could easily become speech radio's *de facto* recorder.

But its recording facility is just part of a much bigger, brighter picture. The Ares-C's onboard editor, while offering only basic cut-and-paste facilities, is simply a joy to use.

Journalists experienced in editing ¼-inch tape, or newer recruits more familiar with a word processor, will find the graphical representation of the soundtrack passing a cursor on the edit screen of the top face 'edit deck' very user-friendly, and it is the work of minutes to confidently manipulate recorded takes.

Edit mode is activated by turning the large rotary switch on the front panel to the Edit position, lifting the lid of the edit deck and following the on-screen labels adjacent to five function keys immediately below the screen. These keys, in conjunction with the jog wheel, now drive the two principle editing operations that may be performed; either joining together various sections of previously recorded takes or inserting a particular section of one take into another. This similarity to ¼-inch tape editing, rather than having the extra facilities that a digital editor may otherwise offer of crossfades or variable slope ramp-ins and outs, for instance, is more than compensated for by the speed and ease of use.

A track number is allocated at the time of the recording, and this is stored in a directory displayed on the Edit screen when Edit mode is active. The directory also details track duration, start time, date and compression mode. Selection of a particular track to be edited from this directory brings two horizontal bands to the Edit screen. The original band of audio, the 'recorded' track, sits above a blank band that will become the 'edited' track. As the recorded audio plays from right to left past the 'replay head', in and out points may be chosen by means of the function keys and fine adjustments made with the jog wheel. As soon as an edit has its in and out points chosen, that portion of audio appears on the bottom edited track, along with the duration of the edit, and the in and out times from the start of the recorded track. All necessary edit information

is kept as an Edit Decision List, and is shown in the directory alongside the edited track number as a small pair of scissors.

It was, in truth, the manual that told me that this symbol was a small pair of scissors —it looked to me like an email 'sad face'.

The Ares-C can talk to the outside world down a conventional ana-logue telephone line from its 600Ω female banana connectors on the right-hand side of the machine, or digitally through its onboard ISDN codec module. The ISDN synchronisation protocol is selected from the main menu tree, as is the telephone level, switchable between 4.4V or 1.55V, and the operation of the dual-function Auxiliary in-Line out level control pot, which is sited next to the two microphone level controls on the machine's front face.

The Ares can direct-dial a call from the numerical pad on the edit deck or from its store of ten regularly used telephone numbers, or it can be left in Stand-By mode, emitting a warning beep when called. Once contact is established, and with the editor in Transmit mode, the microphone inputs become operational and the return signal of the line is heard through the headphones. The machine will even display the running cost of making the call from the Ares, through the digital exchange, in local currency.

A useful facility is the ability to preselect an

edited track from the directory and have it cued ready to play out; ideal for remote inserts into a studio programme. A 2-way live conversation can take place between the reporter and the studio presenter and then at a suitable point the reporter can play-in a prerecorded item or interview, safe in the knowledge that the Ares will mute the reporters microphone until the track has reached its end point, at which time it will then automatically return the microphone live to line. Although the Ares-C does not support MPEG full duplex at present, transmitting in MPEG and receiving in G711 is a reasonable compromise.

But Nagra must address the complexity of the configuration menu by at least providing a schematic similar to the manual for the lid of the machine, to eliminate guess work when the instruction book is not to hand. I would also like to see the editor's Delete function safe-guarded by a simultaneous double button push.

Is this the end of ¼-inch tape? Well, Nagra has a sister machine for the Ares-C, the C-PP, a standard 19-inch rackmounted version that serves as either a seamless ISDN coder-decoder for the remote Ares-C, or as a studio play-in for a journalist's mailed PCMCIA card. The Ares system simply scythes a whole new position for itself in a growing market.

'Radio what's new? Someone still loves you.' ∎

D&R Octagon

Building on its success in surroundable consoles, D&R gives **Zenon Schoepe** an exclusive preview of a multiformat, multidiscipline desk that will be launched at the IBC next month

Centralised digitally controlled switching reduces price

ONE OF THE LAST remaining bastions of independent mixing desk manufacture, D&R has a history that goes back nearly 25 years. In that time the Dutch company has addressed all the standard market sectors and tried its hand at all volumes of production. It's now concentrating on what it considers it does best, relatively low volume manufacture with the emphasis on quality.

All D&R desks are still built by hand at the company headquarters in Weesp near Amsterdam, but even more unusually the company prides itself on its ability to accommodate custom options. And this from a manufacturer that positions itself well below the traditional price cut-in of the custom fix. D&R claims that most middle-market desk manufacturers' reluctance to accommodate custom tweaks is not down to technical impossibility—few things are—but to one of attitude. D&R maintains a philosophy that states if customers want something, and are prepared to pay for it, then they should have it. They leave happy, and D&R draws valuable ideas for future developments.

D&R is best known for strong, and rather distinctive, music-recording consoles like the Avalon, Merlin and Orion, but its most recent triumph has undoubtedly been the Cinemix, as something of a departure that combined a high degree of automation and digital control in a board that could comfortably work in a variety of surround formats.

New for September's IBC exhibition in Amsterdam will be the Octagon, a desk that goes further than the Cinemix in a number of significant areas. To trace its origin you must look to the response generated by Cinemix and Merlin as there is a lot of both in the new console.

The convenience of Cinemix is that it can readily be used to mix-in surround, a feature of Merlin is lots of EQ and auxes. Combining the best of both, and adding a bit more besides, has resulted in a desk that will appeal in film and video circles as well as serving music recording with the sort of open-endedness for multiformat work that is becoming appreciated.

Octagon has dual path input modules, 16 aux sends, a total of 48 buses, 8 master mix buses for all formats, and uses control modules for digitally controlled switching. Four of these can be placed in the same frame for multi-operator film applications giving switch reset augmented by pot recall.

Recognising that the requirements of film, video and music recording are quite specific, Octagon attempts to cater for all these disci-

plines and is in line with D&R's reluctance to produce a console for a single market—particularly as a facility's ability to crossover is increasingly desirable. The Octagon policy has clearly been to leave nothing out lest it offend any of the intended markets, but to present comprehensive facilities in a manner that is flexible. The 16 auxes will seem extreme for film-work, but are a real attraction for recording, while programmable macro keys that can be assigned to switching functions will be welcomed, along with the ability to bring stem returns in to the top of the console on calibrated faders, and access to the master mix buses available on upper and lower faders. Recording will enjoy the full EQ on all signal paths, the number of buses, and may find a use for the surround capability. Either way it is unprecedented for a desk in this price range to be SDDS-ready out of the box.

Octagon will be available in two frame sizes the first holding 48 dual input modules and 8 dual return modules, the second 72 dual inputs and 8 dual returns. The two signal paths in an input module can handle mic-line, and tape-group signals, and have 4-band twin parametric mid EQ with variable high-pass and low-pass filters on the lower path. The 16 auxes are accessed from 12 controls per input module with local mute and pre-post switching, and these can be accessed from the upper and lower signal paths, both of which have automated 100mm faders.

The lower path has 8-way panning with LCR and Front-Rear control while the upper section has LCR panning and discrete assignment to preprogrammed formats such as mono, stereo, LCRS, 5.1 and SDDS.

The two paths can be controlled by 8 master control groups for grouping faders, mutes
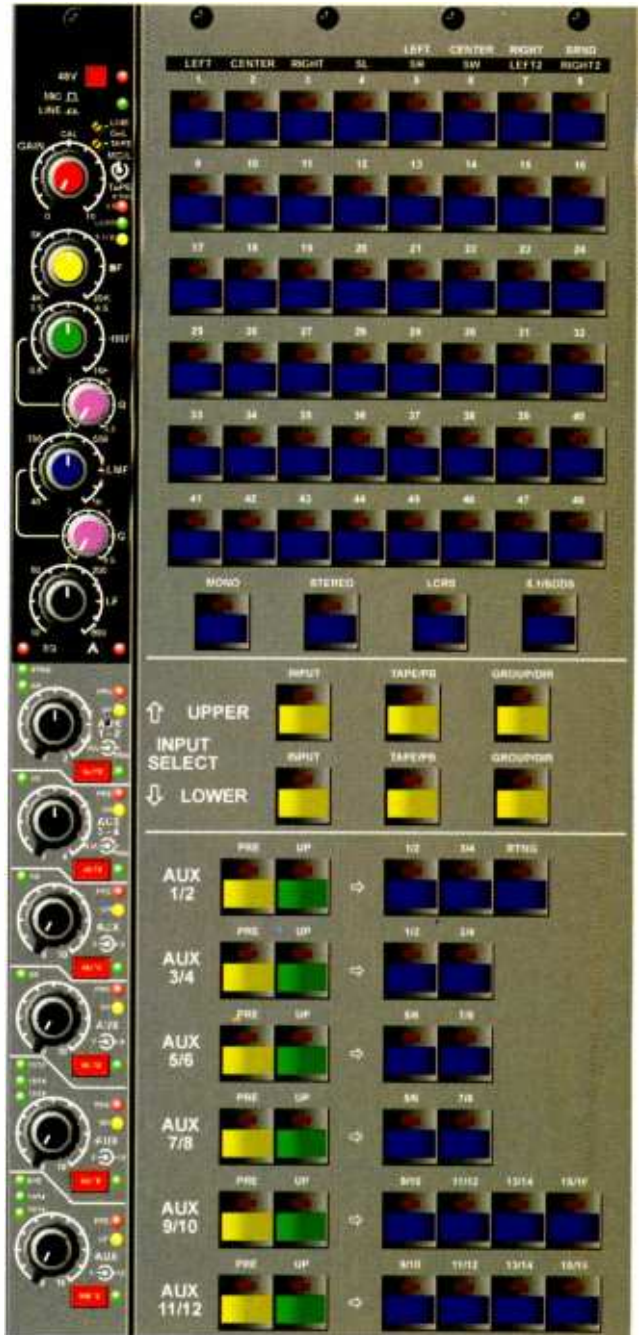
and automation. Input module switching is directed by an assignable control module that influences everything as part of a reset except phantom power, mic-line, EQ, LF bell-shelf, inserts and filters. Each control module has transport controls and four control faders.

The Master section houses a TFT-LCD VGA display working in conjunction with a keyboard and trackball to control automation, computer data, and time-code functions. Two sets of studio outputs are provided with comms and a listen circuit while the 8-track outputs are automated, and an 8-track front-panel switchable encoder insert is supplied. There are also two automated joysticks assignable to the 8 internal main buses, or the 32 output groups in groups of eight. A control room monitor matrix caters for mono, stereo, Pro Logic, DTS, 5.1 and 7.1.

Digitally controlled control room levels are coupled to an elaborate system of selective monitor muting that, among other things, allows nearfield monitoring to be **page 16 >**

Four control modules can live in one frame for multi-operator use

< **page 15** selected separately to the main surround mix for localised adjustment.

Pot recall is performed using an LED bar for position in the channel strip and whole desk configurations can be stored and recalled as well as just EQs and selections of modules only. Recall can be administered across local or complete desks in multi-operator situations.

Onboard dynamics are an option and these are controlled from a 4-row alphanumeric display, and as an interesting aside MIDI control of solos, mutes and faders is possible.

Automation running on PC is D&R's own PowerVCA package with a moving fader option. This is another feather in the company hat as automation has frequently

been an unsuccessful pursuit for many manufacturers who opt instead for third-party supply. D&R originally started on this route, but gravitated towards in-house development to make the automation integral as opposed to a bolt-on mutes and faders arrangement. Work is advanced on a Windows 95 version, but the Octagon has a lot of automation, and the data is compatible with Merlin and Cinemix. The degree of digital control is said by the company to be proven in existing products and given that major cost-contributors on a console are its switches, it also saves money.

Money is what it all comes down to, and when I reveal that when it ships at the end of the year Octagon prices will start at around £40,000 (UK) for a small version with automation this amounts to staggering value for money. The search for a truly multi-discipline console at any price is restricted severely by choice and that D&R should choose to fulfil this expectation at this level says much about its sincerity. I would go so far as to say that Octagon, on paper, promises to be something of a landmark product because it shatters preconceptions about what a serious film-conversant board ought to cost. Currently it looks like the most advanced desk of its type in its league, and it is advanced. It's not a music recording desk with a few film concessions, rather it's a film desk with music recording features. Video post will take to it too.

I have no doubt that D&R will deliver Octagon as promised, it is after all a rethink and rebalancing of its existing technology. Indeed, this desk also contains the core of a very clever theatre installation board. Perhaps that is something they might want to think about.

I can't wait to see Octagon in the flesh. It promises to be one of the most exciting and mould-breaking products of the year. Look out for it. ■

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Sonic Foundry CD Architect

The PC's challenge to the Macintosh's popularity moves on with a powerful American
CD writing package that sits within a bigger system. **Rob James** boots it up for a quick fix

OVER RECENT YEARS, the inexorable rise in processing power and storage capacity of even modest PCs has been leading toward an interesting point for pro-audio users. For several years Macintosh users have enjoyed gloating over their cousins attempts to use PCs for audio and multimedia authoring. This is rapidly changing.

There is a growing band of people who are getting involved in manipulating digital audio for PC applications, games and Internet Web sites. The plethora of 'standards' involved in this area of work make the standards conversion problems we have to deal with in music production and sound-for-picture seem trivial by comparison. Until recently, most sound work for these applications was done on Macs and converted for Windows PCs. But not for much longer if Sound Forge is anything to go by.

Sonic Foundry's Sound Forge is a suite of

Editing is simple and amazingly fast

digital audio solutions for Windows PCs. The core software, Sound Forge, is now at v4.0a. The scope of this suite is now so great this article will confine itself to just one plug-in, the new CD Architect. This, as you might guess, concerns itself with the process of burning CD-R discs.

I have been using Windows PCs for some years and have become sufficiently familiar with the beasts to follow the dictum 'when all else fails open the manual'. I felt it would be a valid test of how far audio software has progressed if I could manage to produce a CD-R without recourse to the paperwork. Knowing the fun and games various friends have had getting this to work and the number of very expensive beer mats produced, this is a pretty tall order.

The Sound Forge 4.0a software installed itself without any fuss or drama as did the CD Architect plug-in. 'Mousing around' enabled me to make intelligent (?) guesses in order to set up the software to talk to my sound card and store audio on the SCSI drive. I recorded 45 minutes of audio as 17 files and opened

CD Architect. Compiling, topping and tailing the sound files into some semblance of order took considerably less than the running time. CD Architect found my Yamaha 400 CD writer, and, since I was feeling brave and impatient, I selected maximum speed write (4x), although I did do a dry run first. Much to my amazement the first attempt was a success. This is no small achievement since it was the first CD to be burned on this system, let alone with this software.

CD Architect passes the test with flying colours. I may have been lucky, but I don't think so. Provided your PC is set up with digital audio in mind there is now no reason not to use it for audio.

CD-R burning is a particularly demanding task since buffer underruns—the inability to output data at the rate required by the CD writer—will result in the production of a beer mat, rather than your magnum opus. To this end the manual, which I finally opened and read assiduously, recommends turning off Auto-Insert Notification for all CD devices on your system, disabling screensavers and closing all other applications including the less obvious ones like Microsoft Find Fast and System Agent. If your machine is on a network, log out or better still, physically disconnect the machine.

CD Architect burns audio CDs only, not CD ROMS or any of the other flavours of CD. It does not support TAO (Track At Once) which allows the recording of single or multiple tracks in several sessions and the final 'fixing', writing the TOC (Table of Contents) data to complete the CD. Although this method sounds attractive there are disadvantages. Each recording uses a considerable amount of extra space for the hidden housekeeping stuff like the run-out sectors created as the Laser shuts off. These can give rise to clicks on the finished disc. CD Architect uses the altogether preferable method (for audio) of DAO (Disk At Once). With the extensive editing and preview features TAO really should not be necessary.
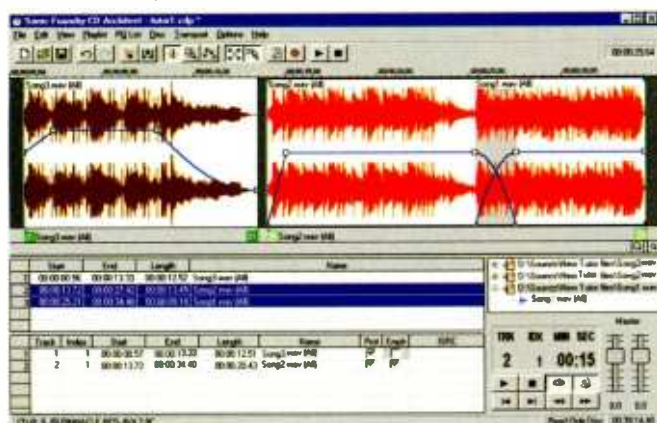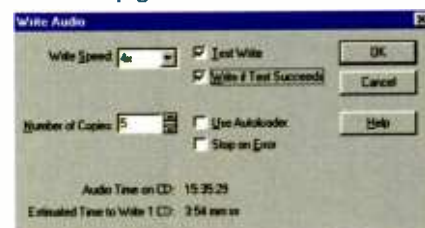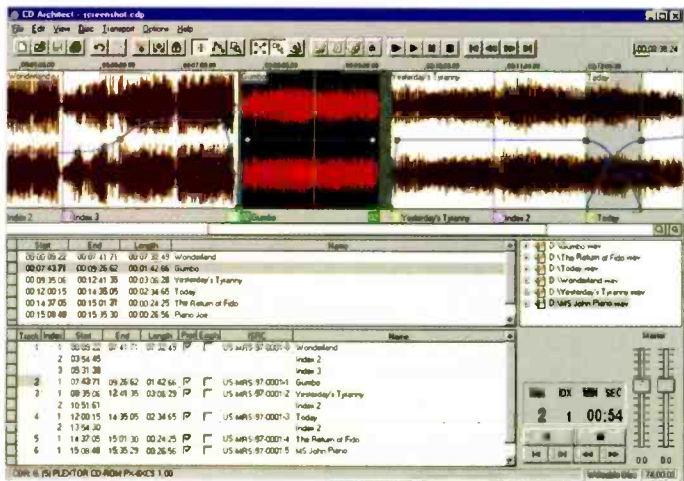
The main window is divided into six sections. The Track View is a graphical representation of the current CD project. The principal element is a wave display with time ruler, track ID and sub index markers. Zoom controls allow the whole project to be viewed or

small sections. There is a small status display that gives the current cursor position either as time, seconds to six decimal places, various SMPTE formats, samples, time and frames (Red Book standard 75fps) or absolute frames.

The Playlist window is a text-based list of audio regions inserted into the CD project displayed in spreadsheet format—allowing fields to be edited in the same manner as a spreadsheet. If several regions start times are selected together, editing the first start will move all the other selected starts by a proportionate amount. The PQ list window functions in a similar manner with exact start and end times, track IDs, sub indices and markers, names, ISRC codes (Industry Standard Recording Codes) and flags to set copy protection and pre-emphasis. The Audio Pool window is a list of all sound files available to the project. Regions already defined in the files show up as separate entries, files can be entered into the project by 'dragging and dropping'. Alternatively you can drag and drop files from Windows Explorer. A pause of two seconds of silence is added between the resulting tracks. This is a preference and can be altered or defeated. A project can consist of a group of sound files or one sound file—such as a live performance with regions defining track start and end points.

The Toolbar is a row of buttons which give quick access to commonly used functions and is fully customisable. The final window is the Transport-Master panel which can function like a CD player or simply as transport control for the software. There is also a pair of graphical faders which affect the master volume level.

Editing is simple and amazingly fast. Mark-

Read audio page

Write audio page

Crossfades can be manual or automatic

ers or tracks can be picked up with the mouse and moved in time. There are a variety of 'snaps' provided to help—'Start of region to track beginning marker' moves the marker as you edit the region. Clicking on the waveform display moves the cursor to the mouse position or there are 'goto next-previous marker' controls. Fades are introduced using the Envelope Adjustment feature. Shift clicking on the centre line of the waveform display places a 'handle' at the cursor point; this can be grabbed with the mouse and pulled up or down to create the fade. Double clicking will add more handles up to a maximum of 32 per region. Fades can be linear or logarithmic. With Auto Crossfade selected equal power crossfades can be cre-

ated by dragging a region over another. If Auto doesn't sound right, turn Auto-crossfade off and the handles can be adjusted until the desired result is obtained.

All of these activities can be carried out while audio is playing. There is a tiny delay while the change takes effect but this does not diminish the power of near instant editing and auditioning.

When you are happy with the project, burning the CD is really the easy bit. You just select Write Audio which brings up a simple dialogue box. The number of copies can be set (1–10,000), the write speed selected, and a choice made whether to go straight for a write, test first, then write, or test only. The total audio time on the CD is displayed with the estimated time to write a CD. Test does everything a write would do except switch on the laser. If a buffer under-run or other error is detected you have just

saved the cost of a blank CD.

If a buffer underrun is detected on a project with multiple source files and fades, one option is to lower the writing speed. Another, perhaps better, option is to make an image file. This will consolidate the project into one file with fades and crossfades rendered and the PQ information embedded which reduces the load on the PC during the writing process. This facility is to be found under the Save As option on the File menu. The otherwise excellent manual mentions image files but fails to point out where the function is located.

In addition to its primary purpose of producing CDs, CD Architect also works well as a fast and accurate playlist editor. The Convert To New command under the Playlist menu works in a similar way to creating an image file but does not embed the PQ information. A further treat is the ability to extract data from audio CDs and convert it into .WAV files.

The software works with a wide variety of CD recorders including autoloaders, which makes a little more sense of the option to write 10,000 copies. There are comprehensive keyboard shortcuts for those who prefer the keyboard to the rodent and a huge number of options for those who really know what they are doing with CDs. The manual is excellent and provides a good primer for the whole subject as well as concise reference information. ∎

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Roland VS-880

Not all manifestations of digital audio workstations need to be big and expensive in order to meet the brief. **Rob James** explores a clever compact DAW with many applications and a big heart

THE FIRST SYNTHESISER I ever played was an early Roland monosynth. In fact, I still own a Juno-6 and I remain particularly fond of Roland's late lamented R880 reverb —although its is no longer made. it outperformed many of the more expensive machines.

The VS-880, meanwhile, would appear to be primarily a digital replacement for the cassette-based analogue 'personal multitracker'. A closer look, however, reveals something more—a compact digital audio workstation with up to 8 replay tracks, a digital mixer with up to 14 inputs. snapshot and dynamic automation and onboard effects. Rear panel connections are sparse: ¼-inch mono jacks for the four analogue inputs take priority over the alternative phonos, and 4 phonos for aux outs and analogue master out. Two further phonos take care of SPDIF in and out and a 25-pin D-connector facilitates connection of up to seven further SCSI hard drives to supplement the onboard 2½-inch IDE drive (the unit can only address up to 1Gb at a time so larger drives must be partitioned—up to 200 Songs can be stored in each partition). There are ¼-inch jacks for headphones, and a footswitch, and a pair of DINs for MIDI In and Out-Thru. The 880's A–Ds are 18-bit, 256x oversampling affairs and its D–As are 18-bit 8x oversampling.

The front panel is pretty sparse as well, as the photograph illustrates.

The VS-880 has four recording modes. Each of these can be at 32kHz, 44.1kHz or 48kHz sampling rate. The manual is rather reticent on the subject of compression, as are the manufacturers, but I am led to believe Mastering mode records 18-bit linear limited to a maximum of 6 replay tracks. The other three modes are compressed. Multitrack 1 uses 2:1 compression. Multitrack 2 is approximately 2.5:1 and Live approximately 3:1. This translates to 202, 404, 539 and 646 trackminutes per Gb respectively at 44.1kHz sampling rate. Roland

is not saying what method is used to compress the audio. but it is the subject of a patent application.

There are quirky things here: 18-bit linear conversion is not a misprint. It's unusual, but, if you are using the analogue I-O, potentially useful. (The SPDIF output is dithered to 16 bits.) Still more unusual, when one of the compressed modes is in use, mixing, equalising, and so on, takes place on the compressed audio. Don't ask me how it is done, but it seems to work.

Housekeeping facilities are accessed from the SYSTEM key and the SONG key. MIDI. sync, time display format and record monitoring can be set. and drives and partitions switched or initialised. Under SONG, Songs can be loaded, copied or erased. New songs can be created and named. and space can be recovered by optimising. Songs can be backed up to an ordinary audio DAT recorder, although it is preferable to archive onto a removable SCSI disk.

Each of the 8 tracks has 8 virtual tracks giving a total of 64. Material can be freely swapped between virtual tracks to assemble a complete track of selected takes. Punch-in can

be achieved manually or automatically, and you can set the VS-880 to repeatedly loop over a section and rerecord on each pass until you get it right. Punch-in recording can use the virtual tracks to keep up to 8 takes stacked. All the usual editing functions are present. The PREVIEW TO and FROM keys and SCRUB key make editing simple. The Scrub function simply loops a section of sound (25ms–100ms) rather like cueing a CD player and, when selected, the display changes to a rudimentary waveform.

There are 8 autolocate memories and the TAP key may be used to place up to 1000 marks in a Song either while recording or replaying. Existing marks and locates can have their position adjusted. Deleting a mark causes the marks which occur after to be renumbered. The position of a mark can be altered in increments down to ¹/₁₀₀th frame or less, or you can work in measures and beats.

The time stretch and expand function can be applied to sections of tracks or whole tracks including, if you wish, the 8 virtual tracks. This is a non-real-time operation and takes a while to crunch. There are three algorithms you can try in order to

< **page 21** achieve the best result. The range is 75%–125% and works well. If you don't like what you get you can undo it.

The Varipitch facility adjusts the sampling rate on replay for the whole machine. Pitch changing can also be accomplished on individual tracks or sections using the time-stretch function, although this does also change the length of the treated section. If you have the optional effects board, this offers further pitch-changing possibilities.

The mixer has three modes. In effect there are actually two main modes one of which gives a layered mixer. Input to Track gives a one-to-one correspondence between inputs and tracks. In this mode the inputs to the mixer can be the physical inputs to the device, tracks in replay or the mix bus. Equalisation in this mode is 3-band with top and bottom shelving filters, and a peaking mid with variable Q.

The other two modes—Input Mix and Track Mix—are used together to enable a total of 14 sources to be mixed (8 tracks, 4 analogue ins and a stereo SPDIF in). The Mixer mode select button pages between control of the tracks and the inputs. In these modes EQ is 2-band.

Internal processing is 24 bits wide, and there is 12dB of boost and cut available on all EQ bands. Adjusting EQ parameters, aux send level, and so on, is achieved by pressing the CH EDIT key for the relevant channel, selecting the parameter you wish to change and dialling in a value with the parameter wheel. It takes considerably less time to do than to describe

and rapidly becomes instinctive.

An optional effects board adds two effects processors to the mixer. There are a prodigious number of presets on offer. Grouped by type they are based on 20 basic algorithms and, generally, they are of high quality 'straight out of the box'. In addition to reverb, delay dynamics and pitch shifting, there is guitar-amp simulation, and graphic and parametric equalisers. There is also a vocoder and one preset—Robot—gives *Hitchhikers Guide to the Galaxy* fans a passable Marvin the Paranoid Android. Bullhorn is good and does as the name implies without spending an age setting up EQ, delay... Two of the algorithms, Reverb and Gated Reverb, and the presets based on them are only available on Effect 1.

Snapshot automation comes in the form of 8 scene memories. These can be set to recall all parameters or everything except fader and pan. Scenes can only be recalled manually when audio is not playing, but they can also be recalled by MIDI program change messages. Dynamic automation of fader, pan and aux settings is available onboard.

The MIDI connection enables the VS-880 to be synchronised to MIDI Time Code. It can also generate MTC or MIDI clock. More comprehensive dynamic mixer automation can be achieved by using an external sequencer.

What exactly do we have here? A package

about the same size and weight of a large notebook PC containing an eight track digital recorder with 1Gb of onboard storage, a 14-channel digital mixer and two effects units. All that is required to produce finished items is monitoring, a source, or sources, of raw material, and something to record the final product.

There is, of course, a downside. When this amount of power is crammed into such a small box the user-interface becomes critical. Don't be deceived by the simplicity of the control surface. If you are going to get the best out of this box it will take a while to get to know it. But, If you can hack a *Tamagochi*, this shouldn't give you too much of a problem.

The VS-880 will certainly perform as a producer of demo tapes or disks, but other possibilities spring to mind: the Mastering mode allows it to be used as a high-quality stereo editor. In radio, using Multitrack 1 mode, it would seem perfectly suited to the production of complete programmes, perhaps in a hotel room, to be sent via ISDN for transmission. I am sure it will also find applications in the theatre where space and finances are frequently restricted. Roland claims PC games developers are also taking an interest.

Overall, treated as a stand-alone 'digital recording studio in a box', the VS-880 is pretty much unique. ∎

"Easy Patch"
PATCH PANEL

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NEW TECHNOLOGIES

24-bit Apogee

Apogee Electronics has introduced an A-D convertor system called the AD-8000 for 8-channel 24-bit conversion, optional D-A and interface cards and the company's Soft Limit and UV22 processes at under $6,000 (US).

Soft Limit and UV22 can be switched in on a per-channel basis and the unit features AES-EBU outputs while four optional interface cards can be installed for multichannel format conversion. Interface cards currently include ADAT, Tascam TDIF (both with bit-splitting technology for recording 24-bit signals on multiple 16-bit tracks) and Pro Tools. AES-EBU and SPDIF inputs permit the processing of external sources, and an SPDIF output follows the channel pairs

selected on a built-in headphone monitor D-A convertor. Optional stereo and 8-channel 24-bit D-A expansion cards enable the AD-8000 to be configured as a complete conversion system while comprehensive source-destination switching provides confidence monitoring, digital track bouncing and overdubbing features. An optional remote mic amp is planned.

The 2U-high rackmount has 6-mode lightbar metering for simultaneous peak and average ballistics, and 2-second, and infinite peak hold. Digital overs are indicated numerically and are user-definable. The device can sync to a range of external signals and sample rates including video with an optional video sync card as well as providing internal crystal locked 44.1kHz and 48kHz sample rates and optional sample-rate conversion.

The company has expanded its recording accessories line with lint-free alcohol-based tape head cleaning wipe that uses medical grade 91% pure isopropyl alcohol.
Apogee Electronics, US.
Tel: +1 310 915 1000.
Net: www.apogeedigital.com

DAR Genesis

DAR will launch new Genesis software at the IBC (September) which will be installed in all current systems with an upgrade path for existing users. Also debuting will be CDAdvance—a means of accessing CD material directly from Sound-Station and Sabre systems.

Genesis software will form the basis for the company's next generation systems and combines the current user-interfaces and multiplatform compatibility with file import-export, networking-Open media compatibility and the ability to work with a wide range of third-party devices. This introduces a new method of project handling that enables users to open multiple reels simul-

BSS Opal DPR422

The long road to affordability leads BSS ever on till its arrives with the Opal range. **Zenon Schoepe** tests this compressor/de-esser

IT HAD TO HAPPEN, BSS had to come out with a series of units that pitched it at the more accessible level that seems to be enjoying most of the activity at the moment [which is described as what?]. That it has chosen to do this with the Opal range, which also includes a gate, is interesting because this unit employs some of the tricks introduced by earlier models from the company. Thus the DR422 dual-channel compressor/de-esser will stir fleeting memories in those who know the DPR402. However, at £495 plus VAT (UK) its around 40% less than the DPR402 which remains a mainstay of many studios and live rigs, but the DPR422 shares the latter's VCA technology.

It's a glorious-looking piece of kit, finished in a fetching shade of jade with a selection of chunky translucent and illuminating buttons that are straight out of a packet of wine gums. Pots are the usual expensive-feel BSS types with superb LED bar-graph metering that deserves an introduction.

Multiple LED output level metering graduates from green to orange and can be flicked to read input level on a nonlatching switch. Red gain-reduction metering, working right to left, is complemented by a green below-threshold display working left to right that shows how the input's level relates to the current threshold setting. An orange Threshold

a majority of sources.

Individual channel bypasses are selected from two centrally located pushbuttons that are reminiscent, at least in position, to the DPR402 and DPR502 gate. Stereo linking shifts Threshold, Ratio, Attack, Release and Auto selection to Channel 1 leaving gain adjustment separate. Channel 1's below threshold meter follows both signal paths, but both gain reduction meters show the same readings leaving output-level metering independent. If stereo de-essing is desired then both channels' pots have to be set to identical values.

Connections are on balanced XLRs switchable between -10/+4 operation and there's the admirable inclusion of balanced side-chain input and ground compensated output available on stereo jacks.

Operationally the DPR422 is something of a delight being extremely easy to get your head around, as well as satisfying in performance. The compressor employs BSS's characteristic 'progressive knee' curve which is nicely soft and unobtrusive on delicate ratios and altogether more brutish when you want to pour it on thick.

The first half of the RATIO pot's travel is dedicated to covering 1:1 to 3:1 but where it differs from other boxes of this type is in the subtlety of variation available. The unit's strong hand is undoubtedly in gentle dynamic

LED positioned between the two sections illuminates when that point is hit. Simple yet extremely informative, and the sort of thing you can now expect from BSS.

There are two other LEDs that work with the separate de-esser section, these indicate that the stage is active and that around 15dB of gain reduction is going on. The de-esser is rather clever because in addition to working with the usual type of more-less control and sweepable (1-10kHz) FREQUENCY pot, control of de-essing can be handed over to the compressor controls at the press of a DE-ESS HF switch that effectively flips operation from the default Broadband mode to high-frequency filter instigated compression. The filter can be auditioned on a SIDE CHAIN LISTEN button and this mode disables the de-ess more-less pot.

The compressor section offers fully variable threshold, ratio (1:1-infinity:1), attack (50μ-100ms) and release (5ms-2s). The chain is finished by a ±20dB gain make-up pot, however there is also an AUTO switch that disables attack and release time adjustment in favour of a programme dependent mode. This is a forgiving setup that sounds surprisingly good on

control where it's presence is practically undetectable. This is a unit you would reach for when dealing with a critical source that is well controlled, but needs a tad smoothing out. It's a wonder on vocals because you can set it to nip the tops off the peaks, and relax in the knowledge that if the singer does pile in, then it will still look after you.

And that's before we come to the de-esser which has got to be a leader in its class. Whether in broadband or HF mode the results are exemplary. There are shades of the DPR402 in here although it's important to remember that these are very different units.

What you're getting with the DPR422 is a good all-round compressor-de-esser for stereo and dual-channel work that has a real talent for vocals and solo instruments in general. The thing about BSS compressors is that they sound nice and contemporary with a real surge in loudness once they tip into the red. The degree of de-essing control is excellent and almost unexpected on a unit of this type. BSS could simply have concentrated on providing a compressor, instead it's added a classy section that is too often regarded as a luxury. ∎

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Empirical Labs Distressor

The expansion of compressors continues, with equipment appearing almost daily. **Terry Nelson** justifies one of the most recent arrivals

THE SHEER NUMBER of compressors that hit the market every month make finding the ideal compressor like a search for the Holy Grail. Obviously, everyone has their own preferences so an 'ideal' unit as such isn't an option. Instead, we're invited to trawl the options for the best match to our requirements and tastes. Here then, for those who understand what compressors are supposed to do, is the Distressor from the American company Empirical Labs.

The front panel presents a gain-reduction meter, BYPASS pushbutton, 8-LED radio indicator with pushbutton, two pushbuttons marked DETECTOR and AUDIO, and four large knobs for Input, Attack, Release and Output. A rocker mains on-off switch completes the panel.

Connection to the outside world is via transformerless balanced XLR connectors and mono jacks. There are also stereo link In and

Out mono jacks for use with a second Distressor. Operation is deceptively simple —cycle around the Ratio settings with the pushbutton, adjust the rotary controls and away you go.

I started off with the recommended 'beginner's setting' of 6:1 ratio and all knobs at '5', and was immediately impressed by the smoothness of the response. Increasingly, the input level provided more compression, but the sound did not take on that lifeless characteristic often found with other units.

I used the Distressor on a variety of material—from microphone to single unprocessed tracks to final mixes—and it has to be said that it handled everything superbly well. Operationally, the drawback is that all functions (with the exception of Bypass) have to be cycled through in one direction only. You cannot go back to, or go directly, to a setting —you must cycle through. In the manual, Empirical explains that this is due to space restrictions imposed by the 1U-high chassis. While I agree that the layout on the whole is clear and simple, the cycling feature is slightly annoying, though not the end of the world.

The Distressor is based on an interesting premise: digital may be pristine at times, but it can be lacking in character and musicality—and this is where this particular piece of equipment steps in.

Compression ratios start at 1:1 and move through to limiting at 20:1. The final setting, Nuke, is a brick-wall limiter —you can hit it with all the level you can muster and the output will not budge. So what use is it apart from transmitter protection? The Nuke setting was developed with room mics in mind and it is amazing how it

will transform a fairly dull drum-sound (with an equally lifeless drummer) into John Bonham playing in a live room—the reverberation fairly oozes out. It is also excellent when used on unpredictable singers where you can set the input level for minimal compression while keeping a fairly constant output for when they decide to go into overdrive—all without undesirable squashing.

The compression characteristic is soft knee, and certainly imparts a musical flavour to the overall sound. Attack and release parameters are extremely flexible and give considerable control over the envelope of the signal to be processed. Empirical regards the Distressor as a mixing tool to help individual elements blend well, or sit well, in the overall mix, and it does this extremely well.

The unit is also more than just a compressor or limiter, however, and features some interesting additions. Using a 1:1 compression ratio will add a certain warmth to the sound you can then decide if you need actual compression or not.

The AUDIO pushbutton controls the two distortion modes as well as a high-pass filter for eliminating rumble and unwanted low-frequency content. The filter can be selected by itself, or in conjunction with two distorted modes.

Dist 2 produces a class-A type of warmth with some 2nd harmonic distortion while Dist 3 adds both 2nd and 3rd harmonic distortions (similar to tape compression). The Dist 3 mode also fires up the '1%THD' and 'Redline' (for 3% distortion) LED indicators, just to keep you informed as to how much grunge you may or may not have.

The DETECTOR pushbutton inserts a high-pass filter into the detection circuitry to avoid plosives. Control can be further refined by putting a 6kHz peak into the circuitry as this smooths out harsh vocals and gritty guitars. Both functions can be used singly or together.

Not surprisingly, comparisons with 'vintage' equipment crop up with this compressor, and it's true to say that the 1:10 compression ratio used in conjunction with the 'optical' settings (slowest attack and faster release) does a very good job of emulating the original settings will also add extra warmth if required.

Overall, the Distressor is one of the most interesting new compressors to come along in some time. It is not a comp-limiter as there is no peak limiting when compressing, but it is not intended to be. Instead you either limit or compress, yet the soft-knee almost allows you to use it as a comp-limiter when necessary. Best of all, the Distressor does not set out to copy other manufacturers' equipment—although it does feature some of their characteristics—but to be a musical tool in its own right. ∎

Contact
Empirical Labs, 10 Schley Street, Garfield, NJ 07026, US.
US: Wave Distribution.
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taneously together with enhanced storage and drive usage.

New editing features include slip, trim, slide, new roll and copy/spot over functions. Sample-rate conversion and segment reverse facilities are included.
DAR, UK. Tel: +44 1372 742848.

Summit EQP200B

Summit's EQP200B programme equaliser replaces the EQP200A and adds a number of features including a low frequency 6dB/octave shelving filter starting at 50Hz which is selectable with a bypass-in-LF shelf switch. The low frequency boost-cut section has been enhanced with the addition of 180Hz to the existing frequencies and the low frequency section offers separate boost and attenuation controls capable of 16dB boost and 20dB of cut.

The high-frequency boost section has added 1.5kHz to the existing frequencies and the same section now has a variable bandwidth control. Meanwhile the high-frequency section has attenuation shelving filters at 5, 10 and 20kHz.
Summit, US. Tel: +1 408 464 2448.

Opcode XTC sync

The Studio 64 XTC synchronises analogue and digital multitracks to PC and Mac-based workstations. Simultaneous wordclock and superclock outputs permit sample accurate sync of Pro Tools to machines like the DA-88, Akai DR4, E-mu Darwin and some DAT machines. It can also control ADATs through MMC.

An internal sync clock can write SMPTE as the master reference or it can generate wordclock and superclock from incoming SMPTE. It also accepts video and blackburst signals as reference and routes MTC and MMC. It can also be used as a 4x6 MIDI interface with patchbay capabilities and can be networked with an Opcode Studio 4 interface or added to any MIDI setup that has a spare serial port.
Opcode, US. Tel: +1 415 856 3333.
SCV, UK. Tel: +44 171 923 1892.

Soundtracs RX8

Soundtracs has released details of an affordable desk, the RX8, designed for recording and live sound applications. Equipped with four stereo subgroups, a normalling system allows monitoring of an 8-track mix through the effects returns and the return of them to the first eight full mono inputs for remix.

Available in 24-input and 32-input versions, mono inputs have 3-band swept mid EQ with an 80Hz high-pass and individually switchable phantom power. Six **page 28 >**

CEDAR for Pro Tools

From elite beginnings, CEDAR's audio processing technology has moved into the mainstream. **Dave Foister** loads it into Pro Tools

CEDAR'S ACCLAIMED audio restoration processes have operated from a variety of hosts. The original DOS-based system spawned a collection of self-contained processors, offering removal of hiss, clicks, crackle and azimuth errors in a set of four rackmount units. The development work on these led in turn to a return to the PC, this time as a plug-in card with Windows software. Multitasking allows CEDAR for Windows to work alongside a DAW system, such as SADiE, in the same machine, but integration goes no further than plugging cables across between

Small	Medium	Large
Model	Threshold	Pre/Post	Gain(dB)

the respective hardware interfaces. The CEDAR system on its own has internal busing between cards to simplify patching, but no common audio bus currently exists within the PC world to allow proper integration between disparate software packages.

Whereas on the Mac, Digidesign's TDM has become as much an essential part of the Mac virtual studio environment as a patchbay in a physical one, and the resulting palette of effects can contain as much as a control room's worth of racks, often from the same manufacturers. Now that palette can include CEDAR restoration processes.

Most TDM plug-ins are pure software packages, using the host DAW's available hardware for the actual audio processing. Sometimes this will utilise the computer's own sound facilities, but more often it will be something like the Digidesign Audio Engine with its DSP Farm cards—the more Farms, the more processes the system can run simultaneously. CEDAR for Pro Tools is different, because the usual hardware can't even come close to supporting the 40-bit floating-point arithmetic necessary for the restoration processes to work. Instead, one or more of CEDAR's own MacDSP cards is required alongside the existing hardware, just like the PC card on which CEDAR for Windows runs. Once in place the cards become a transparent part of the system, with access to the software modules achieved in exactly the same way as any other TDM plug-in from within the DAW environment.

The processes themselves will be familiar to many, but perhaps not to the Pro Tools fraternity this new package addresses. Three separate processes are available; these are purchased individually as software packages, and can be run in any combination on the

available MacDSP cards. A card can support two mono processes or one stereo, and the processes and their control windows are identical to those in CEDAR for Windows.

CEDAR long ago identified three primary categories of sound degradation which required different approaches to their removal: clicks, crackle and hiss. The Declick process removes the kind of click typified by the scratch on a vinyl or 78 record, and three distinct algorithms or models cope with differing lengths and severities of disturbance. User adjustment consists solely of choosing the most appropriate algorithm and setting a Threshold value.

Decrackle deals with the very different problems of more constant signal disturbance, and uses a 'split and recombine' approach to identify the unwanted elements and remove them. An additional control on this module allows you to help the system distinguish between wanted and unwanted signal, and this, in conjunction with two basic algorithms and a THRESHOLD control, deals with all kinds of problems from record surface noise to thyristor buzz—it can even get rid of certain types of distortion.

Finally, Dehiss2 addresses the apparently impossible problem of removing broadband noise from a signal without altering its spectral content or interfering with it in any other way. This, too, is very simple to set up, with controls to help the system identify the nature of the noise and determine how much it should be reduced. A BRIGHTNESS control is not there for artificially adjusting the treble before or after processing, but as a further aid to the system to help it distinguish between wanted HF and unwanted noise. All these processes are quite spectacular in operation, but Dehiss2 in particular is almost spooky. In every case, forget the compromises and degradations introduced by any other noise removal system you may have tried; CEDAR really does get rid of all these problems completely without affecting the wanted signal at all.

Control of the modules is carried out from simple blue panels with rotary controls, numeric readouts, toggle switches and stereo ganging locks. As always, the biggest surprise is how easy it is to make CEDAR processing achieve the apparently impossible and how hard it is to do anything nasty. In every way these processes are the equal of the existing versions, producing equally astonishing results. CEDAR still stands alone in its extraordinary capabilities, and bringing it to a new audience as the Pro Tools version does is not only good for CEDAR, it's good for the business. ∎

NEW TECHNOLOGIES

aux sends are provided along with talkback, 2-machine 2-track playback, a mono output, an additional control room output, and selectable solo-SIP.
Soundtracs, UK. Tel: +44 181 388 5000.

Darwin v2.01

Darwin 2.01 software provides support for the new DSP option card for time compression-expansion, pitch transposition with 'demunchkinisation' and a gain-fade level function.

New features include the ability to control Darwin's internal digital mixer via MIDI, the ability to store locate points on the fly, support for the two front panel assignable keys and the ability to back up to low-cost QIC drives which can be mounted inside the machine.
E-mu Systems. Tel: +1 408 439 0371.

Cheap in-picture

Chromatec's AM-1 low-cost broadcast quality in-picture audio meter is designed for basic 2-channel analogue audio indication and the bar-graph may be half or full screen height and positioned anywhere in the picture. The mix level of the superimposed image can be adjusted from the front panel or via the remote interface and most popular scales and ballistics are selectable together with input sensitivity. A peak hold indicator with variable parameters may also be displayed. PAL or NTSC composite video is auto selected.

Bel's BBC-designed 7410 mix-minus synchroniser designed to eliminate audio delays from off-air cues is now available. Created to provide a clean feed to commentators in OB who monitor from the off-air programme, the unit adjusts delay, gain and equalisation automatically to generate a mix-minus feed.
Michael Stevens & Partners, UK. Tel: +44 181 460 7299.

KRK RoKit

RoKit personal shielded monitors are a passive 2-way design featuring a 6.1/2-inch polyvinyl proprietary long-stroke woofer and a 1-inch silk-dome tweeter. Power handling is 75W with a claimed frequency response of 69Hz–19kHz ±3dB and a sensitivity of 91dB (1W @1m).
The monitor measures 12¹/₂-inch x 10¹/₂-inch x 8-inch and comes in a

Audio Design MIDA System-4

Building a front end for its own mini-mixer has given Audio Design an appealing modular mic preamp-convertor, says **Dave Foister**

IN THE CHAIN between microphone and recorder, two links have come under particular scrutiny in recent years. One is the gain stage between microphone and line level, where not only have improvements in quality been sought but the very placement of the preamp has become an issue. The second is the analogue to digital interface; given that the final recording medium is digital, how good can we make the convertors, and how soon can we make the conversion?

Many have addressed these issues, and the once-unlikely idea of a microphone preamp with digital outputs has become a familiar sight; but few have the additional motivation of Audio Design, whose specialisation in digital boxes has, perhaps, overshadowed its once-famous analogue expertise. One of the company's leading products is the DMM-1 4:2 digital mini-mixer, that boasts a plethora of digital inputs and outputs, but no analogue interfaces. The ideal front end for the DMM-1 would be a high-quality 4-channel microphone preamp with twin AES-EBU outputs,

sometimes required to work out what they mean.

These last three functions are controlled by means of shift buttons on the master section, which give the channel nudge buttons control of the parameters for individual channels, or a pair as appropriate. The only global function is the sample rate, which copes with internal clocks at 44.1kHz or 48kHz or incoming sync either as word clock on BNCs or AES on an XLR. The unit copes well with loss of external sync, defaulting to the nearest internal setting and flashing its display to alert the user to the fact that all is not as it might be.

Since part of the purpose of a unit like this is to go to digits as early as possible in the chain, the provision of remote-control facilities is an almost essential part of the package. RS485 serial control is available as an option, and a purpose-built, locally powered remote control can operate the preamp over hundreds rather than tens of metres. Certain functions, such as stereo ganging, must be preset on the preamp itself, but the remote then provides rather more elegant control of the remaining parameters than the unit's own front panel. For a start it has rotary controls for gain; these are continuous encoders which can increment the

and if this could be operated under remote control then so much the better. This, then, is what Audio Design has produced in the MIDA System-4 microphone amplifier.

The unit itself appears very simple. A 1U-high box carries two sets of controls and displays for the two pairs of preamps, plus a small set of master controls. There are no frills on the microphone preamps themselves; switchable phantom and variable gain is all that's on offer, with no filters or even phase reverse. The only addition is the facility to gang the preamps in pairs, with overall gain and inter-channel balance replacing the separate gain controls. All front-panel adjustments are made with nudge buttons, the resulting gain being shown on a simple 7-segment display.

A sprinkling of LEDs show a bit more about what's going on; red ones show phantom power is on, and take an age to die when it's switched off, while green ones show signal presence. Another shows when a pair is ganged for stereo, while the last indicates that 16-bit output is selected; the convertors are 20-bit, but each pair can be separately switched to 16-bit for direct connection to a 16-bit medium. All the switchable functions are shown on the displays, although as these are only 7-segment a little imagination is

gain in 1dB or 4dB steps depending on how quickly they are turned, and the resulting value is shown on a much more grown-up LCD window, both numerically, and as vertical bars. Phantom can be switched from here, but that is all. It is worth noting that no metering is provided anywhere (unless you count the signal present LEDs) but no doubt the assumption is that the destination device will meter the incoming digital signal anyway. A second 9-pin D-connector allows independent mute or cough switches to be added, further increasing the remote possibilities albeit in a rather more mundane way.

The signal quality, without which none of the foregoing would be of any use whatsoever, is superb; the noise floor is extremely low, and the frequency response comfortably exceeds the limits of the convertors. For those who prefer their signals undigitised, balanced analogue outputs are available, relegated to 5-pin XLRs that curiously do not conform to the 5-pin stereo microphone wiring standard. No problem; the digital outputs are the MIDA System-4's *raison d'être*, and the package delivers the goods in every respect, providing a simple high-quality front end to many systems beyond the obvious Audio Design links. ∎

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NEW TECHNOLOGIES

grey texture similar to the K.Roks. Group One, US. Tel: +1 516 249 1399. The UK Office, UK. Tel: +44 1442 870103.

Korg PCI

The latest addition to the Soundlink digital recording system, the 1212 I-O PCI multi-channel audio interface offers full functionality of multichannel computer based digital recording on Macs. The card offers 12 inputs and 12 outputs configured as a pair of analogue I-Os, an SPDIF and an 8-channel ADAT optical all of which can be used simultaneously. The card is also equipped with wordclock in and out as well as an input for ADAT time code. It ships with Mac drivers for any Sound Manager-compatible program and Steinberg and E-Magic have announced support. The card is compatible with the DRS 880 A–D interface.

Korg has also announced the DL8000R digital multitap delay with four outputs. Delay times of 4.8s are possible and the unit has a large display showing bpm in addition to time display. Features include tap tempo, audio trigger, MIDI trigger and modulation.

Forty ambience effects including delay, echo, reverb and pitch shifting are offered by the AM8000R multieffects processor. The unit boasts an internal mixer that is used to combine effects and eight parameters can be controlled simultaneously via MIDI.
Korg, UK. Tel +44 1908 857100.
Korg, US. Tel: +1 516 393 8520.

Sennheiser ENG kit

The Sennheiser ENG RF receiver kit comprises four EK4015 UHF miniature diversity receivers in a special flightcase that incorporates a power supply and an input for an

external DC supply. The front and back panels of the flightcase are removable and the back cover includes stowage space.
Sennheiser, Germany.
Tel: +49 51 30 600 366.
Sennheiser, UK. Tel: +44 1494 551551.
Sennheiser, US. Tel: +1 203 434 9190.

Cooper Command stations

JL Cooper's MCS3000 series of media command stations provide direct control of computer-based workstations. Control features include touch-sensitive motorised faders, 60 user-programmable function keys, LCD and LED dual displays, five rotary encoders, transport controls, numeric keypad and concentric jog/shuttle wheel.

August 1997 **Studio Sound**

SONY CDP-D500

CD players are at the heart of the debate about dissimilar 'identical' CDs. Yet they form a basis for broadcast, as **Rob James** reports

THE CDP-D500 is Sony's latest addition to its range of professional CD players. It is, in essence, the replacement for the late, lamented CDP-2700. As such it is is a completely new design with styling and ergonomics in keeping with its stalemates—the PCM-R500 and PCM-R700 DAT recorders.

The CDP-500 is supplied as a stand-alone unit with a remote outputs are balanced analogue on XLRs, unbalanced analogue on phonos, AES-EBU on XLR and (SPDIF) on phono. There is a headphone jack and volume control on the front panel and sockets on the rear accommodate a parallel remote for fader start and similar systems, RS232 for connection to computers and a BNC for wordclock input.

The front panel has all the usual CD player controls with a bright dot-matrix display. The familiar Program Play and Repeat modes are available while notable extras include concentric jog-shuttle wheels, a VARISPEED ENABLE button and associated setting knob, a switch to select serial, parallel or no remote, a switch to

select an external timer and a MODE switch for the Auto Cue-Auto Pause function. Less familiar buttons are the AUTO CUE-AUTO PAUSE, and REHEARSAL, and selectors for remote and timer operation.

The CD tray is a chunky, metal device that inspires confidence. The jog wheel can be used to quickly select tracks. Cueing can be from the start of track or from the start of audio. The detection threshold for start of audio can be varied in 6dB steps from -48dB to -72dB. Starting is instantaneous. There is a looping function to aid finding an exact cue point within a track. Pressing REHEARSAL initiates the loop. The start point can be adjusted using the jog wheel or nudge buttons down to frame accuracy. Pressing ENTER ends the rehearsal and pauses the player at the cue point. Once a cue point has been established the player will re-cue when the CUE/STBY button is pressed.

Varispeed operation allows variations of ±12.5% in 0.1% increments. This is accessed by pressing the VARISPEED button and turning the knob until the desired pitch shift is heard or displayed. The shift can be cancelled by pressing the knob.

The Varispeed function is not available when using external word clock. The Fade function enables tracks to be faded in or out over 1s–9s. If the player is in Pause, pressing FADE starts the machine and fades in over the selected dura-

tion, pressing FADE again will fade out over the same duration. The fader functions are not available when using digital outputs.

The CDP-D500 locks the players internal clock to an external reference. This approach has been known to result in unacceptable jitter on the digital output. In this case Sony has employed some innovative circuitry that resyncs the data from the disc using a high-precision clock with a claimed jitter performance of typically 2.5ns which is comfortably within the 10ns limit recommended in the AES3 protocol (Sony's figures). This does mean you are limited to a 44.1kHz output unless you want to varispeed the machine. On occasions this could be useful though it is not a published feature. Thus, if you connect, say, a 48kHz clock the player will run fast.

Practically, and in the absence of specialised measuring equipment, the CDP-D500 appeared perfectly happy when synchronised to the wordclock output of a Yamaha 03D. The A–D convertors are quoted as (note the quotes), 'high-precision BiCMOS advanced sign-magnitude 20-bit with 8x oversampling'. The use of 20-bit convertors with an inherently 16-bit source has several advantages—the top 16 bits of a 20-bit convertor are likely to exhibit better linearity than the full range of a 16-bit device which results in lower distortion. In addition the noise floor of a 20-bit device is lower minimising the addition of convertor noise to the output. The menu functions are accessed by pressing ENTER while in Stop mode. The shuttle wheel cycles through the options to change and the jog wheel changes parameters, pressing ENTER accepts the change.

Menu options are: Key Protect, this allows all controls except Eject, Stop, Enter and Clear to be disabled; Remote Protect allows the infrared remote to be disabled (useful in installations where several machines are in use); Auto Cue threshold level; Cueing Select, track or index; RS232 for which there are four setup parameters.

The CDP-D500 supports the CD-text format allowing text data to be transmitted down the RS232 connection. This allows track names, artist and so on, to be displayed on a computer. In addition, the existing PQ codes can also be checked enabling the CDP-D500 to be used in a CD quality control environment.

The CDP-D500 is a robust, serious piece of kit. From my observations on test it should be well able to withstand the impatience of operators in pressured situations. It is simple and quick to use with a good number of interfacing options. If you need a synchronisable CD for a 44.1kHz environment this machine provides an economic solution. By the way, it also sounds good. ∎

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Tel: +1 201 930 1000.
Fax: +1 201 930 4752.

< page 30 MCS3800 offers 64 moving fader channels each with four programmable switches and 40 virtual encoder knobs.

With 100 locate points and built-in SMPTE reader and MTC reader, command stations have MIDI ports and two expansion slots allow optional interface cards to be added.
**JL Cooper, US. Tel: +1 310 3064131.
Sound Technology, UK.
Tel: +44 1462 480000.**

Audio rack

Targeted at location recording, the Audio Rk2 minirack accepts two Audio DX2020 or DX2000 wireless mic receivers in a rigid casing that protects them. Only two aerials are required to feed the signal to the diversity receivers via custom filtered RF distribution amps. Reliability is aided by reverse power and over-voltage protection up to 30V while connectors include 6-pin Lemo or 4-pin Hi-rose. Phase reverse is included in a unit that weighs 545g and measures 168mm x 151mm x 30mm.
Audio Ltd, UK. Tel: +44 1494 511711.

Oscillator calibration

Wavetek has introduced a family of oscillator calibration workstations based around its Model 9500 Calibrator and available in

400MHz, 600MHz and 1.1GHz bandwidth versions. They come with PC calibration software including a library of tested procedures for commonly used oscilloscopes.
Wavetek, UK. Tel: +44 1603 404824.

Audix DAW mixer

Audix Broadcast's ADD5000 digital desktop workstation mixer is designed for use in news, small production and editing suites and is designed to allows the monitor to be mounted directly above the control surface and the keyboard directly in front of the user.

The ADC7000 mixer augments the company's range of live radio on-air desk systems and can be supplied in in-line or split formats with up to 16 channels. The desk stores EQ, dynamics and routing configurations which can be saved to and recalled from a smart card.
**Audix Broadcast, UK.
Tel: +44 1799 542220.**

Digital analyser

The Alphaton DA1000 digital audio signal analyser is a small testing unit for digital audio signals and transmission **page 34 >**

dbx 160S

If the name doesn't pull you in, then the looks probably will.
George Shilling enjoys an encounter with dbx' latest compressor

WHEN I BEGAN in this business, the dbx 160 was standard equipment in any studio worth its salt. Not for the faint-hearted, I always suspected that this half-rack, 2U-high compressor was working far harder than its meter indicated—if you really wanted savage compression then this black box was the one to do the business. Other

the world, along with the budget 160XT and its successor the 160A.

Now in the late 1990s, dbx is feeling the heat from the likes of Focusrite. To reclaim the high-ground in the signal-processing market it has introduced the Blue range. The centre-piece is this very imposing unit, named the 160S to denote its heritage. This is truly a Rolls

variations with more features such as the 165A came along, and these were popular but less common than the standard 160 with its fixed auto attack and release settings. In the early-1980s came the 160X, a 1U-high, LED-packed box, still black and still fairly brutal, but a little smoother sounding than its predecessor. And even smoother in OverEasy mode, which is the registered dbx name for soft-knee compression. There must be thousands of these superb units in studios and live racks around

Royce model: everything about the build of it is uncompromised: the beautifully finished blue metallic front panel is ¼-inch thick. The smooth, solid-aluminium knobs are hand-crafted, and look remarkably similar to those used by Focusrite on its Red series. They are larger and more damped than the Focusrite ones, making small adjustments easier. Push-buttons are also aluminium and have a positive feel when pushed, unlike the plastic Focusrite illuminating type. The front panel

NEW TECHNOLOGIES

Precert test

The most commonly used emission test instruments have been combined in a single cost effective package for precertification test capabilities.

The ES-Plus comprises the 150kHz–1GHz

test receiver ESPC from Rohde & Schwarz and the 10kHz–3.5GHz spectrum analyser R4131C from Advantest supplied with Rohde & Schwarz Windows software ESPC-K1.
Rohde & Schwarz, UK.
Tel: +44 01252 811377.

VST plug ins

Steinberg has two new plug ins for its VST platform. The stereo Magneto simulates tape saturation and tape overdrive effects while the Prosoniq Roomulator reverb has a set of stereo algorithms including hall, room and gated reverbs. These join the page 36 >

looks slightly cluttered compared to the competition. and is truly feature-packed. The brightly backlit vu level meters are custom made with the Gain Reduction mode using a different scale—linear rather than vu characteristics. All the LEDS have smart chrome surrounds. and are in a variety of colours. On the back. gold-plated Neutrik XLR connectors are used for inputs, outputs and side-chain connections for both channels. There is a large assortment of unbalancing and grounding buttons and connections, so you are not going to have any hum problems with this unit. There is a panel where an optional digital I-O board can be fitted. The box weighs over a stone and is rugged, yet beautifully finished. You really get what you pay for here, for although expensive, the 160S really looks like it will take any knocks and last a lifetime. It does beg the question, who needs build strength like this? I cannot see many situations where you would take a machine as expensive as this on tour— perhaps a few elite rock stars. live engineers might use a couple, but it is difficult to justify such extravagance for most people. In a studio situation it's true that cheap knobs break or wear out, but I don't remember ever having a problem with a front panel not being thick enough. It seems that dbx is just trying to keep up with the, erm, Joneses.

dbx' proprietary PeakStop and PeakStop-Plus are both offered. These are two different types of limiting, the former based on that

introduced on the 165A, and the latest version which monitors the input to predict what it needs to do to keep the level below the threshold set with the STOP knob.

Each channel has a button and LED for external side-chain insertion, and STEREO COUPLE links the two channels, killing all controls on Channel 2.

The first signal I plugged in was a bass guitar, always great with a 160X and good with the old 160 if you were careful not to overdo it. In Auto mode, the 160S sounded closer to a 160 than a 160X, with a similar tendency to fart. However in Manual mode it was easy to set appropriate Attack and Release times and immediately had a wonderful punchy compressed bass. Switch into OverEasy mode and the compression is very smooth, doing everything you want. On drum ambience, it worked a treat. As with the old 160, 2:1 compression seems much more savage than you would expect, even in soft knee OverEasy mode. This is a machine for people who like a lot of compression. I found myself setting the ratio to 2:1 or 1.5:1 in many situations.

The audio specification is quite astonishing and boasts a custom VCA with a dynamic range of 127dB. The ring-binder manual is great, explaining all the principles and operation thoroughly.

It really does sound as good as it looks: there is little that dbx hasn't thought of with this model. I loved it. ∎

Symetrix 606

Retroism takes a curious turn with the dedicated delay line.
Zenon Schoepe reckons that a good idea is always a good idea

THE CAUSE OF NOSTALGIA has clearly driven us to go through the very last of the drawers of old gear if we have come to the reintroduction of the dedicated digital delay line. Once the centre-piece of any self-respecting rack, for many the digital delay line, with it's ability to modulate itself, was a starting point for the weird and wonderful, and also served as the introduction for many to sampling. The disappearance of this type of unit towards the end of the 1980s was one of those curious mysteries, particularly as most

time to bpm), diffusion models for both channels and room size for early reflections. It's at this point that we encounter one of the tricky bits of the 606. Because it has so many parameters available to it, it is difficult to decypher them simply from the display and you have to constantly refer to a large laminated sheet that contains all the details and translations of the LED display. Needless to say, this takes a little of the spontaneity out of what would have otherwise been a natural process.

You really do have to work at this aspect of

who depended on them were happy with what they achieved and how they did it. Nobody actually said it was old hat, rather it was the multi-effects unit that promised to replace it, and a host of other boxes along the way. Delay lines became marginalised, expensive by comparison, and then extinct.

The Symetrix 606 picks up this heritage and adds 1990s functionality, presets and extensive MIDI control. The front panel distinguishes itself by the presence of knobs a-plenty and the sort of low-key 3-character LED display that used to be regarded as the last word in heavy-duty high tech. Signals come in and out on balanced standard jacks, and accompanying these on the back panel are MIDI IN and Out/Thru, and a footswitch socket used to enter tap tempo.

Aside from input level control and stereo input metering, attention is grabbed by a row of nine continuous pots, and four pushbuttons at the centre of the panel are crucial to business. The 606 has 99 user programs and 10 factory presets intended as starters for programming, and these are accessed on the turn of a knob and the press of a LOAD switch. The box works in dual-mono or stereo modes and pressing the MODE switch illuminates an LED to tell you the status.

The row of nine pots is associated with three layers of legending, a SELECT switch scrolls through these layers and the pots alter the relevant parameters in the layer. Parameters that can be adjusted are mix, delay time, feedback and modulation controls for each of the two channels, and an output level control for the lot. These are bolstered by modulation-type selection, modulation source, modulation destination, modulation level, oscillator rate, filter frequency, filter Q, diffusion amount, MIDI clock sync (for tying delay

the 606 to clarify its arrangement in your mind because you will not simply stumble upon it just by playing with it. This is unfortunate, yet essential if you want to get the most out of this box, but once the penny drops then there's no looking back.

A MIDI activity LED indicator shows if the 606 is receiving MIDI data as all its parameters can be accessed via continuous controller, and that means that they can be automated to a sequencer. None of this was ever possible with those glorious old units.

Delay-based effects on a fully adjustable device like this are among the most interesting and creative, and there is a purity of purpose in the 606. The maximum delay time is 2.74 seconds, and there are so many different things you can do. It does wonderful impersonations of early reflections and spaces, produces some of the deepest chorus and flange effects I've heard, and is capable of that great stereo ADT just like the old boxes. One of the peculiarities of older digital delay lines was that the processed signal often sounded brighter than the original—something of a turnaround from analogue delays where the chance would have been a fine thing. The 606 is sweet as a nut in this respect and permits the processed signal to be altered and downgraded using a filtering section.

The 606 is currently the only device of its type that concentrates exclusively on time-based effecting with an emphasis on hard front-panel control. It's a lot cleverer than those early digital delays, and, consequently, there is an initial operational penalty to pay in terms of accesibility.

Even so its great fun to use and so evocative. It's so nice to have some modulation LEDs blinking away in the rack again. Recommended. ■

Contact

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Digitech Studio 400

Some equipment succeeds where manufacturer had not envisaged, other kit is simply wrongly targeted. **George Shilling** wonders why

AT FIRST GLANCE, Digitech's Studio 400 puts you in mind of Eventide's H3000. The fact that its about half the price of an H3000D/SX is reflected slightly in the build quality—the buttons feel fairly cheap, and the data wheel feels plasticky—but the rear panel reveals a couple of things not found on the H3000. Here you'll find no less than four analogue inputs and outputs on both XLRs and balanced jacks, with a +4/-10dB button for the output. The review model was fitted with the reasonably priced digital I-O option board that features AES-EBU and SPDIF with all the usual connectors and sample rates, and software controlled options for routeing in combination with the analogue connections, which could be quite irritating: no sound emerged from the analogue outputs in factory default mode.

Digitech makes a song-and-dance about the 400's two S-DISC (Static-Dynamic Instruction Set Computer) DSP processors, listing in the manual 57 varieties (!) of effects modules that each have a unique set of parameters. Each of

these modules falls into one of three categories, depending on their complexity—Full, Half and Quarter which use those respective amounts of an S-DISC processor. There are then 23 configurations which configure the two processors in different combinations. Pressing FX Edit repeatedly takes you to each module in turn, plus Input and Output modes sections (where you can choose from such options as four mono inputs and outputs or two separate ins feeding effects which return on the same stereo output) and a section called Modifiers that allows control by MIDI Controllers, LFO or signal amplitude. Within each of these FX Edit sections are up to 10 pages, stepped through using the NEXT PAGE and PREVIOUS PAGE buttons, each page containing up to four parameters. These are selected using the buttons marked 1, 2, 3, and 4.

For me, operation is unnecessarily complex and irritating; certainly the display always shows the configuration diagram, flashes the module being edited, and indicates which page of parameters you are on, but the symbols and lettering are very small, apart from the large program number display and reasonably sized Program Name-Parameter Adjust section at the top. So why is valuable space on the display wasted with the unnecessary S-DISC logo? As a consequence, the display is hard to read if not directly in the line of sight, and no use if

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mounted low down in a rack. By far the worst thing is the input-level metering. Having meters on the main LCD is a very bad idea, especially as they are so small. No other manufacturer does it, most opting for the much more visible rows of LEDs.

The 57 effects types are usable and contain all the parameters you would expect plus a few extra things to play with. For example, in addition to seven basic delay program types there are a couple of analogue delays which behave just like old-fashioned units where you hear the pitch change as you adjust delay time. These have a parameter called Smear which diffuses the delay repeats. Reverbs are excellent, smooth and expensive-sounding. There are also Chorus, Flange, Phaser, Rotary Speaker, Harmoniser, EQ, Dynamics and other variations of effect types. One useful feature is the selection of named starting points for each type, which are available as a parameter in effects edit.

Each Program contains up to eight effects modules (the two S-DISC processors split into Quarter-type program modules). Programs are split into two groups: User Programs 1–100 and Factory Programs 1–191. The latter cannot be overwritten, and astonishingly many of the factory programs are set with much of the dry signal mixed in, as though you were inserting the unit across one signal. You have to conclude that in spite of the fact that the 400 is marketed as a 'Professional Studio Processor' it's set it up for a guitarist. There is a solution, but it is very time consuming wading through modules and pages to find the Wet, Dry and FX signal level parameters, changing them, and resaving as a User program. Why not have a knob on the front to balance the wet-dry signal ratio?

The footswitch socket can be used for bypassing the unit, or with a special Digitech footswitch it can also be used to step up or down through the programs. This feature also suggests a market other than the pro studio.

Programs 101 to 123 are the basic Configurations with no FX modules loaded, while thankfully 124 to 191 are all 100% wet.

If you can put up with this feature-packed machine's foibles and deficiencies, then it can be rewarding. While the manual is helpful, it presents everything seemingly in the wrong order; the audio specification is superb, the effects are generally excellent, and there are enough choices to keep you fiddling for hours. And while irritating in operation, it is tremendous value-for-money. ■

NEW TECHNOLOGIES

SD Instrument mics

SD Systems is a specialist in condenser microphones for acoustic instruments with a number of mics designed specifically for particular instruments. The mics can be fixed in the optimum position and have a shock-free hanging system to eliminate mechanical noise.

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Stringed instruments are catered for by the LCM100HL for double bass and a piezo series for guitars. All systems come with pre-amp power supplies including belt clips.
SD Systems, The Netherlands.
Tel: +31 20 6926413.

FF v3.10

Flying Faders v3.10 software will be the final release of the automation package with Encore taking on the development path for the company's systems. The revision adds a stationary Safe Stop facility and a safer disk formatting procedure which is no longer possible via the copy disk window. Also for improved safety, the mix and special files backup and restore utilities have been relocated from the disk menu. The release also incorporates minor enhancements to the handling of mutes and channel events.

Version 3.10 is available together with a copy of the operator's manual to Flying Faders owners for a small handling charge while freelance engineers may download a copy of the release notes and a full manual from the company's website.
AMS Neve, UK. Tel: +44 1282 417282.
Net: www.ams-neve.com
General Traders, Japan.
Tel: +81 3 3291 2761.

Nady Gold

The Nady Gold series Wireless 441 and 442 feature four user-switchable VHF channels for what the company claims is a third the price of other multichannel systems. The new receivers are compact, rackmountable with front-mounted antennae and feature

¼-inch and XLR output options plus Nady's companding circuitry. The 441 has one antenna, the 442 has two and the company's DigiTRY diversity digital processing for dropout protection.
Nady, US. Tel: +1 510 652 2411. ■

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Reverberation units

Turning the spotlight on reverberation units highlights one of the schisms of modern pro-audio—there's valuable mileage in both high-end and budget boxes. **Studio Sound** looks around and lifts the lid

THE TRANSITION from traditional reverberation generation to digitally driven equivalents was one of the most revolutionary to impact on recording. While modern dynamics and EQ devices often strive to emulate or capture the glory of older devices, apart from a high-quality plate with a hot-rodded front-end, or the luxury of a reverb chamber, few can feel sentimental about the demise of spring reverbs and the other clamourous devices that were the backdrop against which the first digital reverbs arrived.

Digital reverbs appeared originally as expensive dedicated boxes, and it is significant that many of the early players, such as Lexicon with the 224 and 480, are still involved at the forefront of the science. However, democratisation of the art came with the arrival of the Yamaha R100, a basic preset unit with a limited degree of variability, that while still fairly expensive by the standards of the day was orders of magnitude cheaper than the domain's alternatives. Yet it was the Alesis Midiverb that changed the world with afford-

able quality and this in turn was followed by the even cheaper Microverb which threw the simulated cubic capacity/pound ratio out of the window.

It's hard to believe it now, but digital reverb was one of the first demonstrations of a new marketing practice—introduce high-end technology, set a low-cost precedent for it, and then swamp all areas between the two.

Reverb increasingly became the preserve of the multieffects processor which offered embarrassing value for money, but often at the expense of accessibility, and certainly at the expense of individual module quality. Even so, there are great number of really good reverbs hidden away inside multieffects boxes if you search for them hard enough. An encouraging trend is a move towards once more emphasising the reverb aspect of processors.

Anyway, here are the votes of the *Studio Sound* Jury.

George Shilling: The Yamaha SPX90 took everyone's breath away when it first

appeared—multieffects at a fraction of the price of most studio reverb units. The long reverb settings were a bit *clangy* and Japanese sounding, but I still use the pitch shifters, panners and chorus effects and gated reverbs. The more recent, and somewhat underrated SPX990, has wonderful reverb programs —Yamaha sensibly employed a British recording engineer and an American producer to oversee the program design. This unit has some marvellous ambiences, and makes use of enhanced processing power to incorporate chorus and delays where appropriate to supply a rich, well-rounded and useable effect.

Eventide's H3000 has always been a superb machine, and now in its DSE version contains all sorts of goodies. The reverb programs are smooth and natural, sounding expensive yet not overly dense, and the operating system is unbeatable—there is nothing complicated to learn—while a large display shows you all you need to know, even in the darkest studio. This machine is far more approachable than the newer

Eventide Ultra Harmonizer DSP4000

tc electronic M5000

Lexicon PCM90 and PCM80

Yamaha SPX990

DSP4000 which has the most irritating user-interface by comparison.

My favourite Lexicon is the PCM80. It takes some learning time to get the best out of this machine, and will sometimes make you wish you had a simple PCM60. However, there are included all the usual Lexicon-type reverbs, smooth and American sounding, plus a variety of useful unnatural, gated-type settings. These are never as spikey as those found on Japanese units, but have a slightly more classy and clean, sound. I tend to use this unit mostly for plate reverb settings, unless I've found a use for one of the many panning-delayed-gated-chorused-flanged-rotating-thunderclap settings.

Dave Foister: The world of reverberators has its classics, perhaps more than any other, and they're generally the expensive ones. Of those I have a particular soft spot for the Quantec Room Simulator, whose slightly wacky approach gives a great combination of programming possibilities and top-class sound. On the other hand the huge numbers of boxes lower down the market have always tended to polarise opinion, spawning personal favourites and pet hates. There's no doubt that some of these have achieved undeserved success while others have gone undeservedly unnoticed.

One of the most impressive of such boxes I can remember was the Peavey 20/20, one of the first affordable 20-bit multi-effects processors. It brought a simplicity to the creation of new effects that we now take for granted, but was then unheard-of; it had more special algorithms in it than you could shake a stick at, including some of the first speaker and amp simulations to be found in a digital box; and the quality of its reverbs alone would have justified buying one. But hardly anybody did, perhaps because it looked as though it belonged in a guitarist's gig bag rather than in a studio.

An explanation of reverberation

John Watkinson gives the whys and hows of the real world

MULTITRACK RECORDING has evolved as a form of production that allows the greatest flexibility and creativity in the mixdown process. Keeping each sound source on a separate tape track allows complete freedom in the mix to change relative level and to position or pan a source anywhere in the stereo image. This only works if the signals on the tape are truly independent and free of any spatial information.

Independence can be achieved by using close miking, and isolation booths to prevent spill so that microphones only pick up one instrument each. This isolation tends to result in a dry acoustic. An alternative is to use a guide track with synchronous recording so that only one track is recorded at a time. This avoids the possibility of spill, but it still requires that the acoustic is dry so that there are no spatial clues in the recording that might contradict the position in which the signal is panned.

A basic dry mixdown may have correct relative levels and good stereo panning, but it will sound pretty dull without reverberation. As a species we are used to reverberation in real life—it tells us a great deal about the acoustic environment, its size, shape and finish. A dry signal sounds wrong because the reverberation is absent and there is no information about the space. Artificial reverberators, then, are designed to remedy the dry sound of a close-miked multitrack mix by producing a simulated space in which the recording appears to have taken place.

In a real acoustic space, the listener hears direct sound from the apparent sources, followed by reflected sound from various directions. These reflections then give rise to further reflections known as reverberation. Reflected sound has different timbre with respect to the direct sound. This is partly due to the large range of audio wavelengths resulting in different absorbing characteristics of materials at different frequencies. Air itself is an absorber of sound at high frequencies, and even if the reflections are perfect the distance travelled after a few reflections is enough to cause an obvious HF loss. Real reflecting surfaces may cause further HF loss.

After a few reflections the waveform gets pretty messy as different versions of the sound are superimposed with time shifts and timbral changes. However from a spatial standpoint reverberation gets simpler as the energy in the room moves from an irregular excitation to a regular structure determined by the standing wave modes of the room.

Creating artificial reverberation requires an effective simulation of the real process, including the time shifts, the timbral changes due to reflection and absorption and the spatial changes. Clearly a monophonic reverberator will be somewhat simpler because it does not have to consider the spatial aspect at all.

The complexity of simulating reverberation has been attacked by several methods. The reverberation room containing a loudspeaker, reflecting surfaces and a microphone works well because it gives a convincingly complex result, but it is expensive to build and needs good soundproofing. A lot of installations cannot warrant that investment. As an alternative the reverberation plate is smaller than a room, but still quite large and requiring isolation. The reverberation plate is capable of producing a complex output, but it is difficult to change its characteristics.

The artificial reverberator effects unit was basically waiting for digital technology. In the digital domain delay it a piece of cake, filtering to produce timbral changes is straightforward and programmability is easy so that a wide range of different 'sounds' can be obtained simply by changing a few coefficients. Commercially available digital reverberators naturally vary in complexity and cost. The simplest have less flexibility, allowing the user to select one of a number of preset sounds whereas the most complex devices allow the user extremely flexible control. In these the dimensions of the virtual acoustic can be entered from a keyboard, and the virtual reflecting surfaces can be made soft or hard as required.

Reverberators generally have a stereo input and produce a stereo output. The input is delayed and filtered to simulate a reflection, and then the reflection is panned inside the unit so that it appears in the stereo image at a particular place. Simultaneously, other delays are applied to signals which appear at other points in the stereo image, giving the illusion of various reflecting surfaces. But the result can never be as accurate as the image from a stereo microphone in a real acoustic. One reason for this is that the position of sound source affects the reverb. With a stereo input, a reverb unit finds it very difficult to establish where the sound sources are. An ideal reverb would be fed with signals prior to the mix along with the panpot setting. This would mean integrating the reverb with the console, and this would naturally preclude the use of an add-on unit.

The choice of a reverb will often be made on the basis of what is affordable, but it is important to carry out some listening tests once a short list has been decided. Often the most critical test is with the simplest input material, perhaps a single sound source panned centre. This forces the reverb to fill in the rest of the stereo image and its ability or otherwise to do that can be assessed.

It is important to have accurate monitoring speakers so that the effect is properly revealed. In a dry panpotted multitrack mix, each source should have negligible width in the stereo image. When reverb is added, it will occupy the spaces between the panned sources. On poor quality speakers, the images will be smeared so they do not have zero width. These extended sources will mask reverberation and give the impression that there is less reverberation than there actually is. There is no point trying to assess the quality of a reverb unit on poor speakers as its spatial characteristics will be masked.

When applying reverb in production it is vital to be able to monitor the results so that different amounts or types of reverberation can be tried. It is easy to produce a mix with too much reverb on poor speakers.

The Guide

Alesis Midiverb

I have similar feelings for the ART Multiverb II, one of the first boxes I bought, and an immediate favourite for the sheer smoothness and natural quality of its reverberation, 20-bit again, and with a similarly easy and flexible front end, it drives me crazy sometimes with its cheap intermittent front panel sliders, but it's worth hitting it to make it go because it sounds more like a real hall than anything else close to the price did at the time. Bolting its various bits together to create effects from scratch is only a step away from the old DDL approach, a far cry from the type of unit where it never seems worth doing anything other than tweaking a factory preset, and most don't even bother doing that.

Yamaha has always been a byword for affordable reverb, and I think they've really nailed it with the REV500. The algorithms have taken a leap forward, and at the same time the user-interface, not traditionally Yamaha's trump card, is finally friendly and manageable, encouraging experiment and fiddling where older models seem to actively discourage it. With its helpful displays, fast controls and logical management, coupled with a sound quality that would have been unquestionably in the multi-thousand-pound bracket only a few years ago, this has to become as legendary as ever any SPX was.

Zenon Schoepe: I would put the reverb capability of tc electronic's M5000 as among the finest available. The core algorithms are incredibly diverse—many are modelled on famous nice sounding spaces—and it's the depth and movement that impresses me most. I've been a fan of Eventide reverb ever since the original H3000, and this has been preserved as the generation of products has progressed. Admittedly, the DSP4000 offers frightening degrees of adjustment and control, but we need units like this to counter for the aversion to presets. The Eventide is the more contemporary sounding where the M5000 scores more highly on realism.

Realism, or rather non-synthetic results, are produced by KT's DN780 and the Sony MUR201, which some will remember in rebadged Ibanez SDR1000 guise. Neither were particularly fancy, but had an uncanny ability to deliver almost plate-like warmth and glow. You can approximate the results with other boxes by discarding most of the top end, but it's never quite the same. The perfect adjunct to brighter units.

I was shocked at the performance of the Alesis Midiverb when I first heard it—ugly, awkward desktop box that it was—and still think that it sounds pretty good today. Preset 11 is a superb drum 'room' even if the phono sockets do soon work loose.

If I had to choose a traditional multieffects for reverb then it would have to be the Yamaha SPX1000, although it's hard for me to decide how much of that is now down to plain old familiarity after years of being bombarded by generations of the product. The SPX1000 is the best incarnation and furthest development of the SPX90, the box that established the concept of the multieffects pro-cessor, but didn't it look unfriendly when it first arrived? The great thing about the SPX1000's reverb, and it's something that was also true of the SPX90, is a capacity to generate extremely skinny ambiences that sound positively limp in isolation, but actually work incredibly well in the context of a mix. It's almost as if you can strip away anything that you won't hear. ■

The Reverb Listing		Studio Quad	Peavey
		V2	Addverb III
		Ensoniq	Deltafex
		DP/2	**Sony**
Alesis		DP/4+	DPSV77
Microverb 4		**Eventide**	**Roland**
MIDIverb 4		DSP4000	SRV330
Quadraverb 2		H3000DSX	**tc electronic**
AMS Neve		H3500DFX	M2000
RMX-16		**Korg**	Wizard
ART		AM8000R	M5000
FX-1		**Lexicon**	**Yamaha**
FXR		LXP15 Mk.II	ProR3
FXR Elite II		MPX-1	Rev100
Effects		**Reflex**	Rev500
Network		Vortex	SPX990
Digitech		300	SPX1000
Studio 400		480	

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Surround Monitoring

The demand for surround-sound work is on the rise. Tim Goodyer presents an insight into room acoustics and speaker requirements for a professional monitoring environment

THE FACTORS that have occasioned the rise of surround sound are numerous and diverse. They represent the confluence of such forces as the recession's recent ravaging of the high-end studio market, the arrival of the Dolby Pro Logic decoder chip, and the ongoing development of the DVD delivery medium. And they extend their influence over areas as diverse as manufacturers' R&D budgets, recording facilities' business plans, the continuing uptake of modular digital multitrack recorders, recording console design, studio acoustics and loudspeaker design. Dismiss surround sound as a novelty at your considerable peril.

Although recording and mixing for surround is not new to the motion-picture fraternity, its increased relevance to other areas of audio has meant that engineers who previously would have been concerned only with stereo (and mono compatibility) are now having to learn how to work to four or more reproduction channels. This involves new console layouts, additional processing, and new listening environments. Producers, meanwhile, are busy working out how surround sound can best be made to serve new areas such as television drama, television ads, soap operas, sport, and game shows, radio drama, and most recently music recording.

One thing that working with stereo and surround sound share is the importance of a suitable monitoring system. And while there is still debate about the principles behind stereo monitoring, similar debate over surround monitoring has yet to begin in earnest. Many of the specific arguments that have raged over stereo speaker systems—such as driver compatibility, crossover frequencies, off-axis radiation and phase coherence—obviously also apply to surround setups. But here too there are additional considerations, not all of which are as highbrow as those in stereo. For example, Dolby Stereo uses a band-limited (100Hz–7kHz) surround channel, a situation that has led many to assume that all surround speakers can be inferior to the main LR pair. Similarly, the centre channel has acquired a reputation as being 'only' for dialogue —though tellingly not with anyone who has worked seriously with the spoken word.

While surround sound owes almost all of its current acceptance to analogue Dolby Stereo— where left, centre, right and surround channels are phase and amplitude encoded onto a stereo carrier—this has reinforced the lower expectation of the centre and surround replay channels. As the surround channel is a band-limited mono channel delivered through two loudspeakers, it has been easy to short cut on the speaker arrangements in the interests of combining a domestic sitting room with a private cinema. Similarly, the use of 'phantom' centre channels and 'Wide' settings which transfer some of the low frequencies from the centre channel to the LR pair, have further relieved the centre channel of full bandwidth responsibilities. Consequently, the average home cinema setup employs lightweight amplifiers, and under-specified speakers located so as to cause the least domestic conflict.

But if analogue surround has given professionals room to shortcut on their working environments, digital surround systems—such as Dolby AC-3 and DTS—will not. Although matrixing can be used with digital stereo as well as an analogue, digital delivery media will offer five or seven discrete full-

Brookside—one of many British soap operas mixed in surround

From the top: KRK's E8, PMC's MB1, Bag End's ELF system, Genelec's 1037B
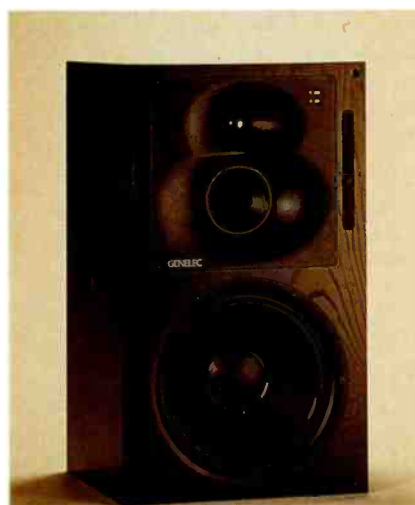
bandwidth audio channels plus a sub—typically these are left, centre, right, left rear and right rear for 5.1, with left centre and right centre added to give 7.1.

The emergence of these formats together with the development of DVD have already seen impressive monitoring systems installed in forward-looking facilities and will force a reassessment on the remainder. JBL reports some 30% of sales of its larger monitors going to centre channel installation in postproduction facilities over the last 12 months.

CINEMA SURROUND BEGAN its rise to stardom in 1975 with George Lucas' epic *Star Wars* and the first use of an analogue 4:2:4 matrix system. Since this auspicious introduction, surround soundtracks have become the norm for heavyweight movies, and their production has followed a consistent line —knowing that the replay systems were the horn-based rigs used by cinema theatres (which were generally bigger than than the 'multiplex' theatres today), the production and postproduction facilities also employed horn-based systems capable of putting out extremely high sound levels. JBL, for example, counts Hollywood sound houses as key business for its 'Cinema' systems. For television (and radio and music) production, however, such sound levels and speaker characteristics are inappropriate, and we need to look to current studio speaker technology for our lead.

Getting away from the misconception of low-spec centre and surround speakers is the first step towards a professional surround monitoring setup. Ideally then, monitoring for a matrixed surround mix requires full-range speakers for each of the left, centre, right and two surround channels although it should be noted that the surround speakers share a common (mono) feed that works best when de-correlated either by room acoustics or, as in many domestic Pro Logic decoders, with DSP. No dedicated effects—or 'boom' channel—is available here as it is with discrete surround formats where it is denoted by the '.1' in 5.1 and 7.1 and has an upper limit of 125Hz (for AC-3). Not that this limitation should necessarily extend to the monitoring chain —Todd-AO's Bag End system has an upper cut-off of around 95Hz.

It should be obvious that the first consideration has to be the size of the room—not simply because it has to accommodate more hardware, but because a physically larger room is necessary if a multichannel system is to be used at relatively high volume.

The next major consideration is that of compatibility between the speakers used. Just as discontinuities in character between drivers in a single cabinet will make the crossover point obvious, discontinuities between the various cabinets in a surround installation will localise the cabinets and destroy the consistency of the soundfield. It is this consideration that makes the choice of centre speaker so critical.

Clearly, anything that is not panned either hard left or hard right across the front of a surround soundstage will find its way into the centre monitor: dialogue, music, and effects. While this does not itself dictate each channel be provided with an identical speaker model—or even that all speakers

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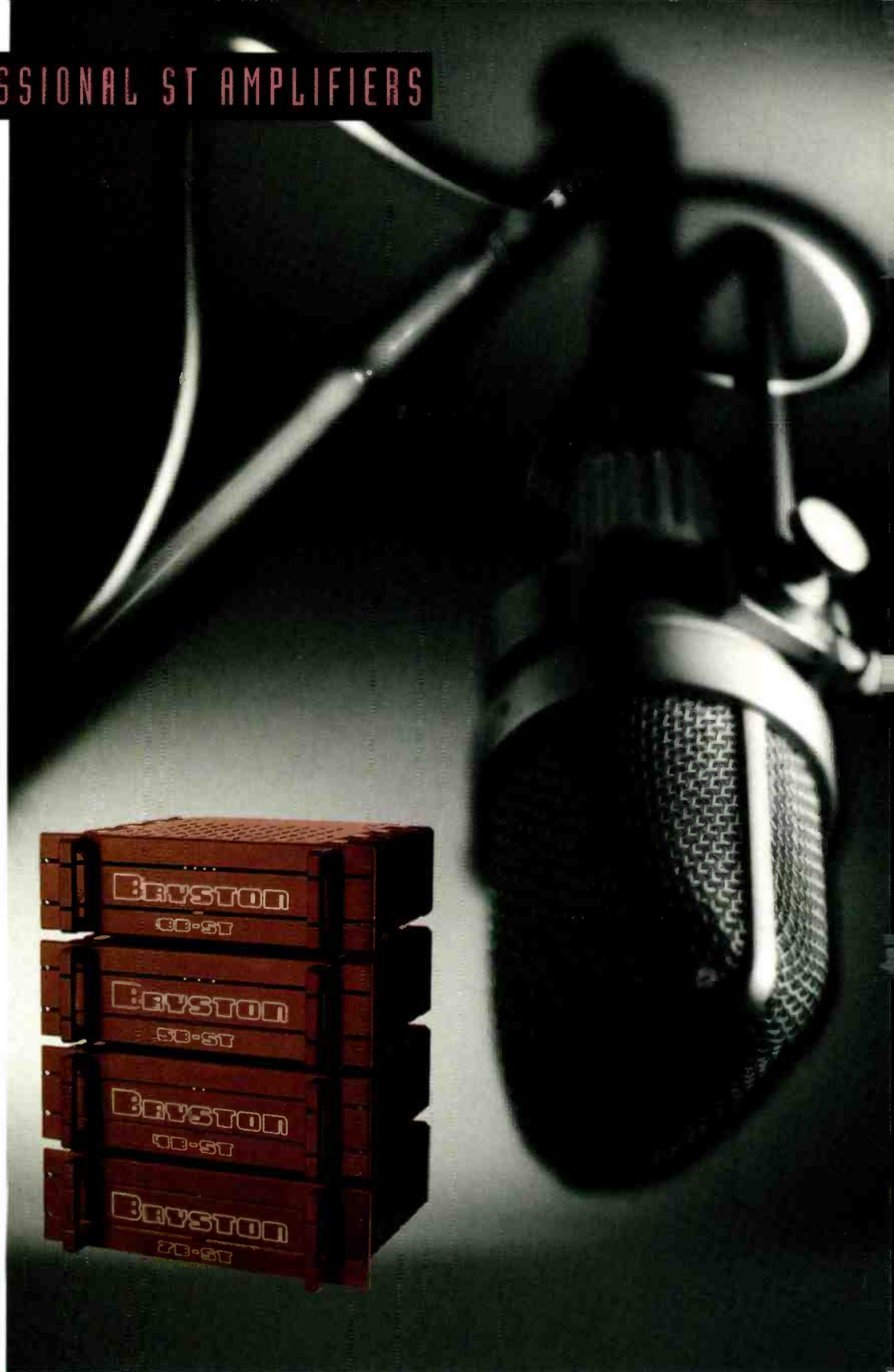
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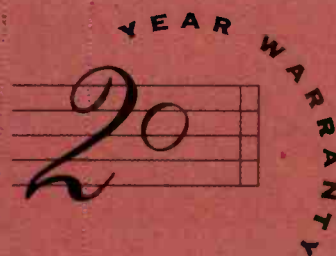
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BRYSTON

FAR SSD Custom system uses horns for LCR positions in large rooms

need to be selected from a single manufacturer's range—it does require that all speakers share a common character. Tannoy's Derek West, who identifies the centre chan-

nel's performance as the most critical of any in a surround setup, highlights the importance of matching tweeter diameters even if selecting from within the Tannoy range. Alan Ainslie puts ATC's philosophy in line with that of Tannoy in that there is no necessity to use identical speakers as long as their timbrality is consistent—he believes that this can be achieved without compromise on the surround channels. In contrast, PMC's Adrian Loader and KRK's Keith Klawitter—whose own background is in film mixing—are insistent that AC-3 and DTS setups should use identical speakers delivering matched sound levels because of the complexity of the soundfield. Consequently, Klawitter reckons all KRK surround installations are set up to within 0.1dB on leaving the factory. On this KRK and ATC concur, as ATC has long considered the off-axis response of a monitor a significant factor in building a stable and consistent soundfield. Ainslie proffers the use of panned pink noise to assess the monitoring environment.

Extrapolating Klawitter's argument, it should be constructive to adopt a speaker that is intended to offer consistency and is available from more than one manufacturer—for small installations this would suggest the long-established and well-respected BBC LS 3/5A. Certainly, the likes of Rogers, Spendor and KEF are happy to see the LS 3/5A considered an option for smaller surround monitoring applications although another LS 3/5A manufacturer sounds a word of caution: Harbeth Acoustics' designer Alan Shore devised the company's P3 monitor to compensate for a

perceived shortcoming in the original LS 3/5 design—that it was intended to be a mono monitor and its profiling of the speech frequencies presents problems when incorporated into a stereo pair. As a result, Harbeth's P3 and Compact 7 offer something of an updated BBC-standard soundfield extended to encompass surround monitoring.

A further word of warning currently accompanies the much-loved LS 3 5A, as KEF has announced that it is to discontinue manufacture of the drivers and cabinets used for the speaker. While this will not immediately affect availability of components to the BBC licensees or the wide availability of the speakers themselves, it is a further factor in the surround equation.

OBVIOUSLY, the main practical problem facing anyone looking to install five or more loudspeakers in a control room is that of their physical positioning—clearly, it is difficult to specify large identical full-range speaker cabinets for all channels if there is no room to site them or if their presence obscures a video monitor or the window through to the live room. The difficulty of addressing these problems has probably helped sustain the myths and popularity of low-performance centre channels and surrounds in professional setups. It is essential to recognise, however, that each channel in a surround setup is important and in a professional monitoring situation, demands as much consideration as each channel in a conventional 2-channel stereo setup. Popular solutions included splitting the for-

ward speakers, leaving the relatively small directional elements to be critically positioned. In this context, proprietary bass enhancement systems such as ABES and ELF may be considered. The ABES, or Active Bass Extension System, is a relatively new development from the European-based DynaudioAcoustics operation, while the Bag End ELF system, with its Extended Low Frequency integrator, comes from the other side of the Atlantic, and has already begun to find favour in West-coast feature film facilities. Bag End's Jim Wishmeyer observes that the attention paid to low-frequency content of the LCR channels in professional environments has increased steadily over the last five years, and that this is now extending to the surround channels.

It is logical to configure the centre speaker—whether it is handling the full bandwidth or only part of it—in the horizontal plane to accommodate a screen, TV monitor, or window. But this may not be as simple as it seems, since certain designs display markedly different dispersion characteristics in a horizontal orientation and others simply don't work at all. At one extreme, we have Tannoy's dual-concentric designs which readily lend themselves to re-orientation off the shelf as the tweeters and bass-midrange drivers share a virtual point source. In the middle ground we have JBL's DMS-1 available as both vertical and horizontal variants, while Zobel's D'Appolito (or mid-tweeter-mid) design gives it dispersion characteristics so highly optimised for stereo close-field operation from the meter bridge that it is totally unsuited to sur-

Coastal Acoustics' Boxers in situ in London's Strongroom studios

round use. It's worth noting that cinema sound systems usually place the centre speakers behind an acoustically 'transparent' screen.

Installation and positioning of speakers is obviously critical but are also dependent upon the acoustics of the listening room. Keith Klawitter's concerns over precise level matching come into their own here, and he advises that LCR placing should be as exact as possible.

Moving backwards through a surround monitor room—a panning technique that reveals much about the consistency of the sound field—we meet the most significant divergence from the stereo music monitoring environment. Where dedicated stereo monitoring rooms require that the rear wall plays a part in controlling early lateral and contra-lateral reflections from speakers mounted at the front of the room, acoustic considerations now have to accommodate speakers located around the periphery of the room. Acoustic treatment, therefore, has to deal with the effect of each local speaker and control the appropriate boundaries.

The positioning of surround speakers, then, is less critical and more circumstantial than the

front in surround terms. Here such fundamental issues as where to put the studio door and how to accommodate a hot spot that may depend on a clear line-of-sight to five or more speakers have to be carefully considered. One of Roger Quested's surround installations saw the LR and S cabinets inverted to ensure line-of-sight between the HF units and the listening area, while Harris Grant Associates' Grand Central installation used the same technique for the surround speakers. Another technique employed by HGA's Neil Grant involves reversing the rear speaker arrays towards a 2-dimensional primitive root diffuser in order to establish virtual sources outside of the physical listening room—this has particular relevance to the problems presented by small rooms.

If the workings of 2-channel stereo have polarised opinion in audio, multichannel working will do so further. Quite apart from the consideration of the mix—where to place the listener with respect to an enveloping soundfield—there is the construction of the soundfield itself. ATC's long-held regard for the off-axis performance of its speakers may find a strong ally

in surround setups.

The conflicting requirements of matrixed and discrete surround channels give rise to specific problems for studio designers: with matrixed surround, the two rear speakers deliver the same signal and may rely on room acoustics to provide a suitably diffuse soundfield. This consideration conflicts with the obvious requirements of discrete surround channels since they can be used to accurately place sounds in the rear field, using signal processing to provide matrix-style diffusion where it is required.

Alongside relatively precise fixed speaker installations are those which use free-standing (often active) close-field monitors for surround work. The placement of these close fields is of particular concern. Excel's Stuart Mitchell, for example. is quick to draw attention to the recognised effect of moving speakers in a stereo setup by even a few inches.

Extend this to a soundfield involving five or more boxes and his point is well made.

It would be a mistake to close the subject of multichannel surround monitoring without mention of 'virtual surround'. The idea of using a pair of speakers or headphones to create a surround soundfield has appeared on a number of manufacturers' agendas—Harman's VMAx and CRL's Sensaura to name a couple. The secret is to create extra 'virtual' speaker sources in order to extend the depth of the soundfield, for which Harman uses its Head Response Transfer Function (HRTF) technology. The advantage of such a system over those requiring multiple speakers will be obvious to mixing engineers and airline passengers alike. But such systems have yet to gain widespread recognition, let alone acceptance. At the moment, if you want to do surround sound, you have to do it with loudspeakers. ∎

Surround Sound Monitoring Systems

Acoustic Energy
AE1 Pro (LCRS); AE108 (sub)
AE2 Pro (LCRS); AE108 (sub)
AmpSpeaker
High-power custom systems
ATC
SCM300A Pro (LCRS)
SCM200A Pro (LCRS)
SCM150A Pro (LCRS)
SCM100A Pro (LCRS)
SCM50A Pro (LCRS)
SCM20A Pro (LCRS)
SCM20 Pro (LCRS)
SCM10 Pro (LCRS)
SCM 0.1/12 (sub)
SCM 0.1/15 (sub)
Audix
Nile V (LCRS)
Studio 1A (LCRS)
Bag End
MM-8/D10E-I with ELF Integrator (LCRS);
D18E-I with ELF Integrator (sub)
Crystal-I/D18E-I with ELF Integrator (LCR); TA12-I (S); D18E-I with ELF Integrator (sub)
Coastal Acoustics
Boxer T5 (LR); Boxer T3 (CS); B5 (sub)
Boxer G3 (LCR); Boxer G10 (S); Boxer B3 (sub)
DynaudioAcoustics
C4 (LCRS); Surround extender (x2); ABES (x4)
M3 (LCR); BM15 (S); custom sub
M2 (LCRS); ABES unit
BM15A; ABES unit
BM6A (LCRS); Surround extender; ABES unit
Excel Audio Systems
201A (LR); 101A (CS)
FAR
SSD10—CR10 (LCR); CR10 (S); 1x15-inch subwoofer
SSD20—CR20 (LCR); CR10Y (S); 1x15-inch subwoofer
SSD80—DBW80 (LCR); CR10 (S); 1x15-inch subwoofer

SSD100—CR100 (LCR); CR20 (S); 2x15-inch subwoofer
SSD Custom System
Genelec
1031A (LCRS); 1092A (sub)
1031A (LCRS); 1092A (sub)
1032A (LCR); 1031A (S);
1094A (sub)
S30CV (LCR); 1031A (S);
1094A (sub)
1037B (LCR); 1032A (S);
1094A (sub)
1038A (LCR); 1032A (S);
1094A (sub)
1039A (LCR); 1032A (S)
1035B (LCR); 1032A (S)
Harbeth Acoustics
P3 (LR); Compact 7 (CS)
LS 3/5A (LCRS)
LS 5/12A (LCRS)
JBL
6208 (LCRS)
DMS-1/4400 (LCR); 6208/Control series
4400 series/JBL Synthesis (LCRS)
6208/4200/Control series (LCR); Control series (S)
DMS-1/4430 (LCR);
8830/4408/Control series (S)
Cinema 4000/3000 (LCR);
8330/3310 (S)
Cinema 5000/4000 (LCR);
8340/8330 (S)
subs available for all configurations
KEF
LS 3/5A (LCRS)
KRK
E8 (LCRS); Rok Bottom (sub)
9000 (LCRS); Rok Bottom (sub)
6000 (LCRS); Rok Bottom (sub)
Meyer Sound
HD1 (LCRS); PSW2 subwoofer
HD2 (LCRS); PSW2 subwoofer
PMC
TB1S (LCRS); XB1 (sub)
LB1 (LCR); TB1S (S); XB1 (sub)
IB1S (LCRS); IB1S Custom (C)
MB1/P (LCRS) MP1/P Custom (C)
BB5/XBD (LCRS); MB1 (S)
Quested
VH2108 (LCR); H108 (CS);
VS1112/1115 (sub)
H208 (LCR); VH2108 (C); H108

(S); VS1115 (sub)
HQ210 (LCR); H208 (C);
H108/VS2205 (S); QSB121 (sub)
HQ410 (LCR); HQ210 (C);
H108/VH2108/VS2205 (S);
QSB121 (sub)
VS2205 (LCRS); VS1112 (sub)
VS2108 (LCR); VS2205 (S);
VS1112 (sub)
HQ210A (LCR); H208 (C);
H108/VH2108/V2205/VS2108 (S); QSB121 (sub)
HQ410A (LCR); HQ210A (C);
H108/VH2108/V2205/VS2108 (S); QSB121 (sub)
Rogers
LS 3/5A (LCRS)
LS 5/8A (LCRS)
LS 5/9 (LCRS)
Spendor
SA500 (LR); SA300 (CS)
QT100/SA200 (S)
SA300 (LR); 200 (C);
QT100/SA200 (S)
LS 3/5A (LCRS)
Tannoy
System 800 (LCR); System 600 (S)
AMS8 (LCR); System 600 (S)
System 15 II (LCR);
System 1200 (S); B475 subwoofer
Westlake
Lc6.75 (LCRS); Lc265.1 (C);
Lc6.75SW (subwoofer)
Lc8.1 (LCRS);
Lc8.1SW (subwoofer)
Lc3w10/12 (LCRS)
BBSM-4/5/6/8 (LCRS);
BB-SWP (subwoofer)
BBSM-10 (LCRS);
BB10-SWP (subwoofer)
BBSM-12/15/15t (LCRS)
Reference TM-3 (LCRS)
Reference HR-1 (LCRS)

Note: Where possible, the table lists example systems with specific speaker applications denoted by LCRS and sub (Left, Centre, Right, Surround, subwoofer); alternative options for certain speakers are also included. Other entries indicate available speakers or custom services.

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Star Trek's Sound Generation

Some 30 years after the original *Star Trek* hit the small screen, the saga continues in three TV series and several films. **Richard Buskin** visits LA's Modern Sound, home to a decade's sci-fi sound

T'S FUNNY. No matter how far you go out into space they all seem to speak English.' says Jim Wolvington, sound designer for all three *Star Trek* television series that have hit the small screen during the past ten years. 'Still, you sometimes want to give the characters some kind of interesting alien caste, and it can be really tough to come up with processes that don't make their words unintelligible.'

In the beginning it was also difficult to devise futuristic sound effects that lived up to some of the well-known noises that we all associate with the original 1960s episodes, featuring Kirk, Spock, Bones and company; the 'wip-wip' of the walkie-talkies, the 'scheep' of the door.

'I have profound respect for what they did on that show,' says Wolvington. 'I mean, it was heavy-handed dramatically and it was heavy-handed sonically too—the sounds on the bridge were so loud, and in some ways I quite envy that. At the same time, they did some incredibly clever things, and when I first started on *The Next Generation* it was quite a challenge to come up with a newer, higher-tech version of the original... In fact, in some cases I think what we ended up with was incredibly bland by comparison'

Modern Sound in LA has been the *Star Trek* post facility of choice ever since the inception of *The Next Generation* back in 1987. Together with the subsequent *Deep Space Nine* and *Voyager* series, at the rate of 26 shows every 10-months, that amounts to more than 250 episodes. Between 1987 and 1990 it also amounted to four consecutive Emmy Awards for Best Sound.

While the facility takes care of ADR, mixing and editing, Jim Wolvington designs the sound effects and relays his work to Modern Sound via ISDN from his home in Vermont. In LA, fellow sound designer and effects editor, Tomi Tomita, integrates Wolvington's tracks with his own work. Until recently, the Foley was also part of the Modern Sound/Star Trek agenda, but this has now been shifted to the new Foley stage at Paramount.

In addition to designing the sound for the aforementioned TV series, Wolvington has served as the supervising sound editor on the last two movies, *Star Trek Generations* and *Star Trek—First Contact*.

'In terms of the way that we approach the sound itself, the movies and the TV shows are actually quite similar,' he says. 'What I do find to be different, however, is that in the film form the picture never stops changing, which

Doug Davey,
Effects Mixer
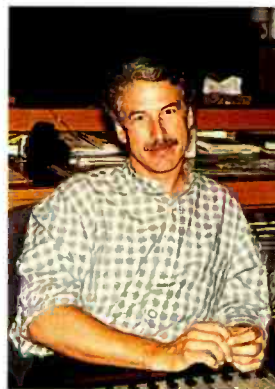
Richard Morrison,
Music and Foley Mixer

Chris Haire,
Dialogue Rerecording Mixer

means that everyday you're cutting literally hundreds of tracks to conform to the new picture. You have to dedicate a large percentage of your labour to basically just keeping old sounds current with those new pictures. With the television show, on the other hand, they've really managed to make it fairly predictable. I get a cassette three to four weeks ahead of when it mixes, and I get the optical animated effects several days before the mix. That never happens with film. Otherwise, it's a remarkable similar process.'

Until 1994 Jim Wolvington worked close to LA, and it was therefore easy for him to go to a spotting session and learn what sounds were required. Now, however, with more than 200 shows behind him, he is able to anticipate what is needed about 90% of the time, while for the remainder he acts according to the notes that are faxed to him.

'We get a cassette of the new show and I just FedEx it to Jim,' says the TV series' supervising sound editor, Bill Wistrom. 'By the time we get in the next morning he's already been working on it for three hours, and so anything we need he sends down the pipe and we're off and running.

'I have a work station here with a Synclavier, Tascam DA-88s, a little mixer and a patchbay.' adds Wolvington. 'It's basically an identical system and database to the one that Tomi Tomita works on in California, and I therefore either FedEx Tascam DA-88 8-track tapes with the effects laid down, or I modem sequences that allow Tomi to recreate what I've done.'

'Jim typically supplies nothing short of superb tracks, and I'm really fortunate to have the opportunity to deal with them,' asserts effects mixer Doug Davey. 'With no disrespect to the other editors, he is a phenomenal sound designer.'

At Modern Sound, Bill Wistrom coordinates the dialogue editing and mixing, ADR, Foley... 'everything but the effects.' Like Jim Wolvington, Wistrom has been involved with *Star Trek* since 1987, since when he states that in some ways the job has got easier, and in some ways it's got a whole lot harder. 'The technology has been growing. We started with Synclaviers and a clunky old Cass computer system for editing dialogue and so forth, and since we've moved over to the Fairlight our job has become much, much nicer.

'We've always recorded to digital, even though the dialogue and so on comes in on analogue. Currently the dialogue is still recorded on ¼-inch, but we'd like to go to DAT soon. For the ADR we record to a Fairlight and an M-O, and our Foley also goes onto M-O, so it stays in the digital domain quite a bit, whereas before the analogue editing was quite a bit slower.'

Nevertheless, while the new studio gear has helped facilitate matters, the post schedules certainly haven't made life any easier. Production values on all three *Star Trek* shows have been extremely high—perhaps higher than anything else on TV—and so there is a direct correlation between producer expectations and the available technology.

'It's a creative process,' says Wistrom philosophically. 'What I like about these shows is that everybody's trying to do the best job possible. That includes the producers, and if

they're not happy with something then it takes more time to get it right for all of us. However, that's okay, because in the end I think the shows come out looking and sounding awfully good.'

In terms of the dialogue, Modern Sound has two resident editors who—as the dailies come in—cut, back-fill and clean across a maximum of eight tracks. The ADR is shot to a Fairlight on the adjacent stage and recorded onto M-O. Then, after the dialogue editors have cut the print takes and everything is in sync on 24-track, dialogue rerecording mixer Chris Haire goes to work.

He, music and Foley mixer Richard Morrison, and Doug Davey, all sit at an 11-year-old, 72-input, G-series SSL 6000 console. Each man therefore has 24 faders to play with, Haire mixes down about 16 tracks of dialogue to produce the composite track, and he then shares the sound effects with Davey.

Basically, the hard effects are built on 20 tracks using Tascam DA-88s: 8 tracks of mono effects and 6 stereo pairs. That doesn't leave Davey with many tracks of his own, which is why he and Haire split the stereo backgrounds.

'That's a typical layout,' says Davey. 'Sometimes I'll build hard effects on the background tracks where necessary. However, because we do surround encode the majority of the background tracks to a slight degree, I try to avoid cutting too many hard effects on there because it requires quite a lot of bus and panning changes. On a fully automated desk— with something other than just a G series automated fader—that wouldn't be a concern, but in our case it requires a lot of mental gymnastics, and it's very difficult to get everything correctly surround encoded and mapped out.

'I also have four automated pan pots—each up to 8 channels—and I rely on them quite heavily for ship-by effects, and effects that are moving overhead, and from front to back on-screen. I use them extensively, and I can still maintain the stereo integrity of the effects and get a nice, smooth even pan until they hit the surrounds, which, of course, are mono at this point in time. Hopefully, we'll get split-surround TV pretty quick.'

The music, which is scored by a variety of composers and produced at different facilities around town, varies from week to week. Still, as with the Foley and effects, the procedure is the same for both shows, save for the necessity to process sounds differently according to the dissimilar on-screen surroundings and the theme of a particular episode.

'With the exception of the main titles the music is entirely different with each episode,' confirms Morrison. 'It's not recycled at all. Only during the first years of *The Next Generation*—and then only on very, very, few occasions—did we go to a different episode to track in cues. Now that just doesn't happen.'

The music is mastered to 24-track 2-inch consisting of a 12-track score submix that is supplied to the stage.

'The 12-track lays out generally in pairs,' explains Richard Morrison. 'There's the high-string pair, a low-string pair, a woodwind /brass pair, a keyboards pair, and **page 58 >**

Right. A tense moment from DS9 as Quark (Armin Shimerman) goes up against Major Kira Nerys (Nana Visitor)
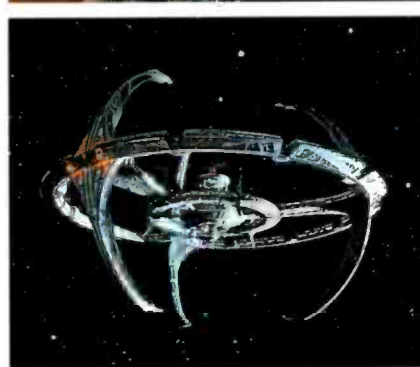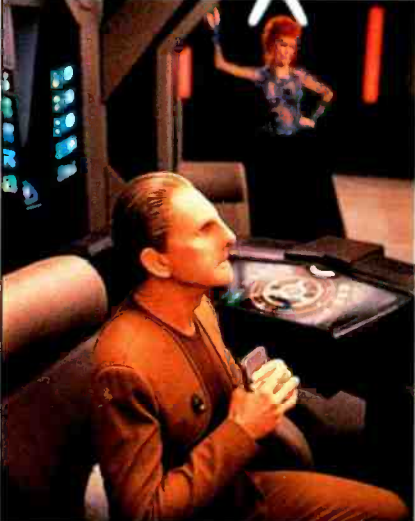
Top: Interstellar intimacy for Voyager's Neelix (Ethan Philips) and Kes (Jennifer Lien). Third from the top: René Auberjonois as DS9's Constable Odo

< page 57 then a mono percussion—sometimes a stereo percussion—an electronic woodwind instrument that is on a mono track, and a pair of room mics. It's laid out for me pretty consistently that way. Meanwhile, we've already gone through and mixed the main-title end-credits, and so that's just on a 2-track DAT that we fly in every week. Then the Foley consists of either 9 or 10 tracks, and that fills up my frame.

'Having just 24 inputs is a problem when there are two 12-track music cues that are crossing, as well as 10 tracks of Foley. In that I case I have to do a quick premix of the area where I can get 12 tracks of the cue mixed down to 2 tracks, and then I cross into that so that I can free up those 12 faders to access the other 12, and then cross back into it. The second set of 12 are on my line 2s—or the second input on each module—so I'm able to cross out and cross back in quickly, and still contain as much independent control as possible. Because, sure enough, the producers will come in and want to change the balance. It's unfortunate if that happens to be in an area that I've already predubbed.

'Most often the changes in the music involve sync. The producers have specific ideas and so we move the music a frame at a time. They're most particular about the start, and we'll go over it until it feels just right. If it happens to be of any great duration that may involve some sort of on-the-spot editing; it may be a move of six or eight frames and it needs to get back into sync within a few seconds. It's rare when I don't have a half dozen of those instances during a show. That's another reason why it's spread out over 12 tracks; it gives the producers the control they need. When it originally came over here it used to be on 6 tracks and they just kept asking for more control. As a result it started growing, and I'm sure they would use 24 if I could handle it.'

Morrison processes the music through a Lexicon 224 XL, Foley goes through a 480L, and then a second 480L is used for any additional music processing, which generally amounts to source cues. There are also a couple of Lexicon PCM42s for delay lines.

'When the tracks score we try to keep them as dry as possible.' Morrison says. 'We therefore add the reverb here on the mix stage so that we have better control over that.'

On a busy episode there will usually be around a hundred tracks to deal with, and it is therefore a struggle to transport that much information into the 72-channel SSL 6000.

'The producers don't like to predub,' says Doug Davey. 'They like as much control as possible down to the very last moment, and for that reason we don't predub. There's never been one show, regardless of how busy it is in terms of sound effects or whatever, where we've ever done any predubbing other than the spots where Richard's referring to for music crosses only. Even though it may mean that tracks are jammed very **page 60 >**

Left: ModernSound's Bill Wistrom coordinates dialogue editing and mixing, ADR and Foley

< page 58 close to one another on a certain channel strip, we do what we have to do to keep all of the information on the desk at one time and not be locked into predub balances.'

'We now have a Sonic Solutions, we have all of these customised programs for the different rooms and locations, and we have a sequencer for our programs,' adds Richard Morrison. 'The Sonic was first used upstairs by the dialogue editors for *Deep Space Nine*, and for the past three years it's been in here with us.'

All the stems are recorded through a 3324. Meanwhile, for monitoring the three men use Westlake BBSM10s at left and right, while in the centre there are BBSM8s. The surrounds are Boston Acoustics. The close-fields are Yamaha NS10s.

'From a dialogue standpoint, the material that we're supplied with is not very good,' says Chris Haire. 'They're working in very difficult locations. There are tracking shots where the camera is being dollied back with the action as they're moving forward down a corridor, and consequently you pick up the shuffling footsteps of the entire crew. It's very unusual to be able to utilise the production sound—usually it's nothing but a cue track for ADR. 'On the positive side the sound effects help us a lot. There's always some kind of ship presence or room tone, and they usually want this massive sound, so there's this low-frequency rumble with maybe a mid-range element in there too. That's the case with anywhere we go, unless it's down on a planet where there's an atmospheric environment. I'm therefore able to hide a lot of production

ills within that background.'

Still, technical problems are not the only cause of dialogue replacement. The producers place a lot of emphasis on the clarity of speech, and that means not only rising above extraneous noises, but also steering through strong accents, masks and false teeth. 'It's a very dialogue-orientated show in addition to the sound effects.' says Haire. 'A typical *Star Trek* episode averages about 150 lines of ADR, and a heavy show will be in the 300s. 300 lines is about half—or maybe more—of the total. That, however, is down to the fact that the producers want every word to be pristine; clean and perfectly spoken. If a word is slurred or its pronunciation is too colloquial, then the actor will be brought in to redo the line.'

Of course, the knowledge that postproduction will fix things can also encourage sloppiness during the production phase. 'The manpower that is used all the way through the postproduction process on *Star Trek* is unlike any other show that I'm aware of,' says Haire. 'However, things are much smoother now than they were during the early days of *The Next Generation*. Our typical dub time now for this process is two 10-hour days. When we started *Next Generation* the minimum was two 12-hour days, and sometimes more than that. So, it has streamlined.

'We've had a little less producer involvement along the way. although in the case of *Deep Space Nine* the opposite has been true.'

'They've done what they can to quieten the sets down too,' adds Doug Davey. 'You know,

changing the floors and so on. When they started *Deep Space* they had a lot of metal surfaces which have since been changed...'

'I remember during the first year of *Next Generation* there were some real obvious things that we had to contend with,' interjects Haire. 'Somebody would walk along and the plywood floor would creak and squeak. Plus it was a raised floor and it was very boomy, and so, what with the boom, the creaks and so on the whole thing had to be looped.'

That's all well and good, but still, isn't it a little surprising that, in this most experienced of TV and film towns, things such as creaking floorboards aren't tested and taken care of well in advance? After all, once the production is rolling it's then costly to call a halt while a man with a screwdriver has to do his job. At that point it's cheaper to leave things to ADR.

'All of this noise could easily have been fixed with a little bit of fibreglass bedding under the floor surface as well as some tighter screws,' says Chris Haire. 'After the first year they did make a concerted effort to do that, as well as to get the balance transformers away from where they were shooting. That worked, but you still only have to look at the sets that they're working with on *Star Trek* and you can tell that they're going to be very difficult, not just in terms of extraneous noises, but also to shoot in and around.'

Which is what helps to keep Modern Sound busy. Still, according to Haire, the *Star Trek* work is now basically a routine. 'We've been in space for over ten years,' he says, 'and we've got it honed to a science!' ∎

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Keith Olsen

From the Summer of Love to the Winter of Content, Keith Olsen has balanced technical suss with musical sensibility and produced some classic recordings. **Dan Daley** shares his view of the changing ages

OMETHING SEEMS to happen to people from colder climes when they get to LA. Something about the palm trees, perhaps, which are as non-indigenous to Southern California as most of the people who live there. More likely, it's the weather. It's manifest in a physical and intellectual freedom that comes with the knowledge that you will never be shut in again by 7-foot snow drifts—the kind commonly found on the plains around Sioux Falls, South Dakota, where Keith Olsen was born.

After growing up in the equally frigid environs of Wayzata, Minnesota, Olsen attended the University of Minnesota where he got hooked up as the bass player with a few local bands, one of which, as he puts it, 'dumped me off in California in the late 1960s'.

It was a fortuitous dumping, indeed, for Olsen went on to become one of the seminal

producers of radio rock in the 1970s and 1980s. His oeuvre is prodigious—Fleetwood Mac, Eddie Money, Emerson, Lake & Palmer, REO Speedwagon, Rick Springfield, Starship, Heart, Joe Walsh, Stevie Nicks, Whitesnake, Kim Carnes, Sammy Hagar, Santana, Pat Benatar, Foreigner and the Grateful Dead. See what getting out of the snow can do?

But how did one transition from journeyman musician to the other side of the glass? In a sense, how did one not, back in the days of the Summer of Love. 'A producer in those days was a guy who used to tell you how good your take was,' says Olsen, sitting in his publisher's office in Nashville, where he is working on getting this town to open up its catalogue coffers for material to work on his next project, multichannel surround mixes for Olsen's KORE Group Records. But more on that later.

After an introduction to production from the late Kurt Baetchler, who produced the Association's classics 'Cherish' and 'Along Comes Mary', Olsen helped form the Music Machine whose radio hit 'Talk Talk' got Olsen on the road again, but at a higher level, and kept him tangentially tied to the evolving LA music loop. Upon his return to LA, he was ready to take a shot at his own productions, but first, he recalls, he realised that, as the equipment of recording was evolving so quickly, he needed to spruce up

his technical chops.

'I had enough electronic training in college to know what's going on underneath the desk and enough music training to know what should be going on on the other side of the glass. And then these good opportunities to go start working with some bands started coming in—this was around 1971. But even though I had learned the musical palette, I needed to know more about engineering. So I hooked up with Gary Paxton's studio in his garage, then with Sound City in the Valley. The board was a couple of equalisers, an old mixer and some wire switches. I think it was a 3-track machine, then a 4-track, a Scully. But I was learning the ropes and meeting people. I met Brian Wilson at his house just after the Beach Boys had done *Smiley Smile*. And he was a little outside even then. But I learned from him to envision everything about a production as you hear the song the first time. You've got to see the whole picture, and get to the point where you can see and hear where everything should be. It takes a while to learn how to do that.

'I also learned that when you get into the studio, you have to be able to modify your vision. But from the very beginning I learned that I always wanted to be prepared when I walked into a recording studio.'

The engineering experience he gained was critical to the success of future

Sleeve notes: The Best of the Music Machine

'IT GIVES US great pleasure to finally make available a consummate package by one of the best rock bands to emerge from the mid 1960s. Because of their aggressive attack and dress—all black, including dyed hair and black glove on one hand only—the group was placed in the vanguard of the punk rock boom. But the Music Machine was much more than that.

'Songwriter Sean Bonniwell assumed various provocative stances and propelled his men through a series of successful experiments, unusual approaches to tuning, use of cymbals, bass emphasis, electronic guitar sounds, an early version of the fuzz box created by bassist (now producer) Keith Olsen, all aided by the production team consisting of Brian Ross, the producer, and Paul Buff the recording engineer. The latter, an electronic genius, invented a 10-track recording machine during a time when most other advanced studios were struggling with four tracks. The records of the Music Machine just might have been the most state-of-the-art of their day.

'The group's very first single, "Talk Talk", provides a good example. An intense whirlwind of disillusion, the song is based on a series of stops and starts that Sean refers to as 'Chinese Jazz'. The only time the title is mentioned is at the four beats at the very end. Right off the group forged its own identity, characterised by fuzz guitar and Farfisa organ, swathed in an aura of mystery. Sean sang and played guitar and wrote the songs. The rest of the band consisted of Olsen, Mark London, lead guitar, Ron Edgar, drums, Doug Rhodes, organ. This is the line-up that played on the Music Machine's better known recordings. Although the group was popular in its native Los Angeles, and while various singles achieved regional recognition, save for "Talk Talk" the band's gems have been long waited to be rediscovered. That time has finally arrived.'

< page 64 productions, enabling him to become a producer in a more traditional manner—switching chairs in the middle of a recording session when Jerry Wexler gave him the green light to finish Mac Rebenack's Dr John's *Gumbo* record in 1972. The credit opened the door for Olsen to hang out his producer's shingle. The first act he worked with was Lindsey Buckingham and Stevie Nicks before they assisted Fleetwood Mac into its pop phase. Both the music and the technology were about to emerge from their respective incubators at the same time, with a synchronicity that was perfectly timed for Olsen and his innate pop sensibilities.

'The technology was starting to get better,' he remembers. 'The consoles, the microphones could all take ever-higher SPL levels, so we weren't distorting the preamps every time the snare drum hit.

'For a while I had been doing a bunch of weird acts—the Grateful Dead and The Sons of Champlain—real San Francisco acts. I did *Terrapin Station* with the Dead. I was really learning to be a producer working with acts like that.

'*Terrapin Station* was an album that we worked on for four weeks before we got our first basic track. Speaking of envisioning a record before it starts—I was looking to hear something tight-sounding, and I picked the wrong band to do that with [drummer] Mickey Hart played on top of the beat and [percussionist] Bill Kreutzmann played behind it, so everything sounded like a flam. After about a week of that, I suggested the possibility of orchestrating the drum parts to Jerry [Garcia]. And Jerry says, "Sure, man, see you later," and walks out. His way of saying, "Just go ahead and do it". So I had Kreutzmann be the kit player and Hart as the percussionist, cause he's the fire anyway. And it worked. Instead of having two drummers playing against each other, we had a little groove thing going.'

It wasn't enough to get the Dead big radio hits—that would have to wait another 15 years for 'I Will Survive'. But the lugubriousness of the San Francisco music that was coming to LA to record was a useful learning tank for Olsen as pop music and recording technology were about to join forces and put corporate rock on the map in a big way. On Foreigner's 1978 *Double Vision* album, Olsen was looking for new ways to cut guitar sounds and stumbled onto the notion of using the inherent fuzziness of wireless systems when guitarist Mick Jones showed a predilection for staying in the control room while playing at New York's Atlantic Studios.

'The long cables running from the control room to the studio were taking the top off the sound,' Olsen says. 'We didn't have amazing cabling then like we do now. It was more like zip cord. But the gain from the transmitter on the wireless gave us a little more input power, and that gave us back a very cool top end. The amps were set up in the studio in a corner—when you corner-load the cabinets you get a really great bottom end out of them. And we miked them with some kind of Shure mics—that was about all we had for high SPL mics back then. And we used the room for natural reverb with some mics placed relatively close around the amp as well as right on the speakers.

'For the drums, I built a riser with cinder blocks from a construction site and thick plywood on top—there was no ring to the drum sound any more. The drums had snap and power. Combined with the angle of the mics on the drums, we had eliminated the ringiness of the kit. I had figured out part of that when I was working with Fleetwood Mac; their drum sound was so tight because the room was so dead. I had been experimenting with recording in more live environments, but I found that if you had a real live environment you get so much destructive interference from reflections that you get a real roomy drum sound but you don't really get any of the power, the snap, punch and crack.

'On Lou's vocals, I used a technique that I had been using since Buckingham Nicks days, processing similar to what George Martin had used with the Beatles. I took a Dolby noise reduction card and started clipping the odd bit off, a transistor or two. You then put the card on the encode side instead of a compressor. Use it as an expander—it's level-sensitive and band-sensitive, and all of a sudden you have this really tight and airy vocal sound without having to go through intense amounts of EQ or compression. I don't think the Dolby people liked me for using their stuff that way.'

Like it or not, Olsen was looking to create the sound in the tracks, not in the mix—a classic approach in the days before remixers had even been heard of, and one that helped define the radio sound of the day.

'You learn about spectrum mixing—not putting everything into the same EQ area. I was layering things, but I was also using the musical arrangements to get the sound together. Not so much using EQ, but arrangement. The one thing I learned was, remember the source—where the music comes from. Great words to live by, because all the gear in the world cannot make a bad guitar player play great. I also didn't like leaving things

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until the very end in the mix. I liked to put it together the way I heard it the first time in my head. If the artist is fine with that, then everything's moving ahead, but if not, then you have to try things a few different ways and the mixes get a little more hairy because you're then trying to take two visions and put them together. That can make for a difficult mix.'

One that wasn't difficult was Rick Springfield's classic 'Jessie's Girl'.

'Here you had a soap opera star who was actually a very creative, musical guy. He was very inexperienced in the studio but we had the basics cut for that track in one day by 4 o'clock. Bing, bang, done. That's how records were often done in those days. Looking back, it was remarkably fast.'

Then there were the ones that didn't go quite as smoothly. Preparing for the follow-up to Whitesnake's hugely successful *Slide It In* in 1987, lead singer and band leader David Coverdale had been working with another producer at Compass Point Studios in the Bahamas. But Coverdale was hung up on the vocals, and he and cowriter John Sykes called Olsen in mid-way through the project.

'He calls me and sends me the tapes and I realised that the problem was that David couldn't sing in tune because every guitar and bass track on the record was, shall we say, outside the window of acceptability, pitchwise. They had printed a lot of effects like harmonisers and somebody went a little bit overboard with the outboard. They were going for a sound, but they weren't looking at the big picture. If you can't tune to it, you can't overdub or sing to it. So I had to go in and rebuild, taking tracks that were half of stereo tracks, but which had less effects on them and put them together to form new tracks. We were looking for things that could pitch references. And the drums were also a bit off timing. I and engineer Brian Foracker was using two machines to do punch-ins and fly around, and offset the beats. I built a 32-track digital master out of all of these analogue tapes. It took a month, but it gave us workable tracks for the album. That and a few new overdubs with guitarist Dann Huff. Then I called David and I said "I think it's time for you to sing now". He came down and on the first day we punched play and I started him off with an easy one, which turned out to be "Still Of The Night". So it worked out quite well in the end.'

So did a few others, including Heart's 'Passionworks', REO Speedwagon's 'Here With Me' and Pat Benatar's 'Precious Time', not to mention Fleetwood Mac's 'Rhiannon' and 'Over My Head', which took the band's eponymously named first album to 9 million in sales.

Olsen has a new project now, based on a new technology. DVD offers over 9Gb of storage capacity, which makes the *Red Book* CD look like a piker, indeed, with its barely 650Mb of data. While name-brand technology has never held any particular allure for Olsen—his 20-year-old personal studio, Goodnight LA, is fitted with a 96-input Trident Di-an console, and of which he laughs, 'I'm the only man in the world using one on a semi-regular basis'—DVD holds some business potential that he finds irresistible.

The rapid growth of home theatre: over 10 million US households have some type of

Keith Olsen mixing it with Ozzie Osbourne

surround-sound system, and another 23 million are equipped with Dolby's Pro Logic matrix surround. This has opened the possibility of delivering surround music mixes on DVD, and a truncated matrixed version on CD, the latter of which Olsen started doing last year with the formation of the KORE Group record labels. The latter of these he is gearing up for as the new disc technology slowly, but inexorably comes on line. Two new artists, one in jazz and one in new age, and two sampler discs of remixed material, have been compiled thus far.

'The idea is feasible and viable technologically,' says Olsen. 'We needed to see if it was also feasible and viable in the market. If the market was there for surround mixed music. Twenty-five million people go down to the video store twice a week, get a video, stick it into their surround-sound home theatres and for 96 minutes they sit there and get whacked by surround-sound effects. Then they turn off their VCRs and if by chance they have left the Pro Logic decoder on when they put in a CD, the centre collapses on the sound. It doesn't work. So I say, why don't we make the mix an event? Make it an experience to listen to music again. So we started mixing a few things I had in the vault and a few things friends gave me—Tom Petty, Stevie Nicks. I mixed that in surround sound keeping the artistic intent of the original mix in mind—remember what I said, remember the source? The same tone and the same placement of instruments, but everything else just spreading around you.'

What is the secret to mixing in 5.1 versus stereo?

'The thing in surround mixing in general is not how much you do, but how much you don't do,' cautions Olsen. 'That seems to be the secret. You use effects in a different way. You don't want to load up the front of the mix with effects; you can use them in the rear channels.'

Olsen gives an example of a Pro Logic 4-channel mix: 'We place the guitars left and right, we put the drums kind of spread out among all three front channels, then move them in just a little bit so it doesn't become all kick and snare. The bass [guitar] is mostly in the centre. The lead vocal? Well, gee, there's a band here, so let's have the lead vocalist take two steps forward towards you, like he is on stage. Then you take a bit of the vocal or its effects and put it behind you to achieve that. Put that into the surround and all of a sudden the vocal is in front of the band, but still part of it. But there are delays on these home systems and people screw with them, so you have to be careful about not placing things too far out and moving them around like crazy. Don't go crazy because it makes people nauseous to the point where they don't want to listen to it. There's the potential for high listener fatigue in surround mixing. The Dolby Pro Logic system gave this a lot of thought because it's also compatible to stereo and to mono, which is very handy for this market.'

The KORE Group's plan is to license existing masters, remix them for surround, and rerelease them, using a variety of direct mail and retail outlets. The project started three years ago and pulled Olsen out of the production loop—voluntarily, he says.

'I took three years out of my life to figure the surround thing out. And I'm not going to do anything else until I figure the business side out because I think this thing is going to be massive. In 1998, GM, Ford, Toyota and Mazda are supposed to be coming out with surround systems in their cars, so that's really going to open the door. DVD is still a buzzword for the next six years or so, but it'll be there, too. To paraphrase Jerry Garcia, it's been a long, strange trip. But I like where it's going.' ■

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Glorious Mud

The swamp is alive with the sound of music. **Kevin Hilton** tells of the broadcasting of Europe's largest music festival at Glastonbury

Soon realising that I wasn't going to buy one of his hideously over-priced carved wooden animals, the stall-holder engaged me in idle conversation. There was no-one to hold up; the rain streamed from the awning above the trestle table and right down the neck of my waterproof jacket. Still, once you get to a certain stage, you just can't get any wetter. 'And he should get rid of the big bands,' the stall holder yelled above the noise of the rain, shifting slightly in his Wellington boots and sinking a little further into the quagmire that had formed behind the nickel candlesticks and sun-moon clocks.

That's the thing about the Glastonbury Festival, it elicits many emotions because it has been such a fixture for so long, becoming part of folklore. We've all heard the stories about Somerset farmer Michael Eavis ('he') and his wife visiting another festival and, thinking that it was a good idea, arranging a small hippie gathering on their land (much to the increasing chagrin of their neighbours). When the Kinks pulled out at short notice, Eavis managed to book the up-and-coming Tyrannosaurus Rex. The small band of attending long-hairs paid £1 for the privilege of seeing Marc Bolan and Micky Finn bash out 'Deborah'—with the bonus of free milk. This perk disappeared long ago, presumably much to the relief of the Eavis herd.

These days there are numerous stages catering for just about every musical taste that may be contained in the 90,000 strong crowd. And then there are the big bands, the names that have taken Glastonbury away from its hippie roots and made it one of the premier festivals in the music calendar. This year Saturday night was the night, both on the main Pyramid Stage and second stage, usually known as the Other One. After the no show in 1970, Ray Davies put in an appearance, along with Chemical Brothers, Neneh Cherry, Ocean Colour Scene, Dodgy, and Cast. Most anticipated of all was Radiohead, riding on the back of their all conquering *OK Computer* album, who headlined in the Pyramid and were broadcast live to 16 countries.

If you couldn't get a ticket, couldn't make it over to the UK, or just didn't fancy the idea of wallowing in mud for a weekend, then the broadcast coverage of Glastonbury was crucial for maintaining a vibe. At one time the Festival only warranted a few slots on radio, while TV managed to hide the event away late at night when programmers thought that nobody was watching. That has all changed: in the UK the BBC started its Radio 1 coverage on the Thursday before, followed by regular visits, some of them live, over the weekend; BBC2 Television had nightly programmes fronted by its music figurehead, ex-Squeeze keyboard player Jools Holland. For overseas coverage, US syndicator Westwood One took feeds so that it could assemble a 4th of July Glastonbury special (giving itself only a 4-day turnaround), while other territories are to take mixes of the acts that they hadn't taken live on manufactured CD.

Deciding to concentrate on only three stages—the Pyramid, the Other One and the Dance Stage—the BBC still faced a big logistical task: covering the three areas, providing feeds to Radio 1 and the overseas radio customers, while also supplying the main audio for the BBC2 coverage. This was arranged as a split between BBC Resources and Radio International, a division of BBC Worldwide.

A Resources Outside Broadcast truck, SCV 5 (equipped with 48-channel SSL E-series with G automation and a small submixer), covered the main stage; the Radio International mobile (48-channel 4000E with Total Recall and Mackie submixer) was on stage two; while the Dance Stage was recorded by a B-type van with a 40-channel Calrec console. As Dave Broomfield, project leader (which

he says meant organising everything from toilets to DA-88s to shovels for the mud) for BBC Resources, explains. the three trucks worked together as a pool. exchanging material for their respective outlets.

Feeds from the trucks came back on a variety of links (cable from SCV5. cable and wireless from the Mobile and wireless from the Dance Stage) to a central control area. housed in a BBC mobile studio (MS3). This housed three Sonic Solutions workstations, which received material from the three trucks, allowing R1's Andy Rogers to work out what was required, assemble it in order and edit out any on-stage profanities. 'The Sonics were for recording, editing and future storage,' explains Broomfield. 'The finished mixes were recorded onto them, applause was faded up and down at the end of tracks as necessary and any swearing was taken out. They're good for editing material into useable chunks, and are convenient tools if any of the music was to be time delayed [that is played out as live after the original recording]. The music would be either played out from the Sonics (or DAT in the case of the Dance Stage) or be relayed live, including an 18½-hour marathon on Saturday, when R1 took just about everything.

'We had the technology to do anything live or record anything,' comments Broomfield. The MS3 area also included three ISDN2 links for transmitting material back to Broadcasting House in London and thence to R1 HQ at Yalding House (one being used as 2-way talkback and an incoming telephone line but on standby as a spare transmission link). 'This was the first year that BT could offer us ISDN,' explains Broomfield, 'although we brought in a V-Sat transmitter as well, which did a lot of the programmes.' Resources used RE660 codecs, which Broomfield says he prefers to the Telos Zephyr because of its automatic functions.

Aside from the ISDN links and satellite flyaway, this enclosure included one of the most sought after items on the whole site —a personal BBC toilet.

Nearby were caravans for Westwood One, Spanish broadcaster RNE Madrid and the R1 production office, which was equipped with its own SADiE unit. The Westwood One van was fitted with a DAT machine, and a small mixing desk to assist in the assembly of material for the 4th of July Special. This was broadcast 'live' to the syndication company's production offices in Los Angeles over ISDN links separate to those used by Resources (again two for transmission and a third as phone link/backup).

'This involved both stages,' explains John Pearson, manager of the Radio International Mobile, 'whatever Westwood One wanted and that we had agreements to broadcast.' ISDN was also used, along with satellite connections, to relay live the Radiohead set. An ISDN line was diverted to the control room at Broadcasting House, thence to Bush House (BBC Worldwide's venerable headquarters) and distributed to the various customers.

Distribution on ISDN or satellite simply depended on customer preference. although the first leg of the link was on ISDN, even if the customers were using satellite. From Bush, the signal went to TV Centre for the uplink to the EBU satellite.

'We looked at all sorts of ways of delivering the music to the US,' comments Pearson. 'The ideal situation would have been to record the material, mix it and then master onto manufactured CD. which would then be sent over to Westwood One. This maintains full 16-bit quality, but the drawback is that it has a long turnaround time. We then thought about DAT, but even that wouldn't have been quick enough. As it's a broadcast application there must be a good master and so we considered that a compressed signal using MPEG Layer 3 processing was sufficiently good enough for the application.'

The application was final transmission on FM radio, with Westwood One mastering the incoming material on CD to distribute it to some 300 affiliates. In this Pearson opted for Telos Zephyr codecs. using 'as little compression as we could get away with.' Despite saying that ISDN isn't yet good enough for real-time transfer of audio data, Pearson adds, 'There is a delay, but it doesn't affect this kind of work, whereas it would be totally unsuitable if it were for overdubbing. ISDN itself isn't the problem, it is the bandwidth that is used on it—it's not enough at the moment. We're also constrained to use lossy compression, but it is good enough for the end use, in
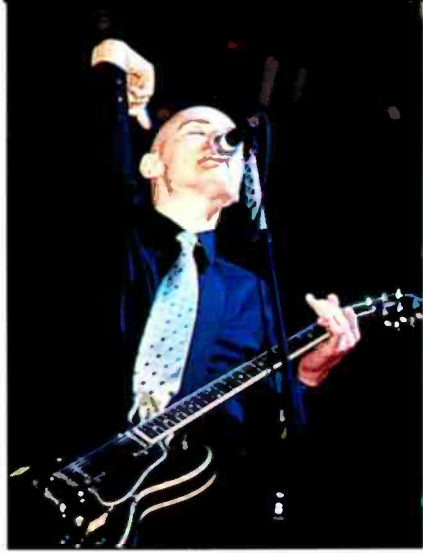
this case FM radio broadcast. It is a way of getting the material to the US quickly and cheaply and cost is the thing, particularly when you think how much it would have cost to send so much material to Westwood One by satellite.'

Radio International's customers, other than Westwood One and those taking the live Radiohead set, take the material on manufactured CD for future broadcast. This gives Radio International enough time to take the multitrack tapes (MTR90s with Dolby SR in the Mobile) back to its remix room at Bush House. equipped with a 48-channel SSL console. This is then mixed down to Sonic Solutions and transferred to the mastering plant on exabyte. For this reason both SCV5 and the Type-B recorded onto multitrack tape (with the Pyramid recordings aug-

AES/EBU?

mented by time-locked DA-88s) for Radio International's purposes, although, in terms of R1 and BBC2, they were relying on live relay, Sonic Solutions or DAT.

Radio International distributes between 40 to 50 CD titles a year and will be releasing sessions from Glastonbury, depending on agreement and approval from the artists involved. The recordings and stage change-overs worked on now time-honoured principles.

'At any one time there was one machine running with the second ready for overlap change overs,' says Pearson, 'there were no hot reel changeovers. In terms of the stage changeovers, we liaise with the crews and organisers. Over the years we've got to know all the live sound companies [this year it was Skan PA]. There's not much time between bands, especially at the beginning of the day— and it seems to get shorter each year.' The A-B system is used, with separate stage boxes and monitor and front-of-house desks for each, flip-flopping between the two.

The Radio International Mobile worked with two crews of three, one on and one off. The trios consisted of the music balancer, an on-stage troubleshooter to check that all the mics and connections worked, and a tape op, who also had to write out lots of labels.

'By having two teams we could have people mixing with relatively fresh ears,' Pearson observes. These crews were made up of five Radio International personnel and a 'guest' from R1. As Pearson points out, dif-

ferent parts of the BBC work in different ways; the Resources truck worked with two main balancers: Miti Adhikari as the primary, working with Mike Robinson. The music mixes assembled in MS3 were linked to the BBC2 TV scanner, where appropriate continuity links were added, plus segments from Jools Holland's caravan.

While the rain and the mud did not affect the weekend's broadcasts in any technical sense, it did disrupt the running order of the second stage on the first day of the Festival. Due to the wrong kind of mud— soupy and squishy as opposed to hard or firm—nine bands were cancelled from the morning to the early evening, but, by some stroke of luck, the broadcasts were not overly disrupted. In the event, Radio International and R1 lost only one band (Sneaker Pimps in both cases), while television lost five.

The other problem was that everything ended up covered in gunge, while the unwanted mire caused its own problems. 'It just meant that you had to find more time to do things,' observes Dave Broomfield, 'and to avoid treading mud in the trucks.' To dry things out, Broomfield waylaid a passing tractor laden with hay and bought bales to spread around the backstage area. 'We used all the Land Rovers to full effect,' he adds. 'It's just the logistics of working in a quagmire; everything worked against the odds.'

The irony is that, amidst all the sogginess of Glastonbury, the only real technical troubles were encountered back at Broadcasting House, where a flood in the control room affected the incoming ISDN codecs, although connections, particularly for the Radiohead broadcast, were not lost. However, asked to name the biggest technical triumph of Glastonbury 97, everybody cites the installation of flushing toilets in the BBC enclosure, complete with toilet paper and pictures on the wall. Evidently, when you're out in the field, your priorities alter dramatically. ■

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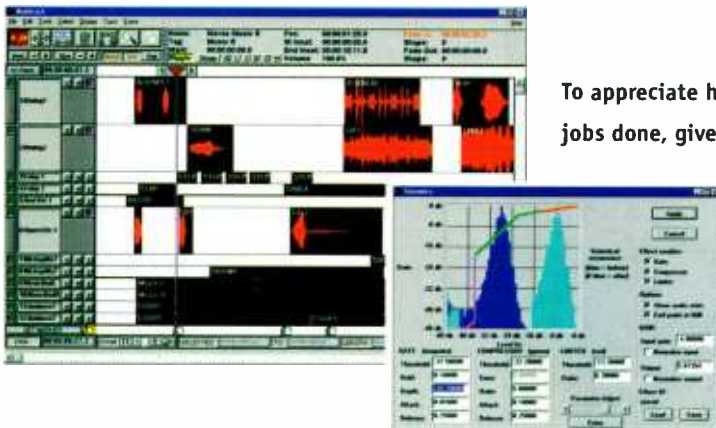
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Goldwyn's Silver Screen

The screening of Oscar nominees requires that the Academy's Samuel Goldwyn Theatre remains state of the art. **Richard Buskin** reports on its latest refit

DAVID GRAY, president of Dolby Labs, firmly believes that there is going to be a major transition within the film industry as cinemas switch from 2-way to 3-way speaker systems. 'After that there will be subtle refinements to 3-way systems,' he opines, 'but I don't perceive that we're likely to be going 4-way in, say, the next five years.'

This may be a prediction worth remembering, as well as one that Gray, as chairman of America's Theater Standards Committee, has good reason to hope for. You see, the basic function of this committee is to supervise the pair of screening theatres at the Academy of Motion Picture Arts and Sciences in Beverly Hills, ensuring that their facilities are state-of-the-art and up to standard. In 1990 this amounted to an extensive remodelling of the larger of the two facilities, the Samuel Goldwyn Theatre. In 1997, it has led to the installation of a 3-way speaker system.

The 100ft x 92ft x 28ft Samuel Goldwyn Theatre accommodates just over 1,000 people and is used to screen every film that is nominated for awards by the Academy, so members can make their minds up on how to vote before the Oscars in March. Throughout the rest of the year, members watch other films there on Sundays, while the 'Studio Series' comprises new releases that can be seen on Saturdays. Studios and production companies also rent the facility for press screenings, premieres, and various industry meetings and workshops that have also take place.

'Late in 1994, about a year before embarking on the 3-way speaker system project, we were in serious need of a subwoofer upgrade,' recalls Gray. 'However, part of the problem for that upgrade was power amplification and clipping. So, what we did as a committee was to test and audition a number of different power amplifiers, and after a total of three days we decided to use a combination of CyberLogic 1200W and 700W power amps for the subwoofers. They're a modular power amp in essence, in that there are 8 channels per chassis, and the modules can be 1200W or 700W depending on the configuration of the chassis itself. We have chassis that are configured for 1200W.

'Therefore, when our attention turned to the screen channels for the 3-way system, from a power amp standpoint it was something of a 'slam-dunk'. We'd already done a lot of testing and therefore knew that the CyberLogics delivered the power that they were supposed to, and that they were clean. Their packaging also meant that they would take up less space. The only real issue was that most of our testing was based between 125Hz and down, and obviously with a 3-way system that would be a different case. On that system we now use 1200s for the low frequency units and the mid-range, and 700s for the high frequencies.

'There are four power amplifiers per screen channel, so there's a 1200 that goes between two of the woofers in the low-frequency cabinet, and another 1200 running the other two woofers in there, as well as the mid-range and the high frequency. The sub-woofer cabinets represent a further 8 channels of power amplification, and those are all 1200W.

'It was a really excellent match-up. They're all JBL components, and the woofer rating, and the amplifier rating, worked out very well. If we were to overload they're basically both going to go at the same time! That hasn't happened yet, and I don't think anybody would still be in the theatre at that point anyway.'

JBL components have made up the main 2-way system since the early 1980s, and JBL had already done a lot of work on the 3-way system at the small Dolby screening room in Los Angeles, and so there was a fair idea as to what would also work in a large room. Nevertheless, what was originally perceived as the starting point did not work.

'The mid-range horn and the high-frequency horn that we started off with have been retained, even though we did make a high-frequency driver change in there,' explains Gray. 'Still, this is a very big room, so we were using two of JBL's standard low-frequency cabinets, and we'd played around quite a bit to try to negate the lobing that that causes in the 2-way system. I'd say that we were somewhat successful in that respect. However, what became very clear as we developed the 3-way system is that we really needed to eliminate lobing. It gives you a hollow type of feel, and it also means that the dispersion in the room radically changes in the low frequency range right at the crossover point. In a big room like this that's very detrimental.'

The end result, according to Gray, "was basically designed on a napkin. JBL computer-modelled it, made it work, and produced one cabinet. We used the right extra channel at the Academy as the sort of testbed, and the difference was fabulous. After that, they just stiffened up the supports for the porting, and that was the way to go.' Forthcoming projects at the Academy include replacing the old BGW power amp setup behind the surround system in the Samuel Goldwyn Theatre, probably with CyberLogic 700s. Then there is the Academy Little Theater, which holds 70 people, and which is often used for documentary screenings, video presentations, and as an overflow from the Goldwyn.

'That room definitely needs an upgrade,' asserts David Gray. 'We have double system capability in there—separate picture and track—but presently we can't run any of the three competing digital film systems. That means that, if you're unlucky enough to be in the overflow, then you've also got the disadvantage of listening in analogue, and I know the studios would like to see that change. So, we hope to replace the current JBL 2-way system in there with a 3-way before too long.' ■
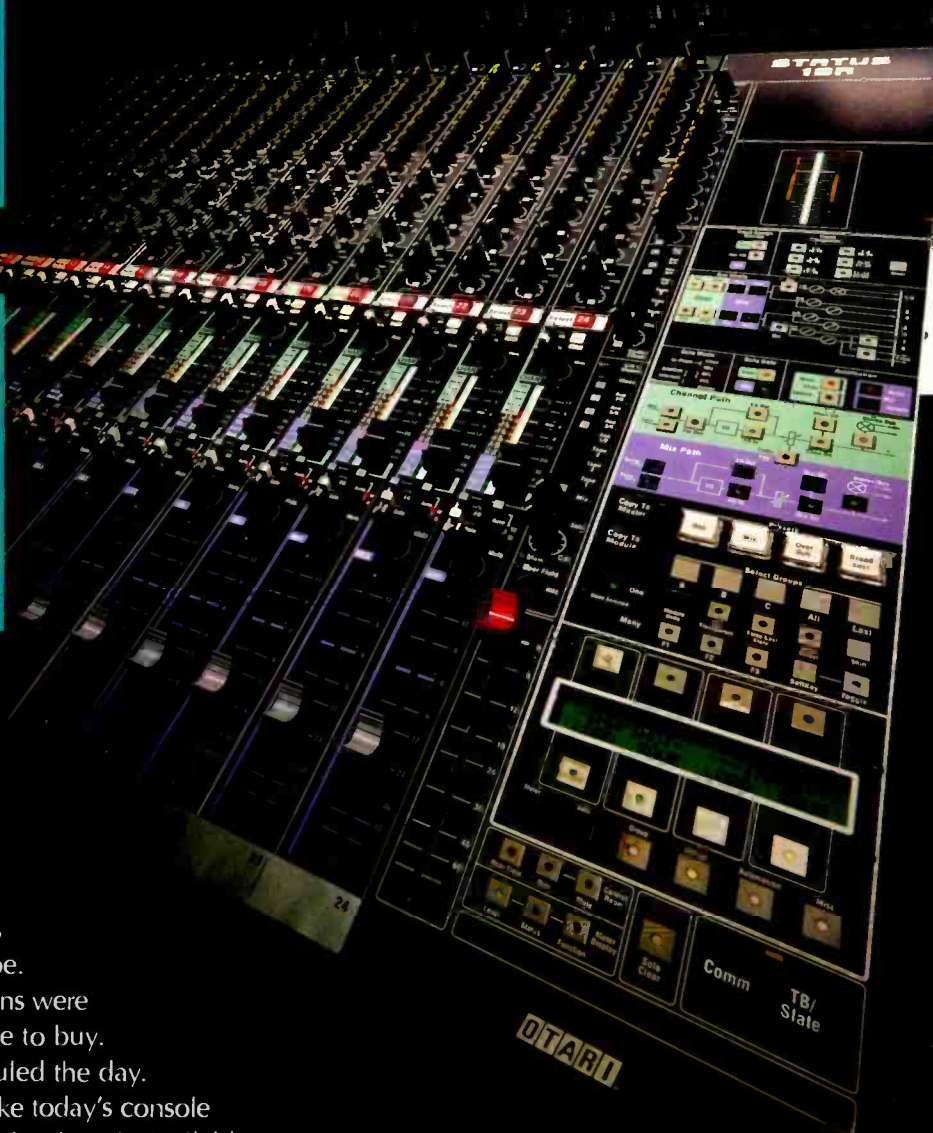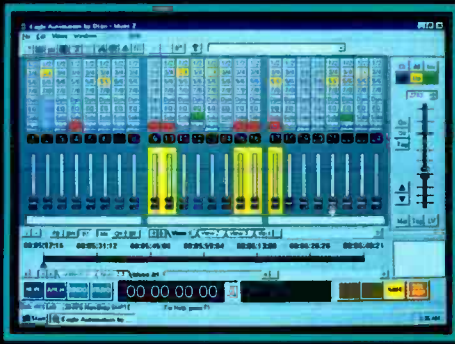
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AND AUDITION A STATUS

The Speed of Sound

If the prospect of 96kHz digital recording caused a stir when it arrived, its position has quickly been challenged. **Tim Goodyer** reports from the first 192kHz recording sessions

WITH THE INK barely dry on reports of the first 96kHz digital recordings, the stakes have been raised again. Now the talk is of a seemingly absurd 192kHz sampling rate—some four times as fast as the current professional recording standard. Can it be achieved? Does it offer any real audio benefit? What will it cost?

The scene was set when Korean electronics giant Samsung approached British convertor specialist dCS with a request for a 192kHz recording system. More than just the system, Samsung charged dCS with responsibility for arranging a series of three 24-bit, 192kHz test recordings. For this, dCS turned to a specialist Dutch recording outfit, Kompas. The reason for all the fevered activity is a presentation intended to convince the Japanese DVD Consortium that 192kHz audio has a legitimate place in the specification for DVD.

The recording schedule involved sessions with a jazz trio in Holland during late June, and a string quartet in Germany on the following day. A choral session in England followed a few days later at the same Queens College Chapel that hosted the pioneering dCS-Meridian 24-bit/96kHz initiative earlier this year (see *Studio Sound*, March 1997).

The systems assembled by dCS for the 192kHz sessions were based around customised versions of the dCS 904 A–D convertor and the Genex Research G8000 magneto-optical recorder. The system was duplicated for security and accompanied by a 24-96 recording chain involving a dCS 902 A–D and Nagra-D recorder running BASF 931 tape, and a 44.1kHz reference laid down via a dCS 900 A–D to a Panasonic SVC3800 DAT recorder. Recording duties fell to Kompas' Bert van der Wolf, acting as engineer, and Ted Diehl, producing. The recording setup was purist and involved four Dutch Sonodore RCM-402 omni microphones and a custom mixer and remote mic-pre setup also built for Kompas by Sonodore.

While dCS' Robert Kelly concedes that the likelihood of a 192kHz sampling option being included in the DVD specification is 'minimal', dCS designer Mike Storey believes that while a limited number of standards best serve computer software, entertainment benefits from as many as possible as long as the domestic player can read the programme information that's on the disc and configure itself accordingly.

'Just because the standard exists doesn't necessarily mean that software and hardware man-

ufacturers have to produce things to work with them,' Kelly asserts. '192k might be regarded as something to be used in n years time.'

'You could take the view that the reason the CD is being seriously challenged now is because it didn't allow for different formats,' observes Storey.

And while Samsung's coordinator, Mr Heo, returned to Korea immediately after the completion of the second recording session, Mike Storey's observations could be all the justification necessary for his efforts.

On the technical side, dCS has moved its efforts from the Nagra-D to Genex' MO machine. Both independently capable of high-bit recordings, the Nagra-D offers two tracks for 24-bit/96kHz recordings while the Genex is able to spread each 192kHz stereo channel across four tracks clocked at 48kHz.

'It uses one AES channel for left odd, one for left even, one for right odd and one for right even,' explains dCS' designer Duncan McLeod. 'Which has the slight advantage that you can pull one out and still get something that sounds pretty decent.' Once in the digital domain, the data management is quietly dismissed by the dCS crew as number crunching. Essentially, after capture by the A–D convertor, the samples are assigned arbitrarily (skewed) to the four tracks serving each of the two channels. Retrieval depends on transferring these samples out of a suitable FIFO interface at sufficient speed to service a D–A convertor running at 192kHz.

'One of the advantages of doing this is that we've had to work out what's had to be improved,' comments Storey. 'As a result, all our analogue stages have a 100kHz bandwidth where previously they were 50kHz–60kHz. As far as the signal-to-noise and so on is concerned, everything's as it was before.'

The key difference in design and performance relates to filtering above Nyquist.

'For 44.1kHz and 48kHz you go for absolutely flat filtering rolling off sharply at the top,' Storey explains, 'but at 96kHz we can roll of more gently. And for 192kHz the filtering is more gentle still.'

If the dCS team had treated the technical aspect of delivering a 192kHz setup as predominantly an exercise in applying the expertise gained with 96kHz (and they had), the listening experience surprised them all—remember that simultaneous 96kHz and 48kHz recordings were available for comparison.

'I listened at the jazz session at Bert van der Wolf's listening room in Holland,' comments Kelly shortly after the last of the sessions had been completed. 'What we noticed was that when things got really busy, and there was a wide dynamic range, the image was absolutely rock solid—there was no tension or indication that it was about to break up, and you could follow all the subtle threads through the music. At 96kHz there was slight evidence of that tension, but we were only able to identify that when we were able to compare it to 192kHz.

'Before we did 192kHz we thought there

would be next to no worthwhile improvement, but we were wrong, and there was a very large improvement. There was a photograph taken when we did the first playback with the musicians and you can see that everybody is amazed—it wasn't subtle at all and that was within the limitations of a less than wonderful playback environment.'

Having been present at the Queens' session, I can endorse these observations, but had noticed an increased sibilance at higher sampling rates in the louder choral passages.

'We concluded that the sibilance in the Queens recording was a product of the room being too small,' counters Kelly. 'When the choir hit a crescendo there was a lot of dissonance caused by strange reflections and the 192kHz setup caught more of that. I don't think it's an artefact of the process itself.

'There's also potential unreleased in 96kHz,' he continues, 'because we found we could improve the sound of our 96kHz hi-fi convertor significantly by tinkering with components. Previously, the limiting factor has been the digital processing, but 96kHz moves that out of the way and you can now hear the limitations of the analogue circuitry which is much easier to sort out than the digital side. So I don't think that people need to worry about having to dash into 192kHz because 96kHz is going to sound pretty damned good.'

For the time being then, the mood would seem to be that the acceptance of a 96kHz recording standard would meet rising recording requirements, but that 192kHz might be a valuable option to keep open—possibly more to ensure the long-term future of DVD than indulge the audiophile fraternity. ■

Above. The first playback—Bert van der Wolf (far left), Ted Diehl (3rd from left) and Mr Heo (seated) with members of *The Toys* jazz band

Polish Radio

Polish broadcasting is upscaling and realigning itself after years of State direction. **Zenon Schoepe** visits Warsaw to report on the broadcasting and music recording scene

THE 1980S WERE an interesting decade for Poland, although it is true to say that most decades this century have been eventful for a country in the unfortunate position of being book-ended by Germany and Russia. However, the 1980s were especially significant for Europe gripped, by an artificial isolation enforced by an East-West divide that is now hard to remember.

As a nation, never the best at self-promotion, the rebellious Solidarity movement focused eyes on the plight of the Poles, and there was briefly a period in which it was very nearly almost fashionable to be of Polish extraction—even the Pope was Polish. Some of the less attractive, but still admirable national characteristics of dogged determination, stubbornness, resilience and a near-genetic revulsion to officialdom shook the Communist tree.

Let's not forget that some were killed as a result, but there can be no doubt that their confrontational defiance of the authorities, and the limited, but symbolic, progress that was attained got the snowball rolling for the monumental changes that took place when The Wall came down. This allowed the world to enter the current decade with a feeling of optimism that had not been experienced this century.

Despite it's tricky locale, Poland had historically always been quite European in outlook, and together with the states that were formerly Czechoslovakia it now presents the most progressive and open face of the former Eastern Bloc. National broadcasters Polish Radio and Polish TV in Warsaw have taken some fairly radical recent steps and are gearing up for an escalation of market sophistication that has already begun. The capital holds the key to all the broadcast and music recording activity in the country.

Polish Radio celebrated 70 years last year and runs five channels including one for overseas. Channel 1 carries music and news, Channel 2 caters for classical music and

drama, Channel 3 generates fast-moving news and varied music output while Channel 4 is mostly dedicated to education. Transmission is selectively in stereo.

Since the opening up of the airwaves private stations have aligned themselves most forcibly with the music and news profile and the national broadcaster now has competition from local and national operators.

Channels 1 and 2 go back to the roots of Polish Radio, the latter having the distinction of being originally wired on a network to villages and played out through loudspeakers in the good old days. Channel 3 was added in 1962 with Channel 4 arriving in the 1970s. Radio is regarded as a viable and expanding medium and is listened to by 20 million people, some half of the population, with Channel 3 accounting for up to 5 million, and Channel 1 drawing up to 8 million.

The hard ties between Polish Radio and Polish TV were separated in 1993 after previously running under one central committee, and now has presidents for the two disciplines. This move was largely political as part of the move away from State to public control.

However, the split means that there are still music recording studios at the Polish TV headquarters that are classed as Polish Radio rooms because this is regarded as the domain of Radio. TV's rooms are dedicated to production.

Polish Radio is also divided into national and another 17 separate divisions each with their own president which represent the main regions. Sitting across this structure is a Radio Information Agency that feeds news across the separate divisions.

Predictably, the biggest change to the broadcaster has been the ability to carry advertising, and in radio this is present on Channels 1 and 3 bolstered by license fees that are apportioned across the whole of TV and Radio.

Money is coming in, and the broadcaster is investing it in equipment and expansion—TV

has added an SSL Aysis OB van, Radio has installed a 9000j-series at Channel 3.

'What we're doing is benefiting from the time saving and convenience aspects of modern technology tools,' explains deputy head of Channel 3 Tadeusz Wisniewski. 'The new console also allows us to support the Channel's tradition of at least one live concert each week which we are hoping to move to CD production.'

The Neil Grant-designed room is head and shoulders above anything else you'll see in Poland and it is apparent that the engineers regarded it as something of a starship, and an equal to anything else that any

other broadcaster has. It works in conjunction with a dedicated live area and a medium-sized stage, and auditorium area that is used for the aforementioned concerts as well as a studio.

As the national broadcaster, Polish Radio sees its role in showing direction for the rest of the industry and to continue as the country's archivist. It alone can afford to indulge in recording of noncommercial material for posterity.

'We hold recordings of the voice of Marshall Pilsudski and (pianist and composer) Paderewski (both were instrumental in re-establishing Poland after the First World War and enjoy national hero status),' adds Wisniewski. 'There are also recordings of many other statesmen, actors and performances. This is seen as a resource and a legacy.'

To visit Polish TV is to see where the national broadcaster is coming from. It also gives a peek into where it intends to go. While the Polish Radio premises in Mysliwiecka Street have an airy and relatively modern feel about them the Polish TV headquarters is typical of the monolithic concrete block approach so characteristic of Eastern Europe. A labyrinth internally, there is a strong feeling that things are being renewed, repaired or upgraded in a manner that cannot disguise the 1960s feel.

There are some 13 studios in total catering from music recording through to drama and the TV's own production facilities.

It's here that you'll find the country's first SSL, a 4000 now fitted with G-series computer, along with an Amek Hendrix, a few Studer desks and a collection of home-grown 1970s Polish-made custom stereo desks dedicated to the large amount of straight-to-stereo recording that still goes on. One of the engineers joked that as many of them had trained on these simple boards and had learnt the ingenuity and resourcefulness required to multitrack with them, the SSL was 'nothing' by comparison.

One of the very real benefits of the building's design is that the live areas are enormous, almost too large for many of the applications and the recording rooms share the unusual feature for the time of shared live room access and a centralised machine room.

The star of Polish TV is a splendid concert hall that hosts live performance broadcasts processed through a Studer desk at the back of the hall in a dedicated studio.

As already mentioned, Warsaw is the hub of audio activity in Poland, and while the broadcaster's music recording rooms can be hired for commercial productions, the capital does have a burgeoning, although admittedly small, music recording community. Top of the tree is Studio Buffo built in to the premises of a former Scouts headquarters and home to the second SSL 4000 ever installed in the country—there are only three in total. Buffo remains Poland's only true top-league independent studio, and it's interesting to note that of the country's three leading music recording rooms, Neil Grant has built two of them.

A large live area with alcoves that offer degrees of isolation for band performances accompanies a large control room that is classic in feel and look. Built four years ago, the studio is the result of the success of the Jerzy Saposa musical *Metro* which continues to run in Poland and tours abroad, and is arguably the most popular crossover piece of theatre in the country's history.

Built using local labour and craftsmen to what Grant describes as 'an extremely competent standard and they followed impeccably what we had drawn', Buffo's in-house engineer Jarek Regulski takes up the story.

'We had looked long and hard to find builders who could build a studio because as you can imagine there aren't that many in Poland. Eventually we found a company outside of Warsaw that was five times cheaper than any firm in the capital, and they were so enthusiastic. However, they quickly realised that there were no right angles anywhere and it was hard for them to imagine what the rooms would look like when they were finished. After the fabrics had been added they were amazed, and some even brought their families here to show them what they did for a living.'

Regulski sees Buffo as the perfect alternative to the 9000 room at Polish Radio, although he says that a lower level of studio is emerging in the city running ADATs and DA-88s. He estimates that there are perhaps 20 studios of an acceptable standard in Warsaw, and less than a handful of old Neve rooms dotted around the provinces.

'As I always try to explain to people, there is no point spending your life in a basement studio if you're going to be selling 500,000 copies and this studio is geared towards that type of act with that kind of money,' he says,

'We've mostly been recording rock 'n' roll here because there is no regular MOR pop market left, that changed fairly rapidly after the end of the Communists. Surprisingly they allowed younger people to listen to rock 'n' roll bands even though they had lyrics that were not exactly proper for the Communists, and that became very popular. A friend and I recorded a band that sold 700,000 copies of an album and that was funny because all the money from it went back to the Communists —they owned the record company and were selling the singles. They were selling something that was aimed against them, but still making money out of it, and a lot of money.'

Since the changes, smaller independent labels have cut into the market with a few acts, and proved to be extremely successful, although the pirate problem in the country, and for that matter the whole of the former Eastern Bloc, also had to contended with ratios as high as 60:40 in favour of the pirate's production. He says the situation has now improved with changes in attitude and action in the Polish Government, although he observes that reducing piracy in your own country is easier than it is to stop the influx of pirated material from other nearby territories like Russia and Bulgaria.

Regulski believes that an increase in recording activity will eventually generate more top flight rooms, but qualifies this by the suggestion that music is not the biggest branch of audio business in Poland, the real growth is in the commercials market.

'You get a computer, some soft- **page 84 >**

Buffo Logic

PRODUCER, MUSICIAN and composer Jerzy Saposa's Metro musical generated the revenue to fund Poland's one and only high-end music studio some years after he had discussed the idea for the first time with engineer Jarek Regulski and dismissed it as impossible.

'Today the situation is completely different because you can rent things, you can invest and live with it, it can work,' says Saposa. 'Our company's structure is such that we have a few people who take care of different aspects of the business. Jarek is the sound man for the studio, the recording and the live sound for the shows. I have a friend who takes care of the choreography and the theatrical aspects while another is responsible for looking for new young talent. It's only because of our travels that we have seen that these things are important and need to be

done here as well.

'The underlying point is that we had to create everything ourselves because the State and the Government hasn't cared about creativity for 40 years,' he continues. 'The State decided what would happen. You're an artist with a qualification from a school or academy but you had to work for the State. Any private independent activity was forbidden—it wasn't impossible, but it was forbidden. My route through the State system was via the student theatre and I was the musical director at the leading theatre in Warsaw for six years. But the system was closed, there were no opportunities for developing larger-scale productions that might go to the rest of Europe. It's one of the reasons why we decided to try something on our own outside of the official structure. It means that our's is the first private theatre in Poland without any support or intervention which runs on pure economic realities.'

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AYSIS OB

POLISH TV'S PURCHASE of one of the world's first SSL Aysis digital desks for installation in a new OB truck came from a requirement for a high density of inputs in a compact space. Digitally-controlled analogue alternatives had been considered but the broadcaster preferred

what it describes as the 'fairly analogue work surface' of the Aysis that did not require enormous amounts of training for its operators. One of the chief engineers in the truck stresses that the greatest difference between this and an analogue board exists at a system level rather than on an operational level.

The 96-input board does not have Disk-

Track but runs with six Sony PCM800s.

Described as 'the first big sound OB trucks' that Polish TV has had, it joins a fleet that includes two smaller sound vans, one big digital component truck, two older picture vehicles and three other small vans. The flagship will be employed predominantly for high profile festival and concert coverage.

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ware, two Beta machines, a marble-topped table, three chairs, a coffee machine, a pretty secretary, and a telephone, and you are in business,' he laughs. 'And you can charge the clients twice what we charge here. We charge less than $50 an hour in the studio which on a 6-hour session is hard to believe for people in the West. And we work nights too.'

Why the commercials market is such a gold mine is apparent when you consider that Poland has been starved of consumer advertising until relatively recently.

'A 9% increase in product sales in the West as a result of advertising would be considered to be really good,' he explains. 'Here you're

talking of increases of 300% because it's a relatively empty market full of open minds. When they first advertised Gillette the response was so enormous that they had to rapidly import extra product because the country literally sold out.'

To this end Buffo opened its own post room at the end of last year with Pro Tools and Regulski's aforementioned list of essentials and the room is averaging two projects a day as well as being used for general editing and mastering work.

Contacts:
Polish Radio, Ulica Mysliwiecka 3/5/7, 00-977, Warszawa, Poland. **Tel:** +482 645 56 68.
Studio Buffo, M. Konopnickiej 6, 00-491 Warszawa, Poland. **Tel:** +482 621 84 52.

Regulski adds that one of the biggest considerations is that building a decent audio post room can be significantly cheaper than achieving the same with a music studio. However, at Buffo they have surpassed these standards.

'I supervised the entire building process and I draw confidence from the fact that I know how the studio was built, and I know that it was built right,' he says. 'The trick is that it looks like it cost five times more to build than it did.' ■

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US: Setting the records straight?

Differing stories from opposite sides of the Atlantic betray fundamental differences in business and philosophy writes **Dan Daley**

RECENT SWINGS through London and New York City produced an interesting and contrasting perspective on the music recording studio business. In London, I thought it was remarkable how the studio owners and managers seemed to have relatively good things to say about record labels lately, whereas in New York, all I encountered was rancour of the most egregious sort.

'The record industry stinks,' one Manhattan studio manager stated. 'The record business went after the post-grunge, lo-fi sound here a few years ago and now they suddenly realised that radio hates it and it doesn't sell records. Record companies weren't developing acts; they were finding independent labels hoping that the guys on the street knew what they were doing.'

That, he said, sent rates into a spiral as studios had to compete with producer-owned studios for clients, many of which were funded from the budgets record companies were giving to those labels and producers to act as surrogate A&R people. I mean, if a label regularly gave out $500,000 to make three records, you'd think a few producers would keep some of that money for equipment and spend a bit less on each project.

That's what happened in New York, and the chickens have come home to roost: several long-time New York recording landmarks have gone into the tank in recent months, including Giant Recording, which had been a fixture there for 13 years. And the studio owners are blaming the record companies' shortsightedness, greed and general, all-purpose ineptitude for the crisis.

If labels do share the blame, then their actions have bitten them in the shorts, as well. US record sales were flat in 1996 and things aren't looking much better this year. Even the folks on the other side of the production line are casting asperitive glances towards the major labels; duplicators and replicators complain that the American record industry hasn't produced a blockbuster album in years, that the creation of former sensations like Pearl Jam were the work of independent labels which the larger record industry paid to act in lieu of actual artist development departments. Record companies got lazy, say the studios in New

York, and it's the studios that have to pay.

In London, the Britpop industry has given the world a bunch of big records in the last year, and it shows in the bottom line: the UK record industry jumped 14% last year. And the studios seem quite happy with their lot. Aside from the usual and globally pervasive grumblings about rates that are endemic to the business at this stage, UK studios seem to regard the labels as allies rather than adversaries.

When asked how this came about, the general response was that, in the void-like wake of techno, the record labels had started acting like, well, record labels once again. Said Carey Taylor of Metropolis Studios, 'It wasn't just the music; it was a very fundamental change in how music was made. During the difficult years, [British] record labels were reluctant to spend a lot of money on recording budgets. That drove a lot of recordings into home studios and also encouraged the production of singles rather than albums, which in turn results in less emphasis on artist development—you wind up with more short-lived solo artists and longer-lived but fewer bands. And that's exactly what happened. But when the bands came back, so did the budgets and so did the sales. And so did the studios.'

Two sets of studio communities and record labels with two very different perspectives on themselves. But a deeper look reveals that the label cultures on opposite sides of the Atlantic are quite different from each other. British labels—granted, some of which are part of large multinationals—tend to focus on making records. US labels, on the other hand, are con-

Home recording makes the news in Europe

Sony is relaunching MiniDisc in Europe, yet again, but this time it has shifted the emphasis onto home recording writes **Barry Fox**

'NEW TECHNOLOGY' read the adverts from major high-street retailer Dixons, 'Now you can record on disc... NEW MiniDiscs... you can record in any order you like—hear it first at Dixons.'

The advert is for a complete hi-fi system, complete with CD player and MiniDisc recorder for under £500. You would have to be blind not to see the clear message—dub favourite tracks from CDs onto MDs. Philips has promised the 'Holy Grail', a sub-£500 recorder that copies music onto either write-once CD-Rs or erasable CD-ReWritables.

The IFPI (International Federation of the Phonographic Industry) complains: 'There have been no discussions between the hardware and software industries. The piracy implications are immense'.

Philips says the new recorder will be 'fully compatible' with the 500 million Audio CD players already in use. This is true only for recordings made on write-once CD-Rs. Recordings made on erasable CDs will play only on next year's CD players, with AGC for the laser to let it read the lower-reflectivity discs. But DVD players already have AGC so

will be able to play CD-RWs as well as pressed CDs, CD-Rs and all types of DVD.

One way or the other, low-cost home disc recording has now arrived. The RIAA warns that it will spawn new 'cottage industry' pirates. A recently issued CD, *Laurie Johnson's London Big Band, Volume 2*, carries an intriguing note on the sleeve. 'To avoid piracy or professional dubbing Horatio Nelson Records and Tapes Ltd have added an identifiable signal to these recordings'.

I bought a copy of the CD (which disappointingly features world-class musicians playing inconsequential TV themes, like 'This is Your Life') to check it out. There is no information on the system used, so I phoned Horatio Nelson Records' Derek Boulton.

Boulton knew nothing about the system used and suggested that it was in some way secret, because if pirates knew what the system was, they could defeat it. Setting aside the issue of whether pirates will really want to pirate the theme from *This is Your Life*, any system that can be defeated once you know what it is called, is hardly likely to serve as much of a deterrent.

Boulton suggested I talk with the BPI, who

suggested I talk to the IFPI. And that's how I heard about Muse, an exciting European research project, jointly funded by the European Union, the major record companies and their trade bodies.

Through Muse (which is not an acronym, but signifies the daughters of Zeus, goddesses of creative arts and performance) the RIAA is paying for the IFPI to test copy-control technology that the RIAA has already promoted as ideal for the job. If the IFPI does not recommend the RIAA's pet, it will be publicly seen to have failed the test.

The IFPI has already developed the International Standard Recording Code, a digital signal that is buried in the subcodes of a CD. The next step is to bury an identifying code in the sound itself.

In September 1996 the IFPI joined with Directorate General III of the European Commission to create the 18-month Muse research project, which is 50% funded by Europe's ESPRIT programme, with the other half coming from Europe's major record companies, BMG, EMI, PolyGram, Sony, Universal and Warner, with Telstar representing the independent producers.

The IFPI's Director of Technology, Paul Jessop, has contacted all the companies round the world already known to be working on embedded signalling. To show due diligence he also placed an advertisement in a science magazine on 7th June, inviting proposals by 26th June from anyone else with a system that is 'available for demonstration now'.

The IFPI is now sending out a music test disc to anyone with a serious proposal. The

stantly throwing away the peanuts from the Crackerjack box in search of the toy prize. Every project is looked at from all sorts of angles, including who owns the publishing to the songs, who's the manager and lawyers (and who else more important do they represent), and its potential to become part of a movie soundtrack, many of which have become chart-topping records in the States in recent years to the point where one new label, Milan Records, does nothing but soundtrack records. Music projects are vetted every which way except usually their artistic merit.

It's not to say that UK labels don't have their share of twits; they do. But the American labels are subject to the American way of doing business, and these days that means playing it safe and dumb.

So the studios of New York are likely to be correct in their indictment of the record industry as contributing to their woes. If there is a silver lining, it's a mottled one. It's inciting a sort of culling of the herd, something other studio communities, such as London, have already experienced. In the long run, the ones that remain will be healthier and it will provide more opportunities for new start-ups. And while it may take some time, American business generally figures things out. Look how long it took Detroit auto makers to realise who really won WW.II in the Pacific. There will once again be blockbusters and decent day rates at studios. But while we're waiting, I suggest you hunker down and pay cash for a while.

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proposer must within six weeks treat the music with their system, embedding a signal and providing the software or hardware needed to retrieve it. TNO, the Netherlands Organisation for Applied Scientific Research, will then try to decode the message after subjecting the music to the same kind of processing which it is likely to encounter in broadcast stations, Internet servers and consumers' homes.

The TNO will also run listening tests in which a panel of 'golden ears' will try to distinguish between the original untreated music and the same music with its embedded signal.

The IFPI is distributing the music as a computer file, rather than a conventional CD, to rule out audible differences caused by CD mastering or pressing anomalies.

The IFPI will only say which systems are competing in its tests if the proponents are making no secret of their interest. The UK's CRL has a notch system that improves on Copycode, as developed by CBS, backed by the IFPI and RIAA, reviled by the audio industry and discredited by US government tests.

The RIAA backs technology developed by US acoustic consultants Bolt, Beranek and Newman, which spreads digital code wide across the sound spectrum, like a thin layer of random noise.

The IFPI's test results should be known by the end of the year. No information will be released on systems which are not selected. But any system not selected will, by definition, have failed the Muse Test.

The secret of great comedy

Many epic films have inadvertently spawned undiscovered comedies in their making—it has fallen to a powerful post job to protect the original brief writes **Kevin Hilton**

MYSTIQUE is a wonderful thing. Some people have it naturally, a mysterious aura that sets them apart from the rest of the herd, but still draws others to them. For the rest of you, well, you have to create it, focusing on a quality that you recognise in yourself and amplify.

While it can be difficult to form a mystique around a personality, it is easier to create one around what people do for a living. There are distinct trades and techniques, but in many instances the difficulty or otherwise of a particular task is enhanced by making people think that it is something that only a select few can carry out. Audio postproduction for television and film is a case in point. Just think about the jargon—always a good way to exclude the uninitiated—that has grown up in this field. Track-laying, dubbing, ADR, Foley—all great buzzwords that mean little or nothing to the great unwashed who think a mixer is used to prepare food.

Which could be one of the reasons why many regard audio postproduction with complete disdain, not to say contempt. I realise that this is a subject I have commented on before but recent events have thrown up how little more vision orientated people think of the discipline.

It all stems from an article I wrote for another magazine, for which the managing director of a leading audio facility commented, 'For most producers, sound dubbing is something that they have to get through, rather than something they need to tell the story.'

This may sound like whinging but it is borne out by a production executive who could only talk about audio post in terms of budgets and the representative of a TV director who said that he was unlikely to comment because 'He's not interested in sound'. It seems all the more bizarre because the director had worked on a production containing action scenes with over 100 audio tracks, including music, dialogue, effects, atmosphere and Foley.

There is a dichotomy between TV and the cinema in this respect, despite the younger medium aspiring to filmic techniques and production values. For as many programmes there are that include sound design, whether straight stereo or surround, there are more that have soundtracks that sound like they were phoned in. I remember a recent episode of a UK soap opera where the quality of an actor's dialogue changed dramatically between the main shot and the cut-away, where there was a distinct alteration in background noise and the equalisation of the voice.

It's all very different from the cinema, which recognises innovative sound design with Oscars and other awards. It has to be recognised that there is an obvious difference in the budgets made available to the two, but when post houses say that they prefer working on commercials to broadcast TV because the budgets and production values are higher, then you know there is something seriously wrong.

AN EXAMPLE of how valuable audio postproduction is was given to me as long ago as 1988, when I attended a lecture in Los Angeles by a leading light at Lucasfilm. It's also one of my favourite stories, one that I have dined out on ever since, so I make no apology for reproducing it here.

To demonstrate the need for audio post, the expert showed two different versions of the opening of *Star Wars*. The first was as we all know and love it: the John Williams score, the sound of giant warships in space, the zap of ray guns and, on top of it all, the stentorian tones of James Earl Jones as the voice of Darth Vader.

The second version was something I had always wondered about. The presence of Vader was played by the huge Dave Prowse, an Englishman from the West-country city of Bristol. I had always thought that the original takes must have been very funny—now I knew for sure. The opening sequence was dull because there was no music, the zapping was replaced by cap-gun bangs but then —O joy!—in came Darth-Dave. He picked up the rebel commander and, in his friendly Bristolian accent, uttered the immortal phrase: 'Where be Princess Leia?'

Just when I thought things could not get any better, the expert returned at the end of the clip with a few explanatory notes. 'You will notice the differences,' he said, 'particularly in the treatment of the voice of Darth Vader. At the sound dubbing stage George Lucas brought in the famous American stage and screen actor James Earl Jones because the actor who played the presence of Darth Vader had a very distinct British "Cockerney" accent.'

It was at this point that I was asked to leave, wiping my eyes as I did.

Despite TV aspiring to filmic techniques and production values, for as many programmes there are that include sound design, whether straight stereo or surround, there are more that have soundtracks that sound like they were phoned in

The resistance movement

Impedance is one of those terms that crops up all the time in audio, but what does it mean? **John Watkinson** looks at principles and practice

IN MANY TECHNOLOGIES, it is useful if something happens. This requires a cause, which might be a force or a voltage and a result, which might be movement or current. Impedance is simply the relationship between the cause and the result.

In simple cases, it is enough to use the concept of resistance. In a flashlight, the battery voltage, and the circuit resistance allow the current and the power delivered to be predicted well enough. In a tape recorder the resistance to tape motion seen by the capstan is pretty stable, and this and the tape speed are enough to calculate the mechanical power deliv-

> A gyroscope is a good example of phase lag in an energy storage system. A helicopter is a gyroscope that can fly. The controls are rigged at 90° to counter the phase lag. To make it go forward the controls actually try to tilt the blades to one side!

ered to the tape by the capstan.

Impedance is a more general form of resistance which is useful in cases where the cause is changing or alternating, and energy storage can affect the result.

In electronics, inductors and capacitors are energy storage devices. In mechanics, moving masses can store kinetic energy and springs can store potential energy. Air can do both because it is elastic and it has mass. All of the above display frequency-dependent impedance.

The difference between resistance and impedance is that impedance allows a phase difference between the cause and the result. It is only possible to calculate the outcome using vectors to take account of the phase. If you

don't like vectors, try shaking a brick. However hard you try, it won't get hot because you are not doing any work on it. Although it resists your shaking (so you get hot) it does so in phase quadrature, just like a perfect inductor. The velocity achieved is always at 90° to the force you apply so there is no net power.

The precession of a gyroscope is a good example of phase lag in an energy storage system. Trying to tip a spinning gyro causes an alternating force on the elemental masses in the flywheel. These respond with a 90° lag so the thing goes at right angles to the direction you try to tip it. A helicopter is a gyroscope that can fly. The controls are rigged at 90° to counter the phase lag. To make it go forward the controls actually try to tilt the blades to one side!

Fig.1 shows a series RC circuit. An ideal capacitor cannot dissipate heat so its current is always at 90° to the voltage. At low frequencies (a) the capacitor is nearly open circuit, but as frequency rises its impedance falls and it shunts away the input signal. Where the impedance of the capacitor is equal to the value of the resistor, the vector diagram is an isosceles triangle (b) and the output phase is 45°. The output will be cos45 times the input which in real money is 3dB down. At very high frequencies, the output phase will go round nearly to 90°.

In audio inputs and outputs it is important that the impedances are appropriate, otherwise the interconnection becomes a filter, a source of noise, or a source of distortion. Fig.2a shows a device with a resistive output impedance of 600Ω. When driving a load of 600Ω, the open circuit voltage is halved at all frequencies so the frequency response is flat and the maximum power is transferred. If the load is altered above or below 600Ω, the power transferred will be less.

Connecting to a higher impedance load (b) will not be a problem provided the next device has a low noise figure. Connecting to a lower impedance load risks distortion as one source has to produce more
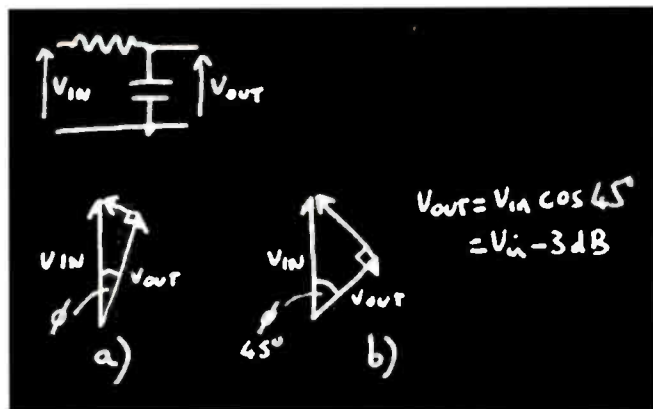

$$V_{out} = V_{in} \cos 45°$$
$$= V_{in} - 3dB$$
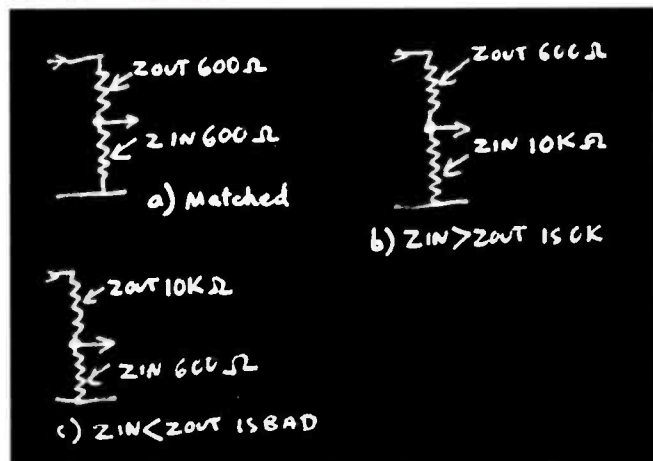
Fig.1: A series RC circuit

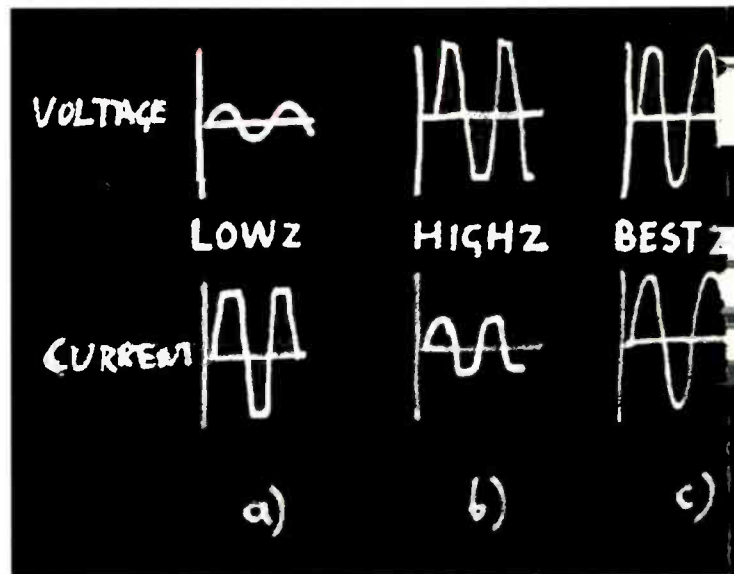**Fig.2: Device with resistive output impedance of 600Ω**

Fig.3: Speaker impedance influences maximum power

current. Consequently trying to connect a professional low-impedance signal source to a high-input-impedance consumer input will probably work, whereas trying to drive a low-impedance professional input with a high-output-impedance consumer device probably won't. As (c) shows, the high source impedance and the low load impedance acts as an attenuator. The problem is worse if the source has a decoupling capacitor. The impedance of this capacitor will be negligible in series with the correct load, but if the load

impedance is reduced, the capacitor becomes a high-pass filter.

When the length of a cable is significant compared to the electrical wavelength, cables can store energy and they have a characteristic impedance. This requires matching at source and load to avoid reflections. Over the distances used in an audio studio, the cable simply isn't a transmission line and you don't have to worry.

In conventional audio power amplifiers, the output impedance is arranged to be very close to zero by the use of negative **> page 96**
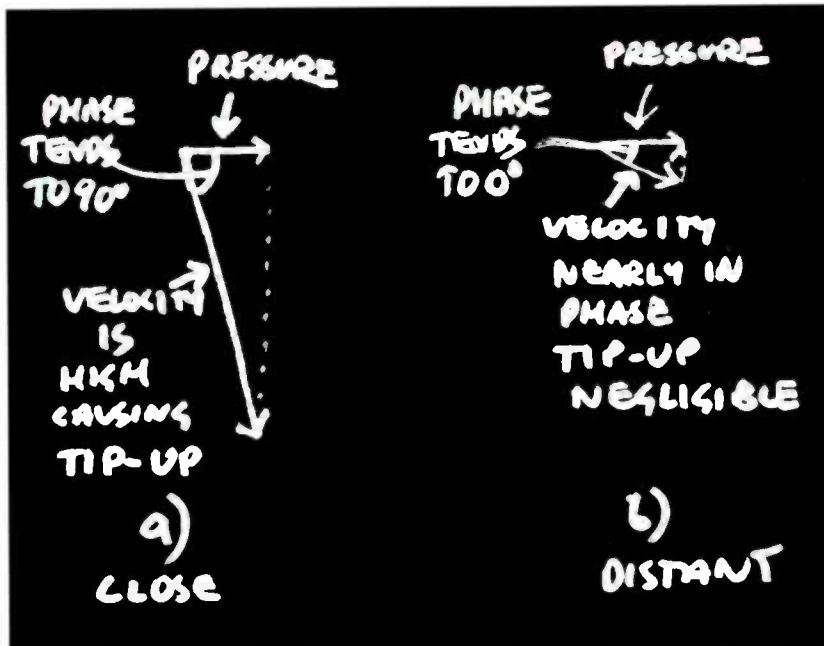
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Fig.4: The proximity effect

feedback. Whatever current is necessary is dumped into the load to make the output voltage equal to the input voltage multiplied by the closed loop gain. In this case matching the load impedance to the source impedance will turn your amplifier into a fuse tester. Loudspeakers are simply not impedance matched to amplifiers and you can connect any loudspeaker to any amplifier and it will work after a fashion.

The speaker impedance will affect the maximum power which can be obtained. Fig.3a shows that if the amplifier sees a very low impedance it will current limit before it voltage clips. Fig.3b shows that with a high-impedance speaker it will voltage clip before it current limits. In both these cases the amplifier is delivering less power than it could. Fig.3c shows the case where the optimum load is being driven so that current and voltage clipping coincide.

Vacuum tube amplifiers have output transformers and it is easy to tap the secondary so that the correct balance of current and voltage drive is obtained for a range of loads. Transistor amplifiers dispense with the transformer so that only one load resistance can be driven properly. Where the supply voltage is limited, as in car radios, the speaker impedance will be reduced, typically to 4Ω.

The situation in Fig.3 is only true if the load is resistive. If the loudspeaker is capacitive or inductive, the current for a given power will be higher so the amplifier will current limit first. In practical loudspeakers the impedance varies dramatically with frequency. With a load impedance between 4Ω and 35Ω and a source impedance of one milliOhm, impedance matching simply doesn't happen. Anyone who tries to tell you a speaker cable is a transmission line is using the word to mean what they want it to mean in the hope of selling you something to solve a problem you don't have.

Clearly, connecting a passive loudspeaker to a general-purpose amplifier is a complete experiment. The speaker manufacturer generally doesn't publish the phase angle or power factor, and the amplifier manufacturer doesn't publish the maximum output voltage or current. No wonder the results are so variable.

As mentioned above, air has an acoustic impedance, and this can have some frequency dependent results. In acoustics, the pressure and the particle velocity are linked by the acoustic impedance z. Just like electrical impedances, the acoustic impedance can be reactive and varies with acoustic conditions. Consequently, the phase relationship between velocity and pressure also varies. When any vibrating body is in contact with the air, a thin layer of air must have the same velocity as the surface of the body. The pressure which results from that velocity depends upon the acoustic impedance.

Consider a hypothetical pulsating sphere. The acoustic impedance changes with radius. If the sphere is small or pulsates very slowly, it will do work against air pressure as it expands and the air pressure will return the work as it contracts. There is negligible radiation because the impedance is reactive and there is a 90° phase shift between the pressure and the velocity. As the frequency or the radius rises, as in Fig.3b, the phase angle reduces from 90° and the velocity reduces. Ultimately, the phase angle approaches zero and the velocity reaches its minimum value compared to the pressure. The impedance has become resistive.

Microphones can measure either the pressure or the velocity component of sound. When z is resistive, the pressure and velocity waveforms in a spherical wave are identical. However it will be clear from Fig.4 that when z is reactive, the velocity exceeds the pressure component. This is the cause of the well-known proximity effect, also know as tip-up, which emphasises low frequencies when velocity sensing microphones are used close to a sound source. Practical microphones often incorporate some form of bass-cut filter to offset the effect. ∎

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Direct File Exchange

After an uncertain start, file exchange between DAWs could be about to take off. **Nick Cook**, director of European operations at Fairlight highlights the initiative and explains some of the issues

ALL USERS WANT a simple, direct sound file and playlist exchange format—read compatibility between systems. Whether this is accomplished by the standardisation of one 'common' exchange format, the direct exchange of information between individual manufacturers or a 'translation' system, the purpose is the same. Everyone has become accustomed to the compatibility offered in film and tape analogue systems, and they want the same compatibility in digital-audio systems.

The benefit of compatibility means users can select products solely based on feature sets, price point capacity and performance. Compatibility is currently accomplished by selecting a common platform on a facility-by-facility basis. Production is, however, a cross-facility activity. This results in transferring materials onto technologies that permit product movement. Unfortunately, these technologies are either 'declining' in the case of analogue multitrack, or 'short-term' in the case of tape-based digital formats. These transfers eat up the potential profits promised by digital technologies. Additionally, 'editorial equity' is lost in the process and the new technologies have replaced progress with frustration.

Until now, most manufacturers have marketed their products on the basis of 'differences'. Although in the past this has been a logical and reasonable market tactic, it has also been short-sighted. Differences are important but not at the expense of compatibility. Each manufacturer wants to hold a unique position in the marketplace. The way to do that is through feature sets, cost point, user-interface, capacity and service—not lack of compatibility.

What manufacturers have failed to see in compatibility is potential. Where the lack of compatibility has forced users to limit their purchase, compatibility will encourage users to commit to the full migration of digital audio technologies in their facilities. The results will be a tremendous expansion of the marketplace for all manufacturers and users alike.

All manufacturers have substantial intellectual and creative abilities that can be better used to create improved platforms and feature sets. This increase in product development, coupled with the user's ability to purchase systems based on performance and features, not compatibility, will significantly increase in the marketplace, and that will work to everyone's advantage.

It would be prudent to examine the situation in a bit more detail. There are four basic problems that must be overcome.

First, you must be able to read the media (hard drive, optical, Exabyte). There are a

The long-term solution recognises that the industry needs a Common Exchange Format designed for audio

number of common formats including Macintosh, MS-DOS and NTFS. Each manufacturer has various capabilities in this regard.

Then playing the sound files as written is a bit more complicated considering the variation of file formats currently in use (PC, AIFF, .WAV, SDII). It's a choice between copying the files to a common format (OMF's solution) or doing the conversion 'on the fly'. Clearly 'on the fly' is faster, but it's also technically demanding. Copying the files is simpler, but it is disc and time intensive.

Next you must be able to understand the instructions about how to play the sound files (playlist). And finally, because of the different capabilities of each system, no matter how this is addressed, maintaining 'editorial integrity' during the translation may be problematical.

The long-term solution recognises that the industry needs a Common Exchange Format designed for audio only that allows each system to easily import and export files. For such a format to become a standard it will need to be achieved with the blessing of the AES, and a working group (AES31) on Audio File Transfer and Exchange, the participants of which reads like a who's who of the audio industry, is already in place. However, to formulate a standard interformat through the AES working group could take a very long time, and it will require cooperation and willingness between manufacturers before this becomes a reality.

The short-term solution is for manufacturers to exchange information about how to play their sound files and the methods to duplicate the performance of their playlists. To this end Fairlight has already entered into 'cooperative' agreements with a number of manufacturers and will enter into similar agreements with any other manufacturer willing to share the information. Work done now on Direct File Exchange will not be wasted, but should be seen as 'creative pioneering' towards the longer term goal of the Common Exchange Format.

Once users have experienced the benefit of sharing files between platforms, and manufacturers have seen the reality of the increased marketplace that this cooperation will bring, more and more pressure will be directed towards achieving the longer-term solution and the process of creating an industry standard will be accelerated.

What can users do? Talk to your preferred supplier, talk to your possible suppliers, talk to your fellow users—encourage everyone to cooperate. Who knows, the short-term solution could be with us for a very long time.

Ultimately, we all share in this problem and we should all share in the solution. ∎

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