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STUDIO SOUND
AND BROADCAST ENGINEERING

THE INTERNATIONAL PRO AUDIO MAGAZINE

Postproduction Paradise
Leading-edge A-V technology at Hollywood Digital

FOUNDATION 2000
Fostex Recorder-Editor-Mixer: an exclusive

Bench Test
Sabine ADF-2400

STUDIO SOUND
AND BROADCAST ENGINEERING

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STUDIO SOUND
AND BROADCAST ENGINEERING

February 1994
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Dynamic action

One of the indisputable areas of improvement in audio recording and reproduction equipment over the years has been in terms of its dynamic range.

Certainly, this improvement is most widely recognised in the performance of digital equipment, but analogue technology has also been consistently turning in better results. Dynamic range: we all wanted it, we have all got it. So what are we doing with it?

If you are involved in the world of classical music, the answer is likely to be that you are revelling in it. After all, the origins of classical music comfortably predate those of audio recording and the music was composed without any consideration for such things as the running time of an LP, the response of a microphone—or the dynamic range of a tape recorder. As such, an improvement in the available dynamic range of audio equipment comes as a welcome improvement in the ability of the technology to best represent the art. Outside of classical circles, however, the answers are less clearly defined and rather less satisfactory.

Dance music, for example, keeps the meters unwaveringly at 0dBFS. Most other forms of popular music fare little better—the dynamic range is nice to have, but other benefits of technological progress (the ability to bounce tracks free from the build up of noise in a digital system, for example) are nicer.

Perhaps it is an opportune time—with popular music obviously in need of refreshment—to remember that greater use of dynamics could be made in both the music itself and production values applied to it.

It is ironic that one of the areas in which improved dynamic range could be exploited to considerable effect is one that currently makes blatant use of compression: broadcast.

Consider two 'givens' of broadcasting: the first is that both radio and television handle a wide variety of programme material which necessarily involves a variety of production considerations, the second is the steady spread of domestic TV installations now using better sound channels than a tiny speaker embedded in a television set—this covers everything from simply feeding a mono signal through a half-decent hi-fi system to ambitious surround sound systems.

Unlike the general consistency usually present in a record release—whatever the programme matter—radio and television stray over everything from the spoken word to the classical symphony and including audio-only drama and fully-fledged feature films.

Obviously the practical considerations of suiting a broadcast signal to in-car systems and televisions still using small internal speakers places considerable restrictions on what can be done. But how much gain riding do we do with our TV sets? Why should it remain the viewers' responsibility. How much more satisfactory it would be to be able to introduce a significant dynamic difference between the music and the bridges music concert broadcasts, whether they be *The Last Night of the Proms* or *Top of the Pops*. And many of opportunities denied to film sound because the best reproduction systems are invariably installed in the highly-reverberant acoustics cinemas would become available if the sound people could rely on the relatively controlled acoustic of domestic environments.

Of course, what is presently impractical may not remain so. The future may see a time when broadcasters can confidently expect the vast majority of television viewers to be using capable audio facilities. And the compression necessary for drive-time listening could be transferred from the transmitting site to the reception site. The only genuinely insurmountable problem I can see is that presented by advertising—who is going to convince the ad agencies that subtlety is a valid policy? ■

Tim Goodyer

Cover: Fostex *Foundation 2000* Photography: Nik Milner

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Bob Clearmountain, Mixer/Producer. Recent projects: Bryan Adams, Bruce Springsteen, Bon Jovi, INXS, Crowded House, The Pretenders, Squeeze, Morrissey.

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If you own Pro Tools version 1.x, and haven't ordered your exceptionally cool 2.0 Upgrade Kit, it's not too late! The cost is just US$49, for residents of the US and Canada, including shipping. Internationally, the cost is just US$69, including express shipping. Pro Tools owners must be registered directly with Digidesign to be eligible to receive the Upgrade Kit. PostView requires some additional third-party hardware and software, for capture and playback of digital video. Spotting to PostView Movies requires a 68040-series Macintosh (Quadra or Centris 650); contact Digidesign for complete requirements. All trademarks and registrations are the property of their respective holders. © 1993 Digidesign. All rights reserved.

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International News

In-brief

● **BBC Wood Norton offer services**
BBC Wood Norton, which for 50 years have formed the backbone of the high technical and operational standards for the BBC, is to make their services available to broadcasters, commercial companies and individuals worldwide.
Tel: 0386 420190

● **Classic FM win Dutch licence**
Classic FM have been awarded one of the first new national commercial radio licences in Holland. Announced by the Netherlands' Ministry of Welfare, Health and Cultural Affairs. The new license will enable Classic FM to broadcast on one of the two important FM transmitter networks in Holland with a potential audience reach of the 80% of the population.
Tel: 071 938 3911

● **SPARS launches LA chapter**
SPARS have established a Los Angeles chapter of the organisation, modelled on the NYC chapter. The first meeting of nearly 100 audio professionals took place at the Mondrian Hotel on 13th January with the Board of Directors. The 1994 SPARS Digital Audio Workstation Conference will take place 21st–22nd May at the Beverley Garland Holiday Inn in North Hollywood.
SPARS Tel: +1 407 641 6648

● **AES Conference and trips**
The AES inform us they have a full set of lectures scheduled for the year covering a wide range of topics. Several technical visits are planned and the UK conference for 1994 will take place in London, 16th–17th May under the title 'Managing the Bit Budget'.
Tel: 0628 663725

● **SSL digital promotional video**
Solid State Logic has released a 20-minute video featuring interviews with owners and users of their digital products. The video , which is shot in England, France, Austria, America and Japan, is available from all SSL offices and distributors. **SSL Tel: 0865 842300**

SSL's video featuring interviews with owners and users of digital products throughout the world

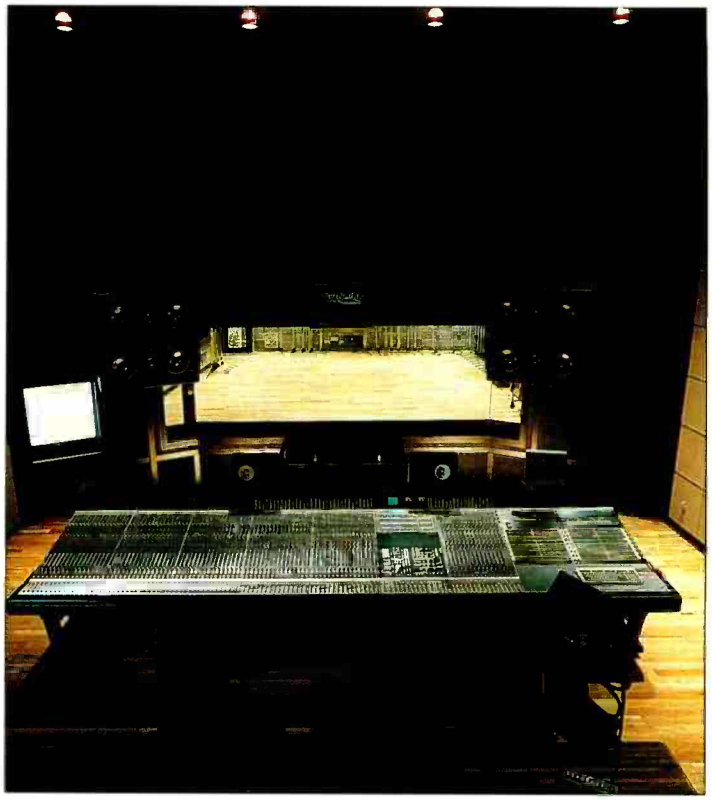
● **Crookwood *Paint Pot* on the phone**
The correct phone number for Crookwood is **0628 528026**

First ever media highway keeps OMF alive at NAB

Industry leaders from a broad range of high technology sectors will participate in the first ever floor-wide network at the upcoming NAB Conference. The OMF Media Highway will demonstrate the use of the Open Media Framework Interchange, an industry endorsed standard for the digital exchange of multimedia information, and a key enabling technology for an all-digital media production environment.

The data exchange will occur over a fibre-optic-based Ethernet network, integrating applications producing graphics, animation, text, video and audio files to simulate a media production environment. A Silicon Graphics *Indigo II* server will act as a focal point for the transfer and storage of interchange files.

Companies involved with the highway at NAB include Avid; Apple; DigiDesign; Studer Editech; The Synclavier Company; Hewlett Packard and TimeLine Vista.
OMF Tel: +1 508 640 3157



The main control room at Consipio Studio, recently opened in Tokyo. Centre stage is the new *Over Quality* console, an *OQM–8196* all discrete 96-channel model with GML automation. Monitors are *M4M(4)* by DynaudioAcoustics. The large tracking room is flanked by two isolation areas and a Bosendorfer piano booth.

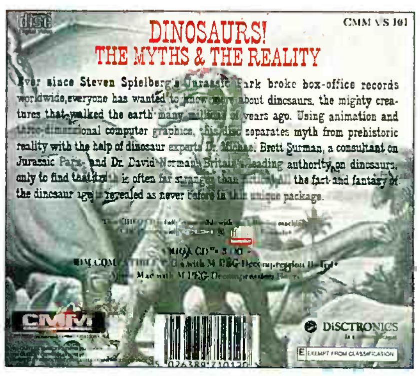
Video CD launches on all formats

At a crowded and noisy launch last month at the Marquee Club in London's West End, various representatives from music publishers, CD manufacturers, multimedia producers, and hardware manufacturers, launched a new format, Video CD.

Video CD promises around 65 minutes of random access video and audio on a 5in disc. A standard has been reached for the format using the MPEG compression system.

Discs will play back on Philips *CD-i* and Amiga *CD*[32] machines fitted with digital-video upgrade cartridges and 386 PCs and Apple *Macs* equipped with CD-ROM XA drives and special MPEG video playback boards.

They will also run on new-generation audio-video CD players scheduled for introduction by other leading consumer electronics manufacturers and the American 3DO games machine.



The Video CD - the myth and the reality

It is expected that there will be 1,000,000 systems capable of playing Video CD by the end of 1994.

On the evening, Castle Multimedia announced a release date of 7th March for their first music and documentary titles. The three titles are the world's first to play on five separate formats.

The titles are *Pavarotti—Nessum Dorma; Dinosaurs—Myths and Reality;* and *The History of Aviation.*

Flashlight PA helps save lives

A potentially fatal incident at an American football game was avoided with help from Turbosound's *Flashlight* PA system.

Thousands of fans surged the field as Wisconsin Badgers defeated the Michigan Wolverines. From 70 rows up, a wave of human energy concluded in a massive pile-up, as fans were pressed up against a fence surrounding the playing area in the packed 78,000-capacity Camp Randall Stadium.

Mike Green, Director of facilities described the scene: 'Once the severity of the crisis became evident, we had to get the ambulances onto the field as quickly as possible. We recently installed a Turbosound *Flashlight* system, and the quality of the system was instrumental in helping us save lives and instruct fans to move to the other end of the field in order to allow access to the injured.'
Turbosound. Tel: 0403 711447

REM sunbathe in front of CYRUS, the Greenpeace solar-powered truck used for recording *NRG*, the world's first album recorded and mixed using only the power of the sun to run the recording equipment. Other artists on the album included U2, The Jesus and Mary Chain, UB40 and Annie Lennox

Sunlight dawns on recording world

Greenpeace International have launched the first album recorded entirely with the power of the sun. The *NRG* album was recorded and mixed using a specially-designed solar generator on a truck called CYRUS (Persian for Sun) and features artists like U2, REM, UB40, Annie Lennox and Midnight Oil, playing live.

The recording team travelled to 14 venues across America, and taped each band's show, using a remote recording truck powered only by the solar power collected, stored and delivered by CYRUS. The tracks were then mixed, either at the venue, or back in Los Angeles, again powered totally by CYRUS.

The generator is a converter 28ft aluminium box trailer, upon which is mounted a 160ft^2, 1,920W solar power array, consisting of 40 1ft x 4ft solar modules. The entire solar array can be hydraulically tilted to track the sun's seasonal variations and capture the optimum sunlight. The power produced by the solar array is stored in two 48V, sealed, long-life batteries totalling 2,500 amp hours or 110,000 watt hours when fully charged.

The album's Co-Executive Producer is ex-lead singer of The Beat and General Public, Dave Wakeling, he commented: 'With the *NRG* album, Greenpeace hope to not only show alternative energy in action, but also to show that the demand for this kind of environmentally-sound alternative is growing by leaps and bounds.'

In 1991 he and Greenpeace coworker Kate Karam commissioned a solar power engineer to design a collector and generator specifically to power recording equipment. The concept came about during the Gulf War when the world's attention turned to energy misuse and the damage it was doing.

Greenpeace. Tel: 071 833 0600.

Yamaha's Sound Works in Wales

The Sound Work's new all digital postproduction suite in Cardiff, Wales, have brought in a Yamaha *DMC1000* to complete the studio.

When Director Simon Jones decided to open a second editing and postproduction suite it was not an initial design consideration that it should encompass the totally digital-audio domain.

'We had an existing analogue suite, but examining the analogue-based alternatives, we quickly concluded that very little had advanced in the last ten years, particularly in terms of automation. With EQs, auxs and effects changing from frame to frame, there has been no automation system running on analogue that can cope with such requirements. The *DMC1000* however can provide completely recallable and instantaneous automation of all parameters.

'Fifty to sixty per cent of our customers are independent producers for S4C, HTV and local BBC. In among those, we have a good number of specialist arts and music programme directors for who sound quality is going to be far more critical a consideration, say for instance with an opera production, and the digital upgrade has pleased then tremendously.

'The *DMC* being digital and software based, it is a future-proofed purchase. We know that it will be in service for many years to come.'

Yamaha Tel: 0908 366700

A Yamaha *DMC1000* digital console looks on under the watchful eyes of Sound Works

Contracts

● ***Delta* becomes Prime Suspect**
Digital Audio Research have sold three SoundStation *Deltas* to the film department of Granada TV. They selected two 8-track *Deltas* and one 16-track, having been impressed by the machine posting *Prime Suspect 3*. **DAR. Tel: 0372 742848**

'Don't quote me on that!'

● **Genesis man endorses Korg**
Apart from a *Wavestation* keyboard and *SR* unit, Genesis' Tony Banks also has a Korg *01/WFD* in his setup.

Banking on Korg: Genesis' Tony Banks

● **The BBC approve of their desk**
Installed in The Music Studio, BBC Television's AMS-Neve *Capricorn* digital mixing console has already 'won its spurs' according to Sound Supervisor, Tony Philpott, (pictured below).

Neve *Capricorn* at BBC Television

● **First SSL console in Argentina**
The installation of the first SSL console in Argentina has been completed recently, with the support of Intervideo Professional, SSL's recently appointed distributor in Buenos Aires. The SSL *4040 G Plus* console with *Total Recall*, has been bought by Panda Studios.

● **Audionics installs Brussels studio** for the BBC World Service to enable journalists to prepare material to file.

Montreux' Palais de Congres also hosts the NAB Radio show

ITS: the future

As might be expected, the announcement made at the end of last year that the International Broadcast Convention (IBC) is to become an annual event, drew a sharp reaction from the organisers of the International Television Symposium at Montreux (ITS).

The organisers confirmed that the next ITS would take place as planned in June 1995 and that support from all areas of the broadcast, cable and satellite industries had been positive. The organisers are throwing down the gauntlet in saying that two large broadcast shows in Europe in a single year are not what the industry wants—or needs.

In a comprehensive press release stating its position, the ITS organisers claim an annual IBC is against the wishes of major manufacturers and furthermore, that it is confusing the issue as IBC is a European event whereas ITS is international—the ITS drew around 32,000 visitors from 105 countries at the 1993 Symposium.

Comparisons of past ITS shows conducted in an unfinished venue are not, claim the organisers, a fair indication of the future: 1993 saw the Palais de Congres finally finished and though still not ideal in terms of layout and navigation, the venue is indisputably better. Special seminars, the increasing popularity of the International Electronic Cinema Festival and the Future Technology Exhibition, combined with a wide conference programme and technical exhibition, mean that the ITS covers a lot of ground—some different in concept to that covered by the IBC.

Parking and transport problems have been alleviated by the provision of out-of-town car parks and shuttle buses. There is also more hotel accommodation than several years ago. Whereas the ITS tariff for exhibition stands has remained static since 1989, what tends to cause the most anxiety is the cost of spending time in Montreux itself—this appears at last to be being countered by the promise of guaranteed hotel prices. However, the fact that Montreux is a resort means that prices will never be low and a show in town will always worsen the situation.

It is encouraging that the ITS organisers are talking more freely with exhibitors, international broadcast organisations and other interested parties. The 1993 Symposium was followed by an extensive mailshot to all participants, who were asked to comment on all aspects of the show and out of the 10% who replied, an average of 85% were fully satisfied with the event. **Terry Nelson**

Exhibitions 1994

● 26th February–1st March, 96th AES Convention, Amsterdam, The Netherlands. ● 9th–13th March, ITA, Tuscon, USA. ● 16th–20th March, Musikmesse, Frankfurt, Germany. ● 20th–24th March, NAB, Las Vegas, USA. ● 12th–14th April, Replitech, Europe, Munich, Germany. ● 7th–11th May, Pro Audio Light & Music, Beijing, China. ● 16th–20th May, AV and Broadcast China, Guangzhou, China. ● 1st–4th June, Broadcast Asia, World Trade Centre, Singapore. ● 7th–9th June, Multimedia '94, Earls Court 2, London, UK. ● 14th–16th June, Replitech International and ITA, Santa Clara, California. ● 22nd–24thJune, APRS, Olympia, London, UK. ● 6th–8th July, Pro Audio Asia, Singapore. ● 24th–26th July, BMF, Olympia 2, London, UK. ● 30th–31st July, NAMM, Nashville, USA. ● 8th–10th September, Leipzig Radio Show, Leipzig Exhibition Centre ● 11th–14th September, PLASA, Earl's Court, London, UK. ● 18th–20th September, IBC '94, Amsterdam, The Netherlands. ● 19th–23rd September, Image World Video Expo, New York, USA. ● 22nd–27th September, Photokina, Cologne. ● 12th–15th October, NAB, Los Angeles, USA. ● 10th–11th November, SBES, Birmingham, UK. ● 10th–13th November, 97th AES Convention, San Francisco, USA. ● 16th–18th November, Tonmeister, Karlsruhe, Germany. ● 16th–18th November Interbee, Makuhari, Japan

Power = MC²

Cofounder and former Technical Director of Klark-Teknik, Terry Clarke, has re-emerged following a two-year research and design programme to launch a new Professional Audio Company to manufacture a new generation of power amplifiers. With codirector, Ian McCarthy, MC² have set up a manufacturing facility based in Exeter in the South-West of England.

Using sophisticated digital management techniques, the new range of power amplifiers incorporate complimentary class AB bipolar outputs driven by a phase compensated floating drive stage. All the dynamic parameters are monitored using a digital control system which intelligently manages load, temperature and all operating parameters and ensures that the power amplifier produces optimum performance under all operating conditions. Specific emphasis has been given to temperature control which will adjust the power amplifiers operating conditions without total shutdown. Limiters have also been included as part of the digital control system which can be adjusted to match the frequency band and dynamics of the load they are required to drive.

The amplifiers are fully configured and wired for external control and will be fully compatible with the various protocols which have been proposed for remote control as standard. The amplifier requires only a small option card to interface its own control system to the host control system.

The power amplifiers, *MC650* (650W per channel) and *MC450* (450W per channel) are to be launched at the Amsterdam AES Exhibition.

MC², Unit 6, Kingsgate, Heathpark, Honiton, Devon, EX14 8YD, UK. Tel: 0404 44633. Fax: 0404 44660

Andromeda 6G

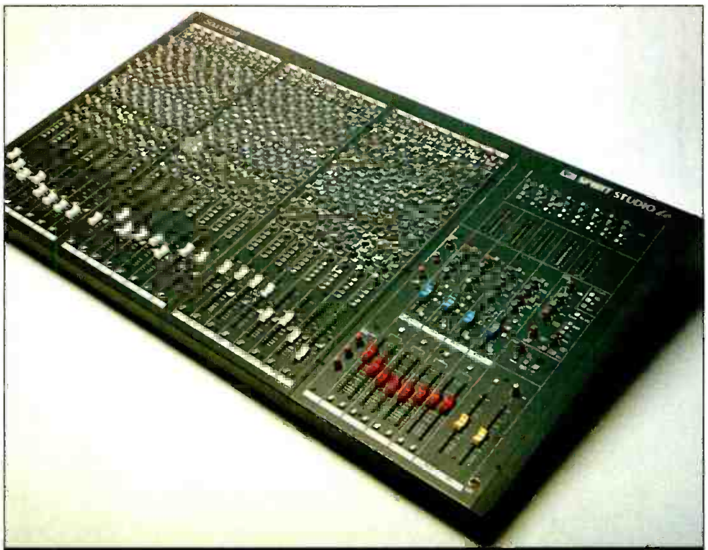
The *Andromeda 6G* digital sound mixing core system has been designed to realise the potential of the 32-bit Floating Point DSP. This core system, which makes use of scalable architecture, offers A–D D–A conversion, digital input and output channels with sample rate conversion, and a processing engine with a revolutionary bus system. The complete system can handle up to 1,024 audio I-O channels in real time.

The most fundamental parameter for musicians and audio engineers in real-time applications is probably the overall delay imposed by A–D/D–A conversion, mixing, filtering, equalising, etc. Conventional digital systems have processing delays of between 10 to 100ms. With a sampling rate of 48kHz, the BitWise approach offers a processing delay of just 42.56 nanosecs. Additional delays due to A–D and D–A conversions come to about 1ms, depending on the conversion technology used. Consequently live performances can be recorded using a single digital mixing console, instead of separate analogue and digital systems.

The system is modular and exists on one or more plug-in core-board modules. At the heart of the sound-mixing board is the *Andromeda 6G* fully customised core board running at more than 5,6 G OPS. Its based on the TMS320C3X/C4X 32-bit floating point DSP from Texas Instruments.

The configuration on view at the AES Convention consists of a prototype console connected via a LAN (Local Area Network) to the mixing core. The eight channels physically present allow the full implementation of a 32 in-line digital mixing desk by multiple channel assignment. The console consists of I-O modules, 128 of which can be coupled to make a single mixing desk, and a master module. The latter is functionally independent of the former, as communications between the two are handled by different software and hardware protocols. Every I-O module accepts a wide range of analogue, digital, mono or stereo inputs, and controls two I-O paths and an assignable path for 'on the fly' redirection of the control to other channels.

BitWise n.v. Digital Design Centre, Van Akenstraat 41, B-1850 Grimbergen, Belgium. Tel: +32 02 270 25 15. Fax: +32 02 270 27 68



***Spirit Studio LC*—breaking new ground at NAMM**

Spirit Studio LC

Soundcraft unveiled their Soundcraft *Spirit Studio LC* console at the NAMM show last month. The desk is available in 16, 24 and 32-channel frame sizes, delivering up to 82 inputs on mixdown. *Studio LC* uses an in-line design for maximum flexibility, with both channel and monitor paths on each input strip, and a Fader Flip facility to allow switching of channel and monitor paths without repatching. Other channel features include 3-band EQ with IN-OUT switch plus swept mid and low bands, a high-pass filter to eliminate rumble and mic popping, eight AUX sends and a 100mm throw level fader.

Both line and XLR mic inputs are provided, with switchable 48V phantom powering. Channel insert points are also provided to allow the use of signal processors, while solo-in-place allows each channel to be monitored with level, pan and effects send settings intact.

Soundcraft, Cranbourne House, Cranbourne Ind. Estate, Potters Bar, Herts EN6 3JN. Tel: 0707 665000. Fax: 0707 660482

RTW DAT remote

The RTW *DC-2* (single machine remote control) and *DC-3* (dual machine remote control) are desktop devices with interfaces for Sony *7000* Series and Otari *DTR-90* machines and *DAE-3000* control line. In addition to a comprehensive set of tape transport controls and a number keypad for locate times and IDs, an illuminated LCD screen displays time code, tape counter or elapsed time. Enhanced features include extended locate facilities, back up and parallel operation and fader start, along ▶

In-brief

● **Audio Kinetics control *F. Faders***
Audio Kinetics have announced a development that allows direct AMS-Neve *Flying Fader* control of its ES.Lock 1.11 system via a new interface using the Adams Smith *2600* serial protocol. Developed in close collaboration with AMS-Neve, this provides users of *Flying Faders* with enhanced flexibility, for a range of machine control applications.
Audio Kinetics. Tel: 081 953 8118

● **Bullfrog introduce speakers**
Bullfrog Inc have recently introduced their new line of trapezoidal speaker systems. Seven loudspeakers and two monitors makeup the line. Designed for small bands, DJ music playback, on-stage monitoring or sound reinforcement applications.
Bullfrog. Tel: +1 219 233 4151

● **Neutrik launch the *NP2RCS***
Key features of the new connector include 2-pole ¼-inch rectangular phone plug; robust make with chuck-type strain relief; and ground termination without soldering.
Neutrik. Tel: +41 75 232 53 93

● **Sound-Link provide ISDN link**
As part of a package with the *DSM100* Digital Audio Transceiver, is the *Pro-Link* ISDN Manager, which synchronises up to three ISDN B-Channels, with a data transmission rate of 384Kb/s.
Sound-Link. Tel:0223 262 765

▼ **dbx extend *Project 1 line***
The dbx *Project 1* processing line now includes the *242* parametric equaliser and the *206* power supply.
dbx. Tel: +1 510 351 0555

● **Crest add to *Century Series***

242 parametric equaliser from dbx

The new Century Series *LM20* monitor console provides up to 20 discrete mono mixes from up to 52 inputs. The *LM 8+4* monitor provides eight stereo and four mono mixes and is ideal for in-ear monitoring
Crest Audio. Tel: +1 201 909 8666.

with regular auto cue, varispeed and chase functions.
UK: HHB Communications, 73–75 Scrubs Lane, London. NW10 6QU. Tel: 081 960 2144. Fax: 081 960 1160.

Trantec *S2000*

Trantec Systems, one of the largest manufacturers of DTI radio microphone systems in Europe, have announced the launch of their latest receiver, the *S2000*.

A quartz-controlled, VHF true-diversity single conversion receiver, the *S2000* features two separate RF sections with fixed telescopic antennas on a durable all-metal enclosure.

As a true-diversity system, the *S2000* constantly monitors the received signal and switches between the outputs of the two internal receivers to find the best one and avoid normal interference. It has an exclusive LED display which indicates power, the RF carrier selected, audio presence and clip.
Trantec Systems, 30 Wates Way, Willow Lane Industrial Estate, Mitcham, Surrey. CR4 4HR. Tel: 081 640 1225. Fax: 081 640 4896.

Three for Valley

Three processors from Valley Audio include the Model *433 Dynamite³*; *730 DynaMap*; and *460 X-Gate/NR*.

The Model *433 Dynamite³* is a dual channel, full function compressor, limiter and expander-gate-ducker, with enhanced stereo linking. Features include selectable key or normal detector inputs, with function-interactive dynamics control using Valley Audio's proprietary Linear Integration Detection circuitry.

The Model *730 DynaMap* offers stereo compression, keyable expansion and gating, look-ahead limiting and sibilance control, and also *DynaMap*, a multiple threshold, multiple segment-ratio true Digital Dynamics Processor available with 500 storage registers.

The Model *460 X-Gate/NR* is a full function, dual channel, sweep-frequency Expander-Gate with integral single-ended Noise Reduction. Features include dual channel, keyable input, selectable frequency-sensitive Expansion, Gating and single-ended Noise Reduction.
Valley Audio Products Inc, 9020 West 51st Terrace, Merriam, Kansas 66203, USA. Tel: +1 913 432 3388. Fax: +1 913 432 9412.

DAT remote from RTW—now distributed in the UK by HHB

AKG release wireless systems for US market

AKG have introduced two wireless microphone systems for the US market and have announced plans for a full line of wireless to be released throughout 1994.

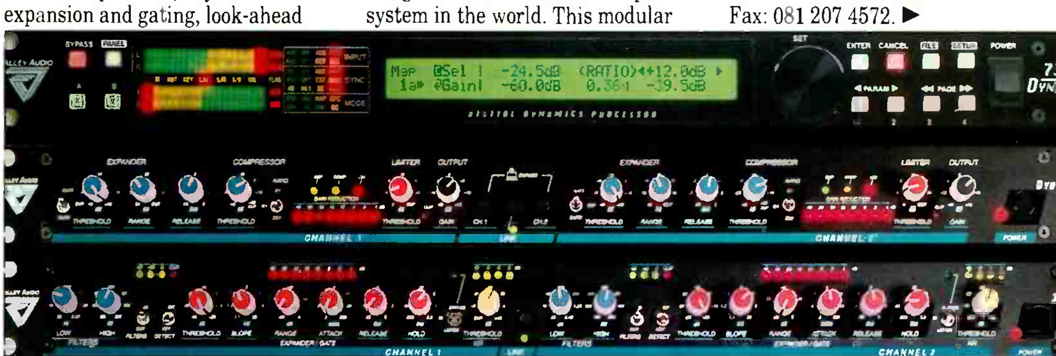
The engineers mandate for the *WMS900* multichannel system was to design the best wireless microphone system in the world. This modular system allows up to 12 microphone channels to be used simultaneously within one UHF television channel. The *WMS 900* system also incorporates RF circuitry that minimises outside interferences.

The *WMS900* has already been used on the Peter Gabriel and Rod Stewart tours.

The *WMS100* is the first in a series of VHF wireless mics that focuses on cost-effective systems. It was designed to meet European wireless communications standards. Dbx noise-reduction circuitry is included for improved dynamic range. The single channel, true-diversity system is available with a choice of two hand-held mic transmitters, as well as a body pack transmitter for mic and instrument use.

Meanwhile AKG's *C451* modular condenser microphone series is being discontinued to make way for the more up-to-date *Blue-Line Series*.

AKG Vienna will hold sufficient stocks of *C451* components to run worldwide spares and service backup for at least five years.
AKG Acoustics, 1525 Alvarado Street, San Leandro, CA 94577, USA. Tel: +1 510 351 3500. Fax: +1 510 351 0500.
UK: Harman Audio, Unit Two, Borehamwood Ind. Park, Rowley Lane, Borehamwood, Herts. WD6 5PZ. Tel: 081 207 5050. Fax: 081 207 4572. ▶



Three from Valley Audio—*730 DynaMap;* Model *433 Dynamite³*; and *460 X-Gate/NR*

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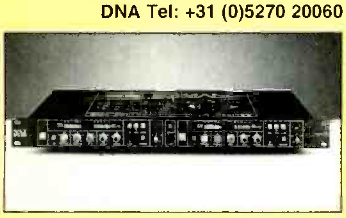
In-brief

● **Son of Francinstien is launched**
Perfect Pitch Music, the makers of the *Francinstien* stereo enhancer, have announced a new range of 3-D sound processors called *OM*. An *OM* encoded recording will create a three dimensional soundfield which totally surrounds the listening position.
Baccus Professional.
Tel: 0234 840 408

● **Audio*Spector* for the Atari**
German software house, Steinberg have launched Audio*Spector* real-time spectrum analyser software for the Atari *Falcon* computer.
Harman Audio. Tel: 081 207 5050

● **A poke in the ear from OSC**
OSC have released two more CD-ROM sound libraries. *A Poke in the ear with a sharp stick*, volume III has over 1,800 sounds, effects, loops and clip-tunes from five different sound designers. *Textural Environments* contains long, evolving atmospheres, soundtracks and beds.
OSC. Tel: +1 415 252 0460

● **DNA shows the *DICTATOR***
Dutch company DNA are introducing the *Dictator* at the AES '94. The *Dictator* is a peak-limiter for mastering and STL purposes. It incorporates a clipper, pre and de-emphasis, and separate HF-limiting.
DNA Tel: +31 (0)5270 20060



DNA's first product the *DYMAND* compressor

Clyde *Presenter 2*

Presenter 2 is the latest offering from Glasgow-based broadcast console manufacturer Clyde Electronics. The design is aimed at the smaller radio station for use in on-air and production work.

Up to 19 universal input modules can be user-reconfigured in seconds to accept nearly any mono or stereo source. The universal concept extends to the equalisation and production modules, which can be connected to any input module, be it configured for mono or stereo use.

Other features of the *Presenter 2 console* include fast acting peak limiters on the main output, distribution amplifiers, and intelligent bidirectional remote interfacing.

Clyde Electronics, 2 Rutherford Court, 15 North Avenue, Clydebank Business Park, Clydebank. G81 2QP.
Tel: 041 952 7950.
Fax: 041 941 1224.

Clyde *Presenter 2* desk

Fairlight *MFX3*

Fairlight's *MFX3* aims to provide a truly 'object based' architecture that allows any audio clip or group of clips to be edited or moved instantly in a 24-track project.

An array of 40-bit floating-point DSPs are used as the basis of the design which also includes digital routing of 24 analogue and digital inputs and outputs, clip-based EQ and comprehensive crossfade control.

A new synch machine-control architecture with optional digital video record and playback is also provided.

The *MFX3* can play back 24 audio tracks from a single hard-disk or 12 from one optical disc. The user-interface is based on Fairlight's *MFX* editing console and scrolling waveform graphics for all tracks.

A new generation tape cartridge subsystem provides the most reliable and fastest backup and restore function available, making *MFX3* particularly suited for multitrack music and general audio productions, while also providing new features for audio post.

Fairlight ESP, PO Box 942, Brookvale NSW 2100, Australia.
Tel: +61 (2) 975 1230.
Fax: +61 (2) 975 1368.
Europe: Tel: 0764 849090.

Nexo *PS15*—part of the *PS* range

Nexo's *PS15*

Nexo are launching the latest addition to their *PS* range of speakers systems at the Frankfurt Musik Messe and AES Convention in Amsterdam. The *PS15* offers the same versatility as the *PS10*, but with greater power handling.

The *PS15*'s unique shape makes it suited to wedge monitor and freestanding or flown applications, where it can also be arrayed. It comprises a 15-inch driver and asymmetrical dispersion horn, and features stand fitting and optional flying rails.

The system uses the new *PS15TD* controller, using the circuitry employed throughout NEXO's processor range. The controller is switchable between PA and monitor mode, providing optimum response for each of these applications.

NEXO, 154 Allee Des Erables, Zac De Paris Nord II, BP 50107, F-95950 Roissy, CDG CEDEX, France. Tel: +33 1 48631914.
Fax: +33 1 48632461.
UK: Network Ltd. Tel: 081 885 5858.

Digital Audio Times

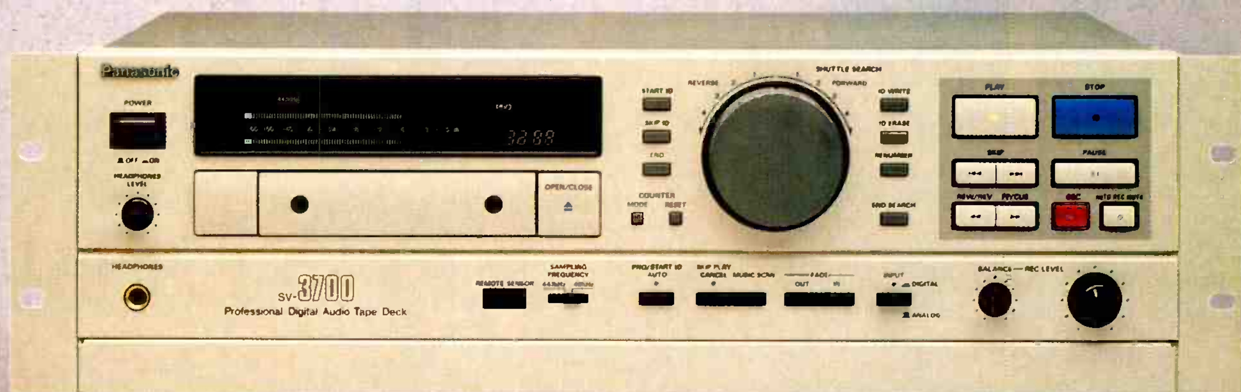
FOR THE LATEST IN DIGITAL AUDIO INFORMATION

WORLDWIDE DAT RECORDER SHORTAGE-BUT NOT AT HHB

When Panasonic UK decided to stop importing the excellent SV3700 Professional DAT Recorder into the UK, the world's leading independent supplier of DAT technology - HHB Communications - went direct to the manufacturers to secure ongoing supplies. As a result, we're pleased to announce our appointment as a key UK and European distributor of the DAT recorder that's already a firm favourite with US audio professionals - the Panasonic SV3700.

HHB also remains tireless in its search for the type of suitably featured consumer machines that offer exceptional value to professional users. Our latest discovery is Aiwa's superb XDS1100 - see below for full details.

HHB SECURES UK DISTRIBUTION OF PANASONIC SV3700 DAT RECORDER



The Panasonic SV3700 is a fully professional DAT recorder featuring high performance 1 bit A-D converters, switchable 44.1 / 48 kHz sampling rates, AES/EBU digital I/O, a shuttle wheel and infra-red remote control. Remember, this superb machine is exclusive to HHB and our authorised dealers. Call today for details of your nearest stockist.

BEWARE-ALL DAT TAPES ARE NOT THE SAME

Although priced competitively with conventional brands, HHB DAT Tape exhibits consistently lower block error rates. What's more, it's available in 6 lengths (15, 30, 48, 62, 92 and 122 mins). Call our mail order hotline on 081 960 2144 for details of prices and quantity discounts. All major credit cards accepted - 24 hour delivery.

AIWA XD-S1100: ANOTHER HHB EXCLUSIVE

With a highly reinforced double construction chassis, 3 motor drive mechanism, coaxial and optical digital interfaces and a wireless remote, Aiwa's superb XD-S1100 is perfectly configured for private studio use. And like everything we supply, it's fully backed by one of the most experienced and respected technical departments in the business.

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HHB Communications Limited
73-75 Scrubs Lane, London, NW10 6QU
Tel: 081 960 2144 Fax: 081 960 1160 Telex: 923393

CEDAR *DC1* DECLICKER

In *Studio Sound*'s October 1993 issue we looked at CEDAR's *CR1 DeCrackler*, the company's latest software-module-in-a-box, and made passing reference to the *DC1 DeClicker*. Since the *DC1* was the first model of this type, predating the *CR1* by some time, and since it represents more closely the image most people have of what CEDAR is all about (however wide of the mark that might be), it warrants more detailed inspection.

To recap briefly: CEDAR recognised some time ago that their complete audio-restoration system was out of the financial reach of most nonspecialist facilities, but elements of what it does would be particularly useful in areas of the business. They therefore decided to produce stand-alone units duplicating the various constituent software modules of the whole system—the first was the *DC1*.

The software in the *DC1*, as the name suggests, specifically targets clicks as opposed to other types of background noise—crackles, hiss, buzzes and so on. CEDAR themselves loosely subdivide clicks into three categories according to their size and nature, which they refer to as ticks, clicks and scratches. The difference is largely defined by the nature of the treatment required to remove them, and translates into three algorithms in the software.

Physically the *DC1* is virtually identical to its more recent stablemate. Operation revolves around a large blue LCD with five soft keys and a large data wheel, and is surprisingly uncomplicated—there is even less to adjust than on the *CR1*. Signal can be presented to the unit in analogue or digital form, balanced or unbalanced, SPDIF or AES-EBU, and all output formats are available simultaneously. Analogue use will require the use of the large, bright input and output meters in conjunction with a single input-level control, and even with digital signals these should be watched as the internal 32-bit processing could conceivably produce output levels exceeding the input levels. This eventuality is coped with by a digital output attenuator.

Making the *DC1* do its job really could not be simpler, but an understanding of the difference between the types of processing algorithm is of help. The algorithms are simply referred to as Small, Medium and Large, but since increasing the size of the signal disturbance alters the nature of the problem and the methods required to solve it, they cannot be considered as increasing degrees of the same thing; they are three quite distinct processes.

The Small setting assumes the presence of a very short click (CEDAR's 'tick') which would appear pretty much as a single vertical line superimposed on the waveform of the original signal. All the computer has to do is remove that vertical line and join up the ends of what is left.

The Medium setting assumes a longer disturbance, clearly spread across several cycles of the waveform. Simply joining up the ends in this case would obviously result in distortion of the original signal, so the software uses AI and modelling techniques to calculate what ought to have been there. Since it can perform 50 million calculations per second, it claims to be capable of dealing with 5,000 such clicks across the two channels every second.

The Large setting recognises that certain types of click may have aftereffects lasting considerably longer than the initial offending noise. The most obvious example is a bad scratch on a record which can induce resonances in the pickup arm; these then show up as low frequencies superimposed on the signal. Simply removing the click (as in the Medium setting) can leave the low-frequency 'tail' intact, resulting in a popping effect when this appears after the corrected gap. The Large setting therefore attempts to deal with any such aftereffects in addition to the basic declicking function.

In practice, of course, the most appropriate setting can quickly be found by trial and error, since it is so simple to switch between them; it may be worth pointing out that the system operates in real time, in case there is anyone who still thinks it is a drawn-out process. Once the algorithm has been decided upon, the only remaining adjustment is the Threshold, which dictates how hard the whole process works.

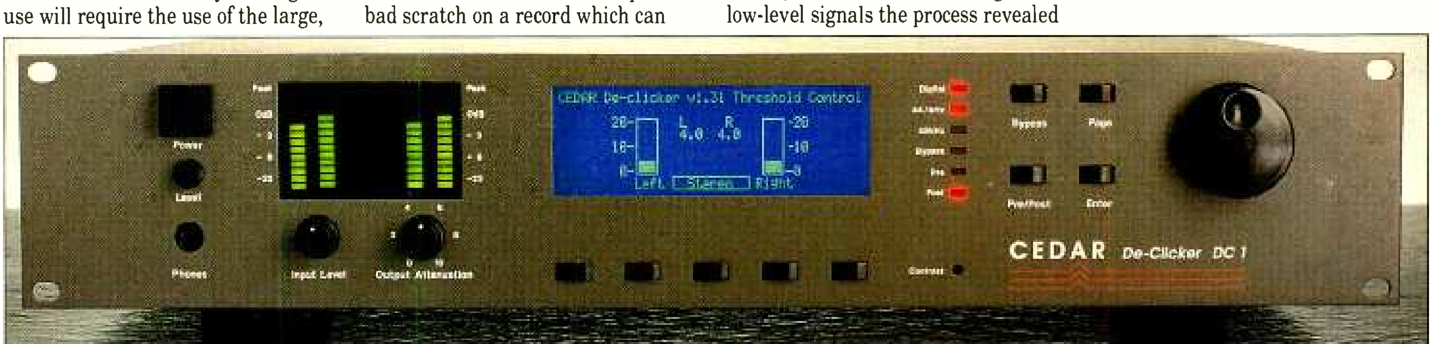
I used the same DAT tape full of 78 transcriptions as for the *CR1* review, safe in the knowledge that this time it would be more suitable as test material than it had been then. The results, with the minimum of fiddling about, were as dramatic as could be hoped. It is quite uncanny to hear how good the recording on an old 78 really can be under all that surface damage, and how low the real noise-floor of the medium is. Setting the unit up is so straightforward that it is quite difficult to introduce offending side effects, and very simple to eliminate them altogether leaving nothing more than the wanted signal and any steady-state noise, which, obviously, this process does not touch. It is possible to get the feeling that some treble is missing, but careful listening always reveals that the main HF component in the unprocessed signal is the surface damage, and that the underlying recording retains any extreme top end it may have had to start with. Nothing I tried—from operatic arias to trad jazz—showed any signs of side effects, and far from disturbing low-level signals the process revealed more ambience and detail than one might credit medium with carrying.

My choice of test material, however, makes me guilty of the same assumptions about CEDAR as most of the rest of the industry. True, their early associations were with archives of early recordings which required restoration, but that kind of work now constitutes a small part —less than 10% by CEDAR's reckoning—of the use to which their products are put. The process is just as happy dealing with LP records, film soundtracks—optical and magnetic—and digital clicks as it is with 78s and cylinders. Current users of the *DC1* comprise mostly of mastering facilities—not just the specialist restoration ones—and broadcast studios. Many radio stations are using them to archive their vinyl libraries to a digital medium such as CD-R, and some even use them live on-air to safeguard against rogue clicks offending their listeners.

There is no doubt that the *DC1* delivers the goods, exactly as promised, in the most effective and user-friendly way possible. Some might say that at that price it had better work—although it is vastly cheaper than the complete CEDAR system it is still strictly for the serious user. Enough of such serious users have already stumped up for one, however, to demonstrate that if you regularly encounter the kind of problems it sets out to solve then the *DC1* is an indispensable tool. ■

Dave Foister

UK: HHB Communications Ltd, 73–75 Scrubbs Lane, London NW10 6QU. Tel: 081 960 2144. Fax: 081 960 1160.
US: Independent Audio, 295 Forest Avenue, Suite 121, Portland, Maine 04101-2000. Tel: +1 207 773 2424. Fax: +1 207 773 2422.



Cost-effective click removal derived from the CEDAR system is the heart of the *DC1*

NEWS FROM TUBE-TECH

LCA 2B

SEE US AT THE AES EXHIBITION IN AMSTERDAM

The new TUBE-TECH LCA 2B stereo compressor/limiter is based on the very successful LCA 2A (released April-93). The LCA 2B features ouput level control and limiter on/off switch.

BELGIUM (08) 941 5278, **DENMARK** (43) 99 88 77, **FINLAND** (90) 592 055, **FRANCE** 87 77 00 00, **GERMANY** (089) 609 4947, **HOLLAND** (02) 0613 1521, **JAPAN** (03) 5489 3281, **KOREA** (02) 741 7386, **NORWAY** (55) 951 975, **SINGAPORE** 748 9333, **SWEDEN** (046) 32 03 70, **SWITZERLAND** (01) 840 0144, **TAIWAN** (886) 2719 2388, **UK** (069) 1658550, **USA** (212) 586 5989.

LYDKRAFT

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The *Audio Gallery*—16-channel multitimbrality and a generous 32-note polyphony

KORG *AUDIO GALLERY*

The quiet revolution of beige-coloured, blank-faced sound modules intended for use with computer sequencers cannot have been missed by anybody even remotely interested in synths.

Korg's *Audio Gallery* is important because it is the company's first offering to this sector and because it represents an attractive starter pack for music departments and beginners as well as acting as a space-efficient and, in places, a very nice-sounding affordable expander module.

The package is strong. The *Audio Gallery* comes in two different versions for IBM or *Mac* computers which are connected via a serial port or modem-printer port cable respectively. Consequently, the unit performs the function of a MIDI interface for these computers with a sound module tagged on. Software thrown into the IBM bundle includes the deceptively able Passport *Trax* sequencer with fairly extensive editing, notation and external sync. There is also a Passport *MIDI Player* program which permits MIDI song files to be played back and edited with controller data using an on-screen mixing console. Finally, there is a MIDI driver, MIDI song file convertor and some tunes to get your feet tapping.

The *Audio Gallery*'s 1U-high half-rack hardware is impressively specified with 16-channel multitimbrality and a generous 32-note polyphony in Single mode. It uses the AI^2 synthesis from the *01/W Series* of workstations to generate 128 GM (General MIDI) sounds and —thankfully—many of the Double mode sounds are not the sort you would want to use on a regular basis (they include such useful timbres as Seashore, Birds, Telephone, Helicopter and Stadium)—sadly, things like this make GM a joke and waste so much potential.

The front panel houses a power switch, power and MIDI indicators and a volume control while the rear has MIDI In, Thru and Out, a socket for the computer cable, a headphone socket, stereo phono outputs and twin phono inputs which allow other sources to be cascaded into the *Audio Gallery* for mixer-less listening.

Other controller data that can be employed in addition to the GM spec include attack, release, brightness and VDF modulation plus chorus and reverb depth so it is actually quite a flexible unit. Sound-wise it is surprising, and Korg's interpretation of GM noises is impressive. There are some very good pianos, some very usable organs, excellent bass sounds and some good washes. The film soundtrack tones are well represented—a lot of them are Double mode and beg the question of how long before they start to sound really dated. There is an argument that says they are clichéd already and you cannot say that about a piano.

People talk about the inevitability of effects processors being hosted in computers but before that happens you can expect to see this same sort of integration.

We must now be within spitting distance of intricate blank boxes, and not just simple playback modules, being widely available and intended to be run exclusively from computers.

Whichever way you look at the *Audio Gallery* it is a damn good deal. Combining a computer MIDI interface, a sound module, a sequencer with notation plus a handful of other useful things makes it a one-stop shop for the beginner with a mute *Mac* or IBM. It is only slightly less attractive as a stand-alone tone module but it has 32-note polyphony and some very workable sounds. The *Audio Gallery* is certainly worth a listen.

Korg UK, 8-9 The Crystal Centre, Elmgrove Road, Harrow, Middx HA1 2YR. Tel: 081 427 5377.

ART *SGX2000 EXPRESS*

I initially suspected that the drift to 'Express' status for the *SGX2000* guitar preamp was little more than a face lift—in fact it is the complete opposite. Externally we have still got those same wacky graphics (not for the myopic) legending and that luke warm LCD, but internally ART have been fiddling around with the vital internal organs.

It is still that the rather good 12AX7-driven guitar preamp with tone knobs on the front and digital multieffects inside but it now sports the lungs of a giant.

Factory presets numbering 475 reside in four banks tagged Classic, Performance, Production and Contribution. The first two are dedicated to guitar effects, the third uses the unit's all-round effects processing abilities and stereo line level input capability to generate recording-type effects which include a selection for guitar, while the fourth concerns itself with slightly more extreme guitar patches. These can be arranged in 200 user slots where your own creations can also be stored.

Assuming a certain amount knowledge of the original *SGX2000* enhancements on the *Express* include warmer reverb algorithms, a dual pitch transposer, an increased regenerated delay time (from 1150ms to 1300ms), a good old-fashioned phaser, and a thicker chorus. The grandly-titled Acoustic Environment Simulator is supposed to have been tweaked but still does not convince and the LED pseudorunes on the original unit's autotuner have been replaced by a vastly improved and understandable version.

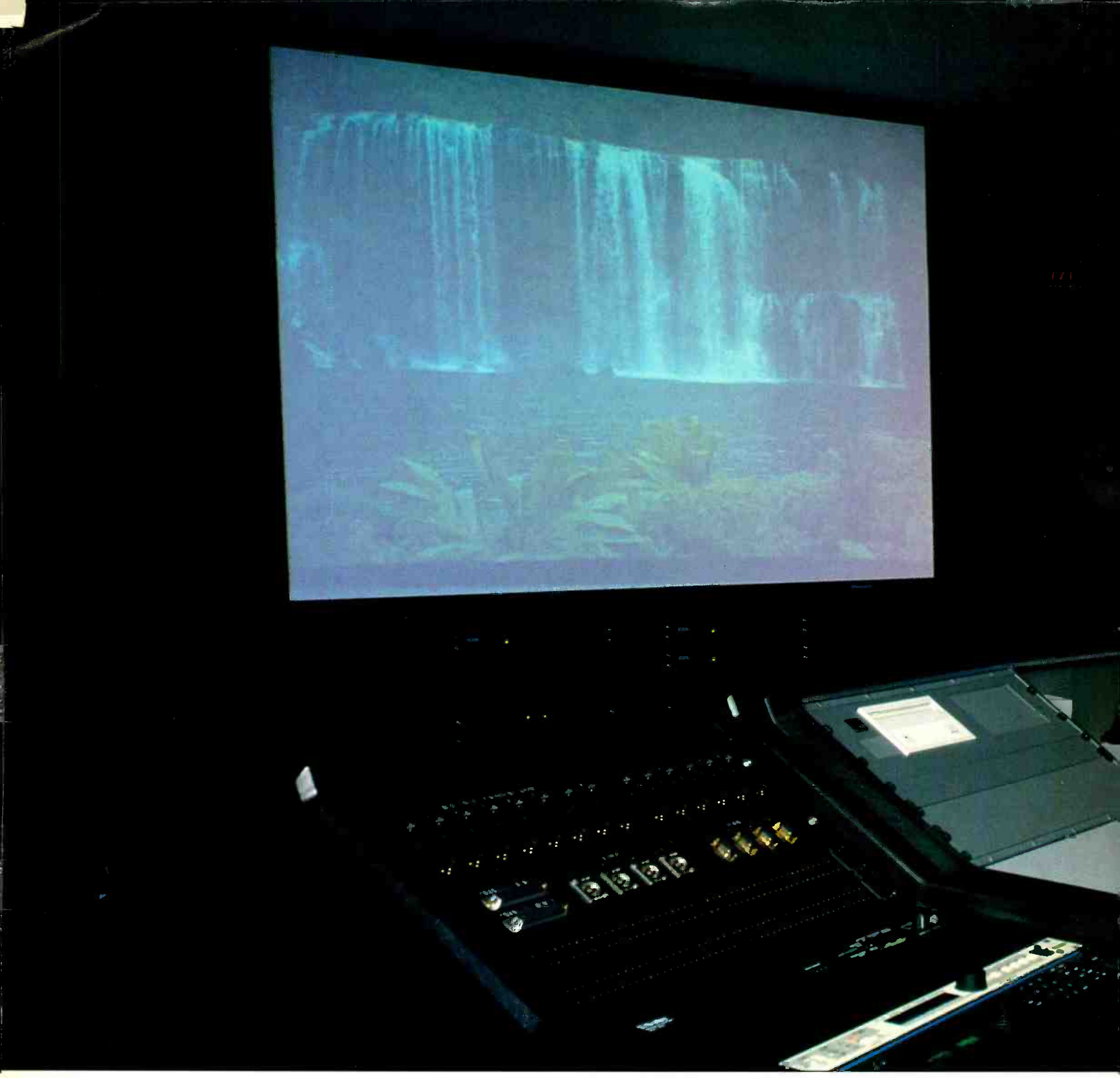
It is still a great sounding unit with a subtlety less heavy complexion that is still able to rip the pile off the carpet when required. There are some fabulously produced presets in the Classic and Performance banks which fill in one of the blind spots of the altogether earthier sounding original *SGX2000*. As a result it is a more complete device. Even the production bank has its use simply because it presents every single effect type in isolation for individual auditioning and use. The old unit was a capable of this but basically did not have the memory to make it easy for the user to access them as single patches.

It is actually a testament to the flexibility of the device that it can fill 475 locations with variations of its processing skills and while I am not saying that every one is totally different, I am saying that you do not get an impression of great similarity. You can not say that of many units of this type. There is a phenomenal palette of tones including many excellent ones from the original *SGX2000*.

The *Express* is a worthy successor and adds real value. A very good device has just become superb. ■

Advanced Research and Technology, 215 Tremont Street, Rochester, New York 14608. Tel: +1 716 436 2720.
UK: Harman Audio, Unit 2, Borehamwood Industrial Park, Rowley Lane, Borehamwood, Herts WD6 5PZ. Tel: 081 207 5050.

Music News is compiled by Zenon Schoepe



James Douglas visits a hot new US facility committing to the use of networked digital audio workstations

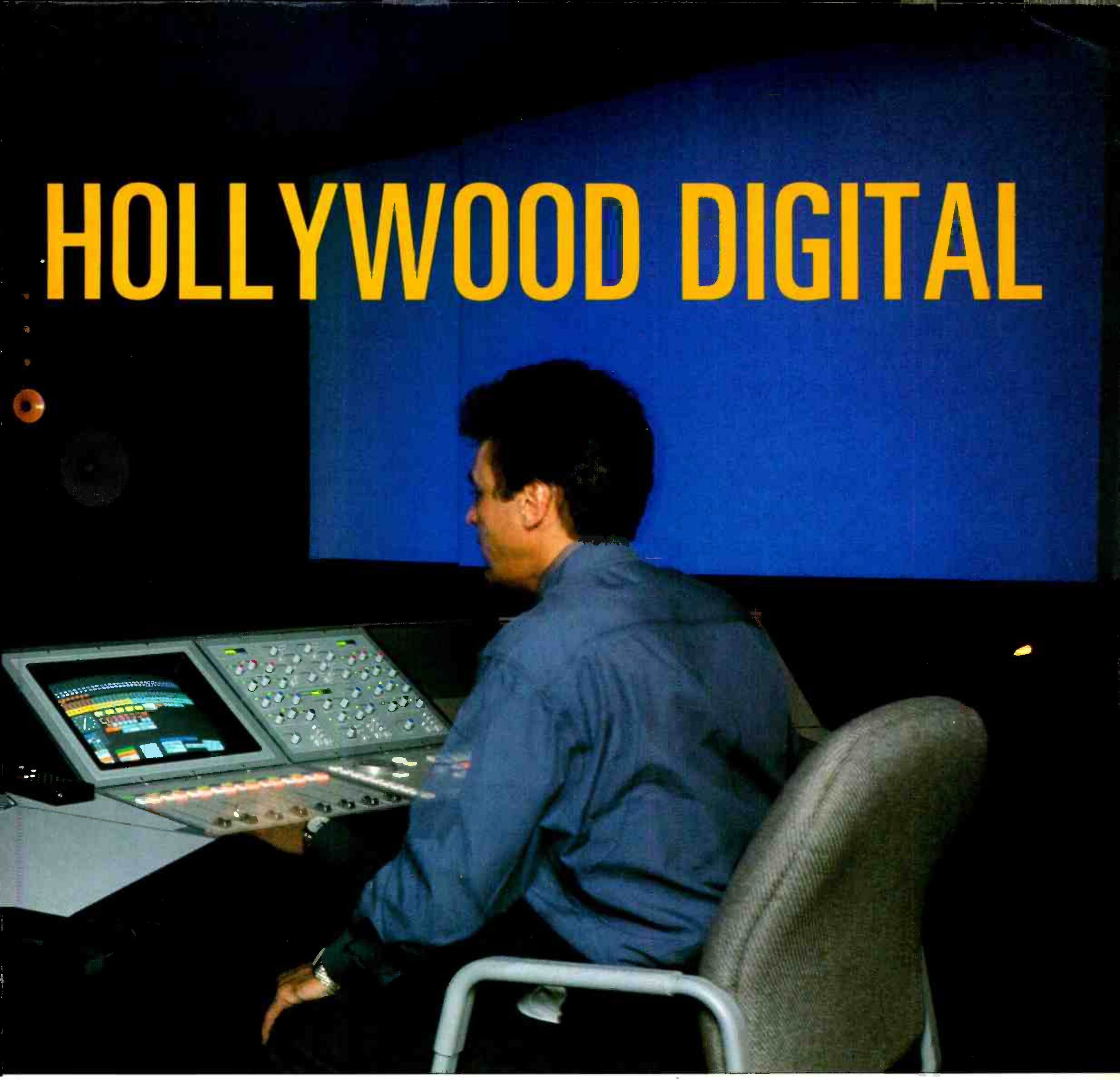
In terms of enhanced audio and video quality, digital recording and editing has virtually eliminated the generation losses that were all to common with analogue technologies. In addition, the nondestructive nature of disk-based systems means that audio editors and mixers can refine and continue to develop the final stereo/surround-sound mix, without having to redub source reels. In just about every aspect, the advent of digital audio represents a dramatic paradigm change for the video postproduction industry.

Hollywood Digital is a new West-Coast-based facility that has made a dramatic investment in digital audio and video techniques. Ground breaking for the 2-story complex, located on Sunset Boulevard in the heart of the Los Angeles music-recording, broadcast and media district, began in August 1992; the facility formally opened for business in February 1993. From the outset, Hollywood Digital's owners, headed by CEO Bill Burnsed, made a firm and unwavering commitment to digital technologies. The complex includes digital component and composite edit bays, telecine, audio editing and mix rooms located on the ground floor, with a traffic department and general office suites on the upper floor. A central machine room houses the various digital video and audio transports, plus digital routers, synchronisation and peripheral equipment.

But, as Burnsed is the first to concede, timing was important to the facility's success. 'Put simply, we could not have built this facility a year earlier—the technology just wasn't available. Our overall *raison d'etre* was to improve the quality of audio as well as video, by implementing one of the first truly all-digital throughputs for video postproduction. Only recently, however, have we seen the introduction of hardware that really allowed us to design and build a state-of-the-art, all-digital facility: a sophisticated digital video routing switcher; second-generation *D2* DVTRs; and fully integrated, disk-based audio editing-mixing workstations.'

For Burnsed and his partners, 'audio is going to be a focus of improvement in network, cable and related video delivery media for the next five to ten years. Responding to those needs, we have created a facility here that elevates audio and video postproduction to a new plateau of refinement.'

HOLLYWOOD DIGITAL



Space is provided on the ground floor for an eventual total of ten on-line digital edit bays, plus a pair of telecine rooms, two mix-to-picture suites, two sound editorial rooms and a central master control-machine room. Systems design was handled by B&B Systems, based in Valencia, CA, with acoustic consultation from Audio Division Director Andre Perreault and Ken Fause. Total budget for the 33,000ft² complex, Burnsed recalls, was close to $11 million.

'We worked very closely with the top hardware manufacturers,' he continues, 'at what we consider to be an unprecedented level of cooperation.' The Grass Valley Group was Hollywood Digital's main video supplier; Solid State Logic supplied the main audio systems; BTS-Sony provided the majority of digital ATR and VTRs.

All signal routing is handled via a GVG *Serial Digital Video Router* for composite and component video, linked to a GVG *Digital Audio Router*. Total audio routing capability is 284-in by 384-out, routing via four channels of AES-EBU-format signal per cross point. Sub-sample accurate audio-video synchronisation for all NTSC and PAL frame rates is achieved via an Nvision *Master Digital Clock*. Nvision *NV4448* sample-rate convertors are also available for 44.1kHz/44.056-to-48kHz digital-domain transfers.

On-line edit suites

Four composite digital *D2/D3* Edit Bays house Grass Valley *Model 251* editors and *Model 3000* switchers, linked to Accom digital-disc recorders. A component digital *D1* Edit Bay features a *Model 251* editor linked to a *Model 4000* switcher. The two digital telecine bays utilise Rank Ursa units with Renaissance *DaVinci* colour correction and a GVG *Model 1000* component switcher. All edit rooms and telecine bays incorporate Graham-Patten Systems D-ESAM Series *Digital Edit Suite Audio Mixers*, which provide edit-system control of audio mixing in a manner similar to conventional video switchers. D-ESAM mixers are designed around a programmable input-routing architecture that can be controlled from an external video editing system, using industry-standard ESAM II serial-control protocols. Equalisation and external processing loops are also available. Any analogue or digital audio input can be electronically assigned as a source to any channel fader and any output —on an edit-to-edit basis if necessary. All channel level settings, channel-machine assignments and audio crossfades can be stored as snapshots, and instantly recalled. As ▶

broadcast and post industries incorporate digital audio and video technologies, D-ESAM products offer complete function control from the video-editor control panel, coupled with a highly sophisticated input architecture employing virtual-machine concepts.

Each D-ESAM *800* handles up to 56 analogue and digital inputs, routing via a Virtual Input Matrix to four analogue Programme, four digital Programme, and four Monitor outputs. Programmable EQ, channel delay and digital output for external processing are also available. Two additional D-ESAM systems have also been installed at an off-site location owned and operated by Hollywood Digital, where Fox Television's *Front Page* series is edited.

'The D-ESAM digital mixers were chosen for four, primary reasons,' Burnsed recalls. 'First, the name "Graham-Patten" is universally recognised as signifying high quality audio for our clients. Second, D-ESAM mixers are very straightforward to operate, and instinctive in operation. Third, Graham-Patten have an excellent track record for delivering its products on time, and fully operational straight out of the box—we just cannot afford downtime at our facility! And, fourth, the D-ESAM is fully compatible with our existing Grass Valley Group *Model 251* editors, as well as the new *Sabre* and *Axial* systems currently under evaluation.

Digital audio and video tape machines housed in the central machine room include: 3 BTS-Sony *Model 300* and 2 *Model 500 D1*-format DVTRs; 9 Sony *DVR-20* and 4 *DR-28 D2*-format DVTRs; 6 Panasonic *D3* composite DVTRs; 4 Sony *D-75* and 12 *D-265* Beta-D VTRs; and 5 Sony *BVU-3000* analogue C-Format VTRs with serial-digital video I-O.

'The D1-component digital edit suite,' Burnsed continues, 'specialises in high-level commercials, music videos and special productions, plus the kind of sophisticated compositing necessary for commercials, graphics and title sequences. The composite bays handle the majority of editing sessions at Hollywood Digital, including episodic TV and drama series, plus *Movies-of-the-Week*. Half of our sessions are for broadcast episodic TV and drama series; 30% for commercials, music video and promos; and the remainder for informational and high-end corporate productions.'

Networked audio-for-video

Because of the sequential nature of audio mix-to-picture and sweetening, Hollywood Digital decided to opt for a fully integrated approach from editing through mixing. According to the facility's Audio Division Director, Andre Perreault, the initial design plan called for a pair of dialogue, music and effects editorial rooms feeding a pair of mix-to-picture suites.

'But we also wanted to make sure that files could be accessed from each area,' he stresses, 'so that we could dramatically reduce the amount of sound transfers required from the initial production reels to the final soundtrack mix. For that reason alone, the SSL *ScreenSound* and *Scenaria* combination, with a *SoundNet* system to interlink them, was the only way to go.'

'It was an easy decision to go with the integrated *Scenaria* system,' Bill Burnsed confides. 'In addition to enhanced digital record-replay quality, *Scenaria*'s random-access replay-editing, digital mixing and [*VisionTrack*] hard-disk video [storage system] has dramatically improved our work throughput.' ▶

HARTS OF THE WEST

Weekly production of one of CBS Television's hit series, *Harts of the West*, produced by Kushner-Locke, truly stretches the outside of the all-digital technology envelope. Starring the father and son team of Lloyd and Beau Bridges, audio production for *Harts of the West* ties up three rooms at Hollywood Digital for the better part of a week. According to Audio Division Director Andre Perreault—and who also mixes the show—'we spend about 14 man-days on editing and mixing *Harts of the West*. Dialogue and ADR editing on *ScreenSound* in the Audio3 Suite takes up about 10 man-days, while Foley, music and effects requires around 4 man-days in Audio4.

'The production dialogue comes to us on time code Nagra reels, 20–40 per show, which we load into *ScreenSound*. Having accessed the EDL information from the video off-line session, we can use *ScreenSound*'s Autoconform function to checkerboard the sound clips against time code. Then the dialogue editor will go back and fix any holes, and also load in any alternate takes that he thinks might be needed during the mix session. The music is supplied on time-code DAT, which is laid in against the work print, and then edited on the second *ScreenSound*.

'Eventually we end up with the material prelaid to four hard drives: one for eight channels of dialogue; another for eight channels of ambiance and background tracks; another for eight channels of Foley and music; and the final drive for eight channels of sound effects. These four drives can be accesses directly from my *Scenaria-OmniMix* system in Audio2, using SSL's *SoundNet*.'

Perreault lays out the mix on his 32-channel assignable *Scenaria-OmniMix* console as follows: Channels 1–8 for dialogue; 9–16 for sound effects; 17–24 for stereo and mono backgrounds-ambiances; and 25–32 for six tracks of Foley and stereo music. In reality, channels 1–24 are located within the *Scenaria-OmniMix* section, and the remaining eight through the built-in *ScreenSound*. 'I can also re-assign sources so that I can use the *ScreenSound* editor to edit an fine tune the tracks; on the *Scenaria-OmniMix* I'm limited to fetching sound files to the desktop, and locating them to picture.'

The various tracks are equalised and blended to produce a final stereo soundtrack mix. Prior to the *OmniMix* upgrade, Perreault needed to route sound cues via effects-send buses to external delays and reverb processors. 'Now I can access everything from *OmniMix*, including reverb programs and room simulations, and store their settings as part of the automation files for subsequent recall.'

OmniMix also means that the engineer can simultaneously output the submix balances that are transferred in the digital domain to a Sony *PCM-3324*. 'For *Harts of the West*, I need to produce a stereo mix plus M&E tracks for foreign language dubs—which are laid back to tracks 1–4 of the master *D2* digital VTR—as well as stereo dialogue, stereo effects, stereo Foley and stereo music submixes for the *3324*.' A single 24-track tape holds stereo submixes for three shows.

'Before the *OmniMix* upgrade, I needed to make separate passes through the system to generate the M&E and four stereo submixes. Now, because of the additional output buses, I can generate them all simultaneously. That saves me a lot of time at the end of a long mix session!' ■

What a concept . . .

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SoundNet comprises a central resource of hard disks, M-O discs and backup tape drives, each of which are made available on request to the *ScreenSound* and *Scenaria* rooms. For assigning drives *SoundNet* uses high-speed, point-to-point SCSI interconnects laid out in a 'star' configuration; a separate Ethernet serial network connects all processors in a 'ring' structure to coordinate and control activities. Such a hybrid ring and star network is essential to fast work flow. Since each *SoundNet* supports a total of up to eight users (including off-line Exabyte and M-O backup), the network in essence offers the equivalent to 64 simultaneous channels—including up to 150% varispeed—from a central storage area, without compromise nor conflicts between individual stations.

SoundNet's storage array resides in a central machine room, along with a variety of time-code-capable DAT machines for inloading production dialogue and other time-code-stamped materials. Also located in the machine room are two Sony *PCM-3324* DASH-format multitrack used to hold final stereo mixes, plus multichannel submixes and M&E (music and effects) stems.

A pair of editorial rooms handle dialogue, ADR, Foley, sound-effects and music editing on *ScreenSound* systems, which were recently upgraded to v.5 specifications. An enhanced processor dramatically speeds up a variety of functions, including screen displays, machine control and scrub editing. Other new v.5 features include high-resolution screen graphics, advanced editing options, plus audio reconform and autoconform. The upgrade also enables *VisionTrack*, SSL's disk-based, random-access video option to be added to *ScreenSound*, in a format that now includes a 2-channel recorder function for off-line sound loading.

The two identical mix-to-picture rooms feature SSL *Scenaria* postproduction systems, equipped with 38 virtual channels, dynamic automation and 24-track random-access hard-disk recording capabilities. Each *Scenaria* accommodates 24 individual analogue and 24 individual digital AES-EBU-format inputs. The systems were recently upgraded to *OmniMix* format, which adds several radical features, including surround-sound mixing, dynamic panning, and multitrack output buses. Also provided on each *Scenaria-OmniMix* is a full-function *VisionTrack* system, which allows one hour of bandwidth-compressed video to be stored to a dedicated hard drive. In this way, the mixing engineer can instantly locate to a selected time-code location, without the inevitable delays associated with VCRs or laser disc players.

'We can also edit *VisionTrack* files,' Perreault says, 'to accommodate picture changes where material has been removed, or to add black in sections where new scenes might need be added. We just make the cuts, and all audio [time-code-referenced] edit points and automation data are updated automatically.'

Edit decision lists generated by conventional video editors can be imported directly into *ScreenSound*, and assigned to tracks for autoconform sequences. At the time of our visit, as mentioned in the accompanying sidebar, Hollywood Digital's *Scenaria-OmniMix*-equipped Audio2 Suite was being used to remix the popular *Harts of the West* TV series. *Harts of the West* is off-line edited on a Avid *Media Composer* nonlinear system, and then on-lined with conventional DVTRs. 'Because we can access the time-code EDL data,' Perreault points out, 'it dramatically streamlines the dialogue and ADR stages in our *ScreenSound* rooms, and saves a great deal of time!' ►

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SOUNDNET AND HIGH SPEED DATA TRANSFER

The majority of current-generation audio hard-disk systems feature an enclosed structure combining a processor and a hard disk. Such systems need to copy, or upload, material onto the disk each time it is required, and before any creative work can take place. While material will always have to be transferred into the digital system at the start of a post session, duplication at later stages is unnecessary. For many facilities, the time required for routine data transfer is a fundamental obstacle to working efficiently.

Simple, 2-channel editing and mixing does not normally form the heart of audio postproduction. Instead, multichannel capability is required, coupled with a process that typically will pass work through a number of stages for preparation, editing and mixing before the master soundtrack is produced. A need for efficiency and cost-effectiveness has resulted in several strategies for moving work through each of these complementary stages, including faster copying and, of greater importance, networked systems.

We might draw up the following list of criteria for a 'real-world' digital audio network:

- It should provide multiple operators with a working area that offers unrestricted random-access control.
- It should allow operators to be located anywhere within a multiroom facility.
- It should provide high-speed intermediate data storage (both short-term and long-term) for work-in-progress.
- It should provide high-speed search and import capabilities when additional sounds are required during a session.
- It should provide off-line backup without tying up an operational facility.
- It should expedite, when necessary, rapid changes of sessions.

These are all functions we take for granted in a conventional office network. However, many assumptions that might currently be valid for office Local Area Networks (LANs) are inappropriate for digital audio networks. A typical LAN communicates serial data via coax cable using, for example, Ethernet and Token Ring protocols, with a theoretical maximum speed of 10 Mbit/sec. (But, due to hardware limitations, this is seldom realised in practice.)

Digital audio data is, of course, extremely bulky and requires much more storage and transmission capacity. A single channel of 16-bit audio at a 48kHz sample rate requires 0.768 Mbit/sec, a number that rises proportionately for higher sample rates or greater resolution. The idea of multiple users accessing multiple channels of digital audio over a LAN simultaneously and reliably is not a practical reality.

It can be argued that faster serial systems with fibre-optic cables might loosen these constraints by offering the same Ethernet and Token Ring protocols, but at a higher raw data rate. Yet all such a strategy can do is to move the boundary without changing the underlying realities. As soon as traffic on the network builds up to a useful level, users' needs for on-time audio will conflict, and someone will be forced to wait for real-time access to a sound file.

But there is an alternative network strategy that achieves the required result much more directly and reliably. Two separate structures form a single hybrid network, providing multiple users with access to central audio storage. Such a system can support a central resource of hard disks, optical drives and backup tape drives, and make them available to a number of users on request. For the disk assignment, the hybrid network uses high-speed, point-to-point SCSI interconnects in a star configuration. A separate Ethernet serial network connects all processors in a ring structure to coordinate and control activities.

Such a 'ring and star' hybrid is fundamental to fast work flow. A total of 8 *ScreenSound* users (including off-line backup) is equivalent to 64 simultaneous channels, including varispeed of up to 150%, with no interaction between users and no theoretical limit to expansion. Distributing audio and project storage resource is simply a matter of assigning devices to users. Projects may be switched from room to room in a matter of seconds, sending material from autoconform to the editor, or from the editor to the mixer, for instance. Working disks can be held by the network for temporary storage of partly-completed projects.
When the time comes for Exabyte data tape backups, these can be managed directly by the network controller. Off-line backups are handled in the machine room, and do not need to involve operational areas. Effectively, the edit and mix rooms have no backup or restore downtime, and can be scheduled to fit the work, not the other way around. ■

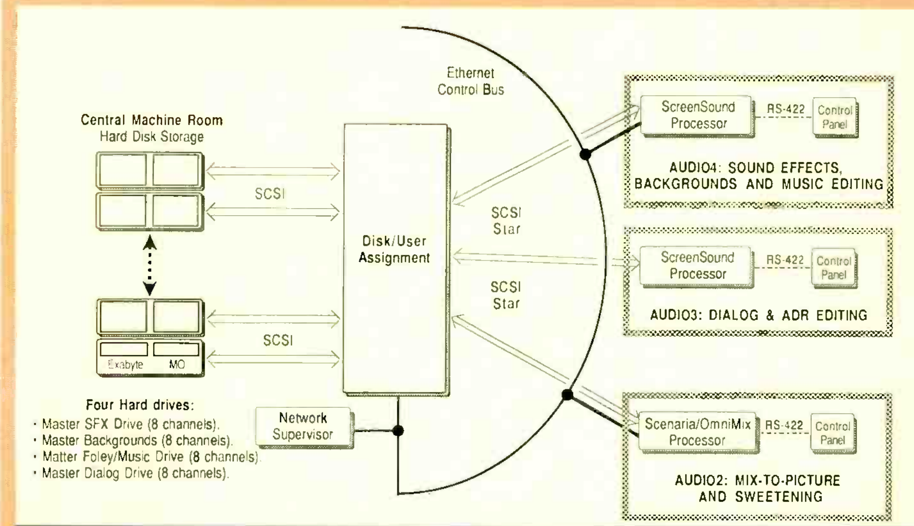

Diagram of *SoundNet* system used to interconnect various editing and mix-to-picture rooms at Hollywood Digital, during the production of the weekly *Harts of the West* series

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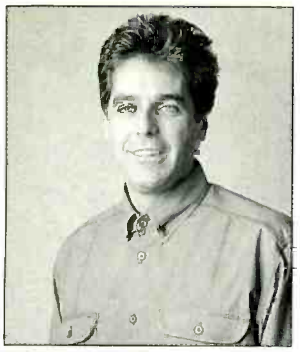


Digital vs analogue editing and mixing

Above: Audio Division Director Andre Perreault. Below: Hollywood Digital CEO Bill Burnsed

For the majority of TV shows produced in analogue or hybrid analogue-digital facilities, audio tracks are prepared by editing production dialogue and ADR to match picture edits, and then adding various music, sound-effects, Foley and related cues. These elements would then be prelaid to a time-code multitrack reel, with alternatives and maybe A-B checkerboarded tracks if the project was particularly complex. Any changes during the mix would then require that the source reels be re-edited and redubbed to the multitrack reel.

Hollywood Digital's combination of *ScreenSound* and *Scenaria-OmniMix* systems means that dialogue, music and effects tracks can be cut to a work print in one of the pair of *ScreenSound*-equipped audio edit rooms, and then the same sound cues stored on hard disk —including mix-automation data —accessed from the *Scenaria-OmniMix* work surface.

Designed to provide full coordination of virtually all editing, mixing and processing functions during film-video post, the all-digital *Scenaria-OmniMix* incorporates a 38-channel audio mixing console, a 24-track audio recorder, and random-access video storage in a single unit. The system's control surface comprises a central VDU showing track layouts and other system graphics; a bank of eight servo-controlled, motorised channel-group faders; a transport control, scrub-edit and bank-switching panel; an assignable 4-band parametric EQ, dynamics and aux-send panel; plus a graphics tablet and pointer. In addition to the 24 virtual tracks provided by *Scenaria*'s built-in hard drives (plus opticals for accessing libraries of music cues, sound effects), additional analogue-digital inputs are available for connection to external sources, including time-code DATs, digital VTR outputs, etc.

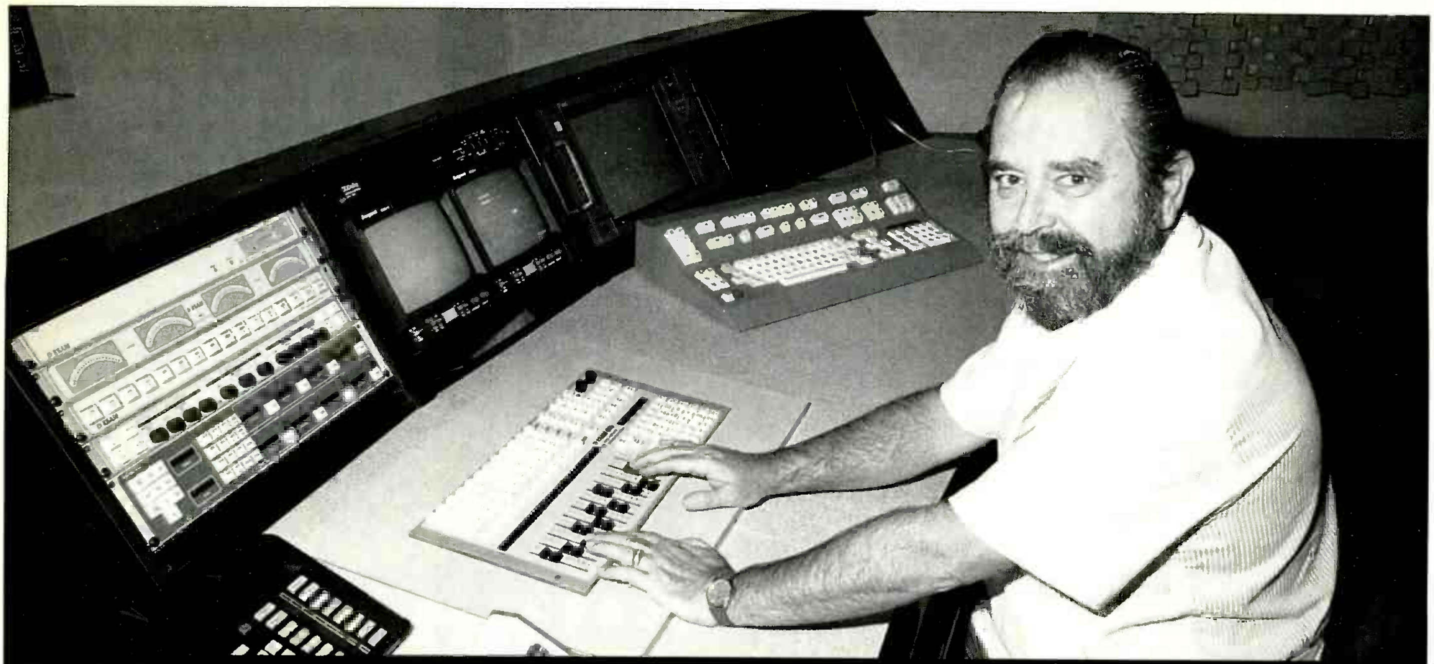
The system's physical faders—8 on *Scenaria* and 16 on *OmniMix*—can be assigned to individual tracks or source inputs, or assigned as nested, freely assignable subgroups. Eight auxiliary sends are available from each input-subgroup. Front-panel settings can be scanned and memorised dynamically, or as a series of snapshots. All *Scenaria* inputs pan between master left and right outputs; *OmniMix* provides full surround-sound mixing with assignable panpot and automated control of signal assignment for LCRS and other formats.

What have we learned?

Summarising his first year's experience of operating Hollywood Digital, Bill Burnsed considers that the facility has been remarkably successful, but in a way that he did not predict. 'I thought that if we offered the best digital audio and video equipment, we would attract a wide range of post clients. But instead, we have found that our state-of-the-art digital equipment attracted a remarkable collection of video and sound editors who, in turn, because of their enthusiasm at what we had here, pulled in the clients. So, instead of the technology attracting the clients, it has been more the case that our talented personnel proved to be the best ambassadors we could have had!

'By and large, our clients love the improved quality that digital systems can offer, but are more impressed by the talent that uses that technology. I also feel that, during our first year in business, digital audio workstations might have been 'frosting on the cake' for some of the projects we've handled. Next year, however, to be truly competitive—and fulfil the exacting requirements of a far more sophisticated client base —we are really going to need these leading-edge systems.

'The number one advantage of the *ScreenSound* and *Scenaria*- ▶



Eldon Phillips at the helm of a *Model 251 Editor* and GVG *Model 4000 D1* switcher in Edit Bay 7

OmniMix systems, we have found, has been their ability to let our clients creatively fine-tune their projects in a time-efficient way. Once a video programme has been edited, we need to provide a way for the director to finesse the audio soundtrack. *Scenaria-OmniMix*, in particular, allows quick and easy changes to be made to the soundtrack mix, and even new or revised sound cues to be added.

'That ability to work quickly and efficiently gives our clients a secure feeling that they are involved in a highly creative process. With older, analogue-based systems, they would have had to compromise in what they could achieve—either because of time-budget constraints, or simply because there was no way of creating the quality they needed. Now everything is possible; there are no compromises.' ■

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James Douglas is a Los-Angeles-based freelance writer and technical consultant

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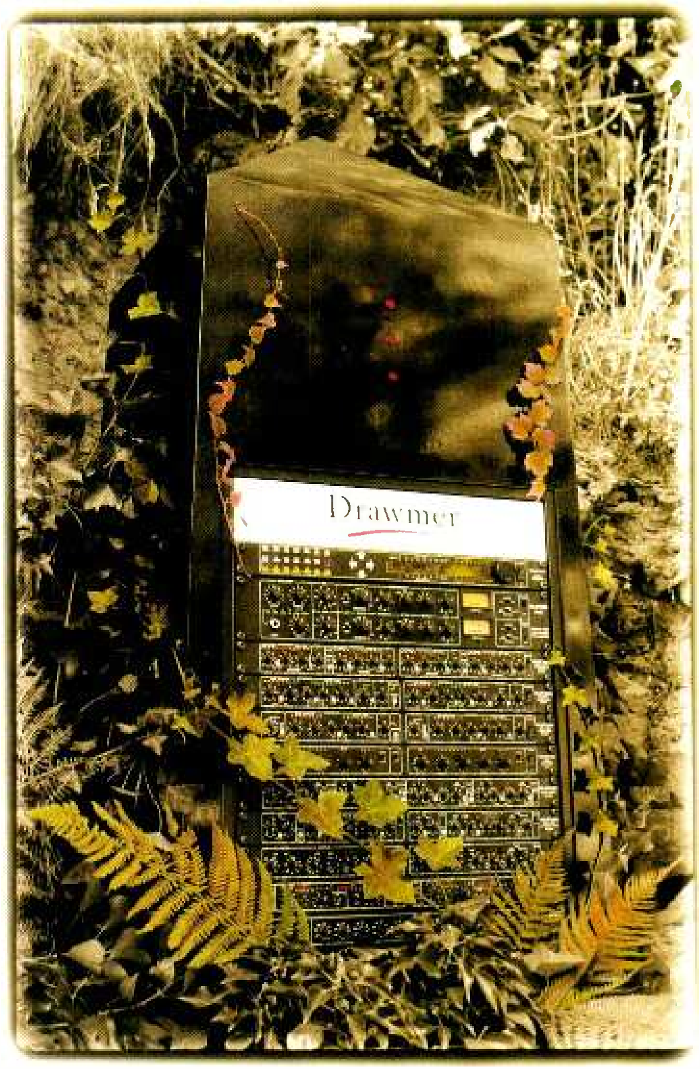

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PRACTICE MAKES PERFECT

Tim Frost talks to Decca's Tony Griffiths about good engineering practices in classical recording

The latest generation of recording techniques introduced by the classical recording world may have generated a certain amount of debate, but what no-one questions is that these 'systems' have grown out of a combination of good engineering practice and the development of noise processing ideas that have been around the industry for several years.

Decca, what one might consider as sister company to Deutsche Grammophon and Philips (being within the PolyGram group), have been developing their own digital ideas since 1975 resulting in the first digital recording from a major label in 1979. This live *New Years Day Concert* LP was released with no reference to its digital nature in the sleeve notes. Only a 'Decca Digital' sticker, hastily added by the marketing department at the last minute gave the game away.

To 'go digital', Decca had to develop their own open-reel video-based recorders, which originally were developed to archive their back-catalogue. But the open-reel video-transport-based machines were hi-jacked by the recording engineers who recognised a good thing when they heard it. Tony Griffiths, General Manager at the Decca Recording Centre, has guided the company's digital progress into the 1990s. They are still using the same digital transports—grown in number to 45—but now making extensive use of all-digital signal routes, high bit-rate convertors and noise shaping techniques. We discussed how 'digital' the recordings have been.

'From 1981 to 1992 all of our recordings have been recorded using digital recorders with an analogue mixer prior to the A–D convertor,' he comments. 'The signal then remained digital throughout the editing process through to the end product. All the postproduction processing was carried out using digital mixers, editors and recorders designed by ourselves.

'Some recordings since 1992 have been made using our small 8-channel digital mixer where the microphone amplifiers directly feed our new A–D convertors. The use of a digital mixer for session recording has only become important with the development of the new A–D convertors where a dynamic range approaching that offered by 20-bit systems is required, otherwise the results might be better with the analogue mixer.'

Although these recorders were developed before CD, they were designed from the outset to be 18-bit-ready and despite somewhat modest comments by Decca, when every element of the signal chain is in operation, its performance measures around that of a 19.5-bit system.

Decca saw the need for better-than-16-bit performance for production work, even though at that time 14 to 16-bit convertors were the norm.

'When we were doing our original development we realised that around 16 bits were required for the final consumer product. We also knew that any professional system should have a higher performance and it seemed, in 1977, sensible to design the digital system to cope with 18 bits even though 16-bit A–Ds were about the limit of the performance in those days. The improvements in convertors from 12 through 14 to 16 bits happened very rapidly so we were expecting the development of 18-bit A–Ds sooner than it happened in practice. Now, of course, we record in 18 bits all the way through the production process.'

With microphone amplifiers nearing their theoretical performance limits, noise from the microphone end of the chain could be noticeably reduced by placing the microphones nearer to the musicians so reducing the gain, and therefore noise, within this last remaining analogue stage.

On record

Tony Griffiths has the evidence of good sounding Decca rereleases from the last three decades to show that it would be wrong to aim for the lowest possible noise floors at the sacrifice of good microphone techniques.

'It is also possible to use a lot of microphones close to the instruments to get a better signal noise ratios than we achieve using more distant microphone techniques, but at the end of the day you have to assess the sound image and balance that is achieved. You only have to see the success of older Decca analogue recordings to realise that while low noise is important, and low noise ▶

is always better than high noise, it is the overall result that the customer is interested in.'

In the early days of digital there was a very limited choice of digital desks—even now they are hardly a mass-market product. Since Decca had both the resources and skills to design a unit to exactly meet their technical and operational needs, that is what they did.

Low-level performance is an especially critical area when processing digital signals, by building and using their own digital mixers Decca have learnt a lot about the importance of small signal performance and dither.

'During the introduction of CD there was a lot of criticism about the harsh sound of CDs,' Griffiths begins, 'this was often due to the commercial studio equipment not looking after the rounding errors following gain changes made with the editing or mixing equipment. With the first generation of commercially available editing equipment no dither was employed, so if the fader was moved from unity gain then quantisation distortion was likely to be generated and distortion would also be generated briefly during the crossfades. Also commercial mixers have been seen to generate quantising noise by not handling the rounding errors produced by digital processing.'

Since the 1980s, Decca have built over 30 8-channel digital mixers. These are used both in house—there is on in each operational room at the London recording centre—and a small number have also been supplied to DG and Philips Classics.

The costs involved in designing and building your own digital desks would seem, on the face of it, to be prohibitive. But with the development costs now spread over nearly 50 units, Griffiths thinks he has got bargain.

'The problem as we saw it was that if you do not make your own system then you have to buy one in, and in the past there has been a very limited choice of digital mixers. The cost per mixer, including the development time is around £14,000—a fraction of the cost of the first commercially introduced equivalent. Developing them ourselves has given us control over the technical specification and the operational features, and in addition this approach has also been cost effective.'

Decca development work continues, recently producing eight new digital editing systems designed specifically to meet the requirements of their classical music editing team.

As far back as 1985 Professor Stanley Lipshitz —now the leading light in all things ditherable —started discussing techniques with Decca who were already using 'Decca Dither' within their mixers. Careful noise shaping can reduce the subjective noise level on a CD to the point where it can seem to have 18-bit performance. Decca have been using one of Lipshitz' 'quieter' curves for some months now and Griffiths is happy with the benefits although is still listening for potential side effects.

'We have done further listening tests here and with the right noise shaping we can get about 105dB out of a 16-bit system,' he says. 'The noise is at the borders of detectability or even below it—it sounds like the 18-bit signal in terms of absolute signal to noise ratio.'

Griffiths does not approach digital systems as an instant panacea to good sound. Even these new noise shaping techniques, which seem to offer a good incremental step forward in CD sound, are being closely monitored, to see (hear) if there are any contra-indications that need to be addressed.

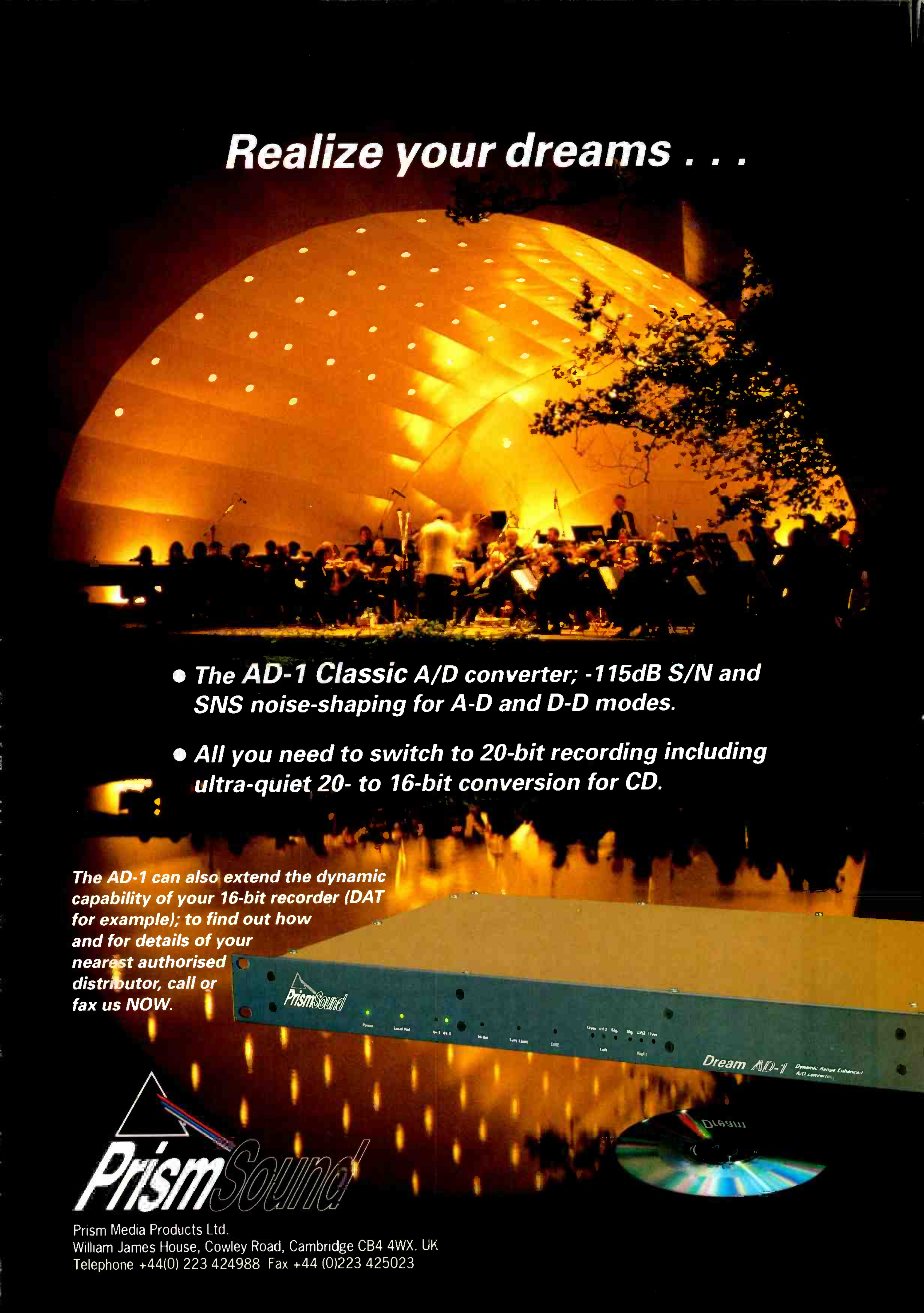
'We have started using one of Lipshitz' curves ourselves but we are continuing to see if there are any side effects, there may be problems with the stereo imagery and these have to be investigated.'

Research from several years ago indicates that an 18-bit dynamic range approaches the limits for domestic listening conditions, where environmental noise will becomes more significant than system noise. However the upper limit for reproduction does not restrict Decca's aims for further increases in system performance.

'Up to 18 bits is of value given the right D–As, replay systems, amplifiers and good listening conditions,' states Griffiths, 'over 18 bits is to useful to solve internal engineering problems. The real advantage is not so much what can be recorded onto the system originally but how much quality headroom it gives the engineers during the mixing stage.'

Griffiths reinforces the point that, like analogue, digital processes generate their own set of problems, which have to be contained during the process of bringing live sound to CD. 'Digital' or any other catchy phrase on the box doesn't guarantee great sound.

'One fact that should be understood is that the postproduction process can, if not properly controlled, destroy the small signal and noise performance of a digital system and that simply stating that some xxx process is involved does not itself guarantee the system specification. There are many pitfalls for the unwary and there is equipment out there being used which does not come up to the basic requirements that we would accept.' ■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FOUNDATION AND EMPIRE

***Foundation 2000*: Fostex anticipate the dawning of a new era**

Fostex have declared the *Foundation 2000* the basis of a new empire, Yasmin Hashmi offers an exclusive technical preview

Fostex describe the *Foundation 2000* as a computer designed to support integrated recording, editing and mixing. They are keen to stress that the system is not just a hard disk recorder-editor, but that DSP and mixing form a key basis of the system. Their design philosophy states that the system should be a 'highly intuitive and efficient tool' to be used in the areas of film, TV, video and music production. Furthermore, they maintain that it has been designed to operate as a self-contained 'recording studio in a box' as well as an integral part of multimachine environments.

Internally, the system supports replay of 16 simultaneous channels, but to the user, operation is presented in terms of eight tracks. Having twice the number of channels as tracks means that all eight tracks can be crossfaded simultaneously, in real time, without fear of dropout or limiting replay to less than eight tracks. Furthermore, the system has been designed such that it requires only one disk to replay all 16 channels. This can only be achieved from a high-performance hard-disk drive, and in order to guarantee 16 channels in worst-case conditions, a proprietary file format optimising disk performance is used. This fragments the audio data as it is recorded to disk and ensures a consistent performance. Optical discs can be used, however the number of channels supported will necessarily be reduced (to around eight channels for current drives).

In terms of inputs, outputs and DSP, the system has been designed to be modular, allowing the customer to extend the basic system simply by adding modules to the main hardware unit (and, if appropriate, DSP algorithms to the software). On a grander scale, *Foundation*'s modularity has been designed so that multiple units (in close proximity) can be cascaded and controlled by one user-interface.

Fostex maintain that from the systems' point of view, cascaded units operate as one logical device (although drive storage and DSP processing are dedicated within each unit) rather than as separate units in sync. Hence from the user's point of view, a 'super *Foundation*' can be built, in modules of eight tracks, with synchronisation and mix output being controlled by whichever unit has been chosen as the master. Currently, the minimum number of successfully proven cascaded units stands at six, which means that a 48-track system (with 96 internal channels) is possible, with integrated mixing and synchronisation.

Hardware

The main hardware unit consists of a 6U-high rack which has been designed for modularity and has a central backplane with room for up to six cards each side. The slots at the front are reserved for up to six generic DSP cards (only one of which is currently supported) while the slots at the back are reserved for *Foundation*'s two computer cards, the master audio module, the master sync card and two empty slots for I-O expansion.

According to Fostex, their proprietary computer has been optimised for professional audio requirements and all hardware has been designed from the outset, to support full 24-bit audio (although this is not currently ▶

PHOTO: NIK MILNER

The men behind the machine. Left to right: Richard Rosensweig, Vice President of Engineering; Michael Geilich, Engineering Manager; Jeff Postupack, Customer Support Services Manager; Eric Richardson, Product Development Manager

utilised). For those interested in processing specifications, the computer uses Motorola 68030 processors, while the DSP is provided by latest generation Motorola 56002 processors which allows 56-bit internal maths.

All inputs and outputs are located at the rear of the unit, while the front sports a removable drive docking bay and a datacard slot. This slot is used instead of a floppy drive, and accepts standard computer datacards (the size of a credit card, but slightly thicker) which use flash ROM and currently have a capacity of 1Mb. A datacard can be used to introduce software updates or new DSP algorithms and can be sent through the post in the same way as a floppy disk is presently.

The docking bay currently accepts a removable hard disk drive with a capacity of up to 2Gb (this gives 6 hours of mono recording at 44.1kHz). In addition, the system supports a single SCSI chain, which allows up to seven external SCSI devices to be connected (one of which could be an archive device). This means that additional drives can be added to increase the overall storage capacity of the system, but not the number of audio channels supported.

Fostex point out that *Foundation* is not tied to any particular drive manufacturer or drive, and the user can source their own drive(s) if desired. However, it should be noted that in order to maximise the channel capacity of the system, *Foundation* relies on high-performance SCSI drives and Fostex therefore publish a list of drives which they approve.

Sync and connection

The master audio module supports 2 analogue inputs, 4 analogue outputs (configured as left and right master and monitor mix) and 2-in/2-out digital (AES-EBU and SPDIF). Convertors are 18-bit, with the A–Ds using 64x oversampling with anti-aliasing digital filters and the D–As using 8x oversampling with digital anti-imaging filters (an anti-imaging filter is used to remove images which are naturally formed by the sampling process at the D–A stage. These images are exact duplicates of the baseband signal and exist at integer multiples of the sampling rate).

In addition to the master audio module, *Foundation* can accommodate up to two analogue multichannel I-O modules, giving a total of 18 analogue inputs (configured as 2 master, 8 direct-track inputs, 4 aux and 4 spare) and 20 analogue outputs (configured as left and right master, monitor mix, 8 direct, 2 aux sends and 6 spare). Although not currently available, a multichannel digital I-O module will soon be offered as an option and will support 8-channel AES-EBU, SDIF2 and *ADAT* format.

The master sync module has several ports for RS422, as well as a port each for SCSI, CPU cascading, audio cascading, MIDI In, Out and Thru and a GPI (General Purpose Interface). It will synchronise to word clock, video sync, LTC (all flavours) and VITC, and the LTC interface is a wide-band reader which will recognise code from 1/30th to 50x normal play speed, both in forward and reverse. In addition, the system will output LTC and there is a through port for VITC.

User interface

Fostex call their user interface the Edit Controller and it consists of a robust, yet relatively lightweight, angled panel. It can be clipped (in an upright position) directly onto the front of the main unit if required, but is more likely to be used lying flat on a surface or on the lap, connected to the main unit by a cable (a 5m cable is supplied, although the Edit Controller will operate with longer lengths. Controls are laid out in function-related sections. The machine control section is situated top left and allows the user to select whether control is of external machine(s) (via RS422) or of *Foundation*(s) or both and allows external machines or cascaded *Foundations* to be brought on or off line.

Below this section are two rows of eight buttons which represent traditional track-ready and track-solo functions. Beneath these are buttons for external time-code lock, auto record, return and play (for setting up auto record, punch-ins, loops and play start point), auditioning edit points (including fades), input monitoring, locate and 'seek' left and right. The SEEK buttons are used to step back or forth through the audio and will jump to the beginnings and ends of cues (or 'events') as well as any edit points in between. The seek function can be used across all eight tracks or will only apply to the selected track(s).

Transport controls are located bottom left and include RECORD, STOP, PLAY, REWIND and FAST FORWARD buttons. A special feature of the REWIND and FORWARD keys is that if they are pressed once, motion is 2x normal play speed. Pressed twice, it is 8x and pressed three times, the beginning or end of the 'reel' is located. Furthermore, if used in conjunction with the PLAY key, forward or reverse CD-like scanning will be heard ('snatches' of the audio at normal pitch while replay speeds through the audio).

The top centre of the Edit Controller is taken up by the main display which comprises a high-resolution, touch-sensitive electroluminescent display, measuring around 5in wide (13cm) and 4in tall (10cm). Given the relatively small size of the display, touch operation is limited to navigating through the various 'pages', selecting options-preferences and using alphanumeric keys which are displayed when appropriate (although an IBM-compatible keyboard can be connected).

If preferred, supplementary scroll keys (located next to the display) can be used, however, the touch method is more direct and the user is quickly reassured that a successful selection has taken place both visually and aurally (the option is highlighted and a small 'beep' can be heard).

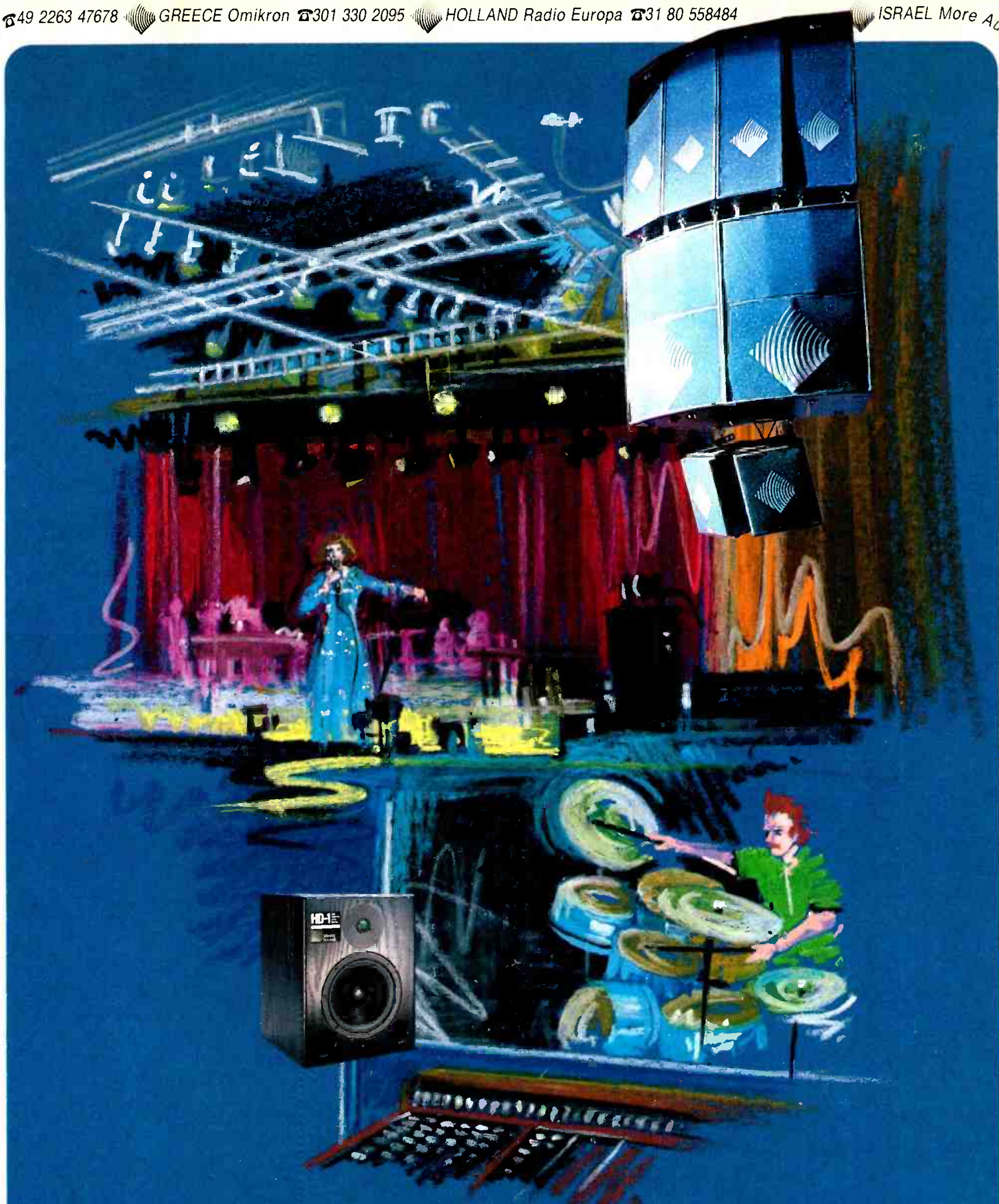
Beneath the display are various edit keys, including IN and OUT keys (for marking in and out points), a GOTO key (which can be used in conjunction with the IN and OUT keys for locating an edit point), CUT, COPY, PASTE, FILL and RIPPLE keys, various keys for event-based editing functions (which is not yet implemented), an UNDO key and a REDO key.

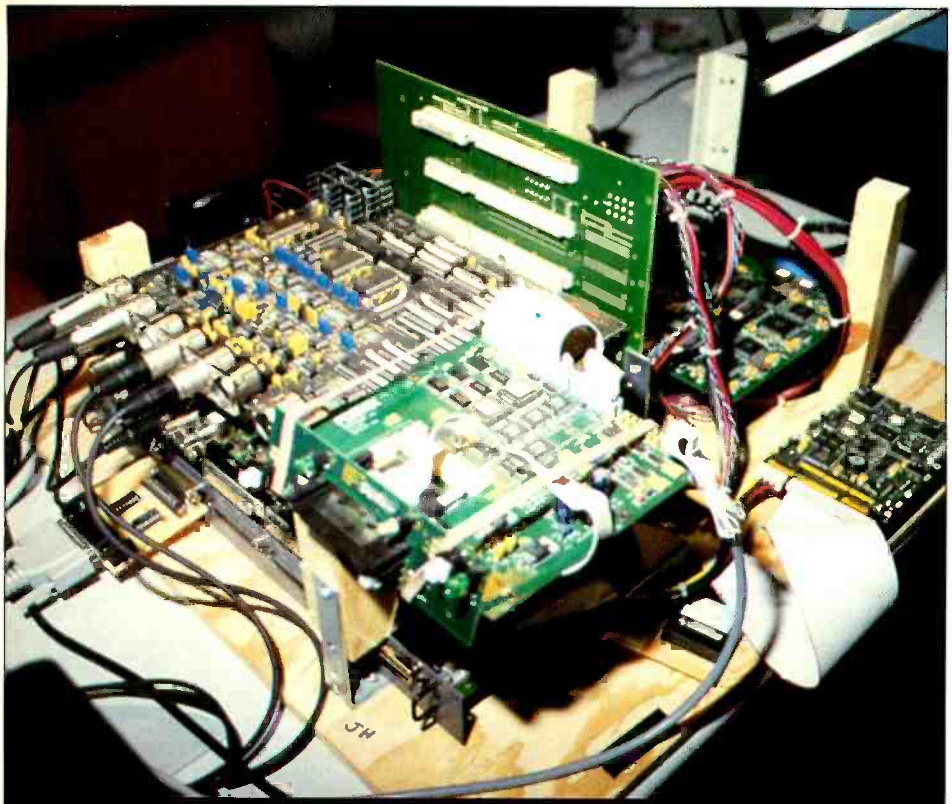
Then there is the multifunction wheel. Apparently there was much discussion and testing before the final design was agreed upon in terms of size, weight and balance. It is a continuously revolving wheel which uses a very high-resolution (256) shaft encoder with an 'intelligent response', such that it ignores accidental knocks. The primary functions of the wheel are motion control and scrubbing (jogging), but it can also be used for shuttling and incrementing-decrementing, and can immediately take over from the transport controls.

The wheel follows different laws, according to the selected mode. In Shuttle mode, one turn produces normal play speed, a second turn produces 2x normal speed, a third 8x and a fourth 32x, with smooth and dynamic transitions between each multiple (an indicator on screen shows what the current speed multiplier is). As expected, play speed can also be controlled in the opposite direction down to 1/32 normal speed. In jog mode, one turn moves 10 frames and in 'super jog', there are 3 frames per turn (subframes can be scrubbed by moving a fraction of a turn).

At the top right of the screen are two 10-digit LED timing displays, the top one shows the current position of play and the bottom one shows a user-defined time for a selected parameter. Timings can be copied from the top display into the bottom one, and there are various associated parameter keys for functions such as offset, preroll, postroll and delay (which, if used in conjunction with auto 'play from' or looping, gives a 'creative' pause before replay commences). Below these are numeric keys, '+' and '-' keys for trimming frames and STORE-RECALL keys for the locator memory (up to ten locations can be stored).

Finally, running along the bottom of the Edit Controller is an arm rest which can be detached if the Edit Controller is to be clipped onto the main unit. In fact, this was demonstrated without the system having to be switched off. The connector was removed and as soon as the Edit Controller was clipped on, it automatically reset to the same condition it was in before it was disconnected. ▶





A prototype *Foundation* seen at FXR

Project management

Fostex' rather elaborate term for a removable drive used by *Foundation 2000* is a *Removable Project Environment* (*RPE*). The reason for this is that the drive not only stores audio, but all other data required to reconstruct a project, including edit data, mix data, DSP data and configuration and project setup data. The *RPE* is not required to store the system's operating software (which is held in *Foundation*'s internal flash ROM) and can be interchanged between different *Foundations*. Every relevant action is auto-saved and if, for example, an *RPE* with a project on it is inserted into a *Foundation*, the system will automatically be reset to the exact state in which the project was left (including the setup and display as it was shown before the *RPE* was removed). In terms of this type of interchange, drives which would normally be left as fixed, are managed in the same way as *RPE*s.

A project consists of up to 99 reels (sessions) and each *RPE* can hold up to 26 projects. Currently, the system will only list those projects which are stored on the inserted *RPE* (or fixed drive), as well the contents of whichever archive device is on line. Preferences can be set up for each reel, or alternatively, a 'default' reel (which only contains preferences, no audio) can be set up for the project. Each new reel created within the project will adopt the preferences of the default reel, and these can then be modified if necessary.

A reel can be copied from one project to another, a project can be duplicated and, as a helpful feature, the system will automatically look at the order of the user's choices and anticipate the next logical step in the copying process. As expected, if a reel or project is duplicated, it is only the relevant data which is copied and not the audio (both original and copy share the same source audio). However, if the original is deleted, its audio will not be deleted if it is used elsewhere.

Operational software

There are five primary software pages, any one of which can each be accessed via its icon which is situated vertically along the right-hand side of the display. Along the bottom of the display are icons representing parameters, which will change according to the page selected. At the top of each page is a display which indicates how much recording time is left on disk, which project has been mounted (selected) and which reel has previously been mounted.

The System page displays details about *Foundation*, including a log of error messages in plain English (which, if accessed further, will display additional data including source code —particularly useful for customer support purposes), the system configuration, time and date, software version and *RPE* details. The user can also activate an Edit Controller test function from this page (which tests all LEDs, buttons and the touch display), format an *RPE* and enter a password for protection purposes.

The Reels page allows the preferences for a reel to be set or modified. These include sampling rate, timing reference, amount of varispeed (up to ±12%) and time-code output offset (an offset for any external device chasing *Foundation*).
The Tracks page is the main recording and editing page, and in this first release of software, provides a multitrack model with eight tracks.

The Mix page displays the routing-patching configuration of the internal mixer. It consists of three subpages, which deal with various input configurations, input level trimming, patching and definition of aux sends and returns. (If appropriate.) Finally, the Meters page displays dynamic, fast response, PPM level meters for each track with peak hold. It also has meters for the main left and right outputs, monitor mix and aux sends and returns, and allows the user to define the meter reference level. ►

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Inside the prototype Edit Controller at FXR

The software under review for this article was v1, beta 1. Software v2 is due for release at NAB.

Recording

In the first release of software, *Foundation 2000* has been designed to operate like a linear multitrack recorder, but with nondestructive cut-and-paste editing. As expected of a multitrack, the system can simultaneously record and replay any combination of eight tracks.

Recording and editing operations take place within a reel, therefore before recording, a project and reel must first be created (or an existing project and reel must be mounted). One of seven sampling rates (32kHz, 44.1kHz, 48kHz and pull-up/pull-down variants of 44.1kHz and 48kHz—in order to correct for NTSC anomalies) can then be selected from the Reels page.

The type of input (analogue or digital) can be chosen from the Mixer page, input levels checked and the input monitored. The Tracks page is then selected and if a new reel has been created, eight empty tracks will appear. The desired destination track(s) is activated using the TRACK SELECT button(s) and the play and record transport controls are then used in the customary manner to begin recording.

During recording (or playback) up to ten marks can be made on the fly, and once recording is stopped, the display is updated and the recording is shown as a solid block, with the begin and end points marked. (Recordings include a 5ms handle at the beginning and end for crossfade purposes.)

If required, the recording can be 'undone', however, this does not initially increase the amount of storage space available. This is because the undo-redo function applies to the last ten relevant record or edit actions, therefore once sufficient actions have been carried out after the recording has been undone, the corresponding disk space will become available (currently, it is not possible to continuously record across multiple disk drives).

Other recording functions include automatic punch-in (a region across the selected track(s) must first be defined), nondestructive punch-in on the fly, auto record within a loop with preroll and postroll and track bouncing (or mixdown) of any combination of tracks not exceeding eight (7:1, 6:2) with real-time digital mixing and EQ.

Editing and mixing

Foundation's editing features are currently limited to track-based instant access multitrack editing, giving the impression of destructive tape-based editing, but with an undo facility. Once a recording has been made, the begin and end points can be instantly located using the GOTO key in conjunction with the IN or OUT key.

Editing operations are straightforward and consist of defining an edit region (using the transport controls and-or wheel and marking with the IN and OUT keys), selecting the track(s) to be edited, activating an edit function (such as cut) and, if appropriate, selecting a destination track (or tracks), a time position and, if appropriate, another edit function (such as paste).

The usual functions are available such as copy, cut and paste, as well as 'fill', which will fill a selected region with the result of a cut or copy (by looping and-or appending if necessary). The cut and paste functions perform a video-style delete and overlay respectively, however if used in conjunction with the RIPPLE key, they perform a film-style cut and insert respectively.

Edits between two sections of audio are automatically given a 10ms crossfade (which cannot currently be adjusted) and are marked for easy location. Fades cannot yet be defined, but can be manually imposed using the internal mixer.

The internal mixer is a 10:2:2 providing left and right outputs and monitor or balance mix. All channels of the mixer include 3-band fully-parametric EQ (including cut, boost, frequency sweep and Q for each band), panning, level fade, mute and two aux sends. Patching allows the left and right mix to be routed to the monitor outputs and vice versa, as well as the aux I-O to be configured (providing the multichannel I-O module is fitted). It should be appreciated that while the mixer appears to accommodate 10 inputs (8 internal tracks plus 2 external inputs), it has been designed to cope with the 16 internal channels. (See 'Design philosophy.')

The mixer is controllable via MIDI (including software-based MIDI mixers such as *Cubase*), with all parameters supported, and was demonstrated being controlled by the Fostex *MIXTAB* control surface. Automation is currently limited to snapshot (100 per reel), although Fostex stress that *Foundation* has been designed for event-based dynamic automation. Furthermore, it was pointed out that while the *MIXTAB* does not take full advantage of *Foundation*'s EQ capabilities, all parameters of the mixer are published for third-party MIDI control, and in any case, a dedicated mix controller is being developed for *Foundation* which will use a high-speed serial link rather than MIDI.

DSP

Foundation's DSP has been designed to be event-based and dynamically allocated, being switchable from one processing algorithm to another within a video frame. The main unit can accommodate another five DSP cards, however, for the time being the first DSP card's role is to support the system's internal mixing capabilities. For the second DSP card, plans are afoot to provide a number of new algorithms such as full dynamics for 16 channels and an expansion of the mixer architecture (making it a 10:8:2 for example). ▶

FXR

Twenty years ago the Fostex Corporation of Japan began manufacturing their own branded speaker products. By 1980, they had begun producing a range of relatively low-cost analogue multitrack recorders, ranging from 4–16 tracks using ¼-inch and ½-inch tape. In 1985 they moved into synchronisation technology and ¼-inch time-code products, and this led to synchronisers such as the *4030*, the first professional time-code DAT machine (the *D20*) and the first professional time-code DAT location recorder (the *PD2*).

In early 1992, the company held a series of strategic planning meetings in order to define the technological future of Fostex. They considered their existing home recording, time code and synchronisation markets as well as the high-end digital tape market, and identified that while there were still opportunities for linear technology for the mid-term, their long-term future would depend on random access and computer technology for audio.

As a bridge to random-access products and following their mid-term plans, Fostex entered the 8-track digital tape-based market by cooperating with Alesis to develop the *RD8*. This is Fostex' version of *ADAT* and includes various professional synchronising features suitable for postproduction. At the same time, they were keen to develop a random-access unit, again by cooperating with a third party, and so the discussions with US-based New England Digital (NED) began.

NED had already had significant experience in random-access technology with their *Synclavier* and direct-to-disk systems, but the discussions with Fostex concerning a possible alliance ground to a halt when NED ran into financial difficulties. However, although NED ceased trading, their research and development team indicated a wish to continue working as a unit and successfully negotiated a deal with Fostex whereby the whole team of 27 people was hired and a new R&D facility (Fostex Research and Development Inc.) was formed in Lebanon, New Hampshire.

The short-term objectives for the new company were to develop a high-end digital audio workstation to retail at around $30,000 US Dollars and to develop core hardware and software for future Fostex products which, in the long-term are to replace Fostex' existing analogue product line. Having defined the specifications for the digital audio workstation in October 1992, the plan was to have a working system by October 1993. This they achieved with the launch at the New York AES show of *Foundation 2000*. ■

Card mock-ups surround the latest design for the *Dancing Fader Module*

Sync and machine control

Foundation has a wide-band time-code reader (the LTC-VITC chip was developed by Fostex in Japan and is used in other Fostex products) which allows it to follow time code in forward and reverse from very low speed to high speed. It will lock within 15 frames and can follow a video scrub, back and forth. It will also flag reference errors or clock conflicts and can use the incoming code as its system clock reference (useful for example, in cases where the external time code and source material are asynchronous in the digital domain).

External machines (such as the *BVU 800*, *BVU 900*, *BVW 75*, industrial S-VHS and Pioneer laserdisc) can be controlled from the Edit Controller. The first release supports control of one machine at a time and this should soon be increased to two. (Although provision has been made for more.) *Foundation* can itself be controlled by external video edit controllers. Furthermore, a special 'twin VTR emulation' feature allows *Foundation* to be split into two virtual 4-track machines and controlled by two separate edit controllers simultaneously.

Archiving

Needless to say, *Foundation* supports background loading and Fostex maintain that this involves 'true' multitasking on a micro level. Either WangDAT or Exabyte SCSI peripherals can be used and loading can be selective in terms of whole RPEs, projects or reels.

One of the drawbacks of disk-based systems has always been the issue of archiving, and while most systems now deal with the problem of archiving all necessary information in order to reconstitute a session, the problem of transfer time still remains. However, there are a number of ways in which this can be tackled, depending on budgets and time management. One solution is to have very large amounts of disk space and then to archive out of hours, but this requires a large initial outlay and the system must be archived on line.

The ideal solution would be to use removable optical disc as both the recording and archiving medium, but due to the cost of media, this could prove a relatively expensive on-going cost.
In addition, optical still does not provide the same performance (in terms of supporting simultaneous channels) as hard disk, so for manufacturers who want their systems to achieve maximum channel capability, there will inevitably be a compromise when it comes to an archiving solution.

The main options are either to use high-speed tape, removable hard-disk drives or a combination of both—the option which Fostex have plumped for. In fact using removable drives serves several purposes—it allows an immediate drive swap, drives can potentially be archived (onto tape or optical disc) off line using a dedicated low-cost 'archive workstation' and work can be quickly transferred to another system in the same way as with tape.

Admittedly, using a tape streamer is the most cost-effective form of archiving, but for applications where archive material is frequently needed, the time involved in transferring can be prohibitive if work has to stop for archiving. Hence the development of background loading while the system is being used. This effectively reduces or even eliminates additional time needed for transferral—the only drawback is that discipline is required in remembering to start the process, anticipating when to start up or down loading and being certain of what should be loaded.

Future developments

A series of major and minor developments have been scheduled for the short, mid and long-term. Most of the short and mid-term developments have been implied throughout this review, however it is worth mentioning that event-based editing and a complementary *Mac*-based user interface are planned before the end of the year, and although waveforms are currently not displayed, the file structure supports embedded waveform storage (at 64 different resolutions) within the audio file.

In addition, a user-track model is envisaged which will allow real-time edits to be traded off for a larger number of usable tracks as, in a film environment for butt-splice track laying, a 16-track workspace could be developed, or possibly 12 tracks with up to 4 tracks being crossfadable.

Conclusion

The basis for the system offered by the *Foundation 2000* to grow is evident, and although (in preproduction) review form there are recording and editing features yet to be implemented, what does it does well.

Foundation is intuitive and easy to use, and its designers have obviously done their research in operational ergonomics and psychology—for example if you do something silly, a message appears with a helpful suggestion. The owners manual is excellent, unpresumptuous and includes a 'getting started' section with tutorials (and unlike in many cases, the service manual is not an afterthought, it is already available and equally informative).

In terms of the competition, Fostex are relative late-comers to the random-access market, but they have already proven their potential by developing a working product in such a short space of time. *Foundation* is competitively priced, but whether or not it matches the price-performance requirements of all of its target markets with its present operational capabilities, remains to be seen. Nonetheless, like the system's name suggests, this is only the beginning. *Foundation 2000* gives the impression of being well-engineered with enormous potential, and if Fostex, with all their resources, continue as they have done so far, the system promises to develop into the recording-editing-mixing workhorse of the future. ■

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FOUNDATION 2000 v2 SOFTWARE FOR NAB RELEASE

Software features to include:

Event-based editing:
- Track display presents track-based and event-based user model
- Align and trim events over underlaying audio and-or handles
- Adjustable logarithmic fades
- Sync point marking for track laying (back timing etc.)
- Event audition—Aud In, Aud Out, Aud In to Aud Out
- Event labelling via screen or ASCII keyboard
- Event label display in event-track screen

Library:
- Named events go to Library
- Library contained within *RPE*
- Library can be searched and edited

Multiple drive expansion:
- Expanded SCSI implementation to format several drives as one indivisible *RPE*

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ULTRA-HARMONY

5G processing—the *DSP4000* is Eventide's fifth-generation *Harmonizer*

Eventide continue the line of development begun in 1975 with the *H910* Harmonizer by launching the *DSP4000*. An eager Patrick Stapley assesses their progress

Empty Swimming Pool, Wooden Men's Room, Wobbly Dobbly, Techno Clank, Zipper Up, and UFO In My Church—just some of the enticing presets supplied with Eventide's new *Ultra-Harmonizer*. Eventide, whose products have successfully and consistently 'harmonised' into studios over the last two decades, have recently released their fifth-generation processor, the *DSP4000,* which supersedes the five-year-old *H3000*.

The *DSP4000 Ultra-Harmonizer* offers a number of new features and enhancements, including a Patch Editor providing a 'build your own algorithm' facility; a memory card for preset storage; a larger LCD; a preset crossfade facility; additional function keys including two user-programmable buttons; and extra rear panel connectors including digital I-Os and RS422 ports. The unit is also now equipped with 18-bit A–D and D–A convertors rather than the more usual 16-bit convertors.

Internal ROM and RAM have been increased and the *DSP4000* comes with over 200 nonerasable factory presets. User presets are stored either in internal RAM (121,816 bytes) or on memory card (128,000 bytes)—the size of a preset will range from 400 to 8,100 bytes. All Presets, whether factory or user are organised into nameable banks, and both the internal and card memories have a total of 100 banks each. A bank can contain up to 128 presets, but, of course, this capacity will be governed by available memory.

The presets themselves are very wide ranging: at one end of the scale are bizarre effects such as 'War With Phaser Guns', while at the other are a selection of extremely high-quality reverbs. In between are programs built around delay, pitch, chorus, distortion, EQ, dynamics and so on. The choice is so wide ranging that one wonders how practical some of the sounds will actually be—the average music studio for instance is going to have little call for a program like 'UFO In My Church' that plays back a loop of the 'Close Encounters' sequence drenched in reverb. For this reason the *H3000* was produced in application-specific versions; however, now with the incorporation of a memory card it is very likely that we will see application-specific cards becoming available to supplement the existing presets. In this respect some of the sounds supplied within the *DSP4000* should be seen as representative of the machine's all-round versatility perhaps more than the program's all-round usefulness.

Another point, which applies equally to other multieffects units, is that with so many presets available (both factory and user), there is a degree of user frustration. For example, I am about to start a mix and I know that within my unit there are three or four sounds that will be perfect: there is a great room sound for the bass drum, a tight gated reverb for the snare, a tap delay I have constructed for the lead vocal, and a classic pitch shift for the brass—but the unit will only allow use of one set of inputs and outputs. The alternatives are either to record the effects to tape, which is not ideal especially as I may already have used up all the spare tracks, or to hire-in additional units. The MIDI program change facility would be of no use in this case as all the sources play across each other.

It strikes me that what is required here is for manufacturers of multieffects processors to produce additional processing expansion modules. These would have dedicated inputs and outputs, be rackmountable and control would be assigned to them from the mother unit, thus allowing multiple presets to run simultaneously and independently. To a degree this is what Lexicon have done with their *480L* which divides internal processing into two discrete 'machines'. With the modular approach, any number of full power processing units could be attached without compromising ▶

stereo operation, turning a single device into a true multieffects system. How about it Eventide?

Operation

The *DSP4000* front panel retains the look of the *H3000* with its 10-segment input meters, four soft keys below the display, large rotary knob, and keypad. Operationally the system also remains familiar, although the increased size of the display allows more information to be shown at one time resulting in improved access to parameters.

As before there are various dedicated function keys or 'Area buttons' that change operating mode, they are:

Program, Setup, Patch, Parameter, and Levels. Selecting one of these assigns a set of associated functions to the four soft keys; for example, selecting SETUP will assign Display, MIDI, segue, and Audio to the four keys providing access to these respective menus. Where additional functions exist for an operating mode, a second press on the AREA button will assign a second set of function-menu boxes to the soft keys. In some cases these boxes will appear overlaid in which case the soft key itself has access to more than one operating menu. Once a menu has been selected, the contents can be scrolled through using the LEFT and RIGHT cursor keys, and parameters changed using the rotary knob or the keypad. Apart from acting as menu selectors, soft keys may execute an immediate operation such as Load or Save.

When installing the unit, the Setup and Levels areas should be checked through to ensure correct operating conditions; these include whether the input is set to source an analogue or digital (AES-EBU or SPDIF) signal, that the correct sampling rate (32kHz, 44.056kHz, 44.1kHz, 48kHz, External) has been set, that the unit is set for Guitar (direct signal-effect mix) or Studio (effect only), that I-O levels are correctly set, the status of the BYPASS switch (DSP Bypass, Relay Bypass or Mute), that the display's contrast and brightness are optimally adjusted, and so on. All setup changes will automatically become the system default, although the original factory configuration may be recalled.

Once the unit has been correctly setup, the Program area can be entered. The display will show Bank 0 and its first three presets. Bank 0 is actually a Utilities store and contains presets such as a white noise generator, oscillator with adjustable waveform and sweep function, and a Thru preset which simply connects inputs to outputs. Actual effects presets are arranged in the next 18 banks categorised under the following headings: Pitch Effects, Delay Effects, Chorus Effects, Ambience, Small Reverbs, Large Reverbs, Alternative Reverbs, Drum Effects, Dynamic Effects, Equalisers, Filter Effects, Simple Effects, Multi Effects, Curiosities. Distortions, Guitar Racks, External Control (preset specifically designed to be modulated by external controllers), and Examples (examples with information on various preset elements). If a memory card has been inserted, its banks will appear after the internal banks prefixed by C.

Banks and presets are separately scrolled through using either the rotary knob or the UP-DOWN controls on the keypad; they may also be numerically accessed from the keypad. Depending on the complexity of the preset, load times can take anything up to 7s; once loaded the unit automatically switches to the parameter area with soft keys providing access to various parameter menus. The menus are all clearly laid out and parameter adjustment is both intuitive and fast; as with the *H3000* certain presets include an Expert menu for the more accomplished user. Edited factory programs are saved as new presets, whereas edited user programs may be either saved as new presets or simply updated. Saved presets can be named by laboriously scrolling through the alphabet and picking out letters, or-and by selecting words from a long descriptive list such as Fat, Thin, Hot, Cold, Hard, Soft, Classical, Grunge, Drum, Guitar, Stereo, Mono, Sexy, Sleazy...

Patch Editor

Apart from modification by adjusting parameters, presets may be more radically edited using the Patch Editor which allows the actual elements that make up the preset to be edited. This facility is not for the faint hearted and Eventide warn in the manual that 'it is not a complete, whizz bang, user friendly, go out and build a big, awesome, destroy-all-planets effect, mind-reading, insta-creativity music maker'. Roughly translated, this means is that the Patch Editor is an extremely complex and powerful facility. It really does put the user in the programming seat for the first time allowing access deep inside the unit—whether editing existing presets or building a new effect from scratch. However, this does not come without a price: time.

I can see few users taking the time to familiarise themselves with the procedure which appears alarmingly daunting in the manual. It may be that a more simplistic tutorial approach would help, but I think the reality is that the Patch Editor will be too complex for most tastes and will only be fully explored by a few brave and persevering souls. However, those who are prepared to stick with it will be rewarded by acquiring a new level of editing skills plus a genuine in-depth understanding of effects processing.

I will endeavour to give a brief and simplified explanation of how the system operates. A preset is made up from modules of which there are a total of 89 available arranged into 14 groups. These groups comprise categories such as Delay, Reverb, Pitch Shift, Mixer, Bridge (converts audio signals to control signals or vice versa), Control Process, Math (perform mathematical functions on audio) and so on. So for example the Pitch Shift group contains six modules—Detune, Diatonic Shift, Frequency Shift, Pitch Shift, Multi-Output Pitch Shifter, and Reverse Shift, while the Math group contains modules such as Add (adds two audio signals), Subtract (subtracts two audio signals), Amplitude Modulator, Quadrature Transformer (provides a 90° phase shift of all frequency components up to 1/8th of the sampling frequency), Audio Comparator, and so on.

When the Patch Editor is first accessed the display will show a block diagram of the modules making up the preset and their audio connections with inputs on the left outputs on the right. The editor allows four operations: Insert a module, Delete a module, Connect modules, and Unplug modules. If a preset is being constructed from scratch, the starting point will be from the 'Empty Program' found in the Utilities bank.

Each module that is inserted will require one or more control knobs and these must be inserted individually. Additionally each knob will require naming (Volume Knob, Delay Knob, Feedback Knob), and must also be placed in a suitable menu along with its correct format (the number of spaces the numerical read out will occupy and the number of digits after the decimal point). It also requires the user to program range, step value, and default value. Once all these parameters have been specified the knob is inserted awaiting connection to the relevant control input on the module.

To make life easier the preset elements can be viewed as Audio Only, Audio and Control, and Control Only. If presets are made up from large numbers of modules the display window will not be big enough to show the entire block diagram, particularly if both audio and control modules are being viewed together. In this case the display can be scrolled vertically and horizontally to reveal hidden areas. The display is actually far from ideal when editing complex presets, and control can also be painfully slow—it should however be possible in the near future to interface the *DSP4000* to a computer for substantially improved operation.

The patch editor also lets users create their own menu structure, assigning a hierarchy of named pages to the soft keys into which parameters will be organised. The menu arrangement is shown in a separate display within the patch editor.

At the back of the manual is a list of all the modules with a description of what they do, plus information on their various inputs and outputs, and minimum-maximum parameter values. To keep track of processing resources, details are also given about the percentage of available processing power each module uses up, although this will sometimes vary as a module's specifications are changed.

Presets

Having created and patented *The Harmonizer* during the early-1970s, Eventide's name became pretty synonymous with pitch changing. However, since the introduction of the *H3000* the company have gained considerable recognition as a multieffects manufacturer, and in particular have impressed many with the excellent standard of their reverb programs. The *DSP4000* carries the reputation forward by offering improved quality and additional stand-alone processing such as EQ and Dynamics.

Presets vary between having stereo inputs and outputs, mono inputs and outputs, mono inputs and stereo outputs, or dual inputs and outputs. Depending on size, two presets may be crossfaded by predetermined fade-in and fade-out times to smoothly merge program changes. The governing factor is that each preset should take up no more than 45% of the unit's processing power, and suitable presets will be marked in the program list with a crossfade symbol. Presets that exceed the processing limit—and these make up the majority of factory programs—may still be 'Segued' to avoid abrupt changeovers, in which case fade times will still apply although there will be no overlap.

As mentioned, the factory presets are arranged into 19 program banks and I will run through most of these to give a flavour for what is on offer, starting with Pitch.

The Pitch Effects bank contains 19 presets which include the Diatonic Shifts first introduced on the *H3000*. These presets intelligently alter pitch by recognising the incoming note and changing it relative to the selected key and harmony interval. Success very much depends on the source and the degree of shift being applied, and the results can sound very electronic or unnatural. Diatonic presets tend to work better on synths and guitars rather than acoustic instruments or vocals unless one is looking for a specific effect or can sufficiently mask the Diatonic output in the track.

Other pitch presets include Dual *910*s which recreates the sound of two of Eventide's original *Harmonizer*s with one pitched up and the other ▶

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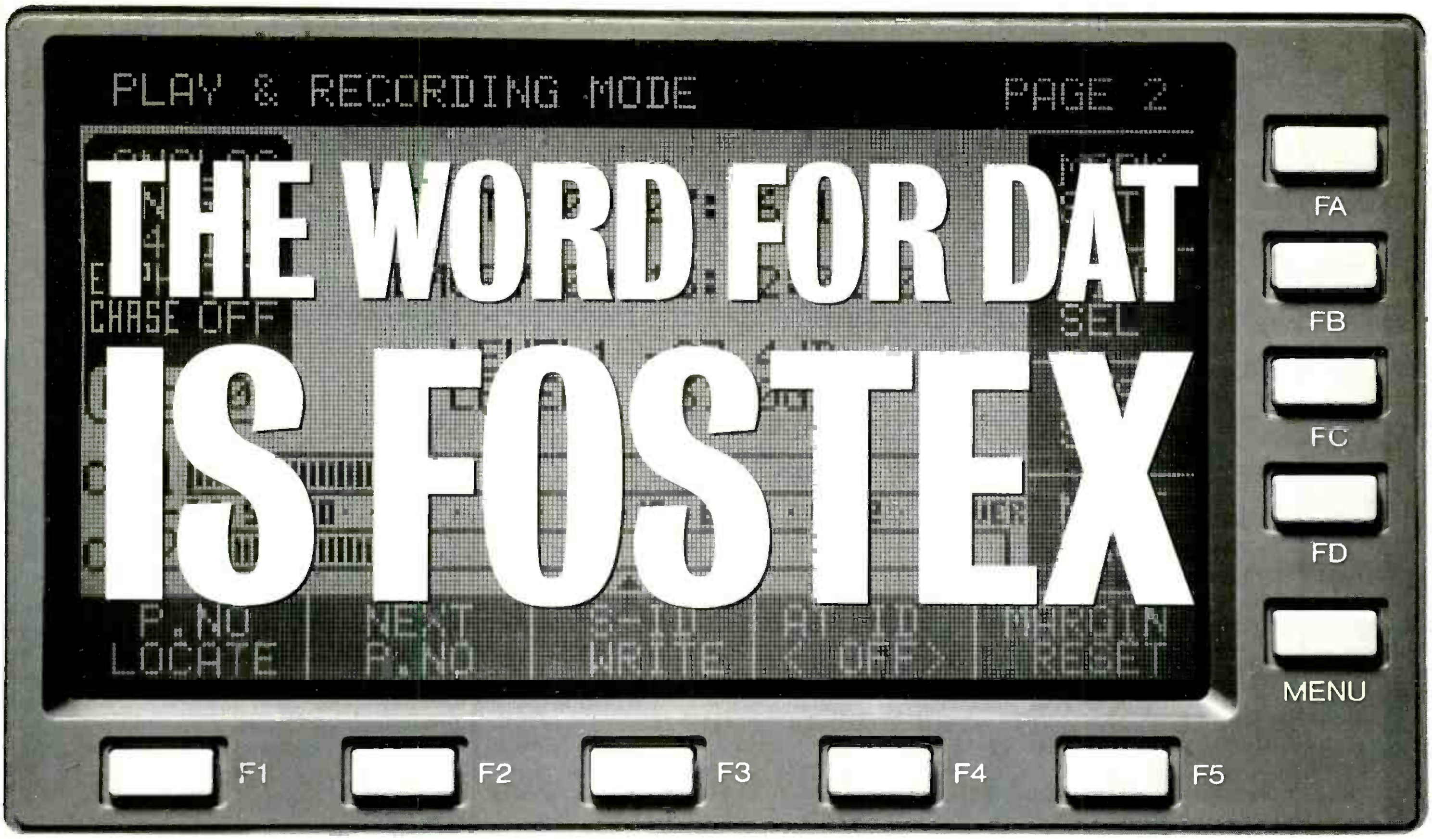

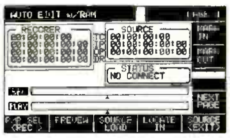

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FOSTEX (UK) Ltd. Jackson Way, Great Western Industrial Park, Southall, Middlesex UB2 4SA pitched down; Shifter Delay that outputs four delayed and pitch shifted voices; Dubbler which uses four micro-pitch shifts for ADT effects; and Poly Shift which breaks the signal into comb bands and shifts each to produce a detuned chorus effect being useful as a thickener. Pitch can be shifted over a ±3-octave range, and up to eight simultaneous voices are available.

The Delay bank has nine presets which apart from standard presets include various multitap programs; Panning Delays which has four delay lines with pan and delay time controlled by separate LFOs; Precision Delay which allows adjustments in microseconds; and a Dual Ducked Delay that only operates when the input level falls below a set threshold. The maximum delay the unit will permit is ten seconds.

There are eight Chorus presets which offer a selection of chorus, flange and phased sounds including a good simulation of manual tape phasing which is controlled from the rotary knob. There is also a Lezlie Simulator with separate control over the two speakers—this preset appears again in the External Control bank where the rotation speed of the 'Lezlie' speakers can be switched externally.

Reverb presets are organised across five banks. The first three divide presets into Ambience, Small Reverbs and Large Reverbs offering a full range of reverberant spaces from Pantry to the diffusor constructed Black Hole. There are some excellent presets here and a lot to choose from.

Reverb effects are stored under Alternative Reverbs and include Zipper Up which is a long backwards reverb that can generate some interesting percussive effects; Swept Verb that produces a very effective phased reverb; Stereo Caves which is designed to give the impression of two caves (a small one on the left and a huge one on the right); and Ghost Bloom which is a delayed inverse ambience with considerable depth. More reverb presets appear under Drum Effects including Drum Chamber, Small Gated Room, Basement Drums, and the off-the-wall Beatbox Reverb which, being constructed from filters configured to produce vowel sounds, seems to speak.

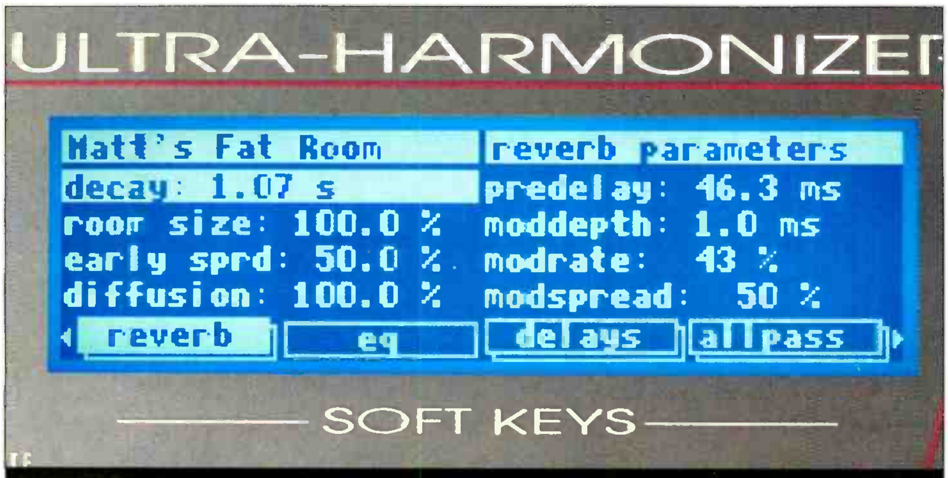
Dynamics presets are mainly connected with compression although there is a Dual Gate and an Auto-Panner (responds dynamically to input). Six types of compressor are available including a basic digital predictive device that is able to respond to signal before it arrives; a stereo compressor created from four separate compressors (two per channel) allowing a knee function; a stereo 2-band compressor that splits each channel into high and low-frequency bands processing each separately; and a Top 40 Compressor that simulates broadcast limiting with a brick wall response. Some users may find the lack of gain reduction metering disconcerting and some form of dynamic display might be a useful addition—alternatively the input bargraphs could be set to read gain reduction.

The Equaliser presets are organised into seven banks. All are stereo with six fully-adjustable parametric bands apart from one that offers eight bands in mono. The stereo presets differ in their arrangement of centre frequencies (although all can, of course, be changed) and some have their first and last bands configured to a shelf curve. The main display mimics a graphic equaliser with vertical level sliders (controlled from the rotary knob) for each band.

Most of the remaining banks contain effects-type programs with the Multi Effects bank offering presets such as Moon Solo that combines pitch shift, phasing, chorus and delay; the Curiosities bank which is home to an underwater reverb sound called Watersized, Duck Soup a dynamically-controlled swept pitch with delays that gives the effect of helicopter rotor blades, the wonderful UFO In My Church, Dr Who Diatonic 6ths, and the curiously named We Know BeatBox TrtMe which the manual flatteringly describes as sounding something between a choir and a washing machine. There are a number of distortion presets designed to emulate transistor and valve amp distortion as well as more involved effects such as Crunchy which produces fuzz followed by chorus and a gate. A bank is devoted to Guitar Racks with various presets built around pitch, delay, chorus and phasing; this also includes an old style octave splitter which outputs a square wave an octave below the source—like the original boxes, this was not particularly successful on all signals. Other effects suited to guitar can be found in the External Control bank including Wahwah and the Lezlie Simulator described earlier. Other externally controlled presets are a chorus with two controllers, one for detune the other for tightness, and a tap echo where the number of echoes and their delay times are externally controlled.

MIDI and external control

As with the *H3000*, the *DSP4000* offers extensive MIDI capabilities. These include MIDI program change which will function for programs within the current bank. If programs are spread across a number of banks it will be necessary to reorganise them on a new bank on the memory card.
The *DSP4000* will output system exclusive to a MIDI sequencer allowing parameter changes to be



***DSP4000* display—soon to be augmented with an interface to assist patch editing**

dynamically recorded and played back; if the sequencer is sending and receiving at the same time, any additional changes will also be recorded. MIDI data dumps of the currently selected preset or the system setup may be sent to a sequencer, or another *DSP4000* whose Device ID matches the send unit.

MIDI and External Control signals may be used to trigger Next Program, Previous program, and Bypass functions as well as triggering events in a preset which contains an External Trigger module. Similarly external devices can dynamically control output levels, the percentage of dry to mixed signal, and preset parameters.

A feature of the *DSP4000* is its ability to simply redirect external control signals. This is achieved by including modulation (Mod 1 to 4) and trigger (Trig 1 and 2) Redirectors in the Setup Menu. These have assigned to them a modulation or trigger source which can be either a MIDI controller or an external controller such as a foot pedal or switch. A Redirector can then be assigned to control a program or setup parameter rather than directly connecting the external source. The advantage of this is that if controllers are changed or the unit is taken to a different location, the Redirectors can simply be repatched rather than having to individually change every program that relies on external control.

Conclusion

The *DSP4000 Ultra-Harmonizer* offers a number of enhancements over the *H3000* including better operation, 18-bit rather than 16-bit convertors, full 24-bit digital I-Os, memory card, and a larger range of preset types. The inclu block' patch editor, although pr unprecedented level of control, users due to its complexity. Th hampered by the confines of display slow access time, but future interface to a computer should help here.

The presets supplied with the unit are very broad ranging and certainly show off the versatility of the *DSP4000*. However, with the memory card facility it will be possible to supply collections of presets that are more application specific to supplement the existing sounds.

As it stands, the *DSP4000* offers a wide choice of high quality presets, including outstanding reverbs, and provides the user with the means of creating some extraordinary effects. Fans of the *H3000* will not be disappointed. ■

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PATRICK STAPLEY began his career in pro audio in 1972 at London's Abbey Road Studios where he worked with artists as diverse as Paul McCartney, The Damned and Matumbi, and was involved in quadraphonic remixes of Tubular *Bells* and *Dark Side of the Moon*. Patrick also ran his own production company and worked as Falconer Studios' Production Manager before beginning writing for *Studio Sound* in 1985.

KEEP YOUR EYES OPEN

Drawing on their experience in TV and video, Tektronix have released test equipment for pro audio. Here Lionel Durant and Jeffery Noah discuss problem areas of digital audio and the value of mixed-signal testing

As the recording industry turns to digital tools for the enhanced control, quality and productivity they offer, a new set of maintenance requirements emerges. Gone are the days when an analogue audio test set provided all the functionality required to keep a world-class facility running on top form.

Ultimately, all audio ends up as an analogue signal but today, as audio spends an ever-increasing amount of time in the digital domain, test devices must offer access to the digital domain as well. Thus the requirement for 'mixed-signal' testing emerges.

Mixed-signal testing refers simply to the need for recording facilities with both analogue and digital production tools to perform measurements not only on familiar analogue parameters, but also on parameters and characteristics unique to digital audio signals.

Having realised the industry's need for mixed-signal testing, equipment manufacturers now offer single-instrument solutions that provide the combined set of digital and analogue tools necessary for maintaining cutting-edge facilities and those in the process of adding digital tools. Test sets designed for this mixed-signal environment supply not only the required battery of tests but serendipitously provide improvements in operating efficiencies over the previous generation of analogue test devices. (More on this later.)

When the first piece of digital equipment arrives, a studio does not necessarily become a mixed-signal operation. With the quality control and high level of customer support offered by most manufacturers today, the chances are that a single piece of digital hardware in an otherwise analogue facility can successfully be treated as another analogue device. However, at the point where two or more digital pieces of equipment are interconnected within a studio, the need for mixed-signal testing becomes a reality.

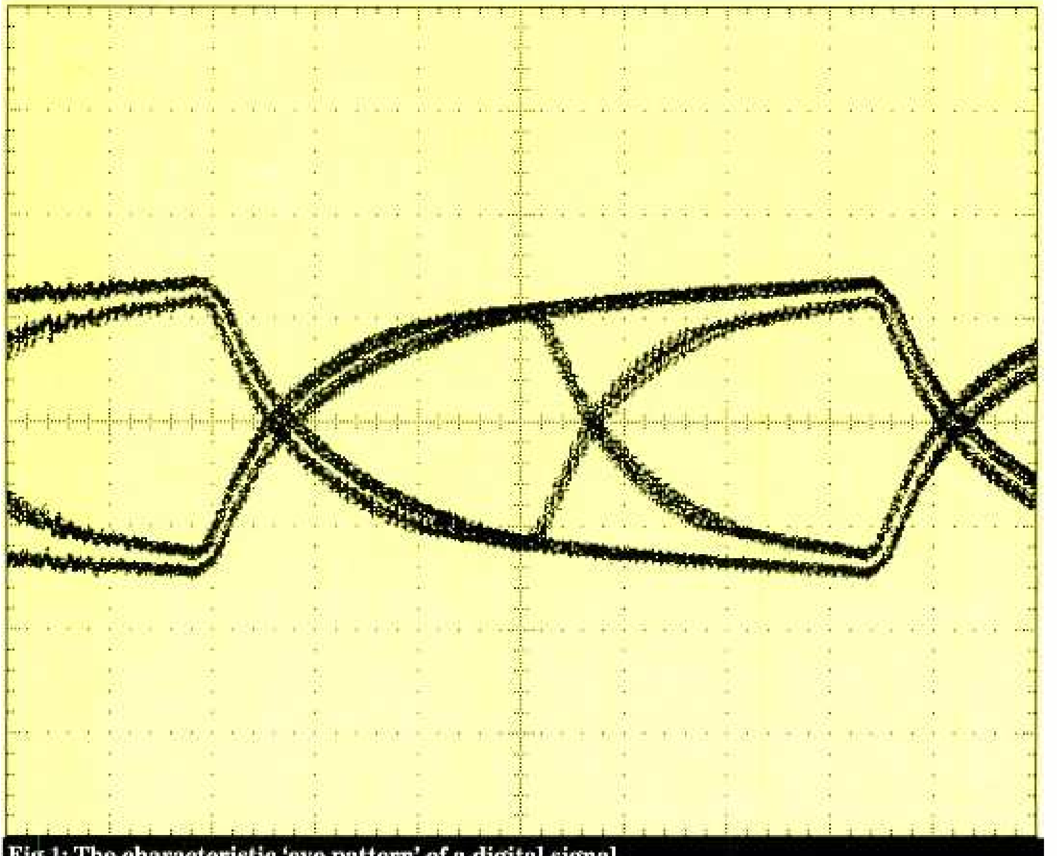

Fig.1: The characteristic 'eye pattern' of a digital signal

Alternatives

Testing a mixture of analogue and digital production tools with analogue measurement equipment can be done but in only one way—by testing the analogue devices as always, and feeding the digital signals into high-quality D–A convertors so that the resultant analogue signals can also be tested. While this method may suffice as a simple quality control test, any problems arising in the digital domain are likely to remain unresolved. Many interfacing problems in digital equipment cannot be solved—or even narrowed down—by use of even the best analogue test devices.

Improved testing

The increasing use of digital audio equipment has not eliminated the need to make analogue measurements. In fact, analogue measurements will probably be with us for many years to come. Traditional maintenance measurements—such as those of frequency response, distortion and level —remain critical to the optimal operation of any recording or production facility.

As alluded to earlier, the processing power and computer control that mixed-signal test sets must possess in order to capture and analyse digital signals yields the unexpected benefit of speeding up and-or automating the execution of analogue tests. Along with mixed-signal test sets, digital-only test sets often employ DSP (Digital Signal Processing) in their operation. These ▶

test sets typically digitise an analogue signal, use an FFT (Fast Fourier Transform) to convert the time-domain audio information into the frequency-domain, and perform measurements on the data stored in the FFT record. The advantages of applying this measurement approach are many.

One advantage is the measurement speed of DSP. While analogue measurement sets must capture the signal for each new measurement, digital sets can make many measurements from one FFT record. Additionally, settling times associated with some measurements add further delays when using an analogue test set.

Automation of mixed-signal test sets can be achieved using programs running on 'outboard' computers via GPIB or RS232 interfaces. Remote control is available by the same means. For those test sets not requiring an external computer, programs can be stored in nonvolatile memory and recalled using simple, front-panel keystrokes or by remote control. As with many personal computers programs (including some for music sequencing), these functions are a type of macro recording and playback feature which capture a series of keystrokes and commands entered via the instrument's front panel. Subsequently one function key can call many others, offering the power of complex programming environments without requiring the use of a high-level programming language.

Since measurements in a mixed-signal test set are performed on a digital record of the audio software by software routines, the addition of new capabilities does not require the fitting of hardware modules. Installing new measurement applications or noise weighting filters requires only a software or firmware upgrade.

An additional benefit of using an FFT as the basis of measurement is a saving in hardware as the frequency-domain information provided by FFTs is easily converted into into a spectral display. With adequate processing power and carefully-written algorithms, the spectral display can be updated at something approaching real time, providing much of the functionality of a dedicated spectrum analyser.

Digital testing

Valuable as they are in testing analogue equipment, it is with digital equipment that mixed-signal test sets are invaluable. In addition to such 'familiar' analogue problems as mismatched impedances, levels and polarities, a number of new and unique potential conflicts exist when dealing with digital audio equipment. Troubleshooting digital interconnections requires more than conversion of the signal to the analogue domain and monitoring for measurable or audible impairments. Whether a digital audio workstation, hard-disk recorder, DAT deck or CD-R machine is the first digital device you press into service, solving almost any technical problem requires a digital analyser—even when utilising the analogue inputs and outputs—and restoring operation to a facility full of new and possibly unfamiliar gear can be a traumatic experience without the tools to fully analyse the digital signal.

It is the flexibility of the AES-EBU format which leads to possible mismatches between the transmitted data and what is being expected by the receiving equipment. Characteristics of data being transmitted—such as the presence or absence of emphasis processing, the length of the digital word, sampling rate and type of data (audio or nonaudio)—are all indicated by bits in the channel-status block. If the channel-status block is not correctly interpreted by the receiving device or does not match what the receiving equipment expects, the results may be catastrophic. For example, although the data mismatch may be as simple as 20-bit data being indicated when 24 bits are expected, some equipment simply refuses to accept a digital audio signal if any data in the status block are not as expected. If the signal were to be accepted, however, the mismatch would hardly be noticeable.

A display showing bit activity of the signal would indicate how many bits were being used; the same display could also determine whether a signal were very quiet or actually digital silence (all bits set to zero).

Potential problem continue in that, for the AES-EBU standard, several preferred sampling rates exist. If two interconnected devices are not set to the same sampling rate, they will at worst fail to exchange any data—at best, serious corruption will be evident.

To resolve these potential challenges quickly, a mixed-signal test set can be used to look at all channel status bits being sent along with the the audio data. In addition to trouble shooting equipment incompatibilities, a quick examination of data rates during an incoming QC check can avoid confusion (or worse) when a digital audio tape does not play.

Analogue problems with digital signals

Problems analogue parameters of digital signals —amplitude, noise and synchronisation problems —can also lead to distortions of the audio signal ▶

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it carries or render the signal useless. An 'eye pattern' display (**Fig.1**) overlays multiple sweeps of the digital signal and provides a wealth of information about these parameters.

Attenuation of the digital signal—easily seen in the eye display—can be tolerated to a great degree before any transmission errors occur. However, any distortion that closes the eye-opening beyond a threshold level can lead to bit errors. One of the major sources of signal attenuation is excessive cable length, which is typically accompanied by high-frequency roll-off. Poor connections at patch bays and multiple terminations are other potential sources of attenuation.

When signals are attenuated below the recovery threshold of a receiver, bit errors begin to occur since the receiver is unable to reliably reconstruct the series of ones and zeros in the transmitted signal. If data bits are not correctly recovered, the probability of audio degradation is high.

The most narrow pulses in the signal will not only suffer from resistive attenuation but will suffer significantly from high-frequency roll-off. In digital audio, those pulses are in the preamble, which must be correctly interpreted in order for a receiver to lock to an incoming signal. The receiving equipment may provide no greater indication of the problem than a status light indicating bit errors, or that it is unable to lock to the incoming signal. A mixed-signal tester makes it easy to pinpoint the location of such an error. An eye pattern display would first suggest that the amplitude of the signal is below the receiver's threshold level.

Phase jitter closes the eye-opening in the horizontal direction. It almost always accompanies the attenuation resulting from a long cable run, and in this case, the jitter often has frequency components related to the programme material. Other sources of jitter could include that inherent in the transport of the transmitting device, power supply-related interference at 50/60Hz and interference around 31kHz from PC monitors.

Whatever the source, jitter closes the eye-opening and, therefore, can result in bit errors. Again, an eye display clearly illustrates the effects of jitter on signal transitions. After determining that jitter is present in a signal, a mixed-signal test set offering a jitter spectrum display would help in isolating the source of the jitter.

If a small amount of jitter (not enough to create bit errors) finds its way into the word clock of the receiver, audio reconstruction will be correct with respect to the ones and zeros. However, word-clock jitter results in timing variations in the analogue signal, since it is the word clock that controls the rate at which data are loaded into a D–A convertor. The resulting sound will be slightly different from the original. To preserve the highest sonic purity, jitter in the signal must be minimised at the point of analogue reconstruction. But to work on eliminating jitter, one must first be able to observe and analyse the phenomenon.

Quality control checks of incoming equipment also create a need for a device capable of providing an eye pattern display. The EBU 'Tech 3250' and AES 'AES3' documents include specifications for receiver performance based on the eye pattern—in both cases a receiver must be able to correctly decode a digital signal that possesses an eye-opening of specific dimensions in time and amplitude.

System timing

The design of many larger digital audio facilities includes the use of digital audio reference signal (DARS) to lock all the equipment to a common stable timing reference. Utilising DARS provides the benefit of ensuring constant phase and frequency between several digital audio signals. Distributing DARS in a 'star' configuration prevents the accumulation of jitter and phase offset which can otherwise occur in a 'daisy chain' of equipment. When the lack of proper phasing is suspected (probably due to occasional pops or unreadable signals), some method of measuring the phase relationship between the signals must be available.

Mixed-signal test sets supplying a read-out of phase or timing differences between signals provides the insight into timing relationships necessary in such a digital audio environment.

Conclusion

The topics covered in this article go some way towards demonstrating the increasing importance of mixed-signal testing facilities in a digital audio facility—even one in which only a few digital devices are employed. The advantages in speed, automation and repeatability of testing such testers bring to analogue testing will reduce the time presently spent in 'traditional' quality monitoring and problem solving. But more importantly, any of the new, uniquely digital problems are likely to remain a mystery without one.

Given existing industry trends, it is inevitable that almost every studio will find itself using digital production equipment. Already, an audio test set offering the advantages of combined digital and analogue signal analysis can play a very important behind-the-scenes role in the success of such studios. ■

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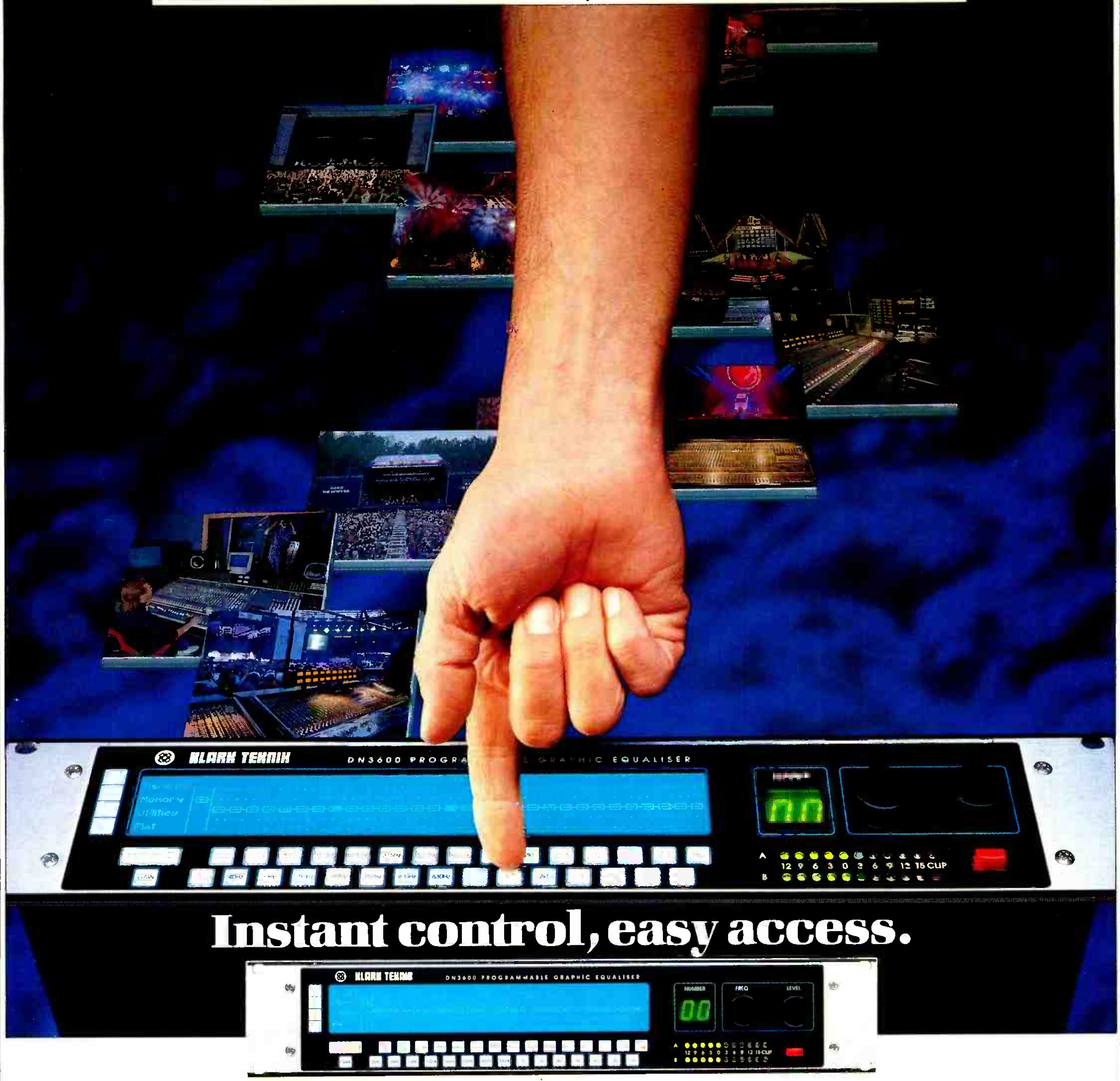

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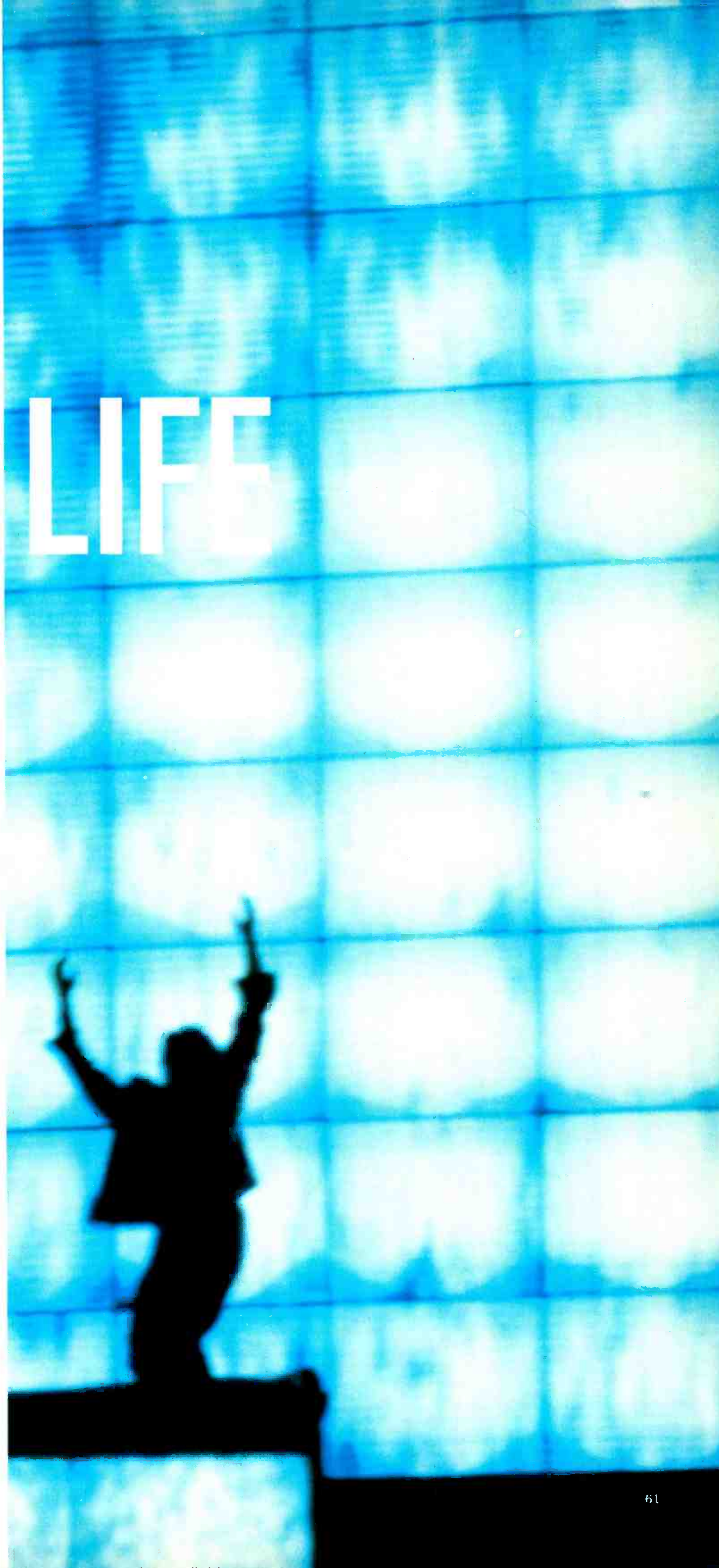

December saw the world's biggest band end their most ambitious tour in Tokyo. Kevin Hilton reports on the challenges encountered

U2 are arguably the 'biggest' band in the world. They have achieved this status through a string of hit albums (*The Unforgettable Fire*, *The Joshua Tree*, *Achtung Baby*) and a gruelling touring schedule made up of emotion-filled shows that engage and involve the audience. All this in under ten years.

The band were renowned for their pared-down, no-nonsense rock 'n' roll, based around drums, bass, guitar and voice. After the *Joshua Tree* tour, Bono, The Edge, Adam Clayton and Larry Mullen said they were going to reconsider their approach. The results were *Achtung Baby* and the subsequent *Zoo TV* tour—bigger, louder and more involved than anything they had done before.

But that was just a try-out. *Zoo TV* played large arenas, bombarding the crowds with music and video images. The *Zoo TV Outside Broadcasts* were the next logical step. The next *illogical* step was *Zooropa*, last year's round of stadium dates, which was hailed as the biggest tour of all time, not only in terms of the number of shows but in the sheer amount of equipment currently on the road.

The production called for intricate planning as two giant steel stages leapfrogged each other from venue to venue. In the venues the main sound rig towered up behind the band, video screens showed enhancing images and satellite links and, hidden beneath the stage, the 'Underworld' was the technological heart of the whole show.

There was nothing easy about the production. The stage position of the band, directly in front of the main rig, puts the sound engineers at a ▶

PHOTOS: PETER STONE

disadvantage from the outset. This called for adaption of existing equipment and the implementation of new technology that was brought out of its cosy development lab to go on the road immediately.

Live sound rental company Clair Brothers have been associated with U2 since the early days and have seen the 4-piece grow to their present position as the biggest band in the world. The company supplied 140 units of its long-established *S4 Series II* loudspeaker (each of which has its own in-built time alignment system) front of house, configured in 2 stacks of 75, left and right, arranged in 3 sets of speakers, 10 high and 7 wide. This enabled the venues to be divided into three distinct areas, foreground, centre and back, each block having its own equalisation setting, programmed into 11 tc electronic *1128* moving-fader graphic equalisers.

Each stadium was 'noised' every morning, usually around 10.00am, using pink noise generators and analysers. After this a tape of the previous night's show was played through the system to see how the building responded and how it affected the natural sound. This gave front-of-house mixer Joe O'Herlihy a foundation on which to work, although, as he notes, an outdoor show is always susceptible to constant change, not least from wind, rain or humidity.

The system was flown from a steel construction in such a way that it looks like one solid mass of boxes. In reality each section was uncoupled from the other, which cut out the problem of bass resonance. Although in technical terms the *S4* has not changed much over the years, its physical shape has. The current unit came from a prototype designed and developed for *Zoo TV*, bringing forward work being carried out by Roy Clair at the company's Lititz headquarters. O'Herlihy comments: 'Three levels of decking is very labour intensive. It wouldn't have been production efficient if the system wasn't stacked.'

It is estimated that if *Zooropa* had been using the previous system, setup time would be anything between 12 to 15 hours. However, it took the crew just four hours to raise the main structure —demonstrably more production efficient. This was just as well because, while there were two versions of the stage, which allowed one to be in transit while the other was in use, there was only one sound rig and getting it broken down and settting up in reasonable time was imperative.

The front-of-house system was expanded by 16 *Servo Drive* sub-bass units and a perimeter system, which was installed under lead singer Bono's position at the front of the main stage and around the edge of the secondary stage B, some 100 feet away, linked by a catwalk. This was the brand new *P4* loudspeaker, a trapezoidal box which carries the prefix 'P' because O'Herlihy, who was involved in its development, thought its operation had a drive-like quality and so nicknamed it the 'Piston' box.

In real terms the whole sound production appears to be working completely against the recognised laws of physics. The *S4* stack was 45 feet behind the main vocal position and some 38 feet behind the drum riser. This made for an open-face production of approximately a 250° spread, which can be seen from anywhere in a stadium, allowing the promoter to sell seats all the way to the back of the venue in the safe knowledge that everyone has a clear sight-line to the stage. The acoustical problem resulting from this was that the sound was not focused.

Four arrays of *P4*s, with six cabinets per array used to form a semicircle, which focused the sound. By implementing psychoacoustic principles, it is possible to convince the audience that Bono's voice was indeed coming from him and not from a point several feet behind him. This procedure is repeated on the B stage where the speakers sat on their sides on the edge of the dais, made possible by their trapezoidal design. The *P4* has its own control circuitry and its own graphic equalisation.

The efficiency of the system meant that there was no need for delay towers, which are a common sight on outdoor shows and are generally thought to be standard equipment. Only one gig on the whole tour needed delays but this was due to the physical nature of the venue rather than anything else. Roundhay Park in Leeds has a 90,000-person capacity and is an elongated shape, with a dog-leg. Birmingham-based SSE Hire was brought in to supply delay equipment on this one occasion.

This extremely high level of production was not reached by accident, it is very much a matter of design and ambition.

'When we were putting the whole *Zoo TV* concept together,' O'Herlihy explains, 'there were certain values to be addressed. The US tour of arenas, which was our baptism, was a standard production. When we did the *Zoo TV Outside Broadcast* shows the flying technology had to be ▶

ready, which was in July of 1992. With the indoor shows the original idea was taken to its optimum effect—we couldn't take it any further. But outdoors, with an open field, we could do anything.'

Because of its massive scale, it was impossible not to notice the front-of-house system. Less obvious but probably even more important in this situation is the monitoring system. And because the band played in front of the main stack, it was imperative that the musicians hear what they want to hear in order to produce a good performance. This became all the more tricky on stage B, some 150 feet in front of the wall of sound. This introduced all sorts of time-alignment problems, not least of which is the sound being on a 150ms lag by the time it reaches a player.

The solution was the Garwood Communications *Radio Station* in-ear monitoring system, worn by each member of the band, all of whom had their ear-pieces moulded by an audiologist to be certain of directing as much signal as possible into their ears. O'Herlihy describes the *Radio Station* as 'an integral part' of the production, as it allowed the musicians to play on stage B without delay problems. 'The reason the whole thing worked is because of the in-ear monitoring,' O'Herlihy states.

However, conventional monitoring was also a priority, with four discreet positions around the stage. Larry Mullen's drum riser used a combination of Clair Brothers' *12AM* wedges and the *ML18* sub-bass units. Bassist Adam Clayton, guitarist The Edge and Bono had dedicated monitor setups and further monitoring positions at the end of each side walkway plus *ML18*s and the *MT4* mid-high units as side-fills stage left and right. As a musician moved away from his main monitors, the signal 'followed' him by means of a digital routing matrix controlled by a joystick.

The usually fraught job of the monitor engineer was made all the harder by the monitor mix position being located beneath the stage in the 'Underworld'. 'Monitor beach'—as it is affectionately known by the crew—housed four Ramsa *5840* consoles, controlled by Dave Skaaf (mixing for Clayton and Mullen) and Vish Wadi (Bono and The Edge), who also have two assistants. The traditional tool of eye contact is replaced by video cameras and monitors, which allow the engineers to see the players in all possible stage positions.

Due to the high degree of mobility exercised by the band during the show, wireless technology is to the fore. A total of 22 dedicated frequencies were used for *Zooropa* and from the outset the crew were not going to take any chances with interference.

'We went into this in depth,' says O'Herlihy. 'Because of the amount of technology that uses wireless we had to have prededicated frequencies, so we went to the licensing authority and purchased the frequencies that we needed. It was essential that we did not have interference so we did the right thing and licensed the frequencies. We didn't have a single problem.'

In addition to the feeds coming off stage, ►

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O'Herlihy had to deal with live satellite uplinks and downlinks, which included a transmission to and from Sarajevo each night. The satellite feed was brought in via a regular dish. The show starts by using the massive video screens on either side of the stage, which require eight stereo feeds as the audio shifts in keeping with the motion of the image. Then there was Bono's nightly phone call, which is brought through the main system by a teleconferencing device, equalised through the tc moving-fader system. Callees included Bill Clinton, Salman Rushdie and a confused pizza parlour in the US which got an order for 10,000 pizzas which were handed out to the audience from the front of the stage.

All this required a total of 128 channels front of house, which were brought through ATI *Paragon Series* and Clair Brothers *CBA* consoles. The ATI is a 40-input board but becomes a 56-channel mainframe by the use of its 16 monitor subgroups. The *CBA* is a 32-input model with 12 returns also used as inputs; 24 effects returns were used for treatments. A standard Clair Brothers control system controlled all processing, in conjunction with a coherent transfer system.

'That's the word for crossovers these days,' O'Herlihy smiles. All treatments and effects were MIDI interfaced through a preprogrammed *Sycologic* system, although there is a manual mode should the band decide to change anything around.

One thing that becomes clear when looking at the *Zooropa* production is that it was not designed by a commercial brain but was the creation of technicians and people with an eye for spectacle. *Zooropa* brought the rock 'n' roll show into the 1990s but whether anyone follows U2 and does it again is doubtful. Joe O'Herlihy's description of the whole shebang as 'Commercially, probably the costliest tour of all time' may just put some people off trying it for themselves. ■

OPERATING LEVELS IN DIGITAL AUDIO

Analogue and digital recorders both record audio waveforms—but there are significant differences in the way that signal levels are established and in the way that overloading is handled. John Watkinson goes back to fundamentals to show the logical choices for digital operating levels

Analogue tape recording gets its name because the intensity of magnetism on the tape is an analogue of the voltage of the audio waveform supplied to the record head. The quality of an analogue magnetic recording is a direct function of what happens between the heads and the tape—tape noise becomes audio noise, tape saturation becomes audio nonlinearity. As a result, the amount of noise or distortion present in an analogue replay is a function of the amplitude of the recorded waveform with respect to the signal range that the tape can handle. In a well-engineered analogue recorder, impairments due to the circuitry are negligible compared to those from the tape-head system.

In contrast, digital recording removes the quality connection with the medium (**Fig.1**). Unless the medium is working so badly that uncorrectable errors are suffered, the replay process returns identical numbers to those which were recorded. If the replayed data are identical to those recorded, the medium has caused no loss of quality and the quality is determined in the convertors. The result is the opposite of the analogue case. Here the impairments are determined in the circuitry and the tape is substantially transparent.

The analogue-to-digital convertor sets a theoretical limit on the sound quality available in the whole machine. The bandwidth is limited by the sampling rate and the maximum signal-to-noise ratio is determined by the wordlength. Bandwidth has little to do with level considerations, so I will not pursue that direction any further. The wordlength of samples puts an absolute limit on the noise performance which an ideal A–D convertor can reach. If the ideal A–D is followed by an ideal D–A convertor, the D–A does not cause any further loss of quality. In practice A–D and D–A convertors are less than ideal, but the actual quality achieved by modern, well-engineered convertors comes close to the theoretical limits.

An A–D convertor has an input voltage range equal to the sum of all of the quantising intervals. Analogue audio signals can be positive or negative, and their long term average voltage is zero. In order to accommodate a bipolar audio signal, an offset of half the quantising range has to be added (**Fig.2**). In practice, a pure binary output from the convertor cannot be used for processing because the analogue offset results in a numerical offset in the output. Instead, all digital audio equipment makes use of two's complement coding.

In the two's complement system, the upper half of the pure binary number range has been redefined to represent negative quantities. If a pure binary counter is constantly incremented and allowed to overflow, it will produce all the numbers in the range permitted by the number of available bits, and these are shown for a 4-bit example drawn around the circle in **Fig.3**. As a circle has no real beginning, it is possible to consider it to start wherever it is convenient. In two's complement, the quantising range represented by the circle of numbers does not start at zero, but starts on the diametrically opposite side of the circle. Zero is midrange, and all numbers with the MSB (most significant bit) set are considered negative. The MSB is thus the equivalent of a sign bit where '1' signifies minus. Two's complement notation differs from pure binary in that the most significant bit is inverted in order to achieve the half circle ►

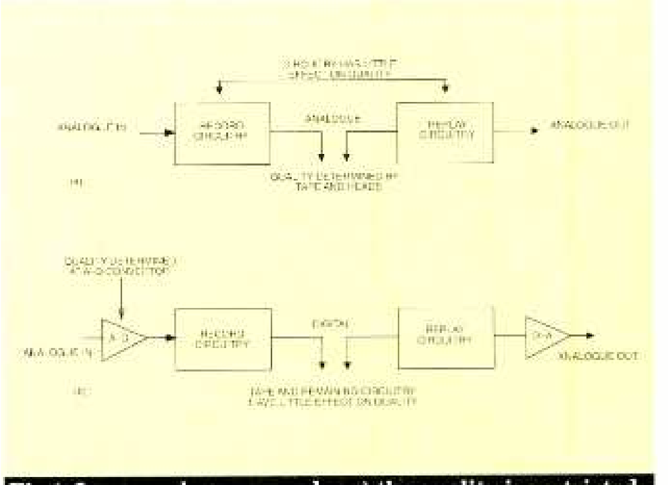

Fig.1: In an analogue recorder a) the quality is restricted at the head-tape system and the degradation due to the circuitry is negligible in a well-engineered machine. In a digital machine, b) the quality is restricted at the A–D convertor and the tape has no effect unless uncorrectable errors occur

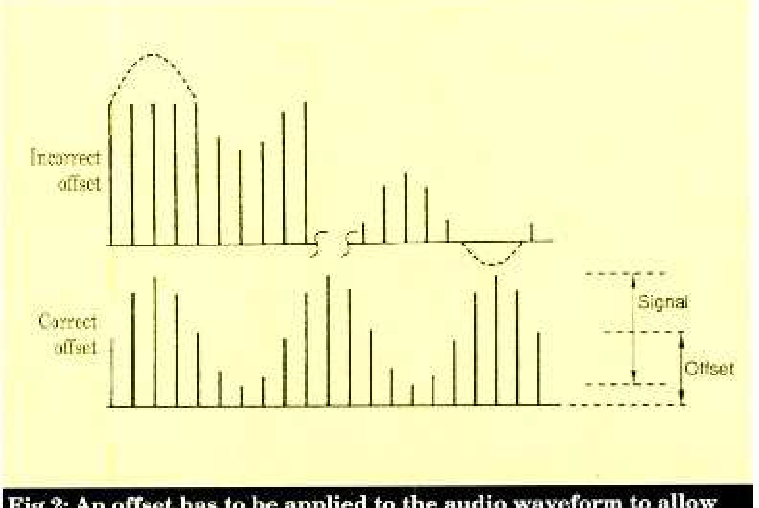

Fig.2: An offset has to be applied to the audio waveform to allow the maximum signal level before clipping

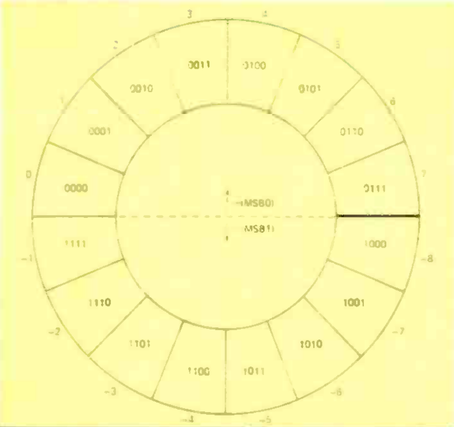

Fig.3: In two's complement coding, the number ring resulting from a fixed wordlength coding scheme is simply considered to start on the opposite side to zero. Note the MSB indicates the polarity

rotation. **Fig.4** shows how a real A–D convertor is configured to produce two's complement output. At (**a**) an analogue offset voltage equal to one half the quantising range is added to the bipolar analogue signal in order to make it unipolar as at (**b**). The A–D produces positive only numbers at (**c**) which are proportional to the input voltage. The MSB is then inverted at (**d**) so that the all-zeros code moves to the centre of the quantising range.

Some audio waveforms at various levels with respect to the coding values are shown in **Fig.5**. Where an audio waveform just fits into the quantising range without clipping, it has a level which is defined as 0dBFs where Fs indicates *full scale*. Reducing the level by 6.02dB makes the signal half as large and results in the second bit in the sample becoming the same as the sign bit. Reducing the level by a further 6.02dB to -12dBFs will make the second and third bits the same as the sign bit and so on. If a signal at -36dBFs is input to a 16-bit system, only ten bits will be active, the remainder will copy the sign bit. For the best ▶

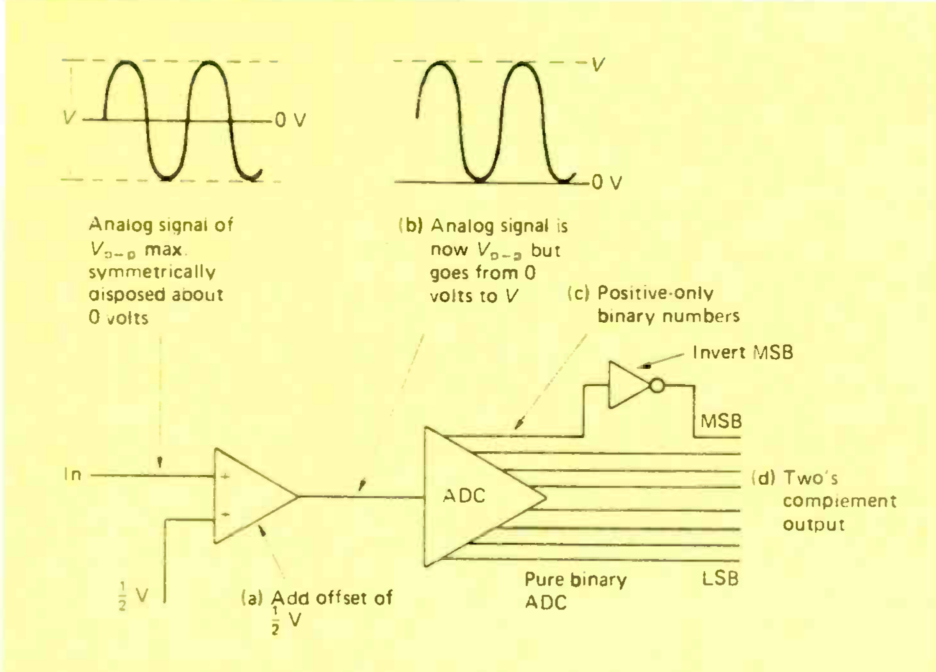

Fig.4: A two's complement A—D convertor. At (a) an analogue offset voltage equal to one half the quantising range is added to the bipolar analogue signal in order to make it unipolar as at (b). The A—D convertor produces positive only numbers at (c), but the MSB is then inverted at (d) to give a two's complement output

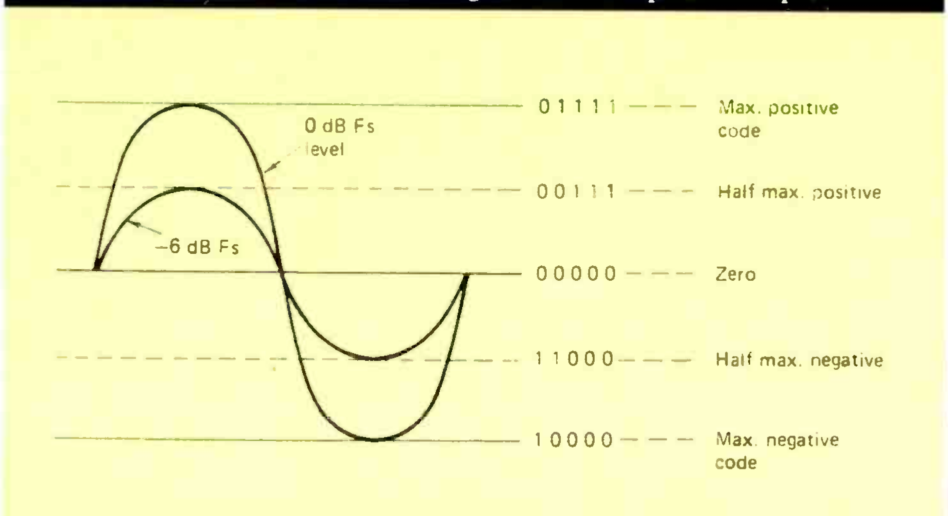

Fig.5: 0dBFs is defined as the level of the largest sinusoid which will fit into the quantising range without clipping. The numerical equivalent of some other levels are also shown. Note levels below clipping have sign bit extension in high order bits

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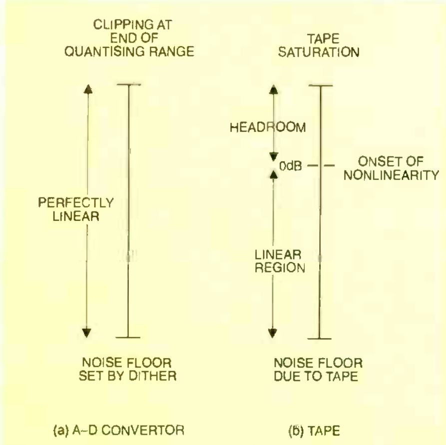


Figs.6a and 6b: At (a) the dynamic range of an ADC goes from the noise floor to clipping and remains linear all the way. There is no natural headroom. At (b) an analogue system goes from noise to the onset of distortion, then becomes progressively more distorted until saturation is reached. The area between the onset of distortion and saturation is the headroom

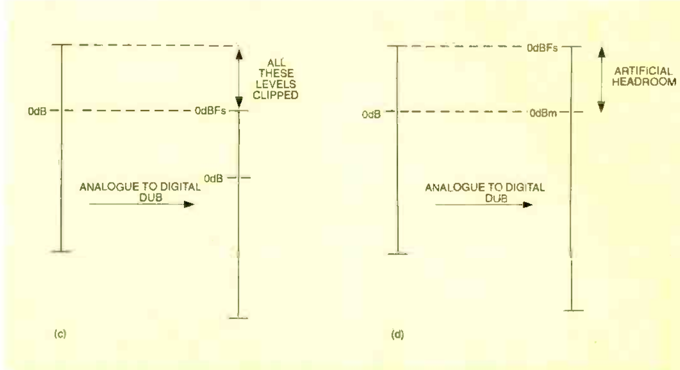
technical performance, analogue inputs to digital systems must have sufficient amplitude to exercise the whole quantising range. In theory the best quality digital recording is made when the largest sample just fails to clip. In this case the signal will be as high as possible above the noise floor of the convertor which will be the set by the wordlength and the nature of the dither and-or noise shaping systems employed.

If the analogue signal has excessive amplitude, clipping will occur. The convertor outputs maximum code value for as long as the input is in or beyond the last quantising interval. The onset of clipping is sudden and the effect on the waveform is surgical. On pure tones clipping in the minutest quantity is irritating, whereas on programme material a certain amount of transient clipping is inaudible because the ear has insufficient time to respond to the harmonics produced. In practice the best subjective recording will be one which allows inaudible clipping.

It will be seen from **Fig.6a** that an A–D has only two levels of interest, clipping level and the noise floor. This should be contrasted with the characteristics of analogue tape shown at (**b**). Analogue tape does not have the sudden clipping of A–Ds, but instead there is a gradual onset of distortion as the saturation is approached. This is partly due to the nonlinear shape of the B–H curve, but also a result of the addition of bias. The peak flux is reached when the bias and signal have the same polarity. As saturation is approached, the bias is clipped asymmetrically and this has the effect of reducing the signal level. Analogue tape thus has a more gentle clipping effect which is sometimes used as a form of compression. Analogue tape has three levels of interest, the noise floor, saturation and the onset of distortion. Practical analogue recorders use standard operating levels which are in the region of the onset of distortion. The range between the operating level and saturation is called the headroom. In this range, distortion becomes progressively worse and sustained recording in the headroom is avoided. However, transients may be recorded in the headroom as the human ear cannot respond to distortion products unless they are sustained.

The PPM level meter has an attack time constant which simulates the temporal distortion sensitivity of the ear. If a transient is too brief to deflect a PPM into the headroom, it will not be heard either.

Operating levels are used in two ways. On making a recording from a microphone, the gain is increased until distortion is just avoided, thereby obtaining a recording having the best S-N ratio. In postproduction the gain will be set to whatever level is required to obtain the desired subjective effect in the context of the program material. This is particularly important to broadcasters who require the relative loudness of different material to be controlled so that the listener does not need to make continuous adjustments to the volume control. It is equally important to balance the relative levels of the various tracks on an album.



Figs.6c and 6d: At (c) a digital recorder is erroneously lined up so that 0dB analogue reads 0dBFs. Any signals in the headroom clip at the A—D convertor. At (d) artificial headroom prevents this happening

In order to maintain level accuracy, analogue recordings are traditionally preceded by line-up tones at standard operating level. These are used to ensure that the gain is correct in various stages of dubbing and transfer along land lines so that no level changes occur to the program material. In contrast, gain drift cannot occur in digital transmission or dubs because neither can change the numerical sample values. In an all-digital system the only use of line-up tone is to set the gain of convertors.

Unlike analogue recorders, digital recorders do not have headroom, as there is no progressive onset of distortion until converter clipping, the equivalent of saturation, occurs at 0dBFs. Accordingly, many digital recorders have level meters which read in dBFs. The scales are marked with '0' at the clipping level and all operating levels are below that. This causes no difficulty provided the user is aware of the arrangement. However, in the situation where a digital copy of an analogue tape is to be made, if the operator does not realise how the digital meter is calibrated, it is very easy to set the input gain of the digital recorder so that line-up tone from the analogue tape reads 0dB. **Fig.6c** shows that this lines up digital clipping with the analogue operating level. When the tape is dubbed, all signals in the headroom suffer convertor clipping with obvious results. If the digital recording is made on an RDAT machine with no confidence replay, the operator may not be aware of the clipping. This is not funny if the RDAT tape is meant to be a safety copy!

In order to avoid such problems, and to make digital recorders behave more like analogue machines, manufacturers and broadcasters have introduced artificial headroom on digital level meters, as shown in **Fig.6d**. This is simply done by calibrating the scale and changing the analogue input sensitivity so that 0dB analogue is some way below convertor clipping. Unfortunately there has been little agreement on how much artificial headroom should be provided, and machines which use it seldom indicate the amount. One famous CD mastering recorder had so much artificial headroom that if the level meter were taken literally the result was a 12-bit CD. On the other hand the analogue desk driving it would probably clip before the end of the quantising range was reached.

If using a digital recorder for the first time it is important to establish how much, if any, artificial headroom is provided. The easiest way to measure headroom is to put tone into the input and increase the level until the A–D is heard to clip. The level is then reduced until the distortion just stops. The reading on the meter then indicates the amount of headroom.

Another way of resolving level problems is to display the actual two's complement numbers from the convertor. This is easily done by connecting the pseudovideo output from a PCM adaptor to a suitable video monitor. Using an audio signal generator, select a frequency which is a multiple of the field rate, and the bit pattern becomes stationary on the screen. **Fig.7** shows how the bit pattern of a *PCM-1610-1630* is interpreted. Once you get the hang of it, the video monitor can be used as a level meter in actual use and it has the advantage that you can see it from a long way away.

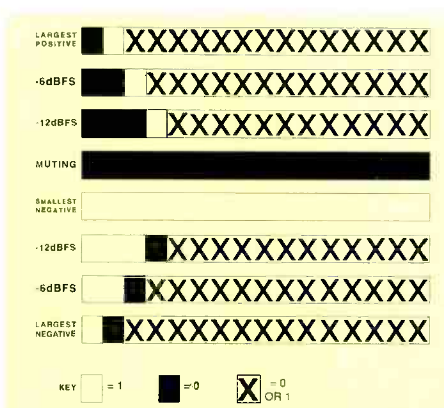


Fig.7: Interpreting pseudovideo to measure sample values is easy once the boundary of the serial sample has been located on the screen. Some examples are shown here. The level is measured by multiplying the amount of sign extension by 6dB

There is an argument which suggests that the amount of headroom should be a function of the sample wordlength, but this causes difficulties when transferring from one wordlength to another. The EBU[1] concluded that a single relationship between analogue and digital level was desirable. In 16-bit working, 12dB of headroom is a useful figure, but now that 18-bit and 20-bit convertors are available, the new EBU draft recommendation specifies 18dB. ■

References

1. Moller, L, 'Signal levels across the EBU-AES digital audio interface' in *Proc 1st NAB Radio Montreux Symp*, (Montreux, 1992), 16–28pps.

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Stereo debate

Dear sir, when someone makes claims that can be easily established as factually inaccurate, it must cast doubt on their other claims and opinions.

In his letter to *Studio Sound* (September 1993, pp 56, 57) Anthony Griffith states badly that when he wrote a detailed reply to an article of mine which appeared in *New Scientist*, 'I then heard from him [Barry Fox] that before publication it would have to be shortened and he wanted all references to himself and Brad Kay removed.'

As copy correspondence will prove to anyone who wishes to see it, this is just not true. The magazine editors, not me, wanted Griffith to shorten his very long letter. I encouraged publication but said that one short passage was factually inaccurate. To try and be helpful I wrote personally to Griffith suggesting that on this point he just deleted four words. (Griffith was saying that I had reported his views on Elgar in stereo, when in fact I had only reported what EMI had given as its reasons for aborting the plan to try and recreate stereo from two separate recordings made by Elgar).

Griffith did not want to shorten his long letter, and told the magazine that he would publish it elsewhere as part of much longer article. I have been waiting with interest to read this, but have never seen it.

Now in his letter to *Studio Sound*, Griffith is factually inaccurate in his suggestion that I tried to get all references to me deleted. And to support his argument that it is not worth even trying to recreate stereo from the Elgar recordings, he also goes on to say that, 'I have at the back of my mind a memory that someone once told me that work had been done on this at Abbey road...'

It worries me to think that EMI abandoned its promised plan to try and recreate stereo from a matched pair of Elgar recordings on the strength of a report from someone who is so casual with facts.
Barry Fox, London.

Come in mic

Dear sir, the DG 4D Audio Recording system for classical music sessions is obviously technically sensible and practical once the signal has arrived at the base of the microphone stand, as reported in *Studio Sound* in the September 1992 issue. But all the comments to and from DG ignore what is still the most important factor in sound quality —namely the microphone. Which type, Which configuration, which make, how many of it and if many, how near they are placed to different parts of the orchestra?

Those of us who use microphones know they are as much the weakest link in the chain as the great variety of loudspeaker-room combinations faced by the listener and on which the final judgment is made. We also know that the differences between microphones are far greater than those between any 16-bit/44.1kHz basic digital system or the effect of using a more esoteric transmission-recording system.

Can't we hear from DG about the microphones they favour, their deployment arrangements and particularly how many are used? Also, as they make their on-site judgments via their completely transparent digital multicore, it would be just as important for us to know which monitors are employed and the acoustic treatment employed in the control rooms. If DG are deceiving themselves in the first place, what about the full-price CD purchaser—whether they notice the '4D' flash on the sleeve or not!
Mike Skeet, Forties Recording Company, Furzton, England.

Design discussion

Dear sir, in my view Manuel Huber has the right approach, and his critics are living in a past age. The late Deane Jensen made great advances in the design of audio transformers but the fact remains that I and other designers (Manuel Huber apparently being among them) have created designs for transformerless mic amplifiers that are vastly more accurate than any transformer could ever be. Moreover, the oft-cited fact that transformers can provide galvanic isolation is a red herring, at least for studio use, where for the most part the only hostile voltage is +48V DC.

For occasions (such as outdoor recording) when safety is paramount, topologies exist that will withstand (and protect the equipment chain and ▶

users from) common-mode voltages of 400v AC-DC, on top of the mic signal. Although transformers of the quality of Deane Jensen's are perceived as expensive, they are actually the cheap and cheerful option, as the best active front ends cost somewhat more. But the price is still a small one to pay for accurately capturing something that can never be corrected, once set down.
Ben Duncan, Ben Duncan Research, Lincoln, UK.

On the cart

Dear sir, in the October 1993 issue of *Studio Sound* Barry Fox enthuses about Sony's MiniDisc, trumpeting claims about how it 'beats' today's best digital cart machines.

Digital cart machines do not primarily use solid-state storage or floppy disks as the recording medium. For example, 360 Systems' *DigiCart* uses 150Mb Bernoulli cartridges. MiniDisc doesn't beat digital carts on running time either. The *DigiCart* delivers 70 minutes of 20kHz stereo.

Barry Fox imagines that MiniDiscs are more cost-effective, but as broadcasters are quick to note, cheapness isn't a virtue when you have a job to do and it can't be done. Far more important than low cost are the traditional required functions of modern cart machines that MD's designers did not see fit to address: follow-on play with zero delay between cuts; instant access to any cut; attachment of readable labels to the cartridge; addition of a new cut to a cart machine without taking it off line; precision editing of audio materials; making backup copies of valuable audio carts.

This is to say nothing of the uphill task of pressing an unproven consumer-format on a notoriously conservative broadcast industry, or asking them to accept ATRAC over the widely-accepted standards of linear recording, Dolby AC-2 or APT-X.

As always, I thoroughly enjoy reading your interesting magazine.
Robert Easton, President 360 Systems, California, USA.

The master's voice

Dear sir, I was disappointed to read your comments in the Editorial column of December *Studio Sound* concerning the general attitude to mastering in the UK.

For very many years we have, at Abbey Road, treated mastering as very definitely the final stage of the creative process and not the first stage of the manufacturing process.

As far back as 1980 we built our Penthouse cutting room which was acoustically designed and very comfortably fitted out with much attention given to clients' needs. A comprehensive EMI transfer console was used plus a fair amount of outboard gear. Later in 1985 we built our DMM cutting room, again acoustically designed in the control room style with floating isolated floor.

We have been making constant improvements and currently have Sonic Solutions hard-disk editing, Sony *SDP1000* digital processing, Studer analogue machines, various analogue outboard gear, and Prism *AD-1* 20-bit A–D convertors in both rooms.

Our engineers, Chris Blair, Steve Rooke and Nick Webb, are all very experienced and have always worked with leading producers.
I hope this clarifies the situation and answers some of your criticisms of UK mastering facilities.
Chris Buchanan, Manager, Postproduction, Abbey Road Studios.

The Editor replies

Agreed, the criticisms I voiced in December might have been more specific. Had they been so, it would have been clear that they were aimed not at those people and facilities actively involved in the processes of premastering-mastering, but squarely at the UK record companies who have consistently refused to grant the mastering industry the recognition and support it deserves.

I applaud the attitude adopted by Metropolis Mastering in an attempt to redress this sorry situation; equally, I acknowledge the efforts of Abbey Road's postproduction in having pursued the same goal for some years—if in a quieter manner.
Tim Goodyer. ■

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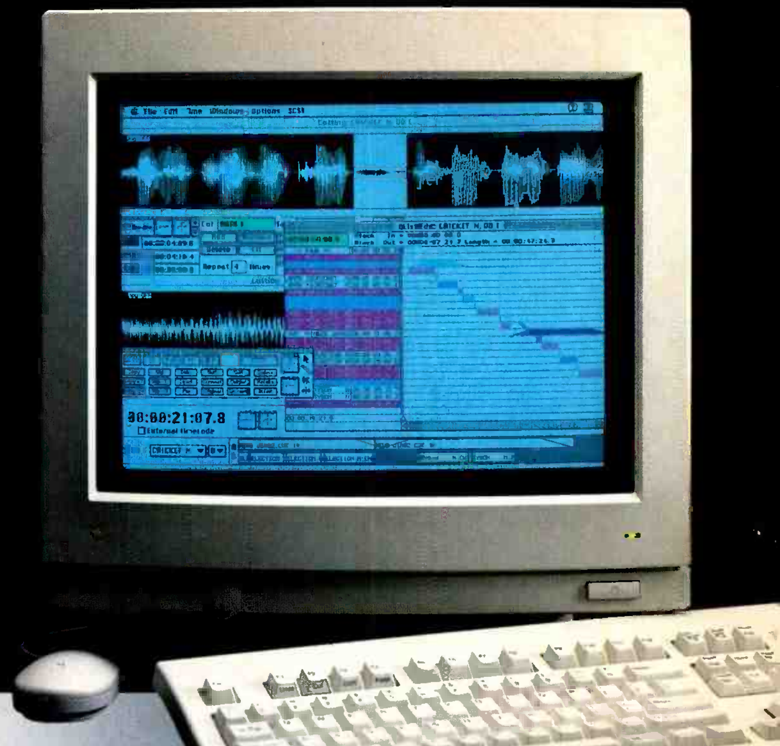

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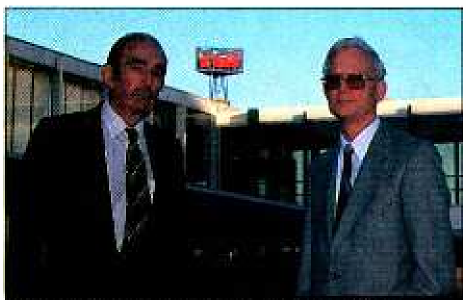
Kevin Hilton

Fighting talk at showtime while news is news for Meridian

The International Broadcasting Convention (IBC) has come a long way since its origins at London's Royal Lancaster Hotel in 1967. This first show, organised by a small group of broadcast professionals from various companies who saw the need for a general technological forum, boasted 30 exhibitors and attracted 550 visitors.

Even then the nascent exhibition was going up against the International Television Symposium (ITS), which had begun in its now firm home of Montreux a few years before. Both have grown considerably in the years since then—estimates put IBC's growth in the ten-fold region. For a number of years the two shows existed in a state of symbiosis, each held biennially in alternate years.

That comfortable balance of life was thrown out at the end of last year by the announcement that IBC is to go annual, although exactly when has still to be decided. 'The gloves have come off now,' said IBC Publicity Manager Tony Lawes—and given the strong ITS reaction, this is very much the case but IBC is far from apologetic.



John Tucker (left), namesake of the IBC/John Tucker Award and a founder of the IBC, with John Wilson (right), chairman of the IBC management committee

Just before Christmas, the organisation flew a number of broadcast journalists out to Amsterdam to see the newly completed exhibition hall at the RAI Centre, which replaced Brighton as the IBC's long-term home in 1992 and is currently gearing up for this year's event (16th–20th September).

Although this was cited as the main reason for the trip, in conjunction with promoting the IBC/John Tucker Award, named after one of the founders of the show, the bulk of the talk centred around IBC's proposed new frequency, which was decided upon at the organisation's last management meeting in November.

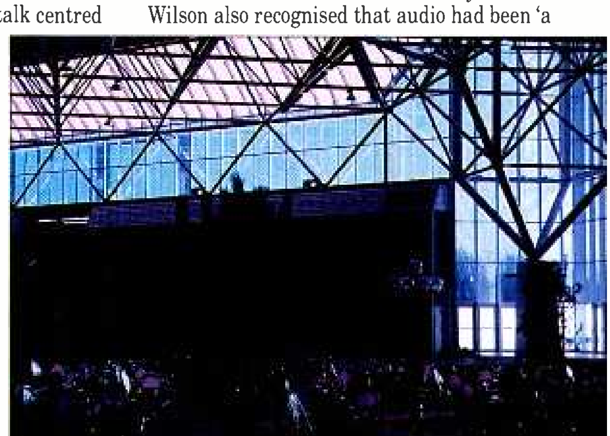
A working party is in the process of collating relevant information at the moment and was due to report to the main committee in mid-January. The decision as to whether annual status will begin in 1995 or 1997 will be taken after that. 'We're going to stick with the decision to go annual and adapt where necessary,' commented Tony Lawes.

Although both IBC and ITS are broadcast exhibitions—and largely address the television market—each organisation sees differences between the two. However, the IBC decision appears to have been prompted by a perceived change in the make-up and emphasis of the industry. 'There are no dividing lines between the various media anymore,' said Lawes, 'and Montreux is still emphasising TV broadcast. Broadcasters are now a very small part of the industry.'

In recent years, the advances in technology have brought a number of previously peripheral companies further into broadcast, including telecommunication operators and computer firms. Conventional optics are also changing, as John Wilson, Chairperson of the IBC management committee, observed: 'Film is a big industry—a lot of TV soaps are now being shot on 35mm stock. We see the exhibition expanding into two specific areas; one is film, the other is computers.' Wilson added that there has already been considerable interest expressed by a number of major computer companies.

This expansion into other areas can be seen as justification for becoming an annual show, as well as trying to offer something different from Montreux, although it is unlikely that ITS will be slow in this area either. Another sector that will see growth at the IBC is audio. 'We've been criticised, quite rightly, in the past for not including enough audio,' conceded Lawes. 'It's a very important part of broadcast generally.'

However, Lawes added that the problem will not be compounded by having what he called 'dedicated radio and audio ghettos.' He continued: 'We will try to have a concentration of audio-radio but it won't be a dedicated area. People will be able to be in these areas of mutual interest if they want to be.' Wilson also recognised that audio had been 'a marketing weakness' in the past and said: 'We are now seriously addressing it.'



The new Hall 11 at the RAI Centre in Amsterdam

As for the decision to go annual itself, the view of the IBC management committee appears to be that it is something they have to do. 'I don't feel too bad about rocking the boat because Montreux is already annual because of the Radio Show,' said Wilson. Backing up his bare-fisted analogy, Lawes commented, 'We believe that we've got our fingers on the pulse.'

Whatever the new elements of IBC, a large part of the convention will continue to be the award for significant contribution to the broadcast industry. This was instigated for the show's tenth anniversary in 1988 and was subsequently renamed the IBC/John Tucker Award in 1992 to recognise Tucker's position as one of the founders of the event.

The significant contribution can be either in the fields of engineering or management, with the award restricted to an individual and not a company or team of people, although it can be awarded to the leader of a group. 'Another category is to give it to someone who has not been presented with a major award before,' explained John Tucker, who had travelled out to Amsterdam to promote his namesake.

When it was first instigated, the award consisted of a £2,500 cheque and a trophy, designed by John Tribe of London Weekend Television. This glass rectangle, set in marble, is intended to represent the marriage between the creative and the technological skills that exist in broadcasting. While this has remained constant, the cash award now totals £5,000. 'The object is to give something back to the industry,' said Tucker.

On a more mundane level, the RAI exhibition centre now offers 11 halls and new restaurant and car parking facilities. Out of a total of 87,000m² of hall space, the IBC will take up approximately 25,000m². 'One of the successes of IBC has been the move to the RAI,' observed Wilson. A RAI management representative confirmed that the centre could accommodate the exhibition for whenever it does decide to go annual.

IBC. Tel: 071 240 3839. Fax: 071 497 3633.

Meridian Broadcasting, the Independent Television (ITV) contractor for the south east of England, is in the midst of completing a new newsroom in the town of Maidstone. This centre is due to go on air during April and will feed stories from the Kent and East Sussex regions of the station's transmission area into the main broadcast complex in Southampton.

The Maidstone news centre comprises a 2,000ft² studio, four edit suites, graphic studio and a single-camera studio for bulletins. The technical installation was largely carried out by Quantel Broadcast Systems, which contracted Pro-Bel of Reading, to supply in the region of £200,000 worth of equipment for the facility. This includes a large number of distribution amplifiers, the main routing matrix and a combined vision and audio custom switcher-mixer for the bulletin studio.

Pro-Bel. Tel: 0734 866123.

Martin Polon

Zen and the sound of one voice singing—a duet

The hot topic on the minds of many in the world studio business is the use of remote digital linkages provided by or provided through the telephone operating companies or other subsidiary common carriers or alternative carriers at each end of the hookup between recording studios. At the same time, the Fall 1993 release of the new Frank Sinatra album, *Duets*, emphasised the use of remote digital telecommunications to allow various artists to perform 'with' Sinatra without being in the same studio at the same time.

The album—which went 'Gold' prior to its appearance in record stores and could achieve 'Platinum' status several times over—pairs Sinatra with some of the musical leading lights of today's record industry. After several disappointing albums and a lightening of his touring schedule, most record industry observers assumed that Sinatra's career was coming to a close. His past records were receiving less bin space in the mall 'chain' record stores and were being sold on television after midnight—signs of imminent commercial death if there are any. But Sinatra had become a legend and an icon after more than 50-years of making music.

He was solicited by a savvy team of producers who assumed correctly that many of today's hottest new music artists like Bono, Gloria Estefan and Luther Vandross and established contemporaries such as Tony Bennett, Aretha Franklin, Lisa Minnelli and Barbra Streisand would leap at the opportunity to sing with Sinatra. They also calculated that Sinatra was ready to reach out musically to an audience one quarter his age.

The project was a remarkably efficient collaboration. Sinatra, eager to reestablish his worth yet again was very cooperative. The new technology of linking remote studio locations via ISDN (Integrated Services Digital Network), 'T' carrier service, Switched '56', coaxial-microwave or a combination of one or more of the above was pioneered by Lucasfilm in linking its Southern and Northern California Skywalker studios. Refined over several years of use, the technology is now offered by companies who 'broker' the necessary services and make all arrangements and was used for many of the 'duets'. Other tracks were similarly laid down at the artists' convenience and merged with Sinatra using digital technology. This use of digital electronic transmission allowed a very complicated album project to be completed without undue physical demands on an ageing Sinatra.

The use of remote digital technology reduced the overall project expense by shrinking the transportation and housing costs for the various other artists. They recorded their tracks while performing in a studio near their homes (or wherever they happened to be on tour). The professionalism of the performers also helped to control vagaries in production with the new technology—since there was none of the 'kids in search of an album' syndrome which regularly consumes days and months of studio time.

Consider also that the use of the digital studio-to-studio production technology reduced the overall studio production 'tab' by many hundreds of thousands of dollars. Also, due in some part to the 'exotic' nature of the new multistudio multinode technology, the record project received extraordinary feature coverage in the nation's magazines and the arts sections of major newspapers. TV entertainment and network news programmes also highlighted the project.

A whole series of objections have been raised to the concept of 'phoning in' a musical project. (These and other objections to the 'digital' liberties now being taken with musical integrity will be discussed in another column.) Suffice it to say that for many in the music industry, and for the public, curiosity concerning Sinatra far outweighs considerations concerning production details.

It is curious fact that the development of new and-or improved technology of digital communications is moving at such a pace that one can only look back at the *Duets* project and say, 'that was then and this is now.' It is equally clear that the future of communications on this planet is going to evolve into a high-quality digital matrix capable of random-access linkage from any point to any other point. The following directions illustrate the dimension that future digital communications will take into the home, into the office and into the recording studio and any combination thereof:

A joint venture has been entered into to create Lightstream Corporation to build the infrastructure for the American nation's data (super) highway.

Time-Warner will deploy their 'full service network' in 1994 to 4,000 homes in Orlando, Florida. The system will include current multichannel one-way interactive communications with the home. Each home will have a video camera and audio origination plus a sophisticated 'setup' microprocessor control unit that will allow audio on demand and video (movies) on demand.

This and other similar systems will incorporate data-telecommunications 'type' switches that will integrate existing public switched copper networks, fibre optics transmission systems, 'T' carrier systems, coaxial head-end-based TV redistribution systems, ISDN (Integrated Services Digital Network), microwave systems and other newer systems as they emerge.

> It is equally clear that the future of communications on this planet is going to evolve into a high-quality digital matrix

The key to the successful implementation of this future technology infrastructure is the use of ATM (Asynchronous Transfer Mode) with multiplexing of voice, music, data, still video and full-motion video. Other systems such as X25, Frame and so on, do not have all of the advantages of ATM.

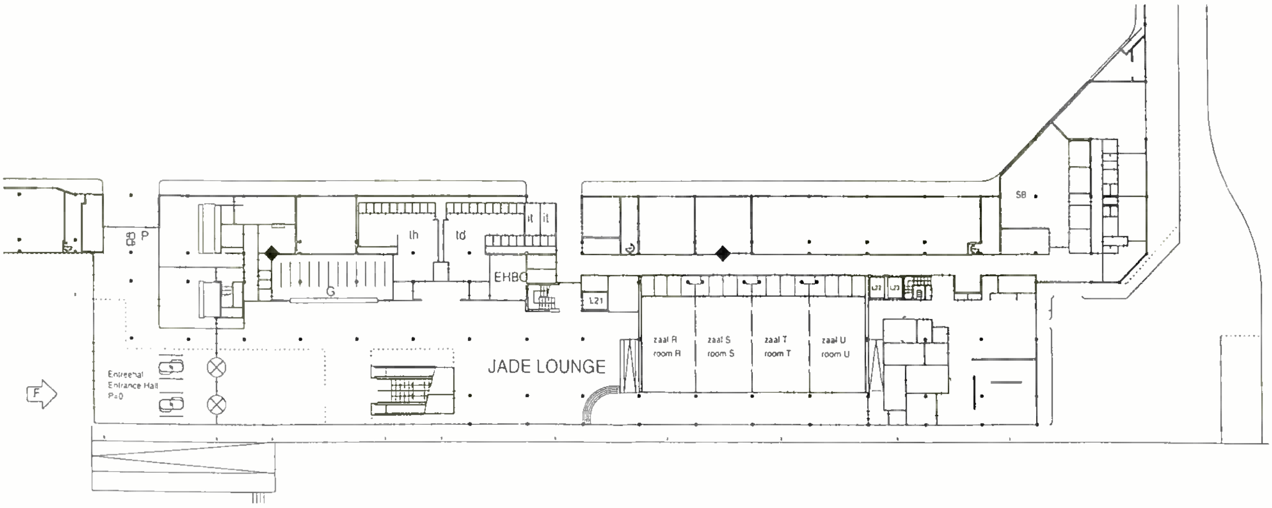
ATM is the international standard for cell relay of entertainment, education and information ranging from megabit to gigabit speed. It is suitable for local-area and wide-area communications services. It is designed for voice, data and multimedia over both public and private networks. It is an international standard, endorsed through the various domestic and international telephone and telecommunications carriers who sought its creation.

ATM is similar to packet systems except that the packets are called cells with a fixed size of 53 bytes (5-byte header/48-byte payload), by agreement among the telephone company originators of the system. This system is optimised for 'bursty' traffic, which the majority of the intended sources are capable of being at one time or another. Synchronous transfer modes such as the T1 carrier system has to have dedicated or reserved time slots to move a fixed amount of information. ATM does not require reserved 'slots.'

To take all of this technology one step further, the introduction of so-called 'smart agents,' will provide users with 'steerable' software that mimic the action of computer viruses and travel about networks and communications nodes to actually do the work of making a suitable path for intrastudio digital connection. These 'agents' will provide all sorts of functionality on the networks to actually handle sending, receiving and set up tasks without the need for human effort and intervention. They have been designed to automate tasks like data retrieval of particular files such as all necessary for the user to indicate what specifically was wanted and the software 'agent' would go into action, circulating about all of the available databases until the job was finished and return with the retrieved data. It would not have any limits in the time dimension—at least in relatively long terms. The 'agent' would 'work' until the job was done.

Instead of studios depending upon 'setup contractors' who arrange for the digital interconnection to be made, 'agents' riding the new ATM networks or the enhanced Internet or the phone companies ISDN matrixes of fibre pipes would be sent out with a specific task and return with links established for a particular time and place. The connection would be enabled when and where it was wanted and the only things left for the user would be to pay the bill for the services used—a result of the connecting 'agent' contacting a 'charging' agent to establish a debit protocol as part of its routine.

There are, however, many studio owners and operators who still earn a pleasing profit from their analogue facilities. And there are other detractors who perhaps should be viewed as being 'temporarily challenged' by digital technology. ■



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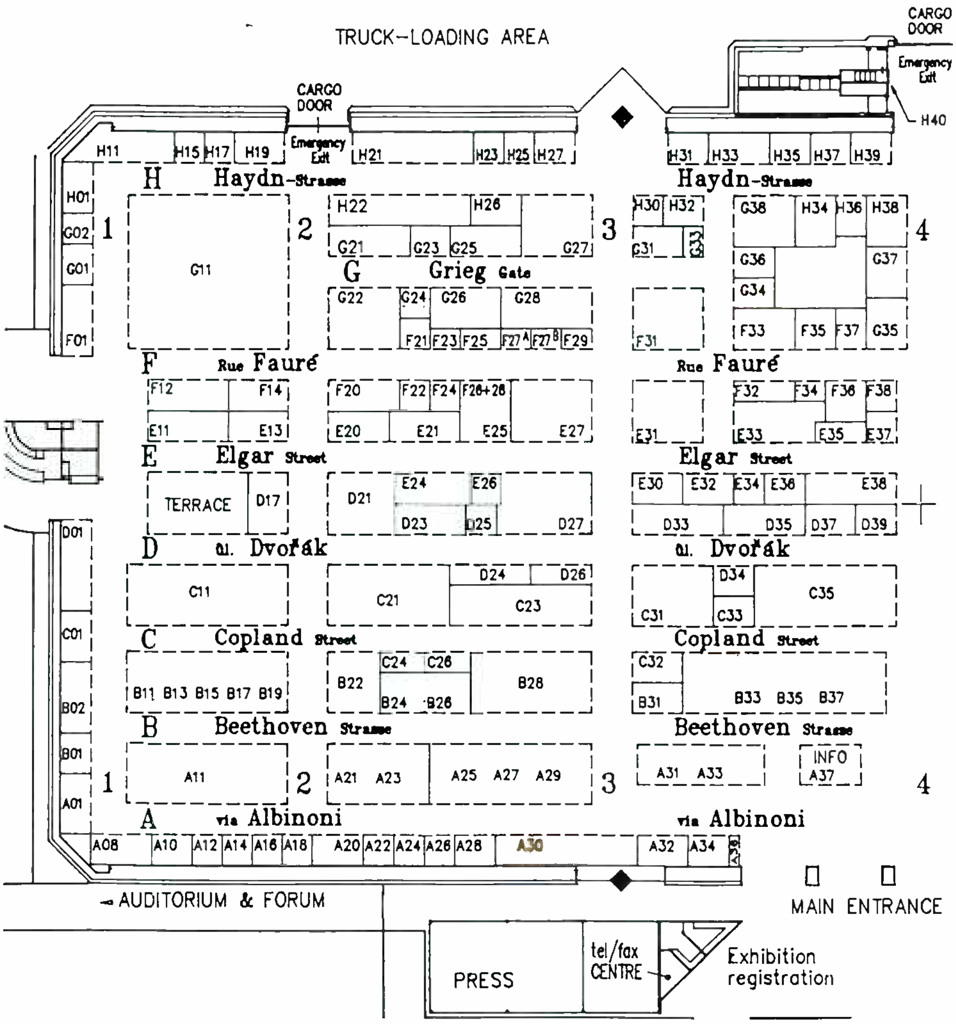
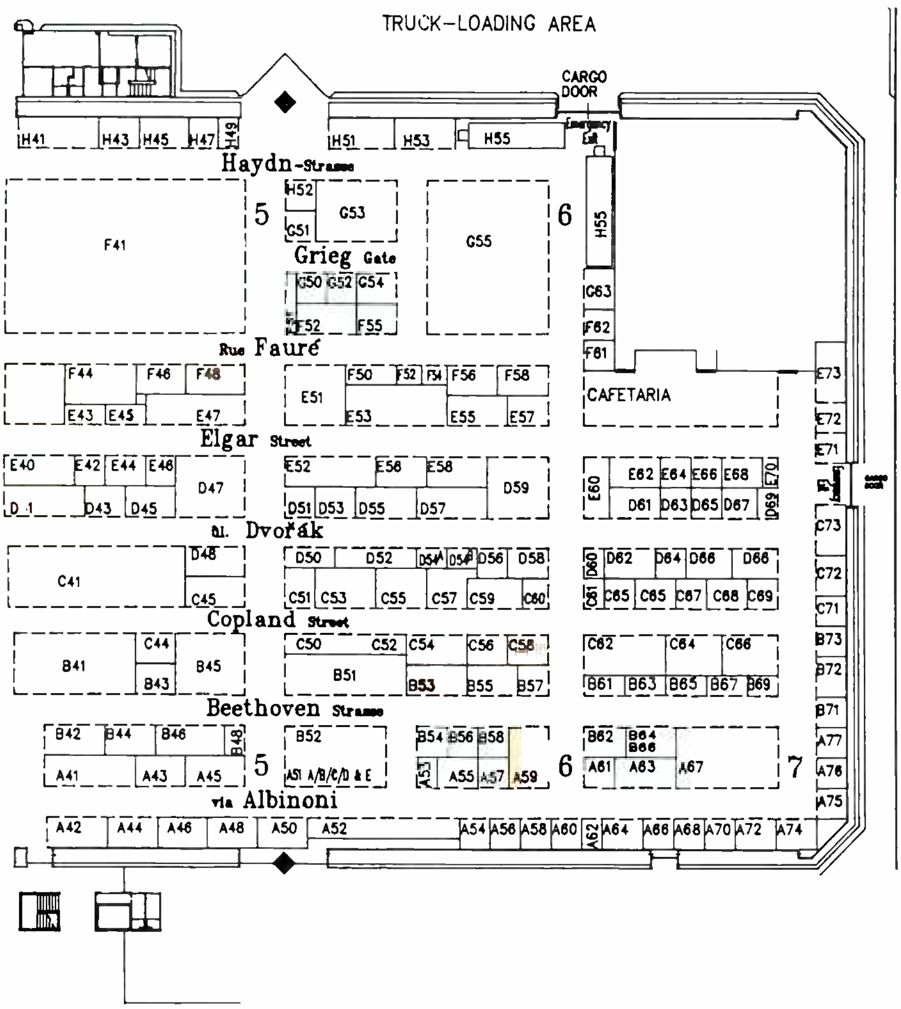

TRUCK-LOADING AREA
CARGO DOOR
Emergency Exit
H40
H Haydn-Strasse
G Grieg Gate
F Rue Fauré
E Elgar Street
TERRACE
D al. Dvořák
C Copland Street
B Beethoven Strasse
INFO A37
A via Albinoni
AUDITORIUM & FORUM
MAIN ENTRANCE
PRESS
tel/fax CENTRE
Exhibition registration



TRUCK-LOADING AREA
CARGO DOOR
Emergency Exit
H41 H43 H45 H47 H49 H51 H53 H55
Haydn-Strasse
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F61
Rue Fauré
F44 F46 F48 E43 E45 E47
E51 F50 F52 F54 F56 F58 E53 E55 E57
CAFETARIA
E73 E72 E71
Elgar Street
E40 E42 E44 E46 D47 D41 D43 D45
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A42 A44 A46 A48 A50 A52 A54 A56 A58 A60 A62 A64 A66 A68 A70 A72 A74

SABINE SISTERS

The Sabine *ADF-2400*—the DSP solution to feedback problems

Sam Wise examines the *ADF-1200* and *ADF-2400* dedicated DSP units intended to reduce feedback problems not only in live situations, but also for broadcast

The Sabine *ADF-1200* and *ADF-2400* workstations are digital signal processing engines designed with a specific emphasis on feedback control. As such, they provide automatic detection and dynamic filtering of sound system feedback resonances, making them useful in any live sound situation and also anywhere where there are monitoring systems which might be triggered into feedback (including radio and TV broadcasting). In addition, the filters, signal delay, and noise gates can be used manually for sound-system tuning, or to get rid of unwanted noises on programme material. They will be particularly useful as an on-line device during sound effects lay-back or mixing, since subtle timing alterations can be made in addition to improving effect quality. In this case the filters and gates can also be used to generate some of the required effect themselves. The *ADF-2400* 2-channel model is the subject of this review. The *ADF-1200* is a similar, single-channel unit.

Quick start

In a typical application, an *ADF* unit should be connected between the mixer output and the input to any processing directly associated with the loudspeaker systems. The *ADF-2400* can be used for various purposes, but its main function is as an 'automatic' feedback suppressor. The basic, but effective operating manual begins with a quick start process. This first requires that a signal is passed through the system with microphones off. The front panel clip-level controls on the *ADF* are adjusted using the front panel LED level display to match the *ADF* internal clip point to the sound system clip point —thereby optimising dynamic range. The use of music, or pink noise for this is essential, since any pure tone will be regarded as a feedback and immediately result in automatic filter action. After the microphones are set up and enabled, the *ADF* is switched into active filter-adjustment mode as the system gain is raised—usually using the mixer master faders. This should be done with an approximate balance between microphone sources. As each feedback frequency is reached, the *ADF* tunes a filter to it and sets its attenuation depth. The manual says to raise the gain slowly, and indeed this is necessary to allow the filter depth to reach the required level before the next feedback frequency starts up, and to allow feedback resonances to be dealt with in an orderly manner.

Fig.1 shows the frequency response which resulted from following these instructions using a 2-microphone setup with a foldback monitor. The starting point for this used the *ADF* unit's default settings for 'global parameters' (more about these later) and filter modes. For this first run the filters had a maximum allowable bandwidth of 1/10 octave.

With filter settings adjusted to allow to widening up to a 1/2 octave, the resulting curve from the same setup is shown in **Fig.2**. A low-frequency resonance was also introduced by holding a hand around the sides of the microphone capsule. Another try with 1/3-octave maximum filter width is shown in **Fig.3**. When filter widths were not altered, the *ADF-2400* repeated similar results.

The *ADF-2400* worked well in capturing feedback ▶

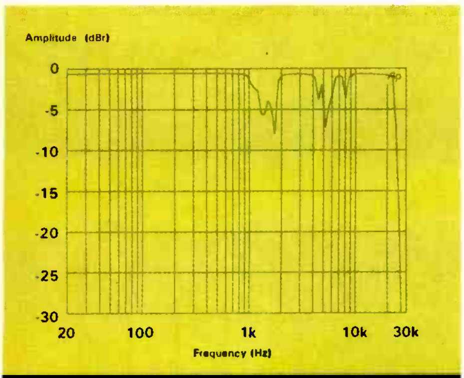

Fig.1: Curve shows the amplitude response of Channel B, following automatic feedback suppression using filters of 1/10-octave maximum bandwidth

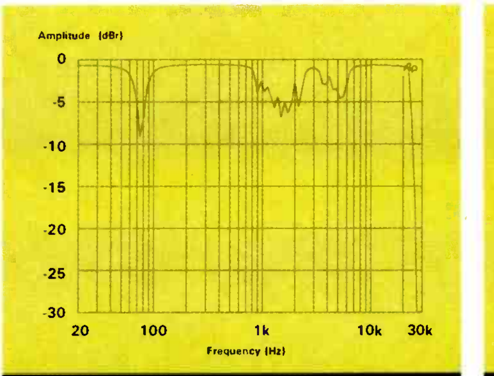

Fig.2: Curve represents Channel B as in Fig.1. This time the filter bandwidth is allowed to grow to 1/5 octave. A low-frequency element has been introduced by cupping a hand around the microphone capsule

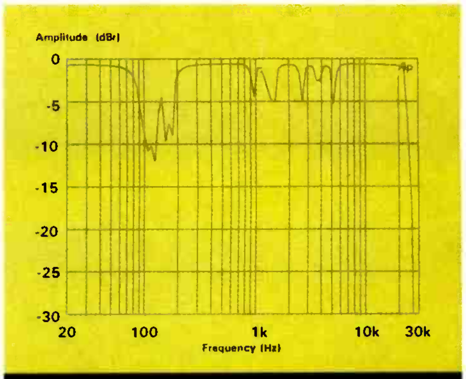

Fig.3: Curve represents Channel B as in Fig.1. This time the filter bandwidth is allowed to grow to 1/3 octave. A low-frequency element has been introduced by cupping a hand around the microphone capsule. The three figures are similar in the mid to high-frequency region, but not identical

frequencies in this manner, which is similar to that used by live sound engineers to manually filter feedback modes using conventional equalisers.

Recordings were made of the resulting outputs so that I could listen to them at my leisure (and when I was not talking into the mic at the same time as listening). The results were good, with ringing disappearing at the required gain level and audibly better quality. Gain before feedback improved 4dB in this case, but this will vary according to individual circumstances.

The 12 available filters per channel can be individually set for Fixed, Dynamic or Parametric modes of operation. The first of these is not quite as it sounds, but rather means that the frequency of the filter is fixed automatically as these setup feedbacks occur, though the filter depth may be automatically altered further later. In fact, my experiments showed that the centre frequencies of the Fixed mode filters could also automatically alter within about ±5Hz of the original frequency.

The Fixed filters are the first to be automatically adjusted as each feedback frequency appears. Dynamic filters are those that remain totally automatically adjustable in frequency and depth, primarily to cope with feedback which occurs during operation. During the initial setup operation, the Fixed filters are used first, followed by the Dynamic filters, until all are used up. After this the first Dynamic filter which was set is reused to cope with the next feedback, and so on.

The unit is quite effective at optimising both Fixed and Dynamic filters. If a feedback frequency is near an existing filter, this might be widened or deepened up to the set limits to capture the problem feedback component automatically without introducing another filter

Filters which are set to Parametric are completely under manual control. The numeric keypad is used to set the frequency, width and depth of the filter. Unlike the other modes available, Parametric filters can be used to boost up to +12dB, as well as to cut. Once automatic filter adjustment has been completed, the filters can be set altered to Parametric mode in order to force the retention of the automatic settings. This might be useful in applications in speech announcement systems, where the microphones and loudspeakers remain in fixed positions. Any of the 12 filters can be set to any of the three operating modes.

One thing not to expect from the *ADF-2400* is that it will prevent feedback altogether—it works by detecting feedback once it has already happened. As mentioned above, it is common for a live sound engineer to follow a similar manual process when setting up systems, but before the audience or performers are present to hear it. In this sort of use, the *ADF-2400* speeds up the process, does it more accurately, and depending upon the exact acoustical characteristics involved can give a more transparent result due to the potential narrowness of the filters. In other applications, untrained users will give rise to feedback under conditions where there is no sound operator present to take corrective or preventative action. In these cases, the *ADF* will not prevent the initial feedback, but can identify it and prevent recurrence without operator intervention.

Furthermore, the *ADF* will not automatically remove ringing from a system. The device is not ▶

MANUFACTURER'S SPECIFICATION

Filters

One channel on *ADF-1200*, two on *ADF-2400*, may be combined. There are 12 independently-controlled digital notch filters per channel.

Each filter: automatic or manual parametric operation 20Hz–20kHz range in 1Hz resolution

Notch depth: from +12dB to -84dB in 1dB steps (manual)
from 0dB to -80dB in 3dB steps (automatic)

High-pass filter: manual operation 12dB per octave slope
frequency range 20Hz–1kHz, ⅓-octave centres

Low-pass filter: manual operation
12dB per octave slope
frequency range 3kHz–20kHz, ⅓-octave centres

Digital delay

ADF-1200—340ms total, ADF-2400—170 ms per channel
Programmable in milliseconds, feet or meters, 100µs resolution

Noise gate

Dwell time: 0–999 ms
Threshold: -1 to -95dB (relative to peak amplitude)

Real-time analyser

31-band, ⅓-octave, 20Hz–20kHz

Memories

8 user-defined, 1 factory default, 1 most recent configuration

Performance

Input impedance: balanced, >10kΩ, XLR with pin 2 hot
Output impedance: balanced, approx. 10Ω, XLR with pin 2 hot
Maximum signal levels: balanced +29dBV peak
Bypass: passive, input to output on power fail or from front panel switch
Frequency response: ±0.25dB, 20Hz–20kHz
Signal-to-noise ratio: >100dB typical (with noise gate active)
Total Harmonic Distortion: <0.02% at 23dBV at 1kHz

Power requirements

120V AC; from 50/60Hz adaptor available in 100V, 120V, 220V or 240V; 22W

Memory battery life

10 years

Dimensions

2-unit rackmount, 19 x 3.5 x 7.5 inches (483 x 90 x 183 mm)

Weight

9lb. (3.9kg) without power supply adaptor

Options

I-O transformer isolation RS232-RS422 serial interface (to be announced)
AES-EBU; SPID digital interface (to be announced)

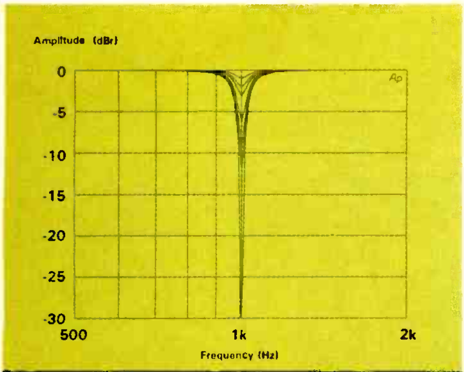

Fig.4: Frequency response of a 1kHz filter at $^1/_{10}$ octave bandwidth and varying depth. Notice consistency of filter width. Filter depth is also very accurate

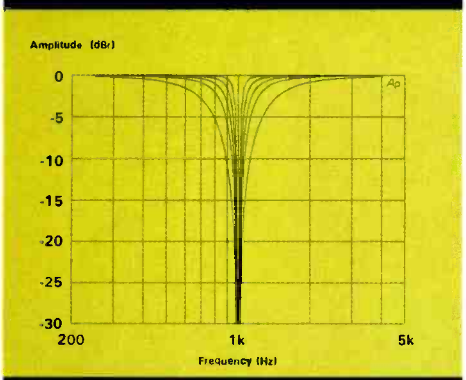

Fig.5: Frequency response of a 1kHz filter at a constant -60dB depth but varying bandwidth. Notice constancy of filter depth

clever enough to identify ringing for what it is, since it is usually provoked only when the system is stimulated and therefore appears transient. If there are spare filters available to remove these, then system gain must be raised further to allow full-blooded feedback to occur. At that point the *ADF* will recognise the problem and set the filters. Reducing the gain will probably produce a more acceptable sound.

Filter performance

The following examples use the Parametric filter mode to show the response and interaction of the filters. **Fig.4** shows the result of fixing the frequency at 1kHz with a $^1/_{10}$-octave bandwidth and varying the depth. The constant-bandwidth nature of the filter is obvious, with little widening of filter shoulders as depth is increased. In **Fig.5** frequency and depth are held constant while filter bandwidth is varied from $^1/_{10}$ octave to 1 octave. Finally in **Fig.6**, width and depth are maintained while frequency is shifted. No parameters interact—you can really believe what the settings tell you.

The internally-generated filter responses can be viewed in an idealised way on the panel LCD screen.

Other filter applications

When not in use for feedback control, the filters can serve many other purposes. Some examples are given for comparison with typical $^1/_3$-octave filter sets in both feedback and other applications.

In **Fig.7**, a set of five $^1/_3$-octave filters of -20dB depth are combined. Notice that not much combining interaction takes place. This is the correct design choice for a equalising device intended primarily for feedback suppression and is equivalent to noncombining conventional $^1/_3$-octave filters. In **Fig.8** the bandwidth is narrowed to $^1/_{10}$ octave and depth increased to 60dB. Here there is no interaction visible at all, feedback would be totally suppressed at the selected frequencies, and the effect on most wanted signals would be inaudible. **Fig.9** shows a mains-hum filter configuration set to 50Hz and several of its harmonics. Bandwidth is now 0.01 octave, and depth 60dB. This creates an effective but inaudible mains noise-filter.

Signal delay facility

The signal delays within the *ADF* are fully adjustable over the range varying from a nominal 0 to 340ms in the *ADF-1200* and from a nominal 0 to 170ms in the *ADF-2400*. Adjustment resolution is 100µs, limiting the range of use in sound systems to low-frequency intercabinet alignment or overall system delay. Since the *ADF* output is normally full range, it is likely that alignment delays for the individual sound system components will be further downstream anyway, therefore this is not much of a limitation. ►

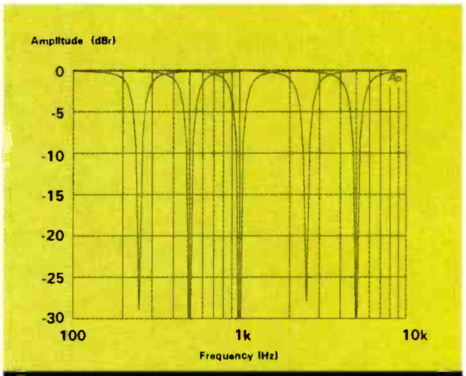

Fig.6: Frequency response of a single $^1/_3$-octave filter at a constant -60dB depth but varying frequency. Notice the constancy of bandwidth

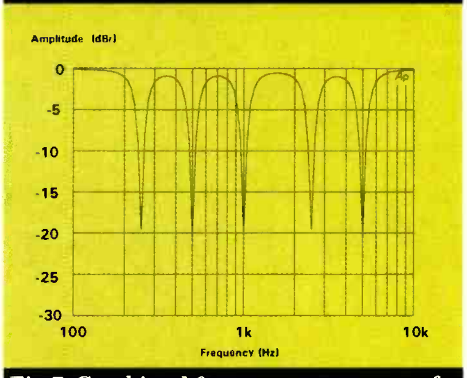

Fig.7: Combined frequency response of $^1/_3$-octave filters at a constant -20dB depth. This is similar to the result which would be obtained using a conventional noncombining filter set

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Load, edit, and dump simultaneously.
No other digital audio workstation offers the multi-tasking capabilities of the Sonic System. You can load and unload to the hard disk(s) in the background while you edit in the foreground. And while you're working, you've got plenty of playback capability — even the most basic Sonic System can play 12 or more channels simultaneously from a single hard drive! With NoNOISE® you can now run *two* jobs simultaneously in the background and keep working in the foreground.

Create radio spots, edit and master a CD, cut sound for film or video, and restore sound with NoNOISE.
The Sonic System is the power platform for a wide variety of applications. Our product line is entirely modular and can be easily expanded or customized to your line(s) of work.

Share soundfiles with the engineer(s) next door.
With MediaNet™ you can share soundfiles, edit decision lists, and processing resources among Sonic systems. MediaNet is the true Power Users' network — it supports playback of 80 or more channels simultaneously, and multiple users can access the same hard disks (even the same soundfiles!) without introducing a drag on the host system.

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Our built-in digital video gives you fast access to picture so you never have to wait for a tape machine to shuttle. SonicVideo plays back smoothly without any interruptions — even when listening to 24 channels of digital audio!

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Sonic Solutions was the first to integrate a workstation with a CD recorder. You can create high-precision CDs which can be used for direct glass mastering, archives, or reference copies. And soon, you will be able to create the new CD-DVs (CD-Digital Video) on a Sonic System!

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No other system offers the breadth or flexibility of the Sonic System. And no other system can match the performance for the price — an entry level system, including Macintosh and hard disk, is under $10,000 (slightly higher outside the US).

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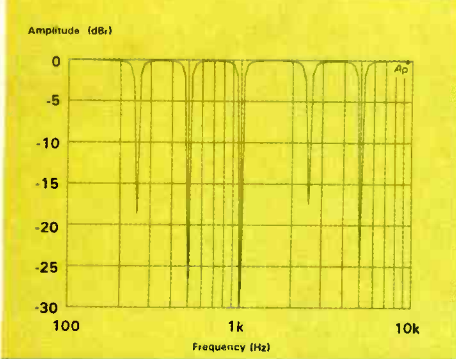

Fig.8: Combined frequency response of 1/10-octave filters at -60dB depth. Notice that compared to Fig.7 the mid-band level is maintained, and the filters do not interact. Feedback at these frequencies is, however, reduced further and the audible effect on wanted signal is also reduced since the filters are narrower than the ear's critical bands

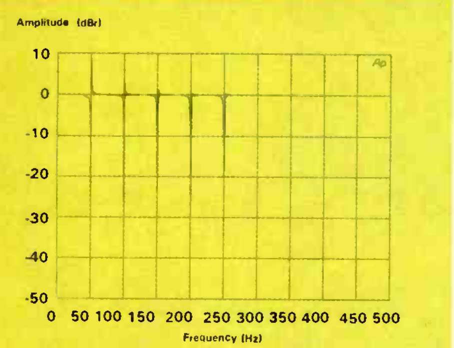

Fig.9: Combined frequency response of 1/100-octave filters at -60dB depth set to multiples of the 60Hz mains frequency. The narrow filters remove the offending noise while having no audible effect on the programme material. There is a 2dB boost visible at the right of the lower frequency filters. This too, should be inaudible

In fact, the nominal '0' intrinsic delay of the *ADF* is not zero at all. As with all digital audio devices, it has an intrinsic system delay, in this case it is about 1.2ms. Live-sound users should therefore be careful to treat either all or none of a group of speakers, since a delay between cabinets could otherwise cause acoustic signal cancellations which would be audibly obvious.

The delay is displayed simultaneously in feet (0–384.6), meters (0–116.8) and milliseconds (0–340) on the *ADF-1200;* and half of these values in the *ADF-2400* per channel. Adjustments can be made in any of these units by simple use of the cursor controls. The distance calculations are based on a fixed temperature, so are only approximate, but close enough for most purposes.

Aside from the intrinsic delay of 1.2ms, the delay settings were found to be accurate within 0.1ms.

Real-time analyser facility

The built-in RTA only depicts the response of the input signal, providing no means of viewing the effects introduced by the *ADF*'s own filters. This can come in handy if levels, and therefore stresses on the loudspeakers, are getting high allowing an informed judgment to be made on additional equalisation. Unfortunately, it is impossible to watch this and adjust an internal filter at the same time. When in RTA mode, filters retain their last settings, but feedback detection is disabled, so no further automatic adjustment will take place. The RTA can be used to examine either A or B channels, but no provision has been made to store any of these observations. This means that it is not possible to compare the two channels in any way. For someone with a poor short-term memory like myself, this is something of a shortcoming. It is easy to understand why the unit cannot cost-effectively display simultaneous real-time information on both channels, but some memory facility would be very inexpensive to implement. When working on the RS232 remote control add-in, it would be good for Sabine to consider this requirement. Another shortcoming is the inability to display the signal in both preprocessed and postprocessed form. Since both input and output audio signals exist within the unit in analogue and digital form, this too looks outwardly easy. The results of filter action are not always easy to infer from the idealised filter-response display provided.

Display resolution adjustment would also make the RTA display more useful. While the large 45dB display range is good for giving a feel of the spectral content of music and speech signals, the present system is rather coarse for use in setting sound-system equalisation. The level indication steps varied, but were generally about 3dB to 4dB per step at the best resolution available. There is no accurate calibration to refer either. The overall result is a general impression of frequency response, nothing more accurate than that. When tested with pink noise, the RTA gave a visibly flat response down to about 100Hz, below which a too-short time constant and-or low frequency roll-off gave results up to 3dB down.

Noise gate

A noise gate on a sound system is typically used to remove the audible effect of noise generated by upstream equipment when it is not masked by reinforced sound. In some equipment, the gates are really included to reduce the self-noise of the equipment, but this is not the case with the *ADF* series, where intrinsic noise levels are already low. When the input signal drops below an adjustable Threshold, the noise gate virtually mutes the output. A Dwell Time adjustment is provided which can be set to determine the time that the input signal is below the threshold before the gate is closed. In the *ADF*, gate closure is accomplished by turning off the D–A convertor and setting the output signal to zero, thus there are no VCAs involved.

While the gate is closed, the digital system continues to monitor the digital output of the input A–D convertor. As soon as the signal level rises above the threshold, the output D–A is enabled again. The dwell time is also controlled digitally, so as designed will be signal level independent.

Noise gate threshold is adjusted relative to the clipping level of the A–D convertors in the *ADF*. This enables the threshold to be adjusted over a 106dB range.

Under test, the noise gate performed as specified to within 1dB of the set threshold. Noise levels of the *ADF-2400* itself were marginally better with the noise gate active as shown in **Table 1**.

TABLE 1

Conditions	22Hz–22kHz RMS	22Hz–22kHz Avg	CCIR 468-3 Unwtd	CCIR 468-3 Wtd
Noise Gate Inactive	-90.2dB	-91.5dB	-86.3dB	-81.1dB
Noise Gate	-92.5dB	-93.7dB	-88.4dB	-84.1dB

Wide-band noise performance at 0dBu clipping level and higher. Noise levels in dB below signal reference level

Memory presets

There are a total of nine memory presets, with one of these assigned to the factory default settings. To save the current setting, the memory Preset Table is selected from the main menu and the cursor moved to the Save field. Pushing the number key corresponding to the required preset number and then Enter completes the function. Loading a preset is similar, using the Load field. Any memory can be given an alphanumeric name of up to eight characters using the + and - keys to step through the alphabet. This is more awkward than necessary, since the alphabetic characters which are most often required are halfway through the list from both ends. It is possible to toggle easily between saved presets to audibly compare the ▶

results. On power up, the *ADF* automatically recalls the last used memory.

Global parameters

Computer jargon is getting in everywhere. Here Global Parameters refers to standard settings the *ADF* uses when automatically controlling the filters in feedback suppression mode. These global settings are stored with and applied independently to each memory preset, aiding with optimisation of presets for different user applications. The settings controlled are as follows: Filter Width adjusts the maximum allowable automatic width setting of feedback suppression filters. The range is from 1 octave down to 1/100 octave. The manufacturer's recommendations are 1/10 octave for music and 1/5 octave for speech-lecture applications. These digitally generated filters are much narrower and more stable than conventional analogue filters. Filter Depth sets the maximum depth that the *ADF* can automatically use in the range from 0dB to -80dB. Threshold sets the level of signal increase which the *ADF* determines as feedback. This adjustment is provided to allow reliable detection of feedback, while preventing suppression of genuine musical tones. Persistence also has an effect on the ability to distinguish between the wanted and unwanted signals. Pure tone generators such as flute or organ will require a lower value than for speech, which is by nature very transient. Mode determines how the *ADF-2400* works. Dual Mono mode allows each of the two channels to work independently, while in Stereo mode the filters track in the two channels. Clustering sets the allowed proximity between two adjacent *ADF* filters. When Yes, adjacent filters can be set only 1Hz apart. When No, the minimum spacing is 5Hz.

Under test, the persistence control was found to have a profound effect on the recognition of feedback. A burst signal was set up with 1s on, followed by 3s off. Under these conditions, a Persistence of 5 resulted in detection. At a Persistence setting of 1, a 3s burst length followed by a 1s pause was detected. Each time the signal repeated, it was identified as having occurred before, and the depth of the appropriate filter was automatically increased. The range of control of this variable seems sensible, but why is a persistence of five more sensitive than a persistence of one? This seems confusing to me.

Threshold sets a relative level before tones are identified as being feedback generated. However, once again the numbers seem backwards, with a threshold of five responding at a lower signal level than a threshold of one. With Persistence at 5, and Threshold at 5, a 1s-length tone 40dB below the clipping level can trigger the unit if repeated several times at regular intervals. Decreasing to four will trigger with a -38dB tone, and so will a Threshold of 3, it just takes a little longer. Curiously, with the threshold set to both 1 and 2, the required signal level is -34dB. This criticism aside, the *ADF-2400* can pick out recurring feedbacks quite reliably.

Other performance matters

Convertor linearity was tested and found to be virtually flawless down to -90dB referenced to maximum input level; within 1dB at -100dB; rising to +3dB in error on channel A and -1dB in error on channel B at -110dB. This is a reasonable performance.

Input stage clipping occurred at +28dBu, with output clipping at +28dBu into 100kΩ, or +27.2dBm (600Ω load). The clipping level of the unit as a whole is set by the front panel CLIP LEVEL ADJUST pots. Their range is from +28dBu to -10dBu, coping with the whole range of professional to domestic operating levels. The controls keep the output stage following in trim, giving an overall input to output gain of 0dB at all settings with a worst case error of about 0.2dB.

At an input level of -10dB, there is a 4dB degradation of dynamic range. The manufacturer's specified dynamic range of >100dB when the gate was closed could not be achieved.

Distortion levels of all types are comfortably low—at least to specification.

The front panel BYPASS switch operates true bypass relays, passing inputs direct to outputs. These also work in the event of power fail. ▶

Build quality

The outboard mains supply seems well made, and does not get overly warm, unlike others we have seen. The keyhole on it makes hanging out of the way reasonably easy. It is a double-insulated design without any earth on the power cable. There were no UL or other approvals in evidence—I thought that was the main point of using such units, the availability of such approvals. This is an AC-only power supply, the rectifiers and regulators are inside the main chassis.

The front-panel legends are clear, and the user interface fairly easy to use, though a bit clunky in my opinion. You can figure it out easily enough without the manual (except for Threshold and Persistence). The operative who assembled the unit has daubed some silicone rubber compound onto components likely to suffer vibration fracture, but this appears somewhat haphazard and could be improved to ensure durability. Internal construction is tidy and clean, with full component legends on all four PCBs. Audio connectors are Cannon plastic XLR types, with incoming low voltage AC power on a latching 5-pin DIN. There is ample room in the unit for future add-ins, for which access openings have been provided.

The lid is held on by Allen-head screws to minimise tampering. With this weight of equipment, the 2.5mm-thick front panel will provide substantial enough rackmounting support in most applications. The covers and sides are steel with a black powder-coat finish.

There are GROUND-LIFT switches on the back, one for inputs and the other for outputs. These do not, I think, work as intended. If both are lifted, the shells and pin 3 of all XLRs float. If output is switched to grounded position, shells and pin 3 of output XLRs cease to float. If inputs are grounded, Input A XLR shell continues to float. Indeed this floats all the time. These connections to ground are only via the chassis, there is never any connection to mains earth. Safety-wise this is okay due to the external mains transformer, but on occasion purchasers may have to earth the chassis especially.

The range is from 1 octave down to 1/100 octave. These digitally generated filters are much narrower and more stable than conventional analogue filters

Summary

The *ADF-2400* and its sibling do what they set out to do well. The use of what appear to be 16-bit convertors may limit their use in some systems, but this is compensated for somewhat by the front panel clip-level adjustment —allowing best use to be made of the existing dynamic range. The weakest link is the user interface, which could do with some further thought. This would be a useful unit to place at the sound-mix position for live sound shows, and in the racks in railway stations and other speech announcement locations without operators. For broadcasters in audience attended TV studios, it will also prove useful. And, for those with a bit of spare cash, it will be a handy equaliser and denoiser for use in sound-effects-type applications. ■

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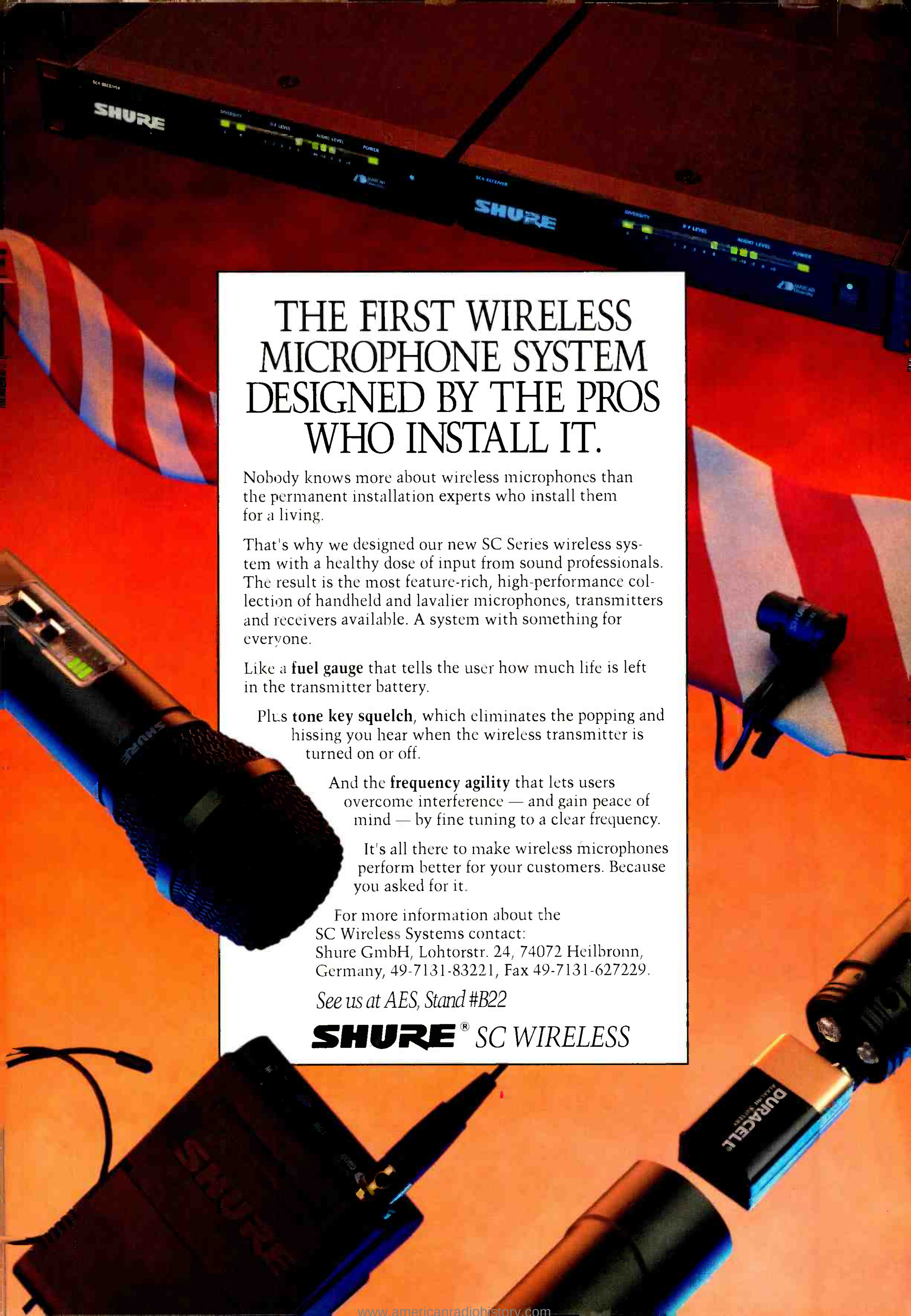
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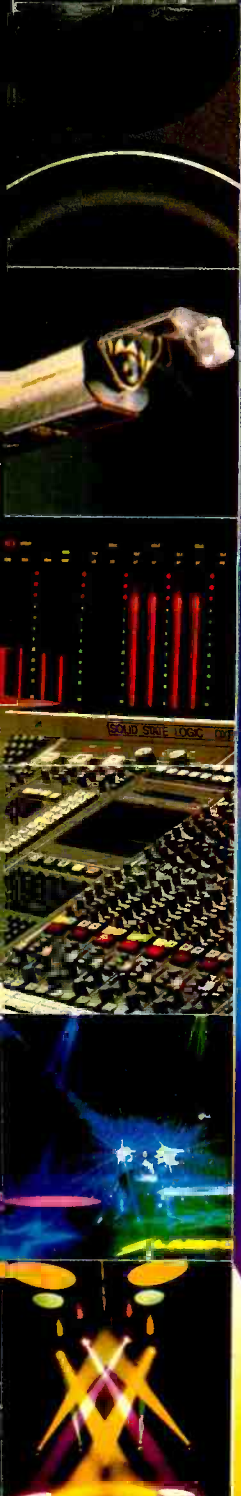

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Present your ID—CD blanks and counterfeit Churchill again

While visiting Kodak's HQ in Rochester, in upstate New York, I got a 'from the horse's arse' update on the company's policy on blank CDs.

Although Kodak bought early batches of dye-polymer blank CD-Rs from Taiyo Yuden in Japan, all Kodak's blanks are now made in Rochester. The only difference between blank *Photo-CDs* and blank data CD-Rs (suitable for audio recording), is in the code pressed into the pregroove.

All types of Kodak blank use *Infoguard*, although *Photo-CDs* do not carry the *Infoguard* label. *Infoguard* is, in fact, a clutch of features. The first, and most important of these, is stability. The laser in the recorder makes marks by chemically altering the dye in the disc. If the dye can be altered by unwanted light or heat sources, or naturally reverts to its original state, the recording is lost.

While being gracefully tactful about the competition, Kodak claim a more stable dye composition than other blanks on the market. This stability follows from using a dye which is stabilised by an added chemical which can itself break down in time.

The dye is tuned to react to light only at the infrared wavelength of 0.78 microns, which is the standard for the laser in the disc writer. Other light frequencies, for instance wide-band sunlight, do not switch the colour of the dye.

The reflective surface is gold, as on all write-once discs, and gold is more prone to scratching than aluminium. So *Infoguard* puts a matte scratch-resistant coating over the lacquer layer which protects the gold. The clear surface is claimed to resist finger prints, too.

Each disc is digitally serial numbered, with production line identification and date of production.

There is also a machine-readable bar code, which the reader-writer can recognise and compare with a similar code encrypted in the data stored in disc. This allows the player to check whether a disc is a genuine first-generation copy or a second-generation dub.

Blank Photo CDs are only sold to Photo-CD processing centres, and they will not work on a data/audio recorder anyway. Quoted price in the USA for 63-minute data-audio blanks in $25 each and for 74-minute blanks $27. List price for Europe is a converted $28. I asked Larry Zimmer, Kodak's Worldwide Manager for CD Media, why 74-minute blanks cost more and why have they been late coming on the market.

'When the CD system records for 63 minutes on a disc (blank or pressed), the linear writing speed is 1.4 metres/second,' he explained. 'For 74 minutes, the linear speed is dropped to 1.2 metres/second. The writer-reader slows the disc speed down automatically, depending on the type of disc loaded. The instruction is in the pre-groove. slowing the speed reduces the size of the data marks on the disc and thus leaves less margin for error in the optics, and the chemistry of the transition. The dye has to heat and cool with more precision.

Zimmer also explained why Kodak now certify disc writers for use with blank discs, especially 74-minute discs. The writer automatically tunes the power of the laser to match the blank media, by making a short test recording on every fresh disc, and reading the test before starting a full recording. But this self-tuning is not enough to guarantee error-free recording for all combinations of blank disc and writer.

Zimmer reminded me how, in the early days of CD-R, Fuji discs only worked with Yamaha writers and Sony writers were tuned to work with Taiyo blanks. Compatibility is now wider, between different brands of writer and blank discs, but Kodak still see certification as necessary, especially for 74-minute discs.

It is easy to forget that we are still at a very early stage of CD-R development, so I asked Zimmer the question everyone wants answered. When will we see erasable dye blank? Will we ever see one?

'I have my doubts,' he said. 'There are inventions to be made before that can happen. Really, to talk about erasable CD, *and* compatibility with the Red and Yellow Book standards, is an oxymoron.'

Although Kodak are cagey about future plans, the company's line of thinking is pretty clear. If there is to be an erasable disc, then it makes sense to build it on a completely new format that uses CAV (Constant Angular Velocity), like a computer disc, with formatting in sectors to speed search access and retrieval. The CD standard is based on CLV (Constant Linear Velocity) which is fine for storing both sound and pictures, but slows data access to speeds which are snail-like by computer operating standards.

Although we are unlikely to see an erasable-compatible CD in 1994, a radical development in the write-once blank market is now on the cards. I know that at least one CD pressing plant now plans to shake up the whole pricing structure by moving into mass production.

The company behind the planned moved is *not* Kodak. But it is already very big in the CD-pressing business. As the prime mover puts it: 'Blank CD-Rs are only pressed CDs, with a dye coating. We can make them for 70p and sell them for £1.50. The current costs are obscene'.

Do you remember the Churchill-Shelley issue? Was Winston really Norman? Churchill made his wartime speeches in the House of Commons and there was at that time no provision for recording or radio relay. So the story goes that actor Norman Shelley took time off from playing Winnie the Pooh and other childrens' radio parts, went into a recording studio and impersonated Churchill for discs which were then shipped to radio stations round the world.

Later Churchill's speeches were released on LP and cassette by Decca, and when Decca folded, the material was licensed to EMI Records.

First Decca, and later EMI, have dismissed the idea that some of their recordings are impersonations. But significantly neither Decca nor EMI have ever provided a copy of their archive documentation for the master recordings.

I have pursued the story, to the undisguised irritation of Decca and EMI, because I spoke personally with Shelley 20 years ago and he told me how the recordings were made. But I did not tape our conversation, and Shelley is now dead. So I have had no way of proving what the actor told me. More accurately, I have until now had no way of proving what Shelley told me.

Just before Christmas, BBC TV produced a rather curious programme, in conjunction with BBC Radio. *Arena*'s Radio Night 'butterflied' over a wide range of arty-crafty topics, but at one time settled briefly on an investigation of the human voice. The programme also looked at impersonations. The BBC researchers had read what I had written about Shelley-Churchill, and asked for more information. I told them how I had talked to Shelley after seeing him on a TV chat show in the early-1970s.

The BBC have now tracked down the original programme. It was *Just a Nimmo*—a chat show hosted by British actor Derek Nimmo. On 11th February 1974, Norman Shelley was the guest. The BBC found a videotape of the programme, and we can now see and hear Shelley, in person, recounting the story that he told me.

Faced with this new evidence it is not time for EMI to release the documentation that came from Decca with the 'Churchill' archive tapes, or for EMI finally to admit that the tapes came from Decca with no reliable documentation? ■

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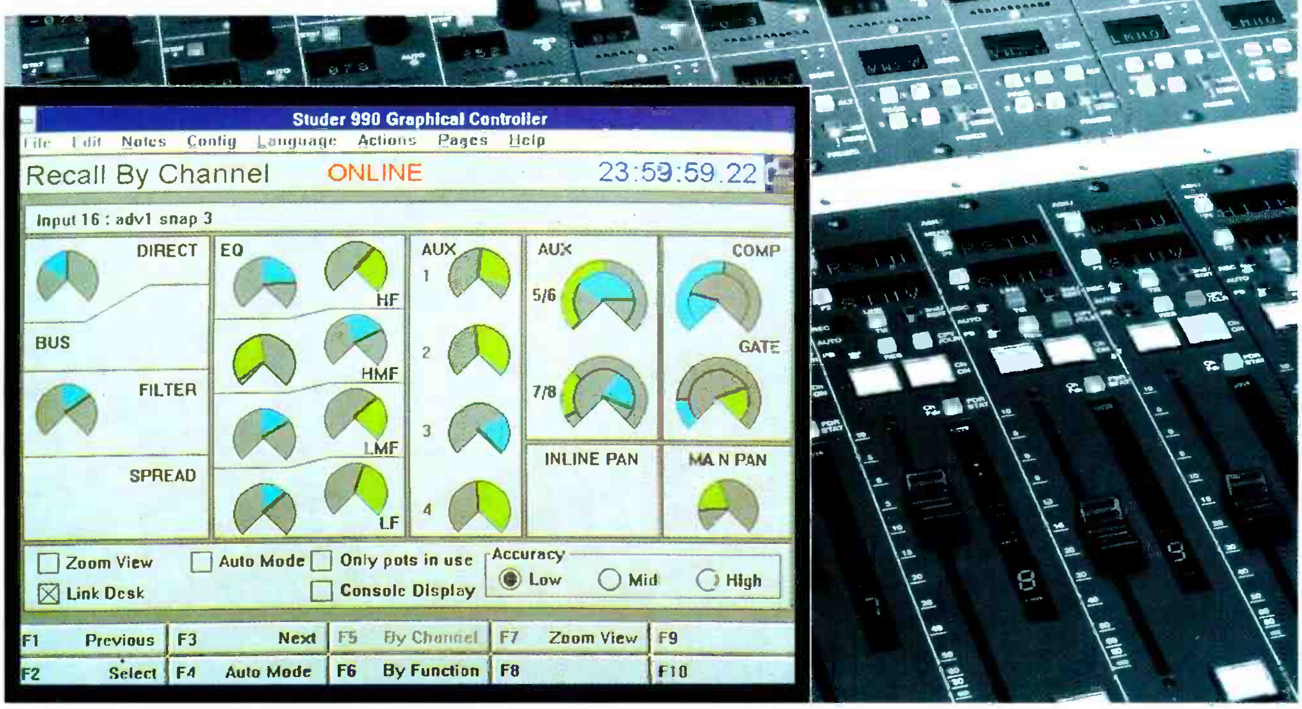

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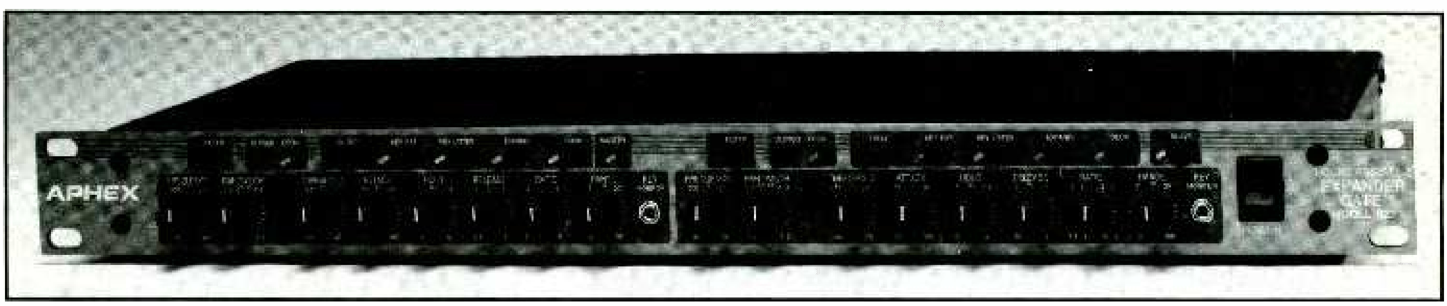

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