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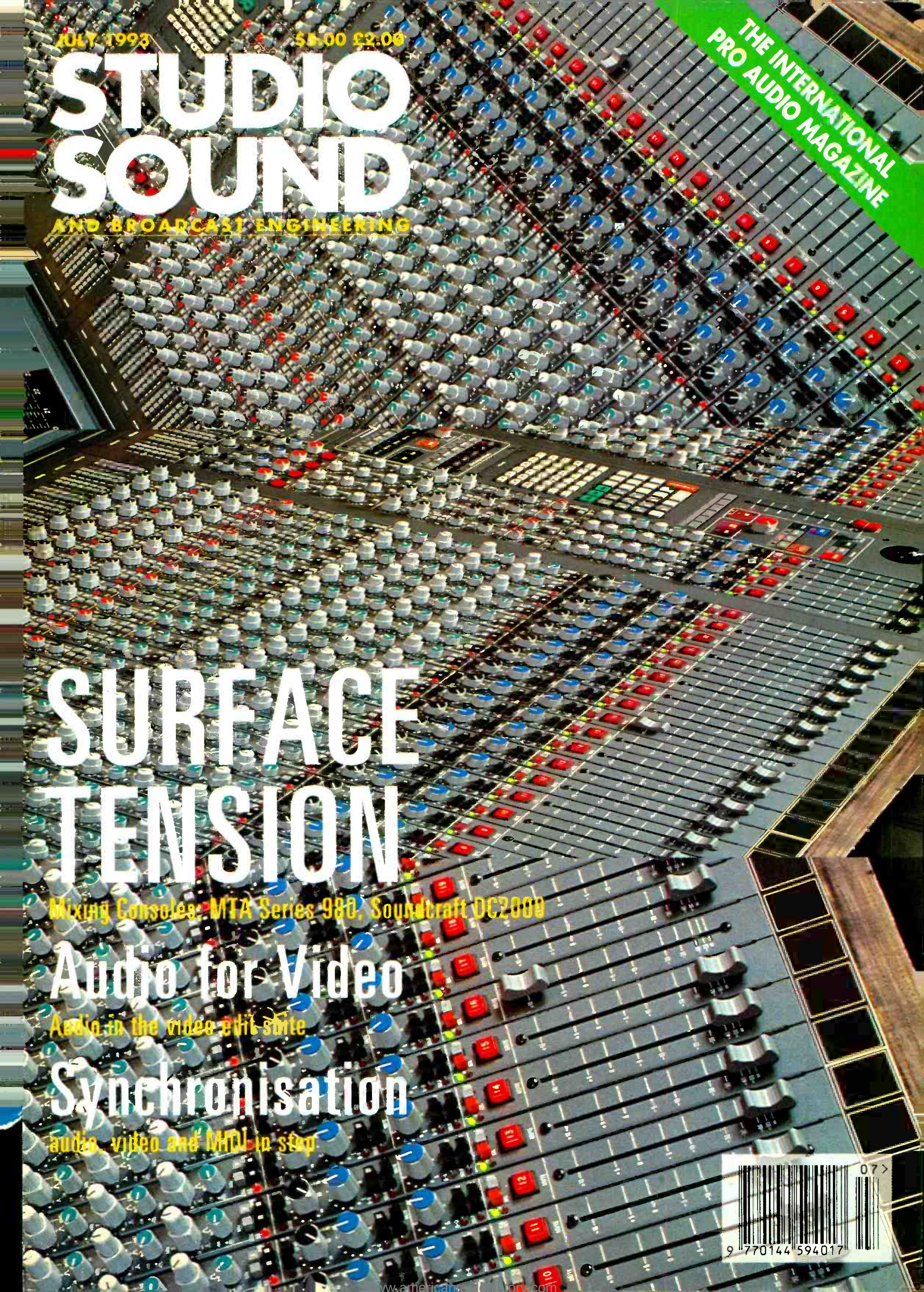
STUDIO SOUND

AND BROADCAST ENGINEERING

THE INTERNATIONAL PRO AUDIO MAGAZINE



SURFACE TENSION

Mixing Consoles: MTA Series 980, Soundcraft DC2000

Audio for Video

Audio in the video edit suite

Synchronisation

audio, video and MIDI in step

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G PLUS CONSOLE SYSTEMS

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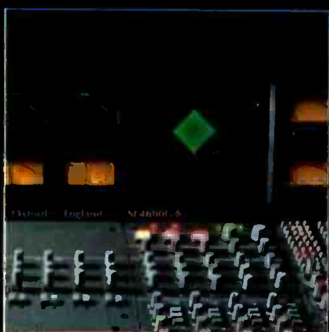
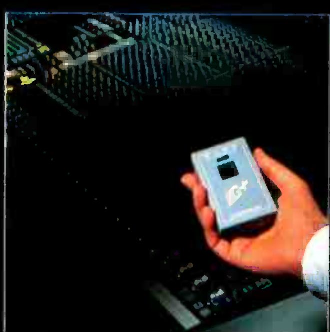


Audio phase scope provides permanent display of amplitude and phase relationship of left and right stereo signals



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STUDIO SOUND

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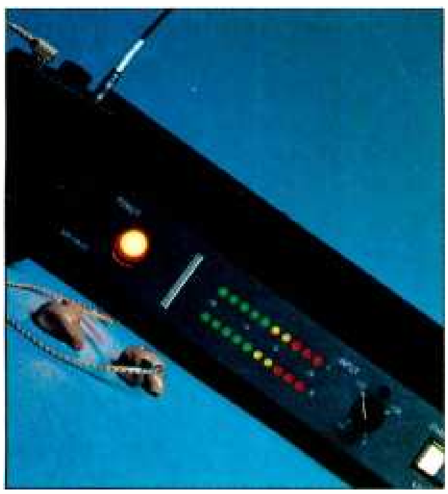

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Nothing comes close...

ADVANCED FEATURES

- Digital mixing and track bouncing
- Time compression and expansion
- Dialogue sift capability
- Automation of gain, pan and varispeed
- EDL autoconform
- Audio 'fill' between mark points
- Hard key control

SoundNet
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- Off-line back-up/restore
- Project sharing
- Fast project changeovers
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ScreenSound

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- Sample rate conversion
- 8 analogue inputs and outputs
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- Automatic cataloging of multiple takes
- Combined keyword searches for sound clips
- 8 audio 'reels' + 8 audio'bins'
- Track and clip slipping
- Variable crossfades
- Copy, review and cycle of sound clips
- Edit and mark points
- Edit and fader groups
- Sound design capability
- 4 autolocate memories
- Varispeed

MAGNETO-OPTICAL LIBRARY

- Re-usable, removable media
- Library of pre-recorded sound effects
- Listen directly off MO discs
- Combined keyword searches for sound clips
- Local or central MO drives

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- Hard lock to film or video
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- Locks to film chain for dubbing
- Multiple ScreenSounds can slave to one Master for multichannel playback

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AND BROADCAST ENGINEERING

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Language is a Virus

Train spotters. We all love to hate them, whether they are the genuine anorak-clad inhabitants of railway stations or some subspecies dedicating its existence to the collection of trade show brochures or Royal Family trivia. Yet one of the most detestable traits of these derided individuals represents an indispensable tool to this, as well as many other industries. It is the supreme ability to rattle off engine numbers and classes, to talk in generally indecipherable terms about their chosen passion; it is jargon.

Apart from having formed the basis of countless jokes, jargon actually fulfils a recognised social function, allowing individuals to identify with a group—and in doing so, set themselves apart from the crowd as part of some informed élite. But is is also an essential element of communication, without which any kind of progress is impossible.

As any area of research uncovers new facts or adopts fresh ideas, it is required to label them is such a way as further discussion, and further progress, is possible. Should it fail to do so, further progress is rendered impossible. Pro audio is certainly no exception to this rule. Admit it: you occasionally enjoy being able to baffle the hell out of Joe Public with your talk of phantom power, sample rates, AC701s and data fragmentation—you are an anorak. We all have to be, and it is not quite as easy as it sounds.

The adoption of standard terms of reference is hindered by the adoption of trademarks. Anyone who uses, for example, terms like Harmonizer, Emulator or Direct-to-Disk without due discretion is likely to attract unwelcome attention from Eventide, E-mu Systems or NED. And this is after such systems have been officially marketed; to enable communication between competing parties in the development of a new area of technology obviously presents even greater obstacles.

Take the heated exchanges between Deutsche Grammophon and their critics concerning the 4D Recording System: without taking sides, I would suggest that, in part at least, this a product of progress outstripping our ability to comfortably discuss it rather than any fundamental disagreement over the technology itself. Where, for example, is the commonly-agreed technical description of a 21-bit A–D converter?

A few months ago, in the wake of our review of the dCS *900B* A–D converter, the company's Paul Maddox raised the issue of standardisation in measurement. He was not taking issue with the test results (nor was he anticipating the 4D debate), he was simply drawing attention to the inconsistencies between dCS' own testing procedures and those adopted by our reviewer. It is a further example of how our inability to communicate fully is hindering our joint progress.

Elsewhere in this issue, Francis Rumsey discusses the issue of the apparent discrepancies between equipment measurement and listening tests. Apart from being another hotly-contested issue in the audiophile's hi-fi world, this actively affects the pro audio perspective on how we create and control sound.

Such examples are easy to find, yet widely disregarded. Perhaps, in a world short on reason, it is a sobering measure of our real eagerness to progress. ■

Tim Goodyer

Cover: Amek *Hendrix* at Mute Studio, London. Photography: Phil Dent

CS2000

Digital Control Studio System

The CS2000 expands the family of Euphonix studio control systems. Featuring state-of-the-art digital control technology, the CS2000 suits applications from commercial music studios to large film dubbing theatres.

The CS2000 provides Total Control of the mix environment. Total Automation™ and the SnapShot Recall™ system speed up the process of mixing, and allow for more creative freedom. SnapShot Recall resets everything in less than 1/30 second. Total Automation allows all controls and switches to be automated to code. The CS2000 reaches beyond the console with MIDI and a high speed interface capability to external effects devices, sequencers, multitracks, and DAWs.

The CS2000 has been ergonomically designed to give the operator instant access to all functions, with central assignability for operations such as EQ adjustment.

The system is fully modular and highly cost effective. Systems can be configured on purchase to suit specifications and budgets. Dynamics, additional aux sends, and film mix buses are just some of the options that may be added whenever they are needed.

The CS2000 is an audio mixing platform that will take a studio into the twenty-first century with a flexibility and power of control that has never before been available.

Euphonix

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International News

In-brief

● **APRS '93—cautious optimism**
At the end of the last day of this June's APRS attendance levels were verified at 5,458 over the three days, with an increase of **9%** in exhibiting companies and 32 new companies

● **Chrysalis put studio up for sale**
Wessex Studio, a veteran recording complex of 30 years standing is for sale. Its owners, the Chrysalis Group, have decided to put all their efforts into AIR Studios and feel Wessex would benefit from new owners 'who can devote 100% of its energies to continuing the historic recording of Wessex.' All enquiries should be directed at **Richard Huntingford. Tel:071 221 2213.**

● **Tannoy seek help with DSP**
Tannoy is searching for recording studios to help in its final phase into the world's first DSP Point Source Monitors. Selected studios will be asked to use Tannoy's DSP monitors throughout the full recording of commercial material and make subjective and objective comparisons against the same material recorded with conventional speakers.
Tannoy Tel: 0236 420199.

● **Ninth Wave counts on area**
Ninth Wave Audio is a new facility based in Birmingham, West Midlands. They will specialise in location recording and digital editing with Nagra and Sonic Solution equipment. The company is run by Tony Wass with 17 years of working with the BBC behind him.
Tel:021 449 3546.

● **Industry people movement**
DigiDesign has appointed Chas Smith as the Managing Director for sales in the UK. Smith was previously Marketing Manager for Roland UK's Audio, Visual and Broadcasting division. Allen&Heath have appointed Bob Goleniowski to head up their international sales operation. He had been previously at Harman UK

● **Unusual move from Wigwam**
Wigwam Acoustics have accepted an offer from Unusual Industries for the purchase of their staging company Staged-Rite Ltd.

● **OMF software kit now available**
The OMF interchange spec, created by over 100 industry leaders, is now publicly available as is a software integration kit. OMF Interchange addresses the need to exchange digital media including graphics, animation, still images, audio and video

Martin Appeal: Where were you?

Thames Valley Police have appealed to anyone that knew David Martin to assist them with their enquiries. The police have charged Colin James with the murder of Martin but are seeking additional evidence, as his body has not been found

The pro audio industry was shocked by Martin's disappearance late last year and the increasing probability since then that he had been killed. No-one is known to have seen Martin since the 29th December last year.

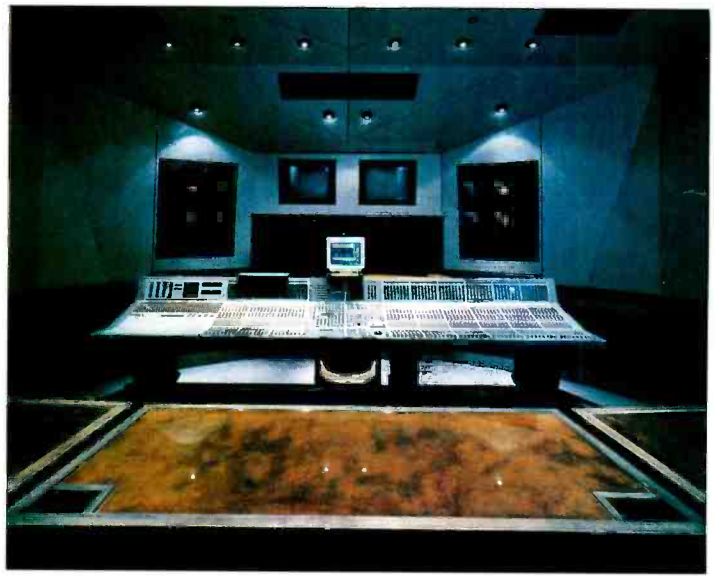
Police are asking anyone that knew Martin to answer, in confidence, four questions:

- What was your association with David Martin?
- Where and when did you last have contact with him?
- Did you have future plans with him? If so, what were they?
- Any other comments?

Martin Audio MD David Bissett-Powell said, 'As a mark of respect to him we would like all his contacts—even those who may have only briefly met him—to inform the police that they have not heard from or had any sighting of him since his disappearance.'

Bissett-Powell said that the police were seeking to establish that no-one had seen Martin since December in order to prove that he was dead. 'Not a day goes by when we don't talk about David—he was a great character and remains very strongly on our minds,' said Bissett-Powell. 'We will have a memorial service in three months. Without a body, we don't to be premature.'

Thames Valley Police ask you to reply with your name, address and telephone number to Dept. Supt. Blair, Incident Room, Aylesbury Police Station, Wendover Road, Aylesbury, Bucks. HP21 7BR. Tel: 0296 396170. Fax: 0296 394786.



Worktops finished in distressed brass are a distinctive visual keynote in the control room at Hitokuchi-Zaka Studio 3. Harris Grant Associates has earned a 1993 TEC nomination for its design work, as has Discrete Systems for the ***BOXER*** monitoring system shown above, complemented by RPG *Diffusor* and *Diffractal* acoustical treatment arrays

AIR six months on

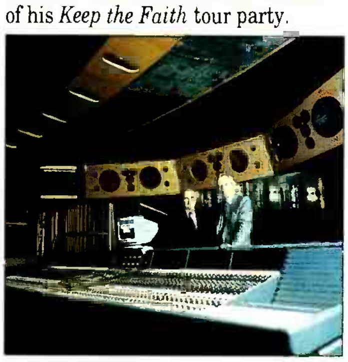
Air Studios new home at Lyndhurst Hall, Hampstead, has already received numerous plaudits from an international clientele.

Air's first clients were Dire Straits who were there to mix their live album. Mark Knopfler said of one of the mix rooms, '...it sounds better than any control room I've heard. I'm certain that Air Lyndhurst will be one of the world's premier recording facilities.'

Henry Mancini, one of the first orchestral visitors who was at Air to record the soundtrack for *Son of Pink Panther*, said , 'the acoustics were wonderful and I wouldn't hesitate to come back here.'

An Air spokesman said, 'The main hall at Air has been an ideal setting for both classical and contemporary music recordings; already the hall has staged several orchestral sessions, for example, with Sir George Solti and the LSO, the London Philharmonic and the Royal Philharmonic'.

Other clients include Gloria Estefan who recorded with full orchestra in the hall, and Jon Bon Jovi who used the venue for the end of his *Keep the Faith* tour party.



George Martin and John Burgess in Air's main hall control room with a Neve *Legend*

It was 60 years ago today

The German celebration of 50 years of stereo tape recording at the AES Berlin, overshadowed a somewhat greater landmark in the history of sound—the 60th anniversary of the invention of stereo as we know it. On June 14th 1933 Alan Blumlein, then working at EMI's research centre, was granted a patent for 'Improvements in and relating to sound transmission, sound recording and sound reproducing systems' which set the agenda for two-channel stereo.

Blumlein's work in audio started in earnest in 1929 when he was head-hunted by Isaac Shoenberg, of the, then, British Columbia Gramophone Company, to develop an electrical recording system that would allow Columbia to avoid paying royalties to Western Electric. Western Electric had so comprehensively patented the electrical sound recording system developed by Bell Labs in 1924, that it had the whole of the record and film industry paying royalties to them.

By the time Blumlein was officially

posted to EMI he had nearly completed the initial theoretical work on stereo and within a month he had applied for a patent which defined a 'binaural' stereo format that could produce a convincing horizontal sound stage using just two audio channels. What now seems an obvious thing to do, the concept of using just two channels went completely against the general thinking of the time. Up to this point experiments with 'stereo', which started as early as the 1880s, had centred on the use of a long line of microphones transmitting sound down multiple recording channels and then replayed through a row of speakers, replicating the position of the microphones.

Blumlein had also considered in the patent the option of using a third microphone to give height information predating the development of the Soundfield mic by 50 years.

But by this time EMI was putting increasing R&D effort into developing TV, so Blumlein transferred his effort where among other projects he designed the landlines and amplifiers for the first TV outside broadcast of the coronation of George VI in 1936. At the start of the war he moved on to the development of radar. Blumlein's death in an air crash 1942, while testing airborne radar, finished further development in stereo until the 1950s, when the LP resurrected interest in the idea for domestic audio systems.

Tim Frost

BBC for *D10*s

BBC Transcription has become the first facility to take delivery of Fostex's new *D10* professional DAT recorder, following the machine debut at last month's APRS show in London.

The *D10* is designed specifically for fast, 2-machine DAT editing and provides instant start and RAM scrubbing for fast cueing. (More equipment details in New Products on page 11).

Pete Fresney, Resources Manager at the BBC said: 'It's the ability to edit in and out on the *D10*, to perform simple assemble and insert edits quickly and easily, that's important to us.'

The first project to be completed was a Happy Mondays concert with announcements on the *D10*.

Fostex UK. Tel: 081 893 5111.

PARIS SATIS
LE SALON EUROPÉEN DES TECHNIQUES DE L'IMAGE ET DU SON
POUR VOIR ET ÉCOUTER L'INNOVATION

Show Report

The French have rarely been short of ideas, and in what may hopefully be seen as the spearhead of a growing trend, the SATIS and APA organisations combined their respective exhibitions into a single event—SATIS-Audio Pro 93.

Held at the Port de Versailles exhibition centre between the 25th–28th May, the joint exhibition met with mixed reactions from the French pro audio community, but a meeting of the APA (loosely, the Association of Audio Professionals) immediately after the show confirmed that the experience would be repeated in 1994, together with modifications to the layout of the exhibition area in order to better integrate the audio and picture worlds.

SATIS is essentially a show for pictures—either television or film —with a touch of audio. However, as the two disciplines merge into one another, the 'Espace Audio' made a lot of sense and allowed visitors to have a more global approach to production techniques.

For most of the exhibitors, it was a confirmation of existing products already launched, although there were several new items worthy of interest.

SCV Audio are now 'Made in England' (who said Europe does not exist?) since the takeover of Hitech Ltd. to become SCV Electronics, thus marking an international commitment—and lower prices!

New from **SCV** were additions to the range of modules for the Universal Buffer range, including a video distribution amplifier, bargraph meters, 'Voice-Over' unit and a distribution amplifier with a +4dBu balanced input and 4 'Hi-Fi' level outputs. Also new is the *NEQ%* Triple Programme Equaliser which has four main modes; EQ of three independent stereo sources—EQ of one stereo source to three separate outputs —three stereo sources mixed to a single stereo output—a stereo source to a stereo output with memories for three different EQ and level settings.

Audio Follow introduced the new *CONTACT* range of digital audio systems for broadcasting, with particular emphasis on 'live' shows. Using proven technology and equipment from Audio Follow, *CONTACT* systems can be configured for a variety of needs from broadcasting, recording, on-air management and network linkups.

IDT introduced the *FM Sound Design*—a computer-controlled analogue multiband processor for FM radio. The stereo unit allows the setup of a 'sound signature' for radio stations and features include an input filter with high and low-pass functions and 4-band parametric equaliser.

Publison introduced the *Oceane II*, a 4-channel recording-editing system using optical discs which can be used as a stand-alone system or with the Publison 8 or 16-channel workstations. *Oceane II* can be used in on-line situations, to a video editor and as a real-time 4-track editor.

The Publison workstations have also received software updates, with one of the most interesting being a biphase interface which allows the workstation to synchronise and control a film dubber or editing table as it would a VTR.

The only sector of the industry that felt a bit out of its depth was sound reinforcement though as the French audio world gets used to the new formula, more interested visitors will no doubt attend the exhibition.

However, this did not prevent the launch of a new office selling Clair Bros audio products and installation.

The SATIS Show is not only an exhibition. The lecture programme, while being mainly video and film-oriented, did include subjects such as the *Evolution of Audio Technology* and *The Integration of Audio-Visual Techniques*. The 1994 exhibition will certainly see a wider range of subjects in the A-V field being dealt with in both the lecture and workshop programmes.

With final attendance figures in the region of 15,000 visitors, an increase of 10% over SATIS 92, and a 20% increase in exhibition space with 100 new exhibitors, the SATIS-Audio Pro exhibition has become one of the most important shows in Europe and looks set to be a successful marriage.

Terry Nelson

Contracts

● **Sweat, Passion and the 4K** Recent orders for SSL's *SL 4000 G Series* console include Keith Sweat's personal studio; Michael Bolton's Passion Studios; The C&C Musak Factory and Walter Afanasieff.

● **Sophie *Jaded* at The Aquarium** Producer Stephen Lipson, currently working on Sophie B Hawkins new album, is using his new Soundtracs *Jade* console.

● **Baumgardner uphill with NRG** NRG Studios in Hollywood, owned by producer Jay Baumgardner, has unveiled its new Studio II with design by studio bau:ton.

▼ **Digital DAWNs in Glasgow**

Dave Murricane with his new *DAWN II* system

The Tyrell Corporation has installed Scotland's first *DAWN II* digital audio system. Glasgow-based Murricane and Murricane have responded to the call for more digital audio north of the border with this purchase.

● **'Eat cake,' says Dead Aunt Thelma** Russ Berger Design Group recently finished the design of a recording studio in Portland, Oregon. Dead Aunt Thelma's Studio was named while owner Scott Parker and others were enjoying a construction break and eating pound cake made from his deceased great Aunt Thelma's favourite recipe.

● **Opera leads with digital desks** Two of Europe's leading opera houses have installed *Logic 2*s from AMS-Neve. Opera Leipzig has celebrated its 300th anniversary this year by installing a *Logic 2* in its control room, while Finnish Opera has bought two.

● **OB full of *Viennese* antiques** Carlton Broadcast facilities, who have been commissioned by the BBC to make the next series of *The Antiques Roadshow*, have chosen a 24-input *Vienna* console from Soundcraft for a new OB truck

● **Packhuskajen sounds like ...** A 39-unit Meyer sound system has been chosen for Gothenburg's new Opera House in Packhuskajen.

CHOOSING THE RIGHT COMPUTER BASED AUDIO EDITOR CAN BE A NIGHTMARE

Photo: Ewing-Reeson Photography

- Windows 3.1* on 486 host computer
- Rapid graphical editing
- clear user interface
- local SCSI drive fast audio access
- all crossfades calculated in real time
- fully non-destructive, sample accurate editing
- up to 8 track playback and real time mixing
- unique Trim Window allowing real time adjustment of audio
- full SMPTE timecode support with chase and trigger lock
- accepts external word clock
- jog and shuttle scrub modes
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- 32 bit floating point architecture
- peak holding metering

Here at Studio Audio, we had many sleepless nights when our customers began asking us for an off the shelf solution to professional audio editing. Then we woke up with SADiE™, a dream of a Disk Editor.

For a start **SADiE™**, runs on a PC, so you get much more computer for far less money. It has a fully functional Windows 3.1* user interface with all features mouse selectable. Our hardware platform is already found in many current audio and video products available from some well-known manufacturers. That means it's been tried and tested out in the field. Studio Audio believes in giving each customer personal service - you'll always get rapid customer support on our helpline. And because we're committed to **SADiE™** and our customers, we'll give you the first 12 months software updates free of charge. Put an end to the nightmare now!

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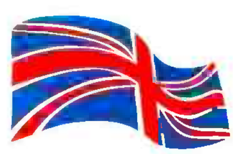
SADiE DISK EDITOR



BRITISH INNOVATION

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P&G MIDI Control

Penny & Giles have introduced a new MIDI control system, the *MM16* MIDI Manager.

Designed to aid the programming and management of complex MIDI setups, the MIDI Manager offers a fully equipped control surface. The front panel features 16 endless-belt faders with integral LED displays, a data wheel, four data entry keys, 16 assignable keyswitches and a further 24 dedicated push buttons.

Available in two options—19-inch rackmounting or as a desktop console—the new system enables rapid real-time access to variable parameters in a MIDI installation, simplifying the task of programming and controlling changes to synthesisers and outboard equipment.

Capable of operating as a stand-alone generator of MIDI codes, the Manager can also Merge, Modify or Replace incoming data. Control assignments are stored in up to 50 Patches and data values in 100 Snapshot memories, while the internal battery-backed memory can be extended by means of a plug-in RAM card or stored externally via the serial interface. The unit can be designated as a system timing controller and tape recorder-style transport keys are provided for MIDI machine control.

Penny & Giles Studio Equipment Ltd, Blackwood, Gwent, Wales. NP2 2YD. Tel: 0495 228000. Fax: 0495 227243.
US: Penny and Giles Inc, 2716 Ocean Park Blvd, Santa Monica, CA 90405. Tel: +1 310 452 4995. Fax: +1 310 450 9860.

Fostex *D10* DAT

The latest DAT machine from Fostex joins a popular team. The *D20, D20B* and *PD2* machines have all found favour with their intended markets and indeed were first with a recognisable time code standard.

The *D10* provides all that is needed for professional use in music recording and broadcasting use. Advanced jog-shuttle techniques use built-in RAM (4Mb x 2) which allows a constantly and immediately available stereo scrubbing facility. Other features include a full complement of Start ID, Program Number, Skip programming and 100 memory location. The *D10* also has two option slots for later expansion of spec.

● Fostex have officially launched their *ADAT* standard eight-track digital recorder, the *RD-8*. They have licensed the Alesis *ADAT* standard and produced a similar machine, apart from the all-white of the *RD-8*'s casing. (The *RD-8* is however fully compatible with the Alesis unit).

Features include up to 100 locate-memory points; a Track Slip function up to 170ms; built-in MIDI interface; and on-board chase synchroniser allowing synchronisation to a master time code with any offset.

UK: Fostex UK, Unit 1, Jackson Way, Great Western Industrial Park, Southall, Middlesex, UB2 4SA. Tel: 081 893 5111. Fax: 081 893 5237.
US: Fostex America, 15431 Blackburn Avenue, Norwalk, CA 90650. Tel: +1 310 921 1112. Fax: +1 310 802 1964.


A Neumann mic for under £1000—its here!

Neumann's *U87* sibling

The Neumann *TLM 193* is a new large diaphragm, double membrane, cardioid condenser, microphone. The microphone's price is set at under £1000 and is designed on traditional Neumann lines, the *TLM 193* features a full metal casing and grille; shaped somewhat like a smaller *U87* the new mic uses standard 48V phantom power and no special power supply or connectors are required.

Dynamic range is claimed at 130dB with a frequency response of 8Hz–20kHz.

UK: Sennheiser UK, 12 Davies Way, Knaves Beech Business Centre, Loudwater, High Wycombe, Bucks, HP10 9QY. Tel: 0628 850811. Fax: 0628 850958.

Rackmount DMP

Yamaha have introduced the *DMP9* digital mixing processor. The new mixer is the latest in the *DMP* series which started with the *DMP7*, but offers new operational facilities, improved audio quality and greater DSP power, some chips are the same as the larger *DMC1000* console.

The design comes in a 3U-high rackmount and is available in both eight and 16-channel variants. The *DMP9-16* features eight front panel 'modules', each with two sets of inputs, configurable individually or in stereo pairs. Each module is equipped with rotary level control and 8-point LED input indicator, channel on-off buttons, assignable parameter select buttons, input trims (-20dB to +4dB) and a 96dB pad.

Probable markets for the mixer would be keyboard rigs, MIDI project suite submixing, post, A-V, location and mastering applications.

UK: Yamaha-Kemble Music UK, Sherbourne Drive, Tilbrook, Milton Keynes, Bucks. MK7 8BL, UK. Tel: 0908 366700. Fax: 0908 368872.
US: Yamaha Corp of America, 6600 Orangethorpe Avenue, Buena Park, CA 90620. Tel: +1 714 522 9011. Fax:+1 714 739 2680. ▶

In-brief

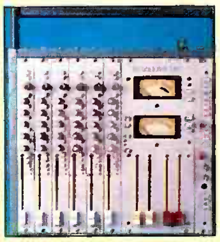
● **Amps approved for THX use**
After successful completion of THX Sound System testing procedures, Ramsa *WP-1000* Series power amplifiers were recently approved for application with THX systems. The series used a dual-voltage supply in a Class H circuit configuration which continuously evaluates the amp's input signal. When the power demands are low the circuitry switches in the low-level power supply voltage helping the cooling.
Ramsa. Tel: +1 714 373 7277.

● **Extended LF for Shermann**
Shermann Audio Systems have launched their *MX-208* nearfield monitor. A 2-way design with extended LF response, greater power handling than previous models and a sensitivity of 93dB.
Shermann UK.
Tel: 0686 622997.

● **Block vote for converter**
Yamaha have released details on the *FMC2, a* 2-channel digital audio signal format converter, configured as three separate blocks, simultaneously able to operate independently of one another. ◑ Block A: Yamaha Y2 format to AES-EBU, CD-DAT format ◑ Block B: AES-EBU format to Yamaha Y2 format ◑ Block C: CD-DAT format to Yamaha Y2 format.
Yamaha-Kemble UK.
Tel: 0908 249194.

● **Studer's new converter**
The *TASFC-20* is a asynchronous sampling frequency converter which adapts to incoming sample rates and generates a precision output at the desired sampling rate.
Studer UK. Tel: 081 953 3533

● **Portable charged performance**
From Soundcraft Broadcast Division comes the *LM1* portable audio mixing console for location work in film and broadcast. The *LM1* can be powered by either a DC input or form internal rechargable cells, providing up to 18 hours continuous operation from a single charge.
Soundcraft Electronics.
Tel: 0707 665000.



PRODUCTS

In-brief

● Cabot—for the ears to come
Cabot Safety have developed the Ultra Tech earplug especially for musicians, sound crews and others who work in close proximity to loud music. Ultra Tech claim to be able to cut out the damaging decibels without distorting incoming sound and can also help people who suffer from oversensitive hearing (hyperacusis).
Cabot Safety. Tel: 0625 878320.

● Less noise from portable mixer
Shure's new version of their *FP32* microphone mixer, the *FP32A*, offers all the original features but adds a dynamic range over 100 dB; 48V phantom and 12V phantom power; and pop-up pan pots; as well as 30dB less noise.
UK: HW International. Tel: 081 808 2222

● PDO announce CD-R media
PDO are now able to offer compatible CD-R media. The disks fully meet the internationally agreed Orange Book standard and in available in 63 and 74-minute formats. They are fully suitable for the storage of CD-Audio, CD-ROM, CD-ROM XA, CD-I, Photo CD and other CD-format information.
Philips PDO. Tel: +31 40 758454.

● Dyaxis enhance into Mk II
The Dyaxis II multichannel audio workstation has been enhanced by integrating autoconform software SMARTLOG, which accepts edit decision list from various video editing systems, plus multitrack editing, time compression expansion, Plug & Play M-O disc storage and scrub editing to picture.
Studer UK. Tel: 081 953 3533.

● Preamp solves bus shortage
Omniphonics audio resolver preamp, the 1U-high rackmounted *PRE 1*, provides nine stereo inputs that can be individually calibrated to a uniform level then switched to the record and-or monitor bus via logic controlled relays.
Omniphonics. Tel: 071 278 8216

Revox's 'Heavy duty' *MR-8*—only available in the UK

D&R shout about optional mutes

D&R will introduce at the Dutch 'Music and Harmony' exhibition in September optional soft-mute automation on its entire in-line recording console range. The *Orion, Triton, Marilon* and *Avalon* consoles will be getting new upgraded modules with integrated soft-mutes.

The automation package will be available as an option with an external remote control for the *Orion, Triton* consoles to store a maximum of 999 mute settings against MIDI time code.

The package is capable of storing mutes in real time while off-line editing is operating. The system is capable of controlling external effect devices through its MIDI channels and it will accept MIDI sequencer information to control all mute settings as well.

D&R say the system is attractively priced and is retrofittable in all D&R consoles with soft-mute switches. Delivery is expected in September 1993.
D&R Electronica b.v., Rijnkade 15B, 1382 GS Weesp, The Netherlands. Tel: +31 2940 18014. Fax: +31 2940 16987.

Soundtracs *Solitaire*

The *Solitaire* is the latest range of music productions consoles from Soundtracs. The in-line format of the new desk is available with 24, 32 and 40 I-O channels, with or without patchbay providing 56, 72 and 88 channels in remix, each with access to the four-band parametric EQ.

MIDI automated muting on all inputs, monitors, the eight auxiliary masters and the four stereo returns is a standard *Solitaire* feature with assignable dynamics on each input offered as an option.

Two on-board fader automation packages are available on the desk, the first a VCA-based version and the second a moving fader option. Soundtracs anticipate delivery to start in September.
Soundtracs, 91 Ewell Road, Surbiton, Surrey. KT6 6AH. UK. Tel: 081 399 3392. Fax: 081 399 6821.

Sound Impression

Asystem have announced the release of *Sound Impression v3.5*, multimedia sound software for *Windows v3.1*. When *Sound Impression* is combined with a suitable *Windows* compatible sound card, your PC is transformed into a digital audio and MIDI production system, design for audio authoring. **Digital Music. Tel: 0703 252131.**

B&K mic strings

The new Bruel & Kjær *4021* instrument microphone uses an unique thick-film preamp with SMD transistors, allowing the use of a *4011* capsule at just a quarter of the length of the standard mic.

The *4021*, which is fixed via spring clips available in different sizes, can be used to close mic instruments from strings to sax to drums.

When miking a stringed instrument, the *4021* would be fixed underneath the strings of a violin, behind the bridge, and pointing towards the violinist to prevent pickup of extraneous bowing noise.

The mic is appropriate for television applications because of its compact size and near-invisibility, but has the ability to handle high SPLs without distortion.
Danish Pro Audio, Hejrevang 11, DK-3450, Copenhagen, Denmark. Tel: +45 4814 2828. Fax: +45 4814 2700.
US: TGI North America Inc, 300 Gage Ave. Unit 1, Kitchener, Ontario, Canada, N2M 2C8. Tel: +1 519 745 1158. Fax: +1 519 745 2364.

Revox launch *MR8* console

Revox have launched the *MR8* console to the UK market. This 8-channel unit is designed for heavy duty use, appealing to a wide range of users in the broadcast, video edit and A-V industries.

The console has been designed to require minimum training and familiarisation by user; for example the meter section is offset to the right where it is not concealed by the microphone. The *MR8* is fully customisable. Facilities include a phantom 48V supply, bus coupling for input expansion and a remote channel control (VCA interface) is offered on user-designated channels. A fader start and comprehensive monitor section is included.
Revox Division, Studer Revox UK Ltd, 1 Berkshire Business Centre, Berkshire Drive, Thatcham, Berkshire, RG13 4EW, UK. Tel: 0635 876969. Fax: 0635 876969.

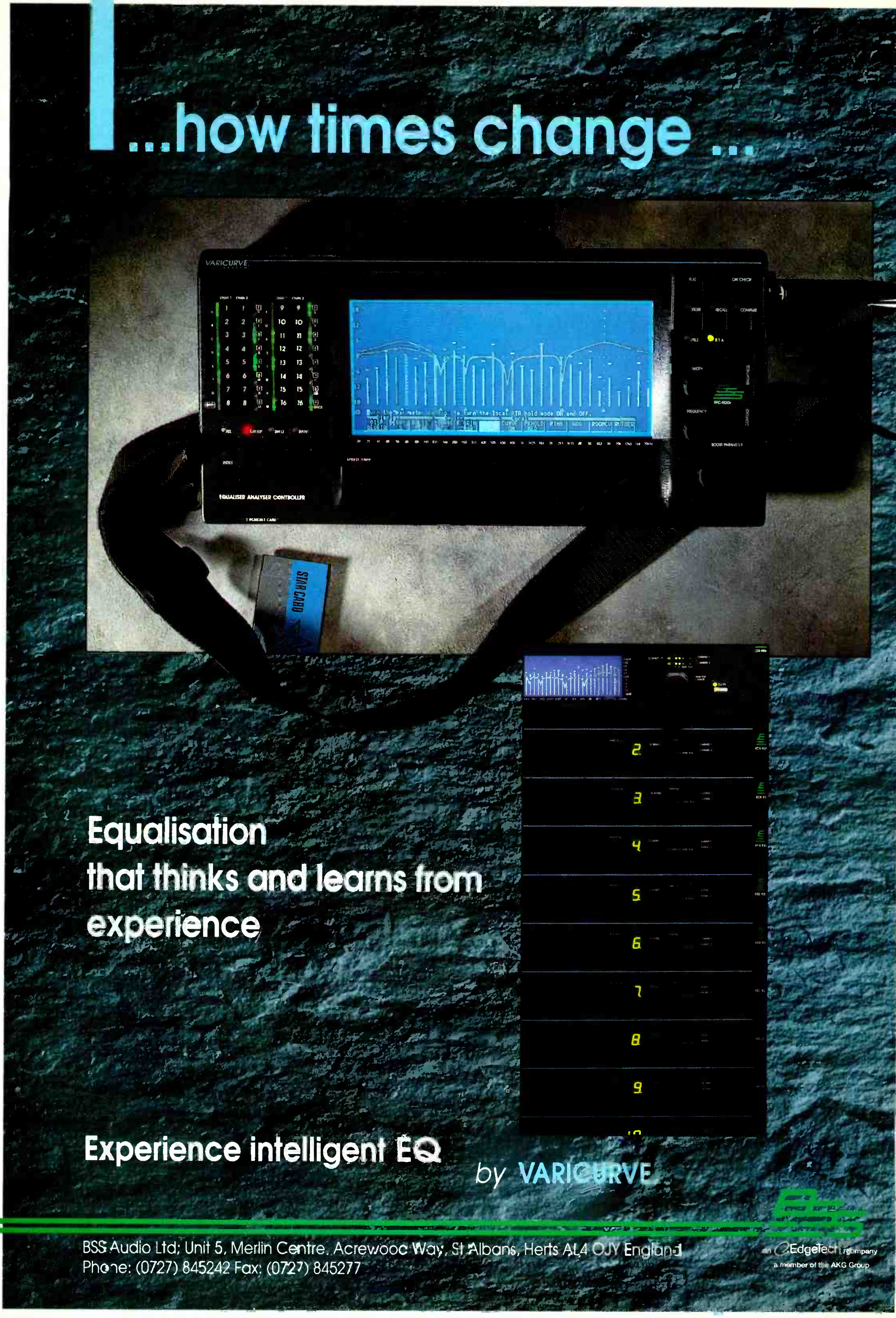
Drake's digital mixer for video

The *DMX1000* is a compact digital audio mixer designed for video postproduction applications.

On the input side there are 20 digital or analogue sources, expandable with a router interface. On the output side there are four mix buses, four monitor buses and two effect send buses.

An edit interface is provided using ESAM II protocol, and timeline editing gives a flexible approach to mixer setup changes.

Other features include a seven-frame delay; 20-input channels (either AES-EBU or analogue); EQ on every channel; effects send buses; compression-limiting and gating on every channel; on-line and off-line operation.
Philip Drake Electronics, The Hydeway, Welwyn Garden City. AL7 3UQ, UK. Tel: 0707 333866. Fax: 0707 371266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a member of the AKG Group

In-ear monitoring courtesy of the *Radio Station*

Radio Station

While not exactly a new product, the *Radio Station* in-ear monitoring system has undergone some changes at base with manufacture of the system now being handled by Garwood Communications in the UK. The system, which has found favour with leading artists like U2, Peter Gabriel, Genesis, David Bowie, Erasure, Michael Bolton, Luther Vandross and Kenny G, is an unusual although obvious approach to the matter of hearing what you are doing in performance.

As stages get bigger and more complex in line with venue size the demands placed on monitoring systems and their operators have also increased. The *Radio Station* simplifies matters by sending the performer's monitor mix via a UHF radio link to a belt-mounted receiver the size of a cigarette packet. The earpieces are connected to this and the artist has control of volume on a single pot. Needless to say its stereo and frequencies can be programmed to choice.

In use the benefits are obvious, the ear plugs attenuate outside SPLs for the wearer while giving a more selective and position independent limited feed. For singers worried about pitching, the in-ear approach is more intimate and more akin to shoving a digit in your shell-like. It is arguable whether the system will actually reduce on-stage levels to a whisper as it can naturally be supplemented by traditional wedges and side-fills and there is always the backline and, most importantly, the drummer to contend with. However, feedback becomes a thing of the past and one of the real bonuses of the *Radio Station* is that the performer can really go wherever he wants without worrying the monitor engineer. This in itself opens up possibilities for a more creative approach to stage and set design.

The downside is, of course, cost as the aforementioned list of well-stacked users would suggest. Around £3,800 gets the transmitter while specially moulded custom ear pieces, available in any colour, cost another £800 and this involves a visit to an ear specialist who will guarantee that they are anatomically correct. Alternatively cheaper and more general shaped ear plugs are also available.

There will always be those who want to feel their pants flapping on stage and there is no reason why they should not continue to flap providing civilised levels are kept in ear. Cost aside, this has got to be the way to go in the long run.

UK: Garwood Communications, Tel: 081 203 5042.
Europe: Personal Radio Systems, Tel: 081 801 8133.
US: Futuresonics, Tel:+1 215 598 8828.

Dennerlein Sampling CD

Hammond *B3* devotees and admirers will be spoilt by the Barbara Dennerlein *B3* sampling CD which not only shows off Dennerlein's extraordinary technique and feel but also captures what can only described as an extremely representative cross section of classic *B3* tones.

The CD kicks off with a jamboree of *B3* clichés—and I mean that in the kindest way—riffs, motifs and breaks that are instantly recognisable as things people do on a Hammond. The majority are rhythmically based and lend themselves admirably to looping. Dennerlein's timing is impeccable with spates of groove dotted in between and leading up to the tight and obvious loop points. These riffs form a sound adjunct to the collection of riffs found at the end of and at various points throughout the the CD. Techno freaks should shoot straight to ID 42 and chill out as Dennerlein has a frightening appreciation of what constitutes a 'kicking' bass line. However, some of the riffs are treated with rather lacklustre slap back echo and reverb which detracts from their usefulness.

Noise is kept pretty much under control throughout but the old beast is hissing in places. Attempts have been made to stifle it at the end of blasts a little more would not really have hurt anybody for the sake of authenticity. There are loads of overdriven sounds in pretty much all permutations and degrees right down to the glassy mellowness that you are not really going to get anywhere but from the real thing.

While the *B3* is inherently a clickey and snappy creature, there are occasions when I would hazard that the artifacts have in fact been created in the recording process on some of the initial peaks. The collection of riffs are invaluable for illustrating Dennerlein's breadth of repertoire and the sort of tonal variety available on the disk. There is a fabulous amount of key noise and a wide variety of *Leslie* and vibrato wobble.

Riffs span styles as diverse as blues, jazz, funk and out-and-out rock signatures and they are arranged in keys. The individual note samples are an education in just how dirty and rich the *B3* is—the sheer amount of movement, harmonic complexity and disparity evident in these single notes and how they change across the keyboard explain why it has remained a difficult instrument to resynthesise.

For multisample collections, a demo of the sound is followed by a full five octaves delivered chromatically. These feature the first three drawbars pulled without percussion; first four pulled without percussion; all out without percussion; third percussion soft; second percussion normal; the first three and five pulled 2nd percussion; first three pulled 2nd percussion with vibrato; first three and seven pulled without percussion with *Leslie*; first three and six pulled 2nd percussion and vibrato; first four pulled 2nd percussion and slow *Leslie*; and the gospel tone of the all out at the sides and dipped in the middle drawbar curve with fast *Leslie*. Add to this an octave and a half of filthy rock organ and two double octave bass tones and that is your lot. I think that is pretty representative. Akai *S1000-S3000* users can indulge in a 23Mb lump of data backup containing multisampled *B3*s and the riffs with keyboard mapping.

Recreating Hammond sounds for use in the MIDI world has for the most part been left to the effort of stand-alone units with the Voce and E-mu *Vintage Keys* being notable in their degree of success. However, armed with this CD, sampling a *B3* (and a very fine *B3* at that), becomes a veritable pleasure. Other facets like no need for velocity and the relatively limited bandwidth required to capture the sound all point in favour of this home-rolled approach to creating your own *B3* library. Matters are helped considerably by the high quality of the original source and the only compromise is likely to be the user's patience and skill at inputting the data.

This CD is a must for anybody even remotely interested in the wonderful sounds of the killer *B3*. ■

UK: Time and Space, PO Box 306, Berkhampstead, Herts HP4 3EP. Tel: 0442 870681. Fax: 0442 877266.

Music News is compiled by Zenon Schoepe

Soundcraft *DC2000*

Soundcraft *DC2000* three years in development

'Our motto is more for less', states Soundcraft's R&D Director John Oakley. And it certainly appears to be fair comment when applied to the company's latest console, the *DC2000*.

For £12,000 (UK price), Soundcraft are offering a 24-input, in-line desk equipped with proprietary moving fader automation, automated switching, machine control and centralised touchscreen co-ordination. A further £3,000 will buy a 32-channel version providing a total of 72 inputs during mixdown. The price is pretty staggering when one considers that the cheapest commercially-available moving fader system cost in excess of £500 per channel.

How is it done?

John Oakley: 'Basically we've attacked every single part of the console with a view to getting the price down. It's been a long hard struggle (approximately three years in development), but we've got there in the end. One of the key things has been to integrate as much as possible into silicon, and we've developed our own ASIC which contains all the logic to control the automation and communications within the console. This, coupled with the availability of a sensibly priced motorised fader, has helped enormously in cutting costs'.

The fader is a belt-driven design from the Japanese company ALPS, who have also expended considerable R&D effort to produce a satisfactory product at a low cost. In fact, there were ten prototype iterations before Soundcraft finally approved the design, which Oakley describes (without giving away any secrets) as 'deceptively simple'. The fader uses 10-bit linear coding which translates to a physical resolution of 0.1mm; its top to bottom speed over a 100mm travel is approximately 200ms.

Moving fader automation is provided on input and master output long-throw faders—these are the monitor faders (small faders always remain in the channel path), master stereo faders and control groups (four faders that can be assigned from the monitor faders to provide functions such as relative trim updates).

The console is designed for multitrack recording, and there are eight buses arranged in stereo using a floating structure very similar to Soundcraft's *Sapphyre* console. However, unlike *Sapphyre*, the console includes stereo group faders providing a stereo subgrouping facility.

EQ is 4-band, arranged in two sections: HF and LF switchable between 12kHz–6kHz and 60Hz–120Hz respectively (±15dB), and two swept mid bands. Each section can be separately switched into the channel or monitor paths, and there is an overall EQ IN-OUT button. A 100Hz HPF is also included, which, in true Soundcraft tradition, appears at the top of the module directly after the input gain.

There are six auxiliary buses arranged as Aux 1–4 and Foldback 1 and 2. The Aux sends source a post signal and the Foldback circuits are switchable pre or post. Aux 1 and 2 are permanently in the monitor path while Aux 3 and 4 share mon-chan switching. Foldback 1 is globally switchable between monitor and channel, but Foldback 2 remains fixed in the monitor path. The two Foldback sends can also be configured as left and right feeding the headphones submixer for stereo foldback. The commonly used method of splitting the controls into level and pan was dismissed in order to keep costs down.

The automated switches in the I-O modules are ON buttons for Aux 1 and 3, and the channel-monitor cuts.

The master section of the console is made up of various modules above and below the touch-sensitive, back-lit LCD screen (320 x 240 dot matrix). Above are the four stereo group faders already mentioned, the Aux and FB masters plus studio loudspeaker level control and headphone submix controls. The Aux masters have automated cuts and AFL; they also include two switches enabling Aux 3 to be linked to Aux 1 and Aux 4 to Aux 2. this facility has been designed for occasions where outboard is restricted and the engineer may wish to send signal to the same devices from both monitor and channel paths.

Four stereo returns are provided with simple 2-band fixed EQ, access to FB 1 and 2, balance control automated cut and routing selection to either the mix bus or a stereo group (so Stereo Return A will route to groups 1 and 2, Stereo Return B to groups 3 and 4, and so on). Also positioned in this part of the console are a 2-frequency oscillator (1kHz and 10kHz) add the talkback mic and controls. TALKBACK buttons would perhaps, be more ideally sited at the base of the master section, nearer to the operator.

Below the LCD screen are the four control groups with red fader heads, the master stereo fader, master solo controls (SIP or AFL-PFL selector, level control and Solo Clear), automation master mode selectors and the control room monitoring controls including two external source switches, MONO and DIM buttons and an alternative speaker switch.

The heart of the console, and perhaps the most exciting part, is the central area of the master section incorporating the touch-sensitive screen. Apart from controlling automation and machine control via a mixture of dedicated and on-screen keys, the console can be set up in various ways using the LCD. For example, to isolate signal paths from Solo In Place muting either by individual selection or by selecting presets; and to control metering functions such as source selection and switching between VU, PPM and VU with peak hold.

Soundcraft have taken pains to make the operation of their automation system as simple and intuitive as possible—important when you bear in mind that many of the console's intended users have little or no previous experience of computer mixdown and moving faders. That said, the system does facilitate sophisticated use as and when the user is ready—configuring personalised setups including status default, glide-back times and so on.

There are three ways to interact with the automation—via the LCD, from central dedicated keys and from local switches. Generally speaking, the more regularly used functions have been allocated dedicated switches; local switching and LEDs independently control and indicate status for faders and automated cuts—there being four mix statuses: Read, Write, Arm (equivalent to update or trim) and Isolate. Automated switches have two associated LEDs—one displays normal ON-OFF status, the other glows amber when the switch is under computer control.

All major time code formats are read, generated and permanently displayed (reader built into meter bridge), and machine control is via the usual selection of transport keys plus a jog wheel. Additionally, there are scrolling cues lists, cycle facilities and auto drop-ins with a rehearse facility. Cues can be named and to do this, the touch-sensitive screen transforms into a qwerty keyboard. The LCD can also be used to switch individual tracks to Ready, and this is confirmed by the flashing of an LED positioned on each I-O module —a master record enable switch is included as a precautionary feature.

The *DC2000* uses a built-in floppy disk drive for storing mix data, cue lists, MIDI event lists and so on—a hard disk drive is optional. All computer hardware is integral to the desk, the only external component being the power supply. A 32-channel version of the *DC2000* with patchbay measures just over 2,000mm by 876mm.

The console is scheduled for production towards the end of August and was shown in prototype form at the APRS show. This is an important product from Soundcraft, offering exceptional value for money and bringing top-end features to the budget market. ■

Patrick Stapley

Soundcraft Electronics Ltd, Cranbourne House,Cranbourne Industrial Estate, Potters Bar, Herts EN6 3JN, UK.
Tel: 0707 665 000.
Fax: 0707 660 482.

Gefell *UM92S*

Recently in these pages I reviewed the splendid Microtech Gefell *UM70* condenser microphone, with the promise of the same company's valve model to follow. That model, the *UM92S*, is now available in the UK.

First impressions are of a very grown-up microphone. The standard kit comprises the microphone itself, a large wind shield, a cat's-cradle shock mount, a long multiway cable and the power supply, all in a sturdy, lockable aluminium flight case. The only omission is the IEC mains lead, which seems a curious and penny-pinching oversight.

The microphone can only be sensibly mounted on a stand by screwing it into its shock mount, which is one of the best I have seen. The mount works by screwing the microphone on to a swivelling ring in its base through which the connector passes, rather than by clamping the centre of the body, which allows the microphone to be rotated in the mount much more easily. There is none of the all-too-common tendency for the microphone to hang where it chooses rather than where you want it; the balance is right and the elastic cording sufficiently tight to hold the weight firmly without sagging, drooping or flopping around. My only complaint is that the swivel joint bolt was not tight enough to support the weight and I had to take a screwdriver to it.

Being a valve microphone, the *UM92S* connects to its dedicated power supply by means of a multicore cable, which is terminated in heavy-duty positive screw-fit metal-shelled connectors. The cable supplied as standard by Gefell is apparently rather stiff and unwieldy, and the main UK importers, Stirling Audio, substitute special Mogami cable which is very flexible and kink-free. There is apparently a possibility that Gefell may go over to the Mogami themselves eventually.

What few controls there are appear on the simple but elegant power supply box. The microphone multiway connector is on the back panel together with the XLR output socket and the IEC mains input, while the front panel carries the on-off switch and the polar pattern selector. There is no pad and no bass roll-off filter. The usual three polar patterns are provided—omni, cardioid and figure eight—by a rotary switch with associated LEDs, which thoughtfully are all different colours making it possible to tell at a distance which pattern is selected.

All the system components convey the same impression of quality as found on the *UM70*. The cosmetic design is elegant in the style of Neumann (as would be expected) and the attention to detail is excellent —close tolerance well-fitting parts, sharply-defined engraving and a beautiful finish overall.

For tests, I rigged up a simple A-B with a familiar top-end FET multipattern microphone for straight comparison, and this produced a surprise. The usual expectation of valve equipment of any kind is that it will have a warmth, a bloom, the aural equivalent, perhaps, of a soft-focus filter. This is decidedly not the case with this microphone, whose top end is present and precisely defined, and whose lower extremes are, if anything, less pronounced than the FET. In fact if these tendencies were any more exaggerated the sound would be verging on the bright, even hard. As it stands, it is hard to decide which was being more truthful, the Gefell or the FET. When they were rigged on a grand piano, I found myself rushing backwards and forwards to check the acoustic sound of a piano I know very well, unsure if the additional bass I was hearing on the FET microphone was actually there to that extent. I eventually came to the conclusion that the Gefell was giving me a more accurate, less flattering account of the piano than the other—which is the reverse of what I had expected. On close-miked voices, the results were perhaps more as predicted; a well-controlled proximity effect gave a rich warmth which, together with the extended and present highs, produced a powerful but clean sound which cut effortlessly through a mix without harshness.

One expects the tonal characteristics of a microphone to change a little as the polar pattern is altered, but this effect was particularly marked with the Gefell. Set to omni, the differences between this and the FET microphone were slight, but as soon as both were switched to cardioid the contrast in the bottom ends became apparent and remained in figure eight. The polar patterns themselves seemed well-defined, with good nulls at the back of the cardioid and the sides of the figure eight, although the omni pattern was apparently not properly circular, being slightly less sensitive at the back. Off-axis colouration seemed minimal, which is particularly important where spill is a problem.

I have to say that all the aspects of the performance I have covered are entirely my aural impressions, since there was no documentation with the microphone whatsoever. Operationally this is not a problem with this kit since there is nothing to misunderstand, but a few facts and figures would have been reassuring.

The pricing of the *UM92S* places it in the lower middle of the valve microphone market, inviting debate as to which end of the spectrum it sits more happily with. The general resurgence of interest in valve gear makes it important to discriminate between bandwagon products, which perhaps offer a bit of nostalgic fun, and genuine professional equipment which attempts to do a serious job using a much-favoured old technology. This Gefell microphone falls unquestionably into the latter category, and as with its FET partner, the *UM70*, the only danger is that its comparatively low price will mislead potential buyers into assuming it is too cheap to be taken seriously.

Dave Foister

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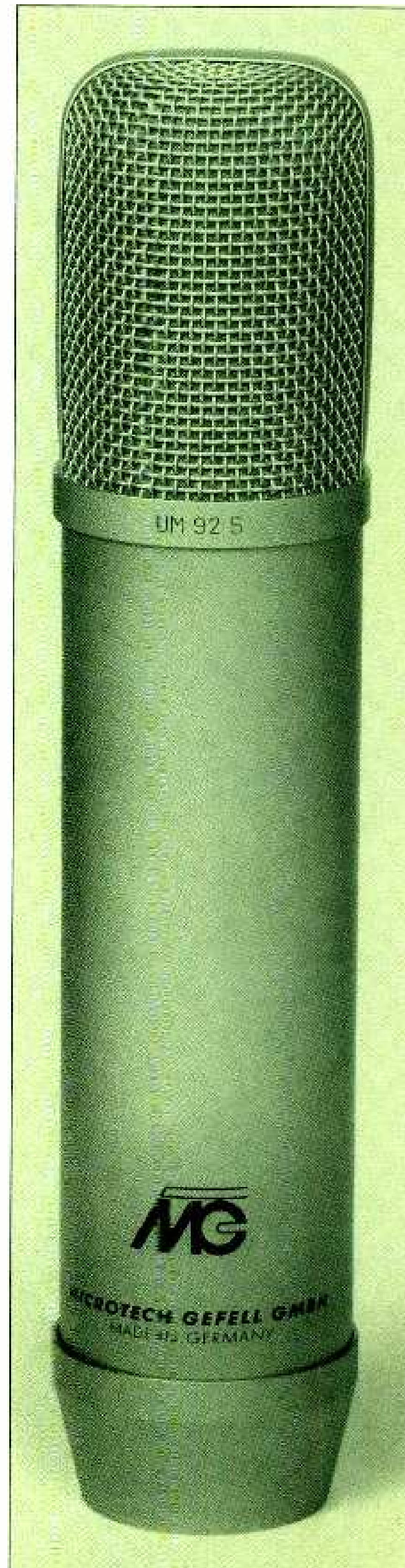
UK: Stirling Audio Systems Ltd, Kimberley Road, London NW6 7SF.
Tel: 071 624 6000.
Fax: 071 372 6370.



The *UM92S*, powerful, rich and warm on vocals

NEWS

NAGRA

News
Nagra-D Events

FILMS

The NAGRA-D has been used on the following major feature films:

FRANCE

«Couples et Amants» directed by John Lvoff, produced by Providence Film and sound engineer Jérôme Thiault.

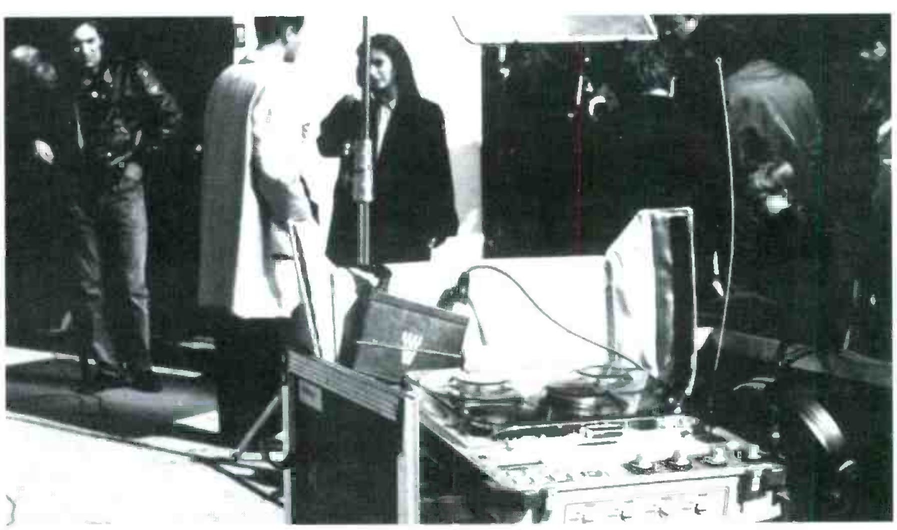
«Une mère de sable», shot in Mali, produced by Atriascop with Philippe Gautier sound engineer.

«La reine Margot» was directed by Patrice Chereau and produced by Claude Berri. The film's stars were Isabelle Adjani and Daniel Auteuil and the sound engineer was Guillaume Sciama.

On location of «Il giorno di S. Sebastiano»

ITALY

«Il giorno di S. Sebastiano» with Pasquale Scimeca as the director and sound engineer Maurizio Argentieri. Cinecitta was chosen for the post-production and the transfer will be made with Dolby SR on 35mm film. The shooting took place in Sicily.



During the filming of «Couples et amants»

Editorial

«SEE HOW SOUND CAN SOUND» is the slogan being used in our publicity campaign this year and invites you to discover the unique quality of recordings made on the NAGRA-D. Today we provide the professionals, whose competence is greater than ours, the opportunity to express themselves about this question.

The one thing that is certain, is that we are assuming the responsibility of putting the highest performance product into the hands of the end users. The question of the quality of digital recordings is still forefront in our minds, and is the reason that we have reserved a lot of space for this subject in the present edition of the NAGRA NEWS.

The quality of a sound can be expressed mathematically and quantitatively measured using ever more sophisticated equipment, but the final judgement is given by our ears. It is the ear which alerts us to audible distortion or wow and flutter.

Twenty years ago one measured frequency response, the wow and flutter and that said it all. Today we find that time (group delay) is of great importance for the sound. The jitter (wow and flutter) is no longer expressed in % but in nanoseconds and is still perceptible to the ear and should not be neglected.

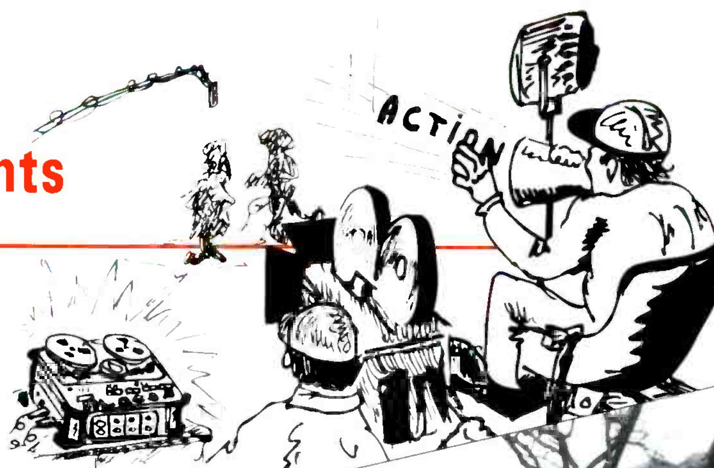
The headroom allows us to fix the surplus bits (or the dynamic range) necessary to guarantee the recording a dynamic of 16 bits to the final listener.

And if we speak about films...

The transmission means available today does not technically allow the maximum benefit from the quality offered by digital recording. However, with the advent of HDTV as well as the digitalization of film, are they not already coming close to reality?

Aldona Mury

News
NAGRA-D Events



USA

«Blue chip» starring Nick Nolte and directed by William Friedkin. The production sound is being done by Kirk Francis and the post production sound by Todd A-O/Glen Glenn Sound. The film will be distributed by Paramount Pictures.

«Quiz show» directed by Robert Redford with the production sound by Todd Maitland. The film will be distributed by Buena Vista (Disney). Shooting began May 17th in New York City.

«Angie, I says» is directed by Martha Coolidge and stars Geena Davis and Stephen Rea. Production sound is by Ed Novick and will also be distributed by Buena Vista. It is being shot in Los Angeles.

NAGRA Diary 1993

JUNE

Showbiz Expo taking place from **5-7 June** in Los Angeles.
Organizer: Phi Technologies, Inc. Oklahoma City/USA

The **18th International Television Symposium and Technical Exhibition,** held in Montreux, Switzerland from **10-15 June.** Stand No. B106
Organizer: Kudelski SA Cheseaux/Switzerland

APRS'93, the 26th International Exhibition for the Professional Recording Industry, from **June 23-25** at Olympia 2, London.
Organizer: Nagra Kudelski (GB) Ltd. St. Albans/England

AUGUST

The Florida Motion Picture and Television Convention and Trade Show FLORIDA PREMIERE taking place from **13-15 August** at the Walt Disney World Resort in Orlando. This event is co-sponsored by the Florida Motion Picture & Television Association and Disney-MGM Studios.

Organizer: Phi Technologies, Inc. Oklahoma City/USA

OCTOBER

The **95th AES Convention** held on **7-10 October** at the Javits Convention Center, New York. Organizer: Phi Technologies, Inc. Oklahoma City/USA

NAGRA-D in TELEVISION

The sound for the following films for television was recorded on NAGRA-D's:

Upcoming episodes of ***«Da costa»*** for Antenne 2 (France) are directed by Bob Swain, produced by Projefi (Christine Gouze-Rénal) and sound engineer is Guillaume Sciama. It will be one of the first police films comparable to the prestigious «Navarro».
The film stars Roger Hanin and Claude Villert.

«Tout va bien dans le service», directed by Charlotte Silvera, produced by Télé Images and with sound engineer Daniel Brisseau.

«South Central» is a TV pilot. Details in the next issue of Nagra News.

MUSIC RECORDINGS

Pierre Verany SA, a very well known French compact disc mastering facility, recorded the music for the film «El Parajo de la Felicidad» on a NAGRA-D. The music was interpreted by Jordi Savall.
Moreover, the majority of the discs in their catalogue, that have been recorded since March 1993, were made using the NAGRA-D.

Each transmission system has its limits and weaknesses.

A T.V. channel can be considered «transparent» if the receiver is capable of re-constructing all the possible combinations of the «draughtboard», from an all white (or all black) screen to one having a succession of black and white squares, the smallness of which is limited only by the bandwidth of the channel (6 MHz for TV) and the number of lines in the picture.

Yet everybody has seen the limitations of the PAL system, with the apparition of the «Moiré» effect, or the «water fall» effect on the SECAM system.

It is the same case with an acoustic system, which suffers from operational limitations within its bandwidth (20 Hz to 20 kHz).

As limitation one can cite:

- Amplitude distortion (harmonic)
- Phase or time distortion
- Aliasing phenomenon (twittering birds)
- Intermodulation (non harmonic distortion)
- Sidebands (micro jitter, wow and flutter) Etc.

Everybody still remembers the aliasing weaknesses of early A/D and D/A converters that were not over sampled. With ΣΔ (Sigma-Delta) systems, the «twittering birds» have (practically) flown away, but one must remain cautious.

Let us stop for a minute on phase distortion

At school we were taught that the ear could not distinguish the phasing between different harmonics of a sound. That is true for a continuous tone.

During transitions, it is completely different. Transitions sound do not support phase or temporal errors (we call these group delay $d\phi/d\omega$).

We can take the example of a telephone line which inflicts a delay of 0.5 second on low frequencies and higher frequencies which traverse the system with a delay of perhaps 1 second. What happens to a «p» or «t» sound made at one end of the line, when it arrives at the receiver? Certainly an unintelligible sound.

To achieve purity of a recording channel, it is therefore extremely important that

The notion of system transparency

temporal errors are reduced to a strict minimum.

A great deal of care was taken with this in the NAGRA-D. Knowing the equalization techniques currently available (group-delay adjustment), it will certainly be possible in an analogue recorder to exploit at 19cm/s an acoustic quality which is presently reserved for recordings made at 38 cm/s.

Thermal noise, and 120 dB dynamic range

The thermal noise of a resistance B is, in Volts:

$$Un_{V}=1.88\times10^{-8}\sqrt{R}$$

which is valid for ambient temperature and the audio bandwidth of 20 kHz.

By way of example, let's try to determine the maximum input sensitivity of a microphone preamplifier to assure 120 dB of dynamic range when it presents a source impedance of 50 Ω.

$$Un_{V50\Omega}=1.88\times10^{-8}\sqrt{50}=0.133\mu V$$

$$U_{INMIN120}dB=0.133\mu V+120dB=133mV$$

If one pushes the gain beyond 133 mV at the input to achieve «0 dB displayed», for example 13.3 mV, one could color the recording with 20 dB of noise due simply to the 50 Ω of the microphone.

Remember that a dynamic microphone has an output of less than 1 mV for at the sort of ambient sound level found in a «quiet restaurant»!

Still on the subject of the dynamic; the difficulty of assuring 20 bits of dynamic range

Remember: 20 bits at 6 dB per bit gives, in principle, 120 dB of dynamic range.

The dynamics of the ear, from the threshold of hearing to the threshold of pain is 130 dB.

The contribution of the noise of a microphone relative to the threshold of hearing in the best of cases is 10 dB.

One can deduce that it is not useful for the dynamic range of a channel to exceed 130 - 10dB = 120dB.

Conclusion: One should be able to make a microphone pre-amplifier without a level adjustment from the moment that a dynamic range of 20 bits is available.

In order not to change the habits of sound engineers we have kept the level adjustment in the Nagra-D, but limiting it to 37 dB.

The «purity» of a digitally encoded sound is determined by the «jitter» of the sampling frequency

The sonic integrity of a sound which has been transmitted via a digital processor is constrained by the accuracy of the sampling frequency:

A/D—> processing digital—> D/A

The maximum sampling frequency jitter which can be tolerated in an A/D convertor to maintain a 20 bit dynamic range must be less than 200 picoseconds (0.0000000002 second). This very low jitter rate can only be achieved by using quartz oscillators of very careful design. The thermal noise of an LC oscillator can barely fall below 1 to 2 nsec (1000 to 2000 picoseconds), if it is externally synchronized, (such as is frequently used in an R-DAT), even if we assume that the powering is perfect. This value has trouble guaranteeing a dynamic range of 16 bits.

Lots of ideas, true or false, turn around digital recording. A lot of other debates will come to light in this domain. Nobody doubts, that an important step will be made once development engineers and sound engineers use the same terminology to confront their ideas. May these few modest words contribute to the improvement of this communication.

To be continued...

Marching in the Margins: Music, Me, and the NAGRA-D

by Albert G. Swanson

It may not have the glamour of feature films, nor does it pay as well as commercial work. And, if you want a shot at true fame, fortune, and glory, hang at the studio long enough for your best rap or grunge (or whatever is «in» this week) buddies to sign with a major label. But some of us can't escape the lure of another kind of audio: remote music recording.

While «remote» often refers to all non-studio work, I am using the term somewhat differently, as comprising the recording of any music originally written for live audiences. This includes all classical music, most jazz, plus pure folk and ethnic music. This kind of audio is distinct from studio (pop-style) recording in that it uses minimal or no post-production mixing, and is generally performed on-site (a concert hall, church, or other venue the acoustics of which are suitable for the appropriate audience participation; note that in the case of some ethnic music, the proper setting may be out-of-doors), although studios are occasionally utilized.

It's a Digital World Out There

How does the Nagra-D fit into the picture? Consider, to begin with, that almost 100% of classical music releases are nowadays digital. The customers for acoustic music recordings are a demanding lot; much of the impetus for technical improvements over the years has derived from their requests. For example, pop-style recordings do not generally encompass a large dynamic range. However, that of classical music often exceeds 60 dB. At such a distance from peak levels, the limitations of the medium become critical: tape hiss, in the case of analog recording onto magnetic materials, or surface noise—which grows worse with repeated playings—with vinyl discs. As these blemishes are non-musical and hugely distracting, the vast majority of listeners prefer at least the release format to be digital, where media defects can be effectively eliminated from the equation—hence, the development of the compact disc and other consumer digital formats.

Since the releases are to be digital anyway, it makes sense, from the engineer's point of view, to perform the original digitizing on equipment under his or her direct control, especially since almost all editing these days is performed at a digital workstation (thus obviating the need for analog-style razor blade editing). This is, of course, precisely why DAT recorders have become so popular for remote recordings.

To be sure, none of this invalidates the artistic validity of high-quality analog originals. Still, the commercial reality is that the **DDD** printed on the compact disc's tray card is important. On the other hand, there are, in the Land of the Audiophile, a few well- (and out-) spoken individuals who will not march in the digital parade, citing the rain clouds of poor performance from available 16 bit products.

These holdouts have a point: After taking into account noise from all sources and allowing for «headroom», a typical 16 bit system actually performs at more like 13-14 bits, or 80-85 dB of dynamic range, thus putting important ambience cues directly where the system has its lowest resolution. The Nagra-D, however, provides (and stores) about 100 dB of dynamic range (or more, depending on the converters chosen), thus providing a true 16 bits (96 dB) worth of audio to the release product. It is noteworthy that many audiophiles have pronounced the Nagra-D as the first digital product they would themselves consider owning!

Seattle Symphony Orchestra Music Director Gerard Schwarz (left) and Grammy Award Winning Producer Adam Stern study session tapes.

'D' Answers

The Nagra-D's exemplary audio quality is, then, certainly a significant consideration for the remote specialist: There is, simply, no other portable machine of its quality. But for the operational versatility required in live-recording situations, it has other, equally important, features, which are, perhaps, best illustrated by example:

Four channels are wonderful. While the possibility of mixing in, after the session, ambience («sweetening») microphones is often touted as a major benefit of four channels, I've personally seen few such implementations. Most engineers who use sweeteners prefer mixing them (to stereo) live. Nevertheless, given the fact that most remote monitoring facilities are either non-existent or uncalibrated to the engineer's sensitivities, proper balancing of solos, for instance, or choruses, is much easier to control after-the-fact. I do a lot of concert recordings for the **Seattle Symphony Orchestra**. These concerts are usually performed in pairs, over two consecutive nights. Soloists and choruses are, for different reasons, notoriously inconsistent; therefore, to edit between performances often demands a good deal of careful adjustment of balances.

A variation on this theme occurs with other kinds of music as well. St. Mark's Cathedral in Seattle is the home of a very large Flentrop organ. Because of its quality and acoustical setting, many organists love to record on it. But it is not always easy to achieve, in recording, the desired artistic result. As the instrument's five divisions cover a vast area of the Cathedral (from the gallery floor to the top of the casework is about 15 meters, as is the lateral distance between the most distant pedal pipes; from the main divisions to the rugwerk is about 6 meters), the ideal microphone position for one registration may sound atrocious for another. Further, artists most often do not discuss stylistic intent (like which divisions they will be using) with the engineer ahead of time. Thus, it alleviates much professional grief to prepare two separate acoustic perspectives, either or both of which can be called upon, as needed.

The extra channels can also give the engineer production flexibility. In a recent recording with the Northwest Chamber Orchestra, we ran into scheduling problems. With only a short time remaining in the session, and an extremely difficult piece left to record, it was decided to record only the orchestral parts; the two soloists would return later, after the first round of editing, to «overdub» their parts. (With the Nagra-D, this procedure requires re-synchronizing during the final editing, as there is an offset between the pairs of tracks. This, however, is a trivial operation.)

Even where live-to-stereo mixing is possible, the extra channels can be quite useful. In concert recording, it is not always possible to set levels properly, because there is no rehearsal (for the engineer, at least). By feeding the same input to both pairs of channels, but offsetting the levels of one pair by, say, 10 dB, you have effectively increased your overall dynamic range by that much. During a recent recording of Tchaikovski's 1812 overture, for example, the acoustic levels of the live artillery were impossible to control, or even guess at, ahead of time. So it was a matter of professional safety to make two simultaneous recordings.

Here is a potential situation for which the Nagra-D is perfectly suited: One of the most interesting places I've ever worked was a very large hole in the ground, a cistern about 60 meters in diameter and 4 meters high. In its original incarnation it held some 2 million gallons of water, but now serves as a recording studio, of sorts—but a most difficult one. Among its characteristics is a 45 second mid-band reverberation time. This feature, along with the lack of a defined artistic aesthetic and the total non-availability of A.C. power, make for some rather nightmarish problems for the engineer. While the commercial products (which involved the improvisatory skills of some excellent musicians) have been well received, the equipment used for past projects was, to be blunt about it, not up to the task. For one thing, the ability to track reverberation down to the bitter end, without the low-level graininess typical with portable 16 bit systems, was missing. Also, the capability of providing alternate perspectives of the same acoustic event would be of considerable interest. I look forward to a new project at this cite.

Deep Listening Band recording in the cistern in 1988.

Other specific features of the Nagra-D are invaluable for remote recording:

• Monitoring off tape saves a lot of professional anxiety, as does knowing, for sure, the dynamic range covered.

• The powering scheme, maligned in some quarters, nevertheless provides a major service: When no additional equipment is needed, the quality of the final product is no longer dependent on the often dubious quality (or even non-existence) of the local A.C. (mains) power. (The new stage hand plugs a 10 kW lamp into—poof—your circuit? No problem.)

• Likewise, another disputed feature, the open-reel tape format, can be put to good use: Some music exceeds the two hour limit of any currently used cassette format (DAT, in its long-play mode, is capable of additional recording time, but at with limited bandwidth). Wagner's Das Rheingold, for instance, lasts for two and a half hours without pause, and Indonesian wayang kulit performances can be longer still. On a more common level, many concert performances last longer than two hours, not including intermission. In these situations, the four hour capability of the Nagra-D is much appreciated.

Troubles

To be fair, it should be mentioned that there are a few problems with the Nagra-D. In the first place, the above mentioned dubious qualities have detractors for understandable reasons: The internal battery is heavy, and its care and feeding can be a nuisance; and cassettes are, often, more convenient than changing reels and threading tape. But the more critical concern for the professional user is an economic one. Most remote music engineers, at least in the United States, are free lancers (who must own, in addition to their remote equipment, a complete editing facility), and the largest portion of their customers are unsubsidized non-commercial organizations, it is often not possible to charge a fee high enough to cover the cost of owning, or renting, a Nagra-D—that is, the very customers who could most benefit from the Nagra-D's virtues are ones who can least afford it!

In passing, however, I would like to say that a possible way out of this economic impasse is beginning to present itself. The Nagra-D was, of course, conceived as a fully professional machine. But as it happens, many consumer audiophile users have expressed an interest it, as there is no other machine anywhere with its qualities. Might it be possible for the professional to provide a premium product to these consumers, so that they, in effect, endow the original recording?

The Bottom Line

Finally, as many of us grow older, we grow to appreciate any and all work we don't have to do. With the Nagra-D, it is again possible to provide a good service with minimal set-up. True, the Nagra-D is heavier than, say, a portable DAT recorder. On the other hand, everything is right there, in one carryable box. One trip in from the parking lot, the Nagra-D in one hand, a pair of portable stands in another, and a bag (for microphones, cables, tape, a few odds and ends, and a good book) over your shoulder....You can't get a better recording, and, should you be accosted on your way to the event, the Nagra-D makes an excellent defensive weapon.

Al Swanson began his association with Nagra in 1972 while still a graduate student in ethnomusicology. Finding academia a commercial cul-de-sac, however, he has since directed his professional efforts toward audio recording and editing, particularly of music. His credits include over 400 albums, too many concert broadcasts to count, several multi-media presentations, and, from a more desperate time in his career, a smattering of commercials, music videos, and feature films. Swanson also edits the newsletter, *Gazette and Digest for the Absolute Nagrist.*

User comments on Sam Wise's NAGRA-D review

Ivan Sharrock and Nigel V. Woodford comment on Sam Wise's article in March 1993 Studio Sound magazine.

«The review of the Nagra-D by Sam Wise (Studio Sound - March 1993) raises some questions about the suitability of the machine as a location recorder. This tone and use of language in his article implies that the machine cannot compare to the currently available DAT recorders. Absolutely true. No car reviewer would dream of comparing a Mercedes to a Lada although they both can get you from A to B!

As a Production Sound Mixer in television and the film industry I have recorded over 200 documentary programmes and 65 feature films, including «The French Lieutenants Woman» (BAFTA), «The Shining», «The Last Emperor» (OSCAR), «Far and Away» (Using a Nagra IV-S with Dolby SR) and recently Bernardo Bertolucci's «Little Buddha» (Nagra-D). The latter was shot in Nepal, Bhutan and the USA (Seattle) over a period of six months all on location and in some extremely remote areas. I was using the first two production-line Nagra-D's and, whilst we had some teething problems with the microprocessor initially, both machines behaved faultlessly. The machines were hired to me by Richmond Film Services but, because this was a «first», I dealt directly by telefax with the factory in Switzerland about the few queries that came up. Their product support is unrivalled.

However, I would like to clear up some «misconceptions» in Mr. Wise's article. He states that:

(1) STANDARD *'DAT has become the de facto standard for master recording and film recording'.* I suggest he checks with the sound equipment rental houses both here and in the USA.

(2) OPEN REEL *'The open-reel format is not as easy to use as the DAT cassette'.* Excuse me!!!!!!

(3) FORMAT *'do you want your material in a format which no-one can replay except another Nagra-D ?'.* Someone has to be first with any new format. When half-track and sub-audio synchronization was the norm, Kudelski introduced his Neopilot system for mono recorders and then the FM-Neopilotton for stereo, neither of which were compatible with other recorders. Although transferring original material was a problem initially for those machine rooms that did not own a Nagra I have seen Studer, Telefunken, Levers Rich etc. modified to cope. Now, of course, any transfer house of note owns a Nagra T.

Whilst the Nagra-D format will never be embraced by the customer market, the professionals will always opt for the best quality machine available. We shall see whether or not other manufacturers have the foresight to follow suit.

(4) PRICE *'for the price of one Nagra-D, two Fostex PD-2 recorders can be purchased'.* However, don't forget the additional outboard piece of equipment you will need to be able to monitor two machines (ever tried listening with two sets of headphones around your neck), plus the hassle of powering and having two cassettes for one complete 4-track. (This also goes for the transfer/copy/mix down suite).

(5) CARRYING *'But there is not even a clip point provided for the attachment of a shoulder strap'.* There is!

(6) BATTERY LIFE As Mr. Wise insists on comparing the Nagra-D to other DAT machines, and specifically the Fostex PD-2, a word about battery life. All portable digital equipment use power hungry ADC's and DCA's. There is not a digital machine available that will run all day on its internal batteries, unlike their analogue equivalents. We, the professional sound recordists and the rental houses, have addressed this problem in various ways when «mains» power is not available. For the Fostex PD-2, Audio Services Corporation of Los Angeles recommend their 12 Volt gel-battery (slightly smaller than, but similar weight as, a small car battery) to give about 8 hours use. For the Nagra-D, I used standard camera battery belts but any source of DC power from 12-30 Volts can be used. I also use an AIWA-HHB 1 PRO DAT recorder for Fx. Although this machine has two internal DC power sources maximum recording/standby time is still limited to 2 hours in practice. To ensure confidence that the machine is not suddenly going to give up an external 7.5 Volt high-capacity Nicad is plugged in. To summarize, all portable digital machines should be treated as if their internal batteries were there to run the internal microprocessors for saving the memory and any serious professional recordist would use an external source of power.

(7) MENU *'Even after several hours use, we found this menu system remained confusing requiring constant reference to the manual. While this might be reasonable when setting unusual things, it gets irritating when selecting between inputs, or changing sampling rate'.* Let me reassure those of you who are not familiar with the machine. All of the normally used settings are stored so that after the machine has been switched off, the next time you come to use it, the memory scrolls through your pre-sets, something Mr. Wise failed to mention. Confusing? Not a bit of it! Admittedly the menu is extensive and will probably become more so as befits this machine. Take a tip. If you worried that you cannot find your way, reduce the manual menu pages to one page of Filofax paper, mount on stiff cardboard and keep with machine; but in practice its no harder than dealing with MS-WINDOWS.

(8) SUMMARY *'Technically, the Nagra-D performs well'* - It is frighteningly transparent! - *'but operationally, we found it disappointing. It is large'* - compared to what? My portable Neve mixer is twice as big. - *'has limited battery life'* - see above - *'a mediocre menu system'* - display maybe, system most certainly not - *'Purchasers also have to weigh up the risk involved with the single source nature of this recording format'* - What does that mean? a). A 'single source' is the original sound source for your master recording. b). A 'single source' is the new Nagra-D format (The Nagra IV-S Pilotton recorder was at one time a 'single source', or c). does Mr. Wise believe that two machines should be used to back-up the recording. Certainly all the DAT users that I have come across in the feature industry back-up to another DAT machine or to an analogue Nagra - says a lot of their confidence.

It is true that the Nagra-D is not perfect. It weighs more than 1Kg; is bigger than a packet of cigarettes and its battery does not last for a week. But it's a darn sight better than Mr. Wise suggests.»

Ivan Sharrock

«Having read Sam Wise's review of the Nagra-D recorder, I feel that I should make some comments. I write as owner of three of those machines and in my position of running a company specializing in the hire of location recording equipment, so I am very much aware of location sound requirements.

I disagree that DAT has become the «de facto» standard for field recording: the ratio of analogue Nagra recorders to DAT machines that we supply is approximately 6:1.

In my experience, the reliability of DAT machines and the DAT system is considerably less than quarter-inch machines (tape tearing in the cassettes, jammed mechanisms and even machines giving every impression of recording but not actually doing so being not uncommon).

Regarding size, weight and battery life: a large amount of location recording is not done with the machine hung over the shoulder, but usually with mixer and recorder mounted on a trolley such as the Urstacart. Most Fostex PD-2 users adopt this method of working - often using a car battery.

The internal battery in the Nagra-D should really be regarded as a back-up power supply, useful when changing external batteries or if a short sequence is to be recorded in a car. On many film locations AC power is available for lighting (either mains or from an alternator) so on feature film or TV drama shoots this should not present a problem.

Many recordists have used stereo Nagras on mono shoots in a two-track mode. Now, with stereo sound being recorded on location a four-track machine is needed to do the equivalent job (have you tried to monitor two stereo machines recording different tracks simultaneously ... much easier with a four-track machine).

For same strange reason, Nagra recorders have always been associated mainly with film sound. To my knowledge, no recording studio has used the Nagra T, a mains powered non-location recorder, for mastering although it must be one of the finest analogue recorders available. The Nagra-D would be an excellent four track recorder for classical music recording, both on location and in a studio. A studio based recorder does not need to be big and heavy...

As regards the price - true, you could buy two Fostex PD-2 machines for the price of one Nagra-D - but if you decide to hire the machinery for your production - two Fostex PD-2 machines would cost you more than double the price of hiring one Nagra-D.»

Nigel V. Woodford

PS: Users so far: Ivan Sharrock (for «Little Buddha»)
Royal Opera House
Glyndebourne
International Classical Music Awards

And if we talk about headroom?

Is headroom necessary with digital recording?

This would appear to be a reasonable question, however, it is clear the term does not have the same meaning for everyone.

However, before we start a dispute, let us clarify the situation.

Luckily, we discovered in an article by Frank Beacham entitled «Professional DAT on Location» published in the May 1993 issue of the TV Technology magazine, an interesting debate between Richard Lightstone and Jerry Bruck on this subject.

We'll let them speak...

«No tape noise

Richard Lightstone, C.A.S., a Fostex PD-2 owner, said the 16-bit DAT format offers dramatic improvements in dynamic range and tape noise over traditional analog tape recording. «Most conventional film sound has been recorded on mono Nagras at 7-1/2 ips. Very few production mixers record at 15 ips,» Lightstone said. «Now we've moved to the 16-bit digital domain where we have no tape noise at all to deal with.»

Having used his Fostex to record location sound in hot, dusty Zimbabwe for the feature «Bopha,» Lightstone said the headroom of DAT is more than adequate for the dialogue and effects work that dominates most motion picture location recording.

The headroom advantages of 24-bit recording advocated by Nagra-D proponents are «bogus,» said Bill Peugh, director of International Audio Technologies, Ltd., Chantilly, Virginia, the U.S. importer of the Stelladat recorder. «The headroom is the same for both formats: 0,» said Peugh. «You don't have headroom in digital. What they are saying is they can put more signal on tape. Headroom is the wrong term. That's a word for analog guys who are giving these things attributes that have nothing to do with what they are doing. I don't know anybody who's found true 16-bit to have any noise or dynamics problem.»

Persistent terminology

Jerry Bruck, New York's Nagra-D representative, responds: « Bill Peugh is technically correct when he says there's no such thing as headroom on digital. Because when you've run out of room, you've run out of room; there's nothing left.

«Nonetheless, the term persists in digital,» said Bruck. «Headroom refers to how many dB you are going to let there be above the signal level that you nominally want to put on tape. Most people want to leave at least 10 dB of headroom for the signal that comes along that they don't expect. All that happens in digital is that subtracts from the overall dynamic range of the recording». Bruck said the Nagra-D allows the mixer to run the recorder at a considerably lower recording level than DAT without sacrificing dynamic range. «If you record to minus 10 on your DAT you could record to minus 25 on a Nagra-D and have just exactly the same signal in terms of resolution and dynamic range,» Bruck said. «That's what is meant by headroom.» ...»

We get letters

We receive many letters from Nagra customers all over the world, here are some recent examples.

PIERRE VERANY S.A.

I would like to say how impressed I was with the NAGRA-D, both by the analogue and the digital quality. Every nuance which occurs in the studio is conveyed to the listener. It is truly a revolution in sound recording technology.

This revolution for us, who convey musical emotion via technology, is of primary importance. The NAGRA-D today makes its entrance into recording history in the same way as the most prestigious musical instrument.

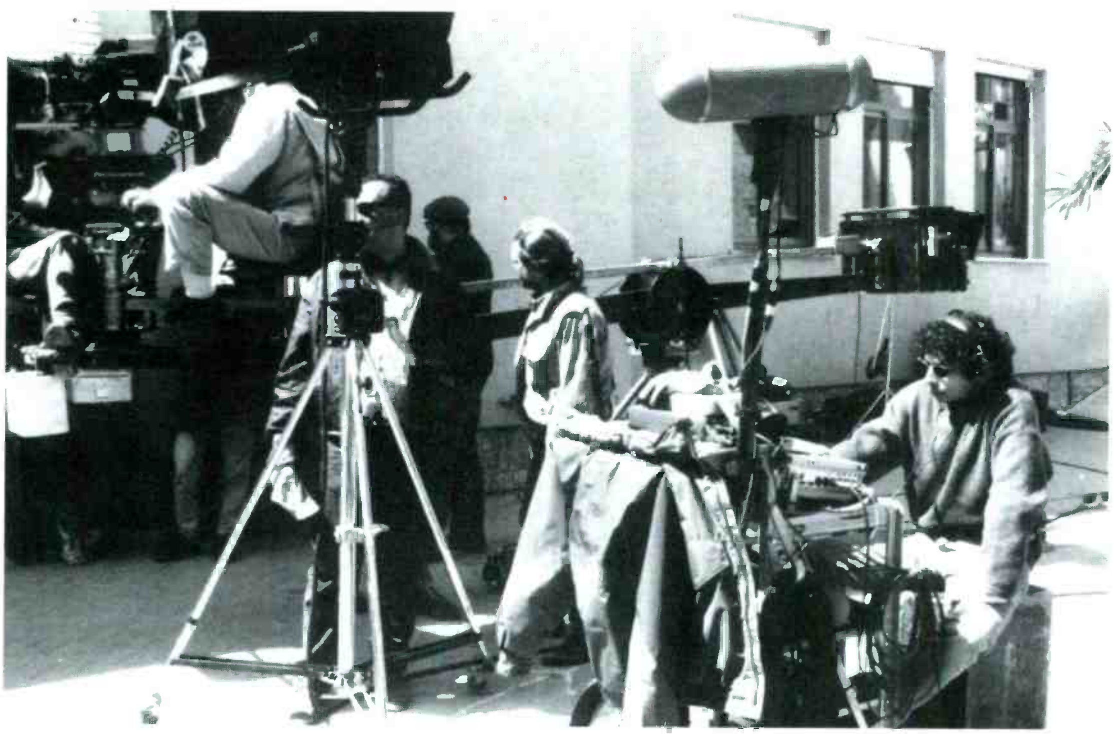
MAURIZIO ARGENTIERI

Sound mixer on the film «Il giorno di S. Sebastiano» recently shot in Sicily.

Finally! a star is born. A recorder that has the possibility to change the manner in which recordings are made.

In this film which is set at the turn of the century I found myself recording the conversation of two people with the background of a carriage and horses passing behind their shoulders. I absolutely didn't touch the input level adjustment and I had no fear of saturating and yet I was able to make a recording which was true to life. It should be added that the input pre-amplifiers are of exceptional quality.

The great advantage of this machine is not only the dynamic range but also the four channels which allow the independent recording of sound, be it dialog or the ambient effects recorded in MS stereo. This saves precious time in production and have the synchronous effect with the scene at the same time. Previously this was only possible with two machines, one for the dialog and the other for the stereo effects.



On location of «Il giorno di S. Sebastiano»

The recordings were made in far from ideal conditions, in forests, in the pouring rain and up in the mountains with temperatures as low as 0°C.

I really appreciated the possibility to make this recording using the NAGRA-D, a machine that will without doubt revolutionize the way films are made in Italy and the world.

Congratulations NAGRA!

KUDELSKI SA
Nagra Tape Recorder Manufacturer
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THE

We hear a lot about advancements in signal processing and in the electronics that make things work. This algorithm speeds up the process 10%, that computer, at twice the price, works 100% faster. Value for money? Certainly—if it is useful and brings a unique level of improvement over existing technology, we can even call it progress.

In contrast, we hear very little about control surface design. The layout of the control panel is not generally regarded as greatly affecting the speed of operation. If all the controls are within reach of the operator, once you learn where they all are, they are each just one action away. And so, the difference between surface designs is largely aesthetic.

The controls that lie beyond the reach of the operator are quite a different story, however. On a traditional long console, they are at least one push of your chair away. On a virtual, or fully assignable, console they are physically removed, and only the correct sequence of keystroke commands will bring them into view and within reach.

This is a problem. Each long console employs out of reach controls similarly while each virtual console uses differing and sometimes complex keystroke commands.

The technology behind any use of digital remote control of audio is staggering and must be acknowledged with the highest respect. The human-digital interface that translates the operator's intent into the necessary operational language seems to be becoming clumsier with each advance in signal processing.

If we consider a variety of tasks that every sound engineer must perform in the course of any complex music recording or mix, and count the number of actions, eye movements, time taken, and amount of operating instructions that must be retrieved from memory to complete these tasks, we can, in effect, measure the console's ergonomic resistance.

It is fair to assume that there is a limit to human mental resources available to us through the duration of mix. Subtracting the console demands from a larger, fixed number will give us a picture of the relative Creative Space each console type allows. It is within this creative space that listening, balancing, and fader moves occur.

Fig. 1 shows the test results for 'long' consoles. This large surface requires many eye movements though very few actions are needed as all controls are dedicated and are only one or two actions away. The simplicity of operation is reflected by the minimal operational instructions leaving us with a healthy degree of creative space (69.13%).

Assessment of identical tasks on a variety of virtual systems appears in **Fig. 2**.

The breakdown of eye movements is surprisingly similar in each case (22.73% and 18.94% respectively) considering the significant increase in keystroke activity. Creativity time is, however, reduced to 57.77%. Given sufficient operational familiarity, the time required for operations is also virtually the same. This explains how at trade show demonstrations the operations seem quick and efficient although, we, as observers, cannot see the mental gymnastics involved. The demo looks impressive, so long as sufficient attention is devoted to the command sequences.

Ideally we want a surface we can operate in our sleep—while continually daydreaming, immersed in the creation of an acoustic image.

When new technology demands more from the user to accomplish the same task as older, cheaper technology, we must question: 'Is this progress?'

The reasons for these additional actions and keystrokes are quite understandable. An axiom of console development states that if you reduce the number of controls on a panel, thus necessitating reassignment of those controls, you ultimately and inevitably increase the number of actions on the part of the operator to compensate. The logic behind this statement is clearly obvious and has pervaded console design work for over 50 years. Designers accept this law and work within these bounds to keep this additional activity to a minimum.

> On a virtual, or fully assignable, console the controls are physically removed, and only the correct sequence of keystroke commands will bring them into view and within reach.

Ergonomic testing

After compiling a list of relevant tasks for testing assignability, the local 'expert operators' of a variety of current virtual consoles were located and asked—in the comfort of their own studios—to perform each test in 'slow motion'. This enabled assessment of the minimum number of eye movements, key strokes and other operations necessary to complete each task. The tests were then repeated in real time and timed to ascertain the fastest speed of an experienced operator.

Since it was specifically the surface demands upon the user that were to be tested, this approach measured surface demands and not electronics—indeed, it confirmed that the weak link is an ergonomic one.

It should be noted that since the reaction time of the operator was always slower than that of the electronics, even a life-sized photograph of a test surface would reveal identical results. Consequently, a life-sized architectural drawing of the *VSC60* design was used to test its methodology.

When various consoles are compared in terms of ergonomic efficiency, it is difficult to quantify or measure significant differences once an active control comes within reach. From this moment, the operator's familiarity with the control surface results in one further action to manipulate the control. For this reason, these tests are concerned with the physical, mental and visual demands of the operating surface in bringing 'out of reach' controls within the reach of the operator.

The following are three exercises representing a section of the tasks regularly performed in the operation of a 60-channel desk used for sound balancing.

Fig. 3 shows the comparative number of actions, eye movements and time (in seconds) required to re-configure ten audio channels into a particular order ready for control while sitting at the centre of the console.

Since long console operators are used to sitting off centre, the may not need to configure a number of channels in the test. **Fig. 4** shows the same test where the long console operator does not need to reconfigure six of the ten channels. **Fig. 5** shows comparative numbers of eye movements required to address Aux Send 1 of 10, 40 and 60 audio channels. It should be noted that visual interrogation of a control surface is a distraction from the creative task and is a recognised source of fatigue.

A very small console

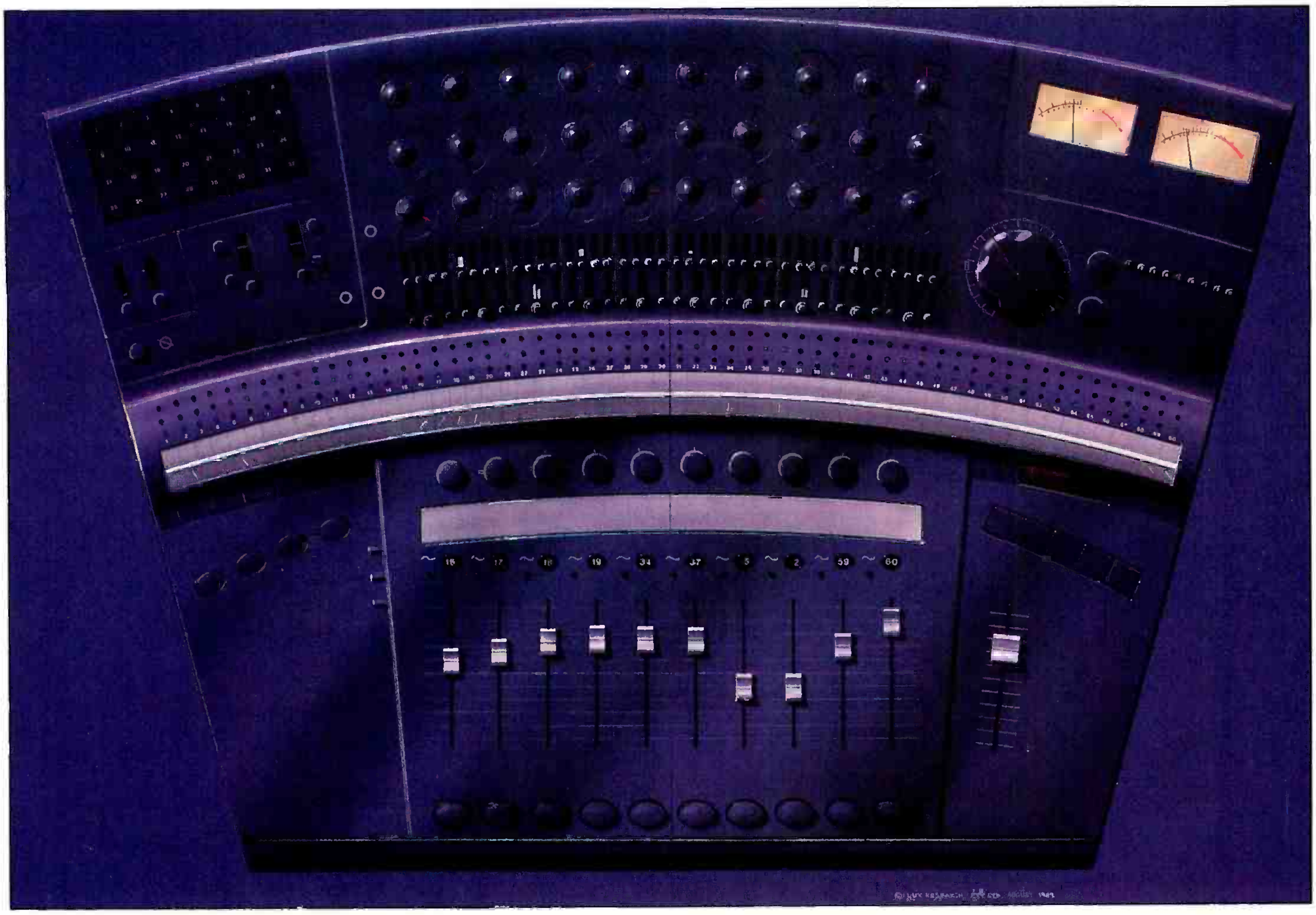
The experimental design presented here—the *VSC60*—is for use as a tactile controller for computer-assisted remote control of analogue or digital signal processors located remotely. The electronics beneath the surface is known technology and beyond the scope of this discussion.

It was only after discarding every human-digital interface known to man that something interesting happened. We redesigned a tactile control surface from scratch and replaced the mouse, trackball, keyboard, and touch screen with a simple strip. To perform the same tasks as the long and virtual consoles we see a radical shift in allocations of human resources.

The situation represented in **Fig. 6** seems to go against the axiom of surface design, yet these are the results given by the *VSC60* design. Eye movements are reduced to some 0.39% of operating time, and creative space is increased to 94.7%. ▶

On the page opposite, and still only a blueprint, Michael Stavrou's *VSC60*

VIRTUAL



REALITY

Michael Stavrou presents his designs and findings on possible directions in which console ergonomics may proceed. The results may shock some supporters of virtual technology

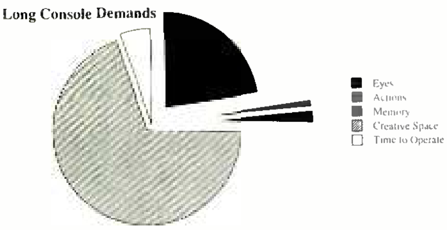


Fig.1: Long console demands

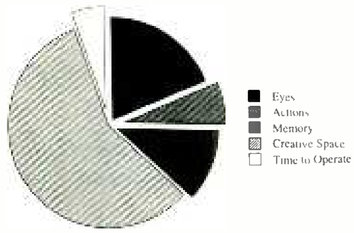


Fig.2: Virtual console demands

At first glance the very small console appears as a 10-channel desk. We understand the methodology so far: channel strips from the bottom: fader, mute, channel number, scribble script, pan pots, cue & echo sends, and equaliser section on top. Secondary controls are placed in a self-assigning Ancillary Block: routing module, high and low-pass filters, compressor-expander, and PHASE button. Note the lack of multifunction controls. Even from across the room, each knob does what it looks like it should. Each knob is dedicated to a specific function while the channel strips are reassignable.

The Main Selector Strip represents a long 60-channel desk to the operator. All inputs are stretched out before the engineer providing a field of view similar to that while looking down upon an orchestra from a balcony.

Repositioning your chair before a long desk would access approximately 220 new controls. Touching the Selector Strip alters the entire face of this 10-channel desk (220 knobs) to appear as though you have moved to that relative position. For example; you are sitting in front of the drums; touch the strip and you are before the violins. Some multifunction designs require 400 keystrokes to bring all these parameters to within reach, as opposed to this single action.

A redundancy of assignment indicators stabilise your orientation. As the new faders snap into position, channel numbers appear within the mute buttons, your own handwriting materialises along the fader scribble script, and slashes along the selector strip confirm assignment.

Perhaps the most popular method of assignment will be the Priority Wipe mode. Instead of your finger representing your chair position, the order with which you touch or wipe the strip is the order which the channels line up. You might wipe across six drum tracks, touch the bongos, timpani, and vocal to achieve an ergonomic spread (a customised selection of channel strips arranged to provide interesting, convenient, and creative combinations of instruments or a combination of faders that fit the hands enabling quick or complex moves to occur that would otherwise not be possible) of nine channels with only four actions. The equivalent performance on many other virtual consoles would take up to 28 actions.

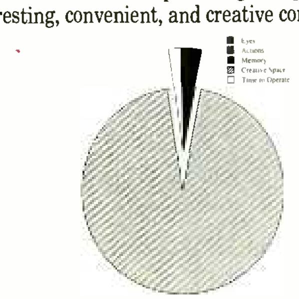


Fig.6: *VSC60* demands

A sense of overview appears to be sadly lacking in many of today's virtual systems. Here we can see at a glance, via overview indicators above the strip, the presence of a signal, its overload status and so on.

The curved nature of this tactile strip adds further to the engineer's orientation in the same way that you can easily (even blindly) locate the two o'clock position on the steering wheel of your car.

These improvements in recording console efficiency are solely the result of surface design and not that of advanced electronics or software. This new approach inspires a host of features that you, no doubt, are contemplating already.

Today, studios that possess the current crop of assignable virtual consoles rely on the in-house engineer to operate them, as they often require months of practice to master. But control surfaces for musical applications should appeal to the musically minded. Is modern technology enhancing our creative abilities or alienating right-brain-dominant artists in favour of left-brain-dominant technicians? We could be at a turning point in ergonomic design that will influence the very nature of the music that we produce. While the listener is often oblivious to the methods by which the music is created, he or she remains sensitive to its energy and feel.

The union of man and machine depends on the two components 'understanding' each other. The adoption of a curved, touch-sensitive tactile selector strip may prove to be the missing link between engineer and software that restores a more traditional relationship between the conception and creation of music. ■

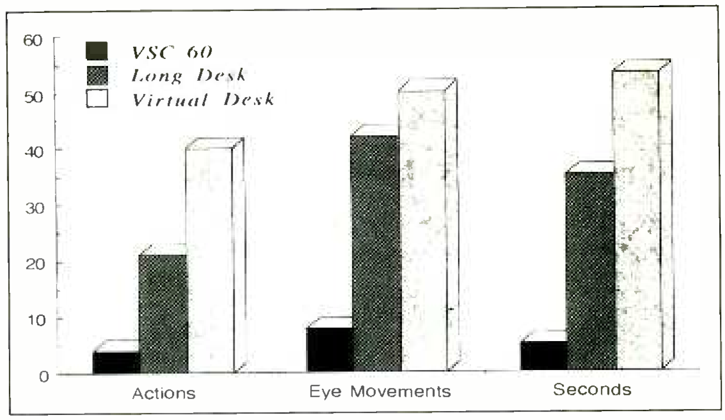


Fig.3: The comparative number of actions to reconfigure ten audio channels while sitting at the centre of the console

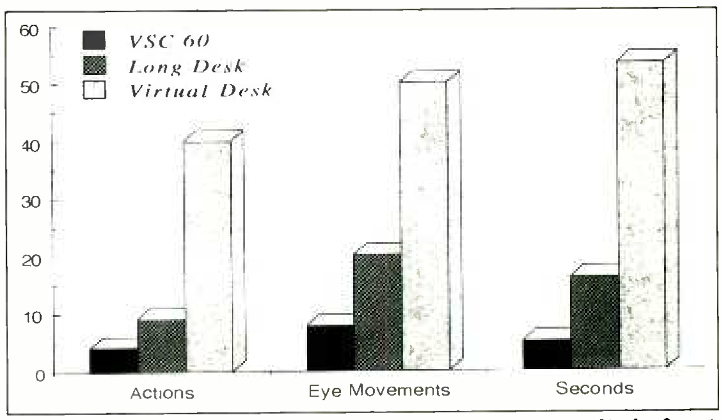


Fig.4: The same test when the long console operator sits in front of the left monitor to avoid patching six of the channels. Here the long console operator patches only three channels while the VSC and Virtual designs reconfigure all ten

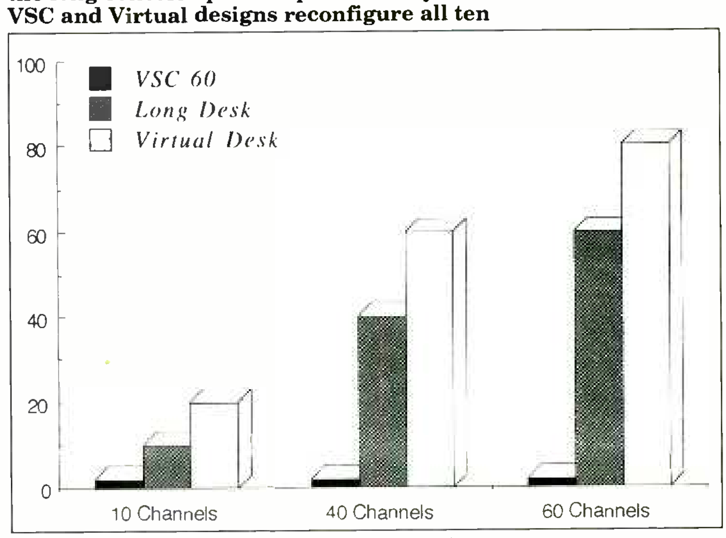


Fig.5: The absolute minimum number of eye movements required by each surface design to focus clearly onto the retina, an increasing number of parameter controls

Michael Stavrou is a recording engineer, producer and console designer who worked for ten years as a staff engineer at AIR Studios in London and Montserrat. Stavrou presently lives in Sydney, Australia where he is a Principal of Flux Research Pty Ltd, an audio research and development firm.
It should be noted that the *VSC60* embodies licensable technology with patents granted and pending. Any response to the ideas presented here at PO Box 397, Mosman, NSW 2088, Australia.

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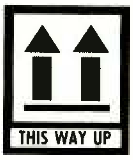

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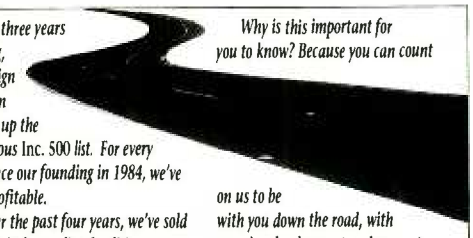

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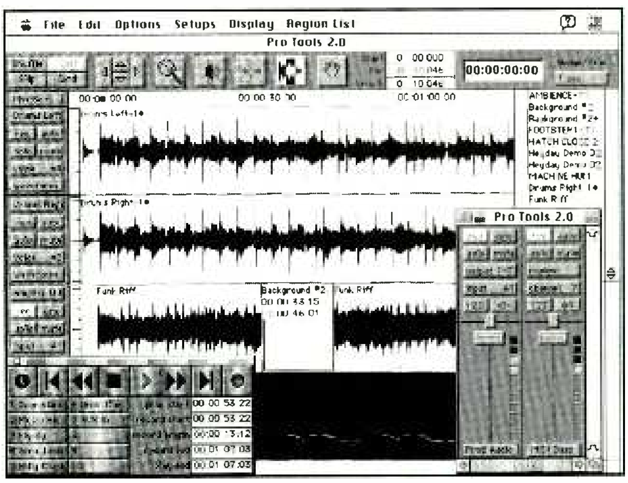

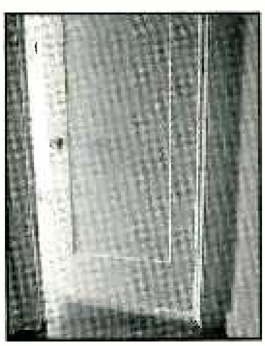

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1994
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24 tracks from one disk
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16 tracks from one disk
Scrolling waveforms
Frequency Domain time compression
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1991
MFX
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8/16 track playback
Audio Freeze frame
16 Zoom scales

1990
Series III Rev 8
Dynamic Voice Allocation
8 track disk recorder

1988
Series III Rev 7
Fast disk transfer card
Caching of software display
32-bit processor added

1987
Series III Rev 6
Cue List timecode sequencer
CAPS music sequencer
Integrated mono disk recorder

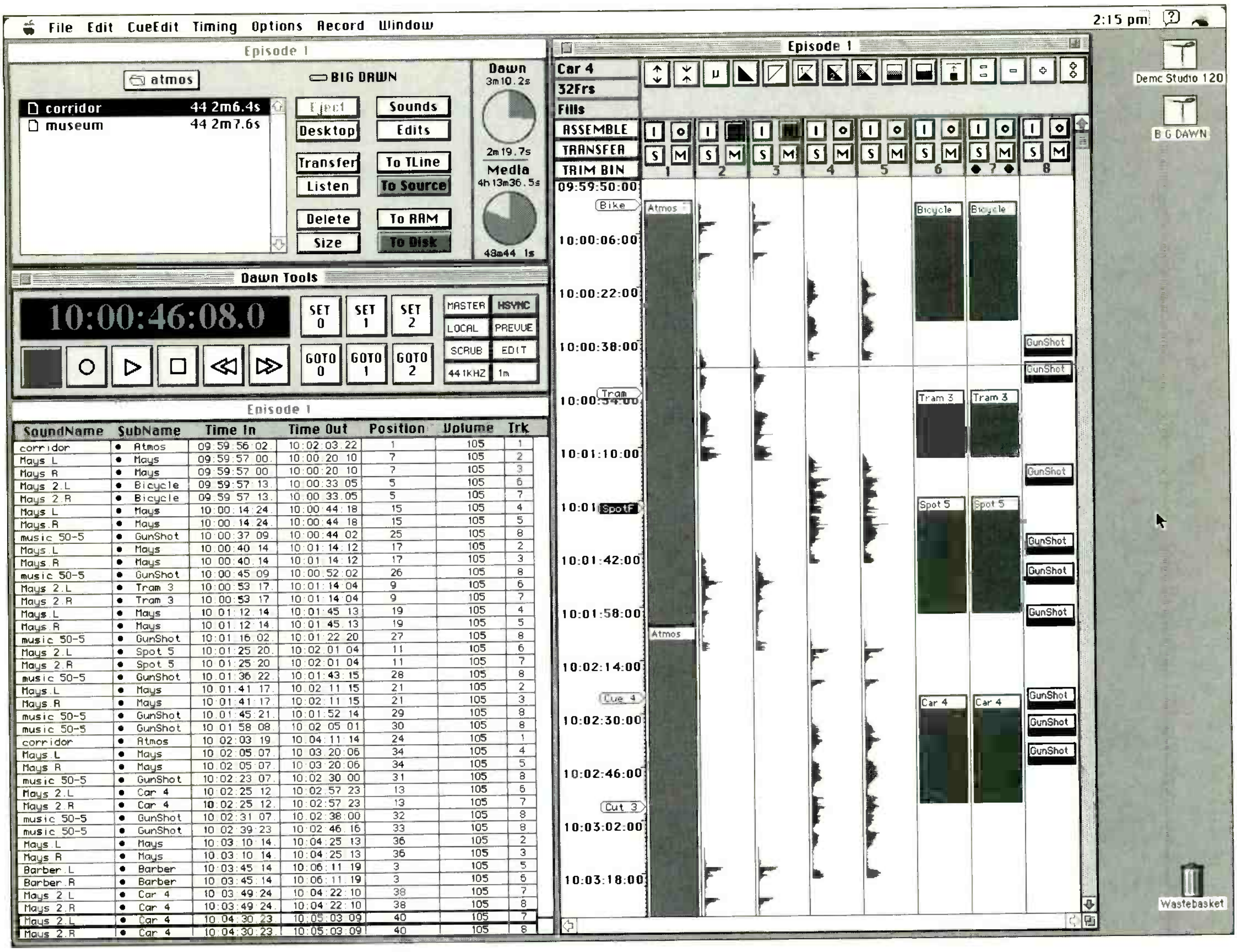
1985
CMI Series III
First 16 bit sampler
16 Voices
Shared Memory Architecture

1982/3
CMI Series IIX
HiFi sampling
MIDI implemented
Page R sequencer

1980
CMI Series II
First Real Time Graphics based Music Sequencer

1978
CMI Series I
First Sampling Digital Workstation
Waveform Editing

1976
QASAR
8 channel Fourier synthesis
Light pen interactive graphics
Dual CPU architecture



***DAWN II* makes screen work more durable**

DAWN II

Aimed at film and video postpro, *Dawn II* now offers integrated sync, digital I-O and 48-channel operation. Patrick Stapley reports

Dawn II is a multitrack digital audio workstation aimed primarily at film and video postproduction and broadcast. Its applications include, tracklaying, ADR, dubbing, on-line video editing, and drama production.

This second-generation system was launched last September, since when 70 systems have been sold putting the total number of *Dawn* sales at around 200 worldwide. Apart from ongoing software revisions, the major changes made to *Dawn II* have been hardware-based —time code synchronisation has been integrated within it, the system is now fully configured for digital I-O, and a more modular approach has been incorporated using accessible pull-out cards.

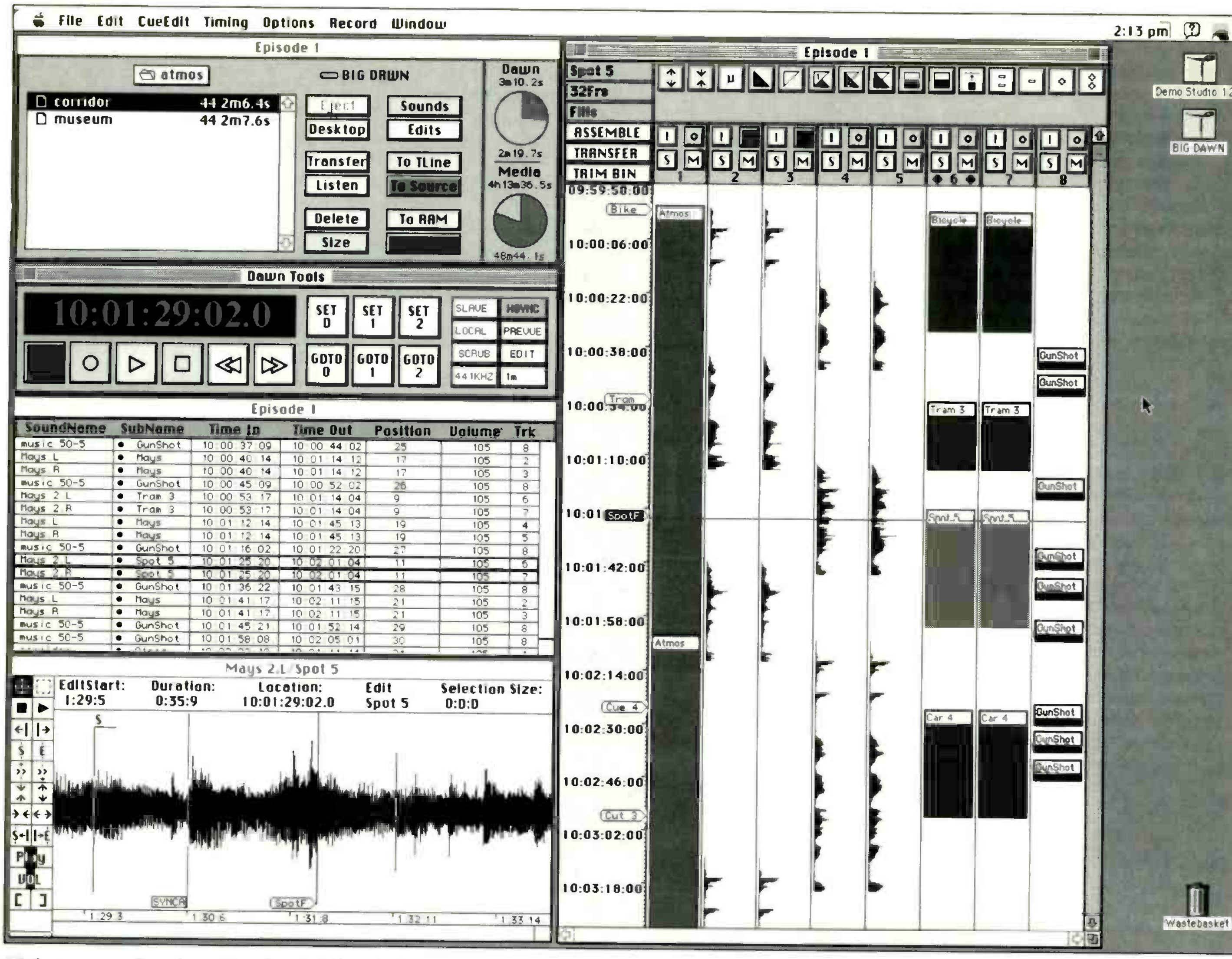
The hardware components of the system consist of a 3U-high rackmounting audio processor (up to six of these units can be linked together to provide a total of 48 channels all with individual I-Os), a *Dawn II* storage rack operating with hard disks on pull-out trays and-or optical discs (up to six racks can be connected giving a maximum of 36 track-hours per eight channels), and a *Macintosh* running System 7 or higher (a *610* or *650 Centris* is recommended). User interface is via keyboard, mouse, trackerball or pen-and-tablet; or optionally a comprehensive dedicated hardware controller, the *SP-1*, can be connected.

Each audio processor provides eight analogue I-Os on balanced XLRs, and eight digital I-Os selectable between AES-EBU or SPDIF. A–D and D–A conversion is Delta Sigma 64 times oversampling. Up to four external machines can be independently controlled from *Dawn II* via Sony 9-pin and VPR3 Motionworker protocols, and the system itself can be controlled by external devices via the Sony 9-pin interface. VT emulation and a sophisticated autoconform package are also included with the system.

Mixview and editing

The main interface display in *Dawn ii* is the Mixview window. The display has been designed to emulate a film dubbing chart with audio Cues marked out and scrolling vertically rather than horizontally as with the majority of other systems. The maximum number of tracks that can be displayed at one time will vary depending on the size of the monitor—a 20-inch screen, for example, will show 20—additional tracks are viewed by scrolling the display.

At the left-hand side of Mixview is the time ▶



Main screen showing waveform edit window

code scale column which marks out regular intervals of time code and can be rescaled by the operator to expand or zoom-in on the tracks. Nameable mark points may be placed down the column to provide useful location points for editing and positioning of audio Cues. At the top of the Mixview window are a number of regularly used function buttons for CROSSFADE, NUDGE, CUE MUTE, and so on; below these are groups of individual track status buttons—SOLO, MUTE, INPUT MONITOR (toggles between the selected input and disc), and RECORD ENABLE.

Cues are graphically represented in two ways, either as block or waveform displays. Both types can be mixed between tracks and individual Cues can be assigned different colours—thus dialogue, music, effects, and atmosphere may be separately colour coded and subcoded. Additionally, tracks and Cues can be named, and Cues can have a comment attached to them.

Each track displayed on Mixview has its own analogue and digital inputs and output on the corresponding processor rack. Audio can be monitored and recorded directly to tracks, plus dropped-in and out either manually or automatically, or it can be recorded to the system's Trim Bin for editing and storage for later transfer to a destination track. Stereo audio is positioned on two tracks but is treated as a single Cue—this relationship can, however, be broken allowing each track to be manipulated separately.

The majority of editing functions are accessible directly from Mixview, although a separate Edit window is also included and certain functions can be implemented textually from the EDL. Functions such as time slipping a Cue or group of Cues in Mixview is a straightforward select and drag operation. Similarly, moving or copying a Cue to a different track involves a horizontal drag—as a safety precaution it is impossible to slip Cue sync while moving track. Additionally, Cues may be time slipped on-line by quarter or whole frame increments (mimicking the sprocket time for 16mm and 35mm respectively) by using + or - NUDGE buttons. This facility allows sync to be accurately matched while still locked to picture, providing a much faster method than the more traditional offset techniques. A Nudge Within function is also available that slips the audio while keeping Cue edit points anchored.

The system uses a Time Line that runs horizontally across the tracks acting as a 'playback head' with the audio scrolling past it. This also provides an alternative method for positioning Cues rather than dragging, nudging or retiming them; by using the Capture command the start of the selected Cue (either from Mixview or the Cue list) will move to the line. Alternatively the end of a Cue, or a Sync Mark (two available per Cue) can be positioned at the line. This Snap facility enables Cues to be quickly and accurately positioned in relation to video events.

In a similar way the Time Line can be located to the beginning and end of existing Cues to help position new Cues on other tracks. For example, a series of dialogue Cues may require a particular background ambience such as traffic noise, this can be edited to exactly follow the dialogue cuts by locating the relevant start and end points and cutting the new tracks in and out as necessary.

Another useful function is Strip Silence, which recognises silent passages in a recording and automatically divides the audio up into individual Cues, rather than having to manually separate each section.

The system has not been designed to mix, it should be viewed more as a mag or multitrack replacement. However, individual Cues can be level adjusted using a pop-up fader window, and a future enhancement will allow dynamic fader movement. This facility has been met with a mixed response—one has to remember that a lot of *Dawn II* systems are used for tracklaying and that traditionally any dynamic level adjustments are left strictly to the dubbing mixer.

The Edit window provides a horizontal waveform display for the selected piece of audio, and can be zoomed in-out both in terms of time and amplitude to the required resolution (from sample level to the duration of the selected audio). A Time Line is also used in the display and corresponds directly to the Mixview play line–thus audio positioned on a track can be viewed scrolling vertically in Mixview as a block and horizontally in the Edit window as a waveform.

Cues are also shown in a text format in the Sound List window. This EDL gives details on source, name, time code, track position, and MIDI information. Each Cue can be assigned a MIDI ▶

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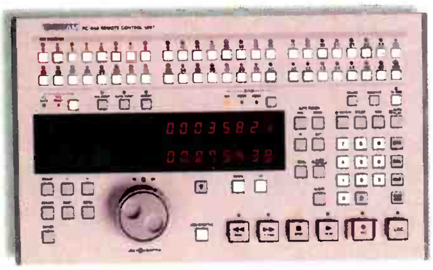

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note allowing it to be triggered from a MIDI keyboard as well as velocity controlled—this facility is popular in America where it is used for applications such as manually triggering Cues during live game shows.

Text edits can be performed directly from the Sound List window—these include altering time code start-points and changing destination tracks. Cues may also be selected from here, rather than from Mixview, for edit commands such as Splice (butt-editing two Cues together). A temporary 'clip-board' is available which allows Cues to be cut or copied from the list and looped and-or inserted into existing audio (either replacing a designated section, or lengthening a Cue at the specified point), and Merging (overlaying one sound on top of another) which will permanently modify the selected sound file. Using the Edit window, parts of Cues ('Bites') may be pasted to other tracks or copied back to the original track to create a number of repeat sections.

Extension can be added to any edit up to the full length of the original recording allowing facilities such as extended fades. The procedure for adding a fade-in pre the start of a Cue, for example, is to position the Time Line at the desired point before the start and issue a Fade In command—as with all fades, the ramp will be indicated in the display by a line following the time and shape of the chosen fade.

Crossfades are created between two Cues by placing them on separate tracks and adding appropriate fade-out, fade-in times and curves. The resultant crossfade can then, if required, be transferred to a single track, which takes a few seconds to compute. If adjustment is needed at a later stage, the crossfade will have to be split back into separate tracks.

Multiple Cues can also be combined using the Assemble facility. This can be useful where a selection of Cues have been used to build-up an effect over a number of tracks, and the operator wants to free-up space or simply create one easy-to-manage sound file. The composite will appear as a new Cue placed on the same track as the selected Cue with the earliest time value.

Dawn tools

The Tools window displays the current time code (can also display feet and frames), the TRANSPORT button, and AUTOLOCATE SET and GO TO buttons (three autolocate points are available including one reserved for the start of the mix). Jog-Shuttle modes are also accessed from here—rock and roll editing tends to be less smooth than some other systems and occasional artifacts are audible at low speed playback.

Other function buttons in the window include Master-Slave mode selector, House Sync selector, Local-Remote control mode selector, Preview (record audition) Mode selector, sample rate selector (32kHz, 44.1kHz, 48kHz), Record Duration Set button, and Edit window selector.

Files

Audio in *Dawn II* is filed using the standard *Mac* folder structure. The Transfer window lists the contents of the active volume (hard disk, M-O disc, and so on) in alphabetical order and by clicking on selected items, contents of folders can be viewed and files accessed. The display also shows the sampling rate, length and mono-stereo status for each file. A utility such as 'Boomerang' maybe added to help search for files.

Audio is further divided into two groups —Sounds and Edits. Sounds are pieces of 'raw' audio—this could be the complete slate from a film job, for example. Edits are specific Cues that have been lifted off the Mixview Window and placed in the Transfer Window—these could be a number of alternative takes that need to be saved for possible reinstatement later in the session. The Edit List and the Trim Bin (mentioned earlier) differ in that the Trim Bin can contain Cues from different jobs while the Edit List is specific to the current mix. Selected files can be transferred to the Mixview window by either offsetting them to the Time Line or positioning them relative to their original source time code.

Mix files containing all audio, edit and events data may be backed up to optional Exabyte or Data DAT archiving systems at four times real time. The system does not offer background loading or downloading.

Autoconforming and ADR

As mentioned *Dawn II* will import Sony and CMX EDLs and conform the original source audio accordingly either directly from external machines or from reels already recorded to disk. Once an EDL exists within the system it can be text edited before autoconforming, for example Cue start times and destination tracks can be changed. Depending on how the operator prefers to work, EDLs may be sorted by event, reel, or running time. To keep a record of sources, each Cue is identified by an event and reel number. Events can be auditioned before autoconforming, and partially conformed lists can be saved allowing the list to be completed at a later date. Audio handles may be specified by the user during the autoconform process and will typically be three seconds long.

ADR lists can also be imported or constructed providing loop recording. Each take can be stored sequentially in the Edit List or discarded as the operator wishes, and unlike some systems, takes can be aborted half way through rather than having to wait for the loop to finish. The system also generates ADR 'bleeps' to cue in the artist.

ADR can, of course, be executed in a more manual sense, with the operator dropping-in and out at the required spots.

New features

At the time of writing some new features were about to be, or had just been, implemented. These include sample rate conversion, pitch shifting, time compansion, dynamic level control, and nonlinear video. The nonlinear video option allows sound and picture to be randomly accessed simultaneously, providing instant locate and playback plus full jog and shuttle facilities for both components. The user can set compression ratios to either 25:1 or 50:1; with a 50:1 setting, storage will be approximately three minutes of picture per 100Mb.

It has also been announced that Doremi are participating in Avid Technology's Open Media Framework (OMF), that will allow *Dawn II* to directly import Avid files. ■

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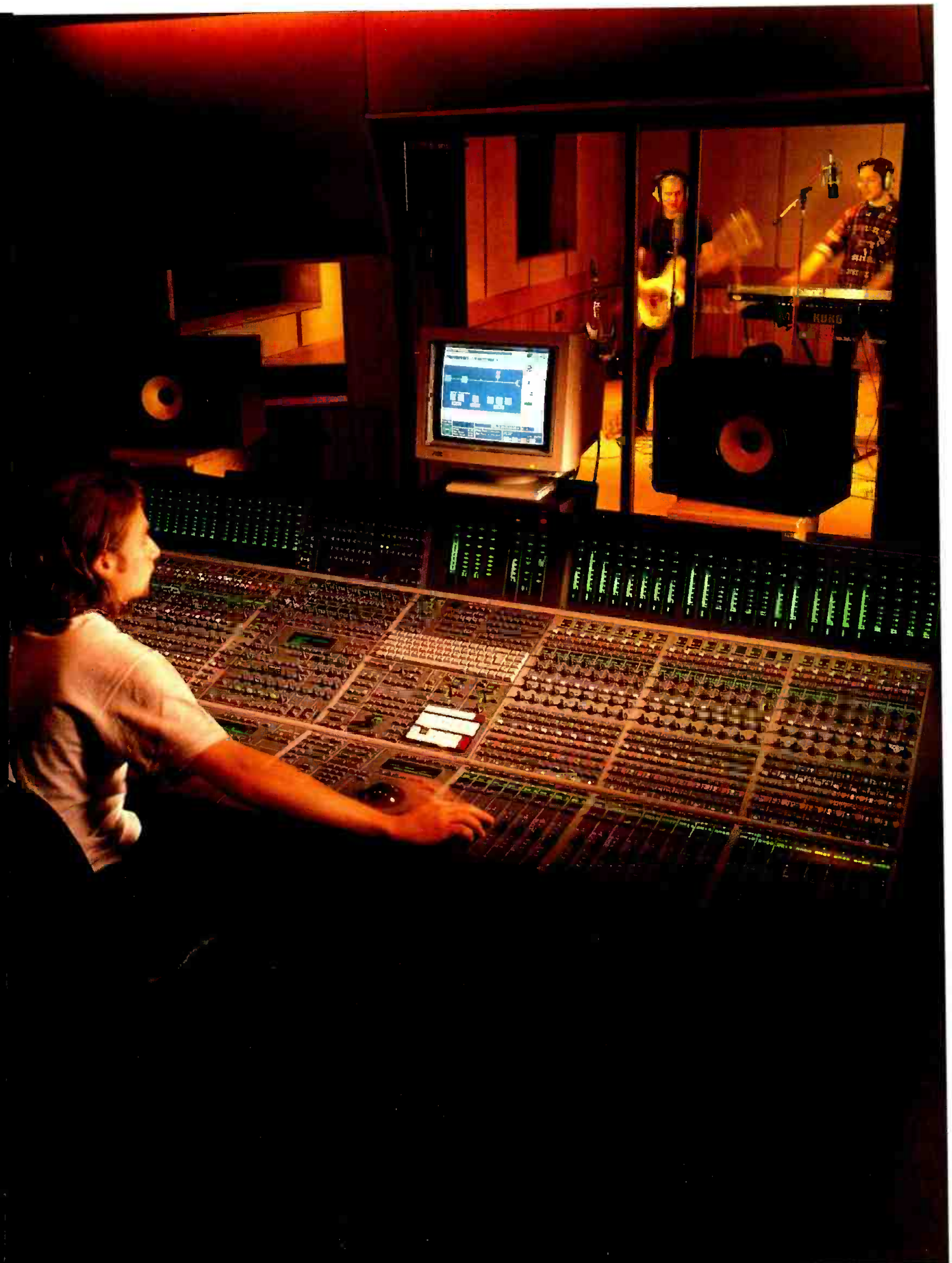

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T H E   C A P R I C O R N

Video editing, by its very nature, can be an extremely complex task. Today's state-of-the-art edit suites might be equipped with as many as half a dozen analogue and-or digital VTRs, routing via a sophisticated video switcher to several mastering VTRs. Having developed a series of time-code-based In-Out edit points on the various source reels—the Edit Decision List produced during off-line editing—for the on-line session these same reels will be laced up in the companion machine room, and then commanded to their respective SMPTE H:M:S:F locations under serial control from the editing controller. As each video edit is performed, the sources will either be hard cut, dissolved or otherwise combined to produced the master videotape. At the same time, sophisticated video special effects might be added, along with captions and titles.

But what is happening to the audio as the video edits are being made? For larger productions, the master reel will simply receive a rough audio mix from the source reels. If other audio elements need to be added—a music track, for example, or maybe extra sound effects, dialogue and Foley elements—after on-line editing the project will move into the postproduction suite. Here, the various soundtrack elements will be assembled to a time-code-striped multitrack, and then mixed against the edited picture as various mono, stereo and surround sound balances are produced.

But for most edit sessions—particularly for news and current-affairs programming where there is often no time to move the project through a post room—the various audio elements will be mixed and processed in real time as the video edits are being made. Dialogue narration might be pulled from a separate VTR, while actuality, effects and other soundtrack cues will be taken from the appropriate video source reels.

Multiple analogue and digital audio sources

As editors extend their creative options, it will come as no surprise that in recent years the number of audio signals being handled in video editing suites has increased dramatically. Quad machines started with a single track, one-inch B and C-format offered two, plus a time-code stripe. All recent digital formats, including component D1, composite D2, D3 and D5, feature four or more audio tracks. An added complication is the mix of analogue and-or digital I-Os available from these newer format machines.

During a typical editing session in a 4-VTR suite, there might be as many as 30 or 40 analogue-digital audio signals, including a tone generator, CD player, DAT machine, and maybe a small multitrack ATR. From these various sources, the editor must be able to quickly find the specific material to be added into the final video product. As well as the large number of source tracks to be handled, in many cases there will be up to four tracks to record on the master D1-D2 VTR.

To handle the multiple analogue and digital inputs from VTRs and other sources (including time code DAT machines holding, for example, music cues or sound effects), a new generation of compact, digital-capable edit suite audio mixer is required. Conventional audio consoles are often too large and cumbersome to be used in an video edit suite. What is needed is a design based on a small, assignable control panel, combined with flexible input and output assignment, level control and signal processing. Such a mixer allows all functions to be integrated within the editing process, and initiated from the editing controller.

> in a 4-VTR suite, there might be as many as 30 or 40 analogue-digital audio signals

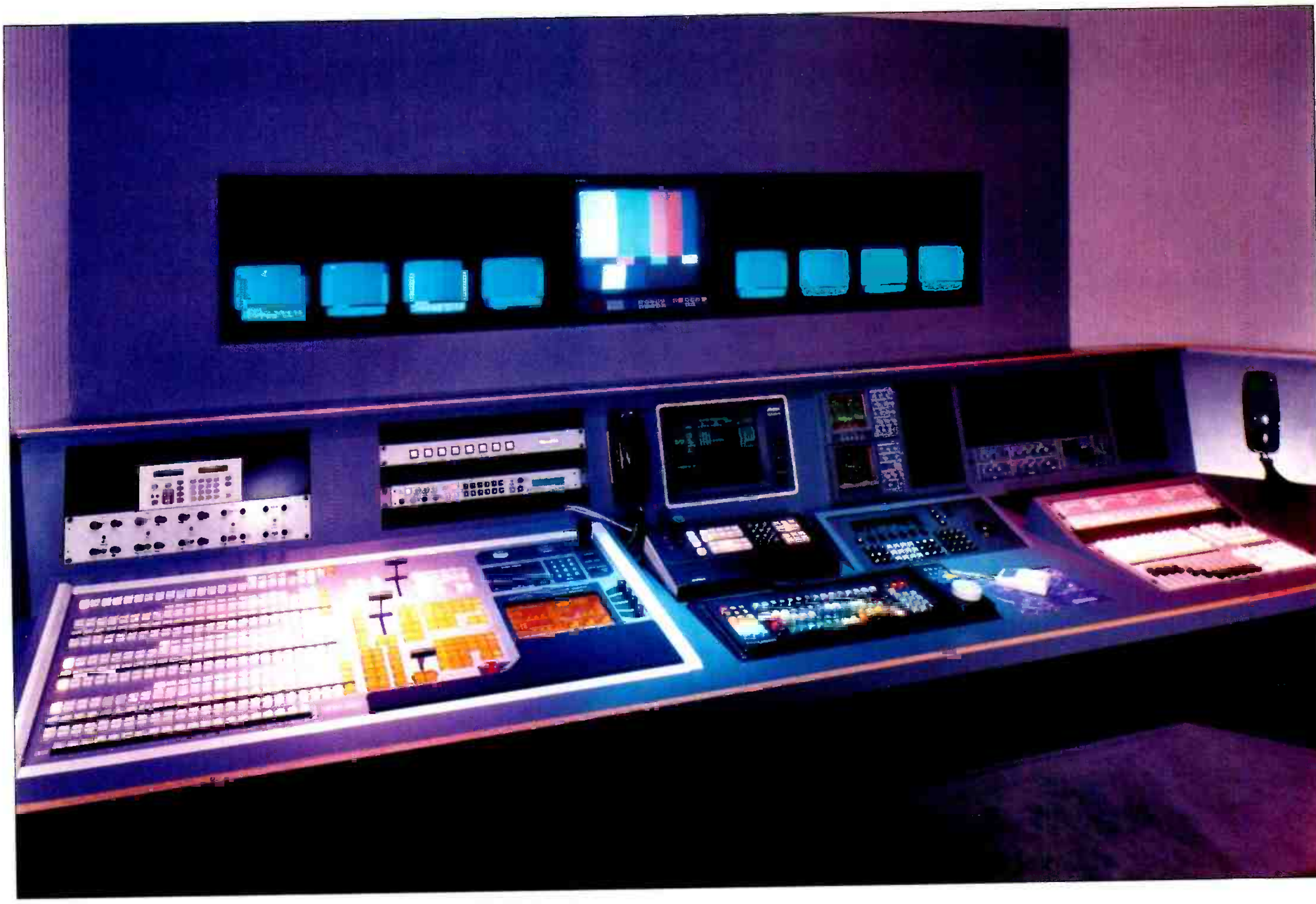
Graham-Patten Systems' *D-Esam 800*, digital edit suite mixer

And, unlike the architecture of a conventional audio mixer, it is very useful if the edit-suite mixer emulates the operating style of a video switcher. Now audio sources can be assigned in a Program-Preset fashion, as 'From' and 'To' sources. Levels can be trimmed in the conventional way using faders, and transitions performed by crossfading between buses—just like a video switcher—allowing precise level matching and repeatability. And, like video switchers, a separate Preview bus performs preview and monitoring functions, independent of the Program mix.

During a typical on-line edit session, the operator will assign video sources via the switcher to Preset and Program buses, and then perform a transition or dissolve at the edit point. A separate Preview Matrix allows various video sources to be fed to a dedicated video monitor to check, for example, the progress of a parallel reel that is being run in time code sync with the main feed. (While editing the product of a multiple-machine shoot, it is often useful to have all reels running concurrently, and then punch between them as the edit progresses to add, let us say, close-ups and medium shots.) The preview monitor displays the output from selected source VTRs.

Input-output assignability

A new approach is required for both the mixing hardware and the user control panel. The hardware must handle a large number of analogue and digital inputs; the control panel must combine flexibility in a compact size. Recent developments in A–D converters make it practical to digitise each analogue audio source within the edit suite prior to mixing. Concurrent advances in DSP



The D2 mixing suite at Edit 1, in New York's HBO Studios

SUITE TALK

technology now allow real-time EQ and mixing.

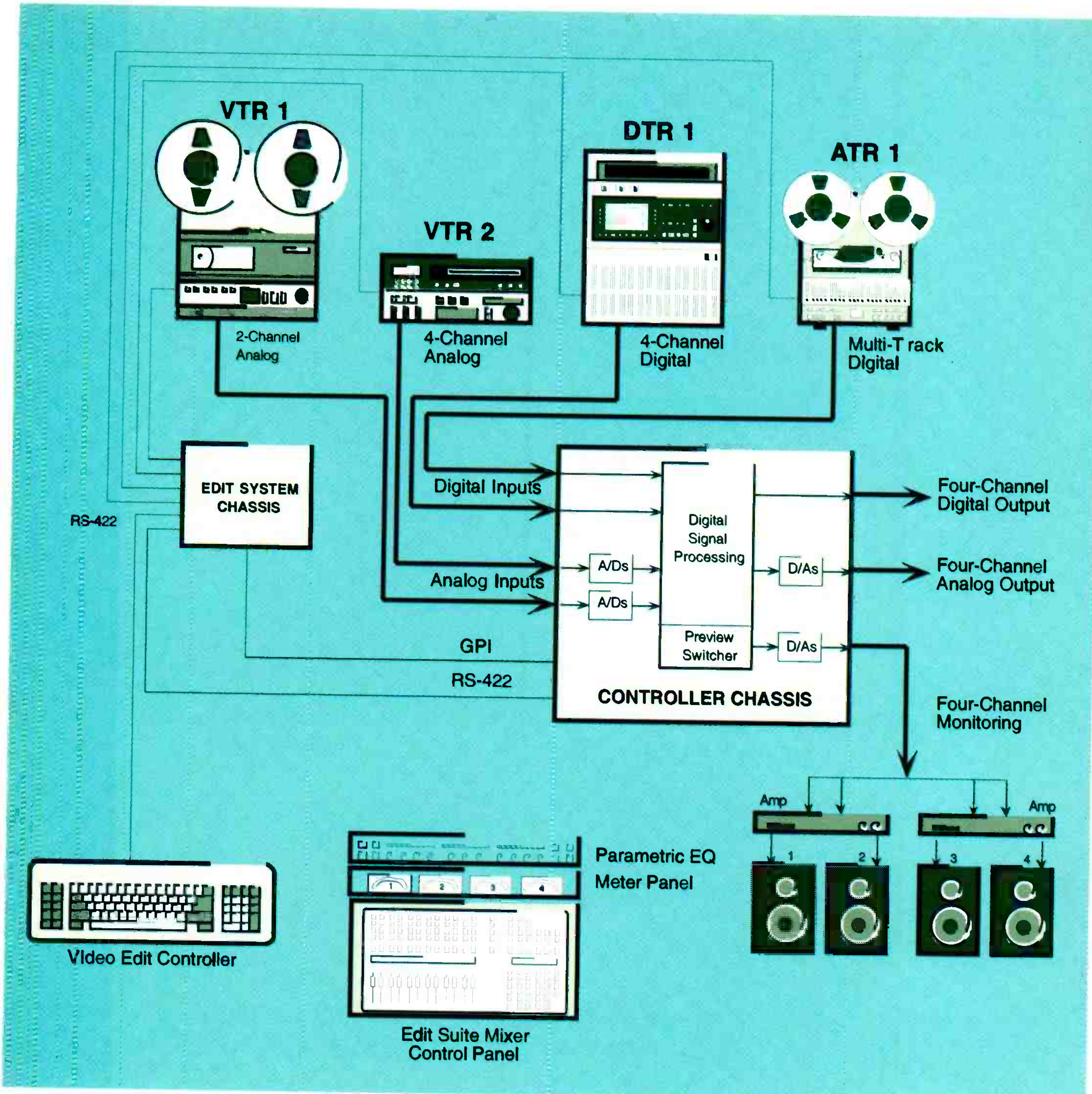
Audio mixers designed especially for video postproduction can benefit from a compact operator interface that mimics a video switcher, with Program-Preset functions plus a dedicated Preview and Monitoring section. To provide full individual control for each input, yet still reduce panel size, implies the use of assignment. But how should the unit be designed so that the mixer is fast and intuitive to use? Fortunately, one of the real benefits of using DSP is that the audio processing itself can be implemented totally in software. We now have the flexibility to manipulate our audio architecture in ways that would be quite impractical in a fully hardware-based implementation.

Common, assignable controls will reduce the amount of panel size. To make this work for, instead of against, the editor, each input-output bus assignment must relate precisely to the elements being used to create the edit. To better understand what is going on, let us consider the basic process of video editing.

A reel of tape is loaded onto a VTR, and the machine assigned an edit controller name: 'A-VTR'. The Video Editor wants to dissolve from video already on the record VTR to a segment on the new reel. The original source, from the 'C-VTR', is reconstructed exactly on the switcher Program bus for the match-frame cut. The new video segment, from the A-VTR, is set up on Preset ready for the dissolve. After the time code In and Out points are marked, the edit is made. (This is, of course, a gross simplification of the process, but it serves to illustrate the elements involved.)

First, notice that the 'From' configuration (on Program) and the 'To' configuration (on Preset) are handled at separate times. Using the same controls for both, through assignment, is therefore a logical initial step. For video systems the savings would be minimal. But, with dozens of audio sources and four or more buses, literally hundreds of audio I-O bus selection buttons are eliminated. ▶

Current generation video postproduction suites have to accommodate a wide variety of analogue and digital sources, routing to multichannel destinations. Michael D. Patten looks at the audio mixers designed for use in such an environment



Block schematic of audio requirements for a 'typical' video postproduction suite, incorporating analogue and digital signals

Second, once an edit controller name —'A-VTR', 'C-VTR', and so on—has been assigned to a physical VTR, that machine is accessed only by its 'logical name'. If the tracks required in an edit could be assigned to a limited number of fader channels using the same logical name, there would be no disruption to the editor's train of thought. Assignment can be performed via familiar assignment process in the audio mixer, rather than in the edit controller; now the audio mixer can associate a logical track with its correct physical track.

Our goal of assignment has been realised. By sharing the same selection buttons between Program and Preset, we can reduce by 50% the number of bus rows required. And the large number of input tracks from physical machines are assigned to a much smaller number of fader channels. In both cases, assignment blends seamlessly with the normal video editing process.

Virtual machines

While editing audio with video, tracks are quite frequently handled in groups that do not always have a 1:1 correspondence with physical VTRs or ATRs. Instead, they might involve only some of the tracks from a particular machine, or include tracks from more than one physical machine. (Consider the case, for instance, of a quarter-inch, centre-time-code ATR slaved to a VTR.)

A mechanism that allows these groups of tracks to be handled as a single entity would be very useful. Let us refer to such a device as a 'virtual machine', by which we mean any collection of input tracks being routed to the mixer; once defined, it can take the place of a physical machine in all operations.

What might seem to be introducing an additional level of confusion for the video editor turns out to be a real benefit. Assignment of a logical machine to a virtual machine in the audio console replaces the logical-to-physical assignment which, as we have already seen, has always been performed at the edit controller. Also, we now have an unlimited number of virtual machine groupings of physical tracks that can be preconfigured.

In essence, a track of a virtual machine (a virtual track) is simply a pointer to a mixer input, or physical track. Any number of virtual tracks can point to the same input, allowing a single physical track to be a member of several virtual machines. Groupings of tracks used most frequently during a session will depend on the type of work being edited—a sports programme, promos, news stories—and the way in which the client designated the tracks during the shoot.

Because of the large number of available virtual machines, and the flexible association between ▶

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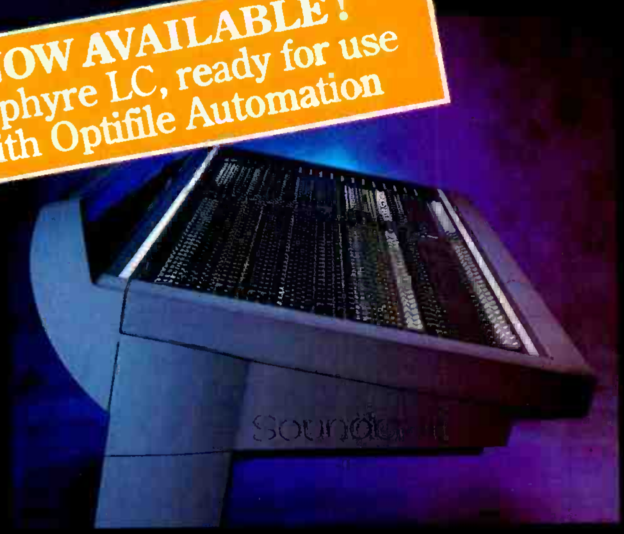

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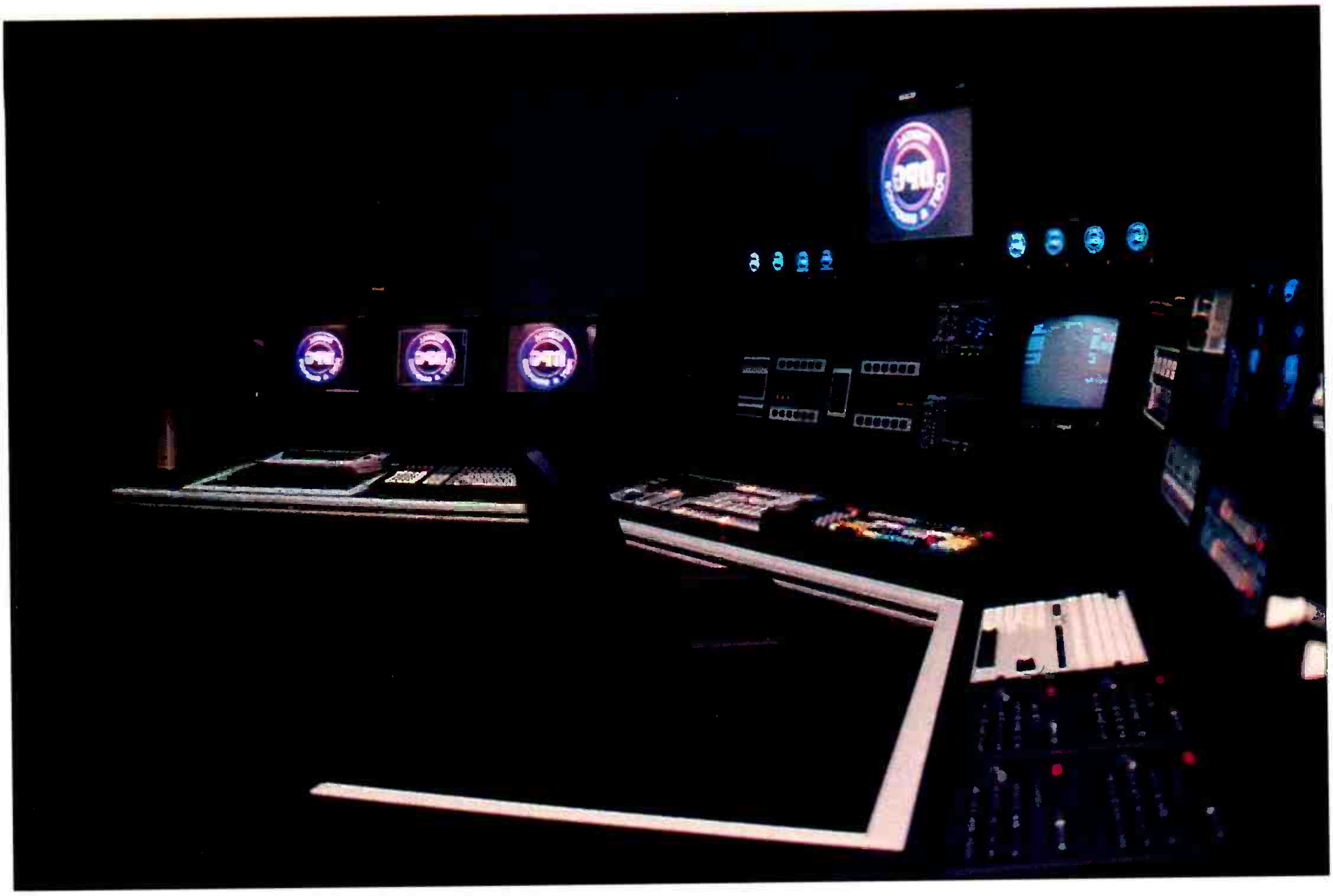


Inside the edit suite at Digital Post & Graphics in Seattle

inputs and virtual tracks, the VTRs in an edit suite can be configured to handle all requirements of its current client base. As these requirements change, so do the virtual machine configurations. A virtual input matrix combines the flexibility of an electronic-jack field and an analogue-digital audio routing switcher. Any audio input can now be assigned to any fader—and any or all output channels—on an edit-by-edit basis if necessary.

Client A might use Tracks 1 and 2 of D2 VTR for stereo music, and Tracks 3 and 4 for commentary and natural sound. It might be useful to define a virtual machine which has just two tracks; let us refer to it as 'DVTR1-AM', to use when editing the music. Later, Client B comes in for a similar session, but prefers the music to be recorded on Tracks 3 and 4. By defining a new machine, 'DVTRl-BM' that uses Tracks 3 and 4 of the same physical VTR, the music is still accessed in exactly the same way as for Client A.

This simple example provides an idea of what can be achieved with virtual machines. In practice, far more complex situations can be reduced for the editor by constructing the suite's complement of virtual machines to match current requirements.

PRACTICAL IMPLEMENTATION OF A DIGITAL EDIT SUITE AUDIO MIXER

An audio mixer incorporating all of the concepts discussed in this article, both in processing hardware and control interface, comprises an audio processor, housed in a 3U-high frame and capable of accepting up to 56 audio inputs. These are grouped in blocks of eight inputs that can be either analogue or AES-EBU-format digital sources. Digital inputs, although paired via the same cable, internally are treated completely individually. Motorola 56001 DSPs are used for mixing and monitoring the audio signals, controlled from a Motorola 68001. Additional 56001 DSPs can be added to perform audio equalisation. Four serial ports handle data from the editing system, control panels, and maintenance terminal.

The control panel provides 12 fader channels, four Program and four Preset buses, plus four-bus monitoring. Controls are available for performing independent transitions on all buses, selective edit enables, plus necessary machine and track assignment functions.

A total of 255 virtual machines are available, each having up to nine standard tracks and a cue track. A total of ten logical machines are supported, A–H, plus Aux and R. Continuous display of both program and preset status is supported through the use of dual-colour LEDs.

In addition, complete logical machines, as well as individual logical tracks, can be assigned to a fader channel, allowing a preset balance between tracks to be maintained. Also, adjacent faders may be linked together to provide simultaneous control of any combination of logical tracks. Each fader channel supports separate levels for Program and Preset, and individual fader channels can be separately disabled from edit system control for manual adjustment. Monitor buses can also be trimmed individually for level, and muted. ■

> a track of a virtual machine (a virtual track) is simply a pointer to a mixer input

Additional uses for virtual machines

Let us further examine the use of virtual tracks, which can be imbued with special characteristics without constraining the physical track. An individual input to the mixer might be assigned to two virtual machines, one making use of the special characteristics and one that does not.

By way of an example, consider the case of a special virtual track that has the ability to replace the audio of selected tracks when the machine is in shuttle or varispeed play. Now, if the analogue cue track output from a D1 VTR is connected to a mixer input which, in turn, is assigned to one ▶

ESAM I AND II SERIAL MIXING CONTROL PROTOCOLS

If the video editor is the source of how each of the various source elements, including audio, will be used to build the master reel, how will this information be relayed to the audio mixing console? Obviously, all of the operator's attention will be concentrated on driving the edit controller, and adjusting video crossfades and so on. Some means therefore needs to be devised to relay details of audio crossfades and transitions from the edit controller to the audio mixer.

Fortunately, there is now a standardised serial interface protocol which includes lists of multiple-byte commands for controlling audio assignments, crossfade profile and other instructions. Developed in 1983, and first used at the 1984 Winter Olympic Games, the ESAM (Edit Suite Audio Mixer) protocol has now become an accepted standard for controlling audio mixers in the video postproduction environment. The ESAM protocol is a simple software language for communicating directly between edit controllers and audio mixers. Source selection, transition control, and audio previewing can be accomplished entirely from an edit controller.

Concurrent with increasing demands for audio creativity and sophistication in edit suites, the original ESAM protocol has evolved into the widely accepted ESAM Serial II. This enhanced protocol, along with recently added extensions, addresses both basic and advanced mixer capabilities, such dynamic fader control, 4-channel monitoring, memory register upload and download, and much more. Because of its inherent power and straightforward implementation, ESAM Serial II is now used by virtually all major editing system manufacturers.

For the video operator, ESAM Serial II provides an intuitive approach to postproduction audio; for editing system manufacturers the protocol represents a logical pathway to added new features, and an expanded customer base. ESAM II is designed for easy implementation, with commands expressed in editing terms. ■

of these special tracks of a virtual machine, the mixer would automatically switch back and forth between the cue track and the normal digital outputs while searching for edit points.

Just as many virtual tracks might point to the same physical input, several logical machines can be assigned to the same virtual machine. Such a setup is useful for automating special audio effects under edit-system control that would previously have been done manually. The advantages are that such sequences are now repeatable, and can be previewed as many times as is required. (They are also documented, to some extent, in the Edit Decision List.)

Using this capability, different EQ settings can be set for the tracks of two Logical Machines, both of which are assigned to the same virtual machine. A smooth dissolve between these two settings could then be performed, fully controlled by the editing system, allowing the editor to adjust the start point and duration of the change from the edit keyboard. The alternative would be some kind of dynamic register recall, using a GPI switch action, for example, and trimming both the edit controller and the mixer.

Audio level changes while preparing voice-overs and so on can be performed in a similar way. The additional benefit is that, because the settings are made on two independent Logical Machines, match-frame levels are never lost.

Of course, the individual benefits of the logical, virtual, and physical machine hierarchy can be combined in a variety of ways. Many of the difficulties encountered with conventional audio control concepts can be overcome by configuration and manipulation of the logical and virtual machine parameters.

Multibus monitoring

Fast, flexible monitoring is of prime importance when editing audio with video, primarily because a large percentage of time in the suite is spent locating material and previewing edits. A simple but effective approach is to group the tracks to be monitored by physical machine. Each group can then selected with a single button that has programmable track-to-bus configuration. If logical machine assignment is available at the mixer, and the virtual machine concept is added, we can now monitor groups of tracks by selecting a logical machine, each of which has already been assigned to a virtual machine as part of the normal mixer operation. (Because of the greater numbers involved, track-to-bus configuration is a touch more complex, although the principle remains the same.)

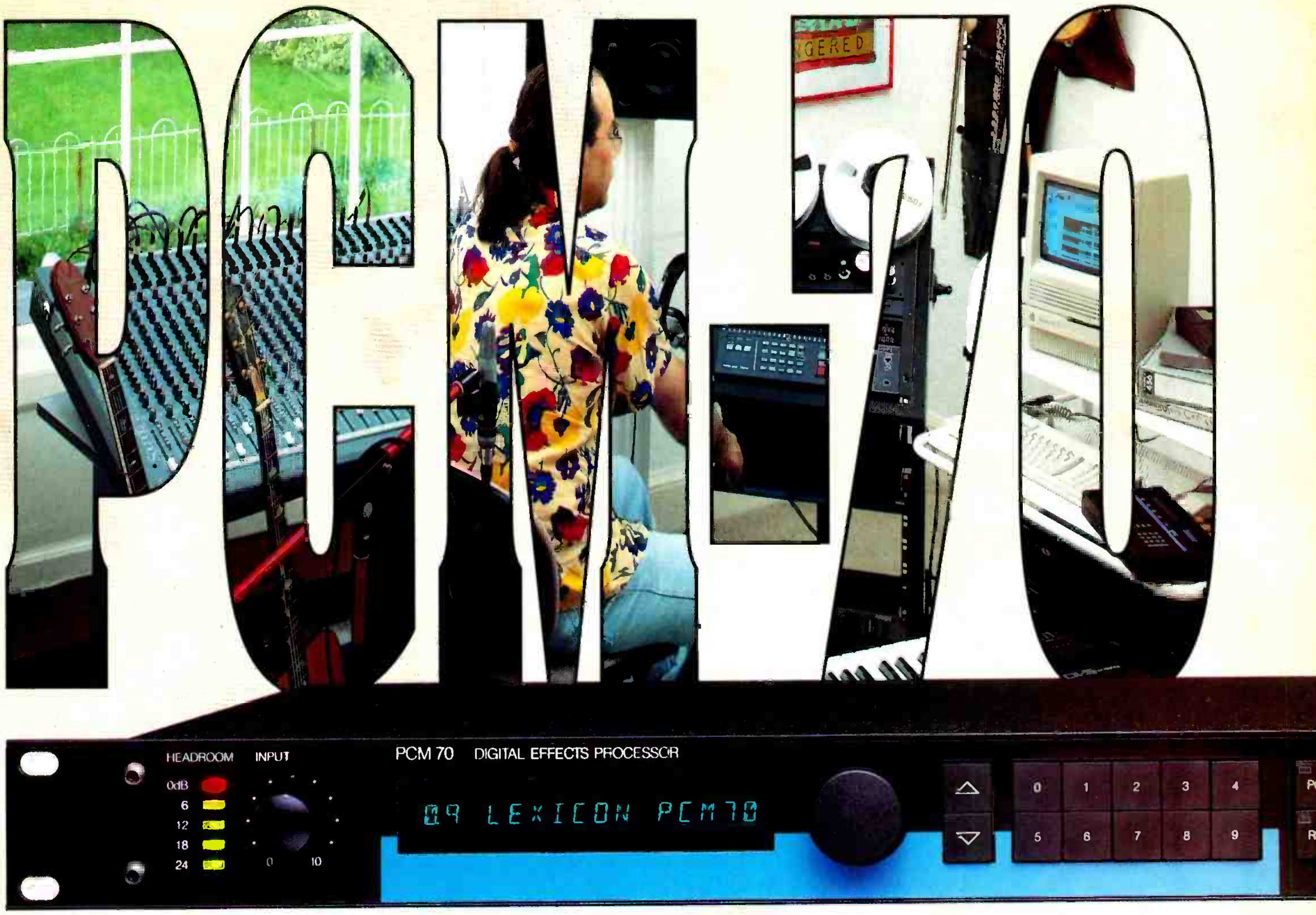
The only new requirement is to condense the number of buses to the number of monitoring outputs, and add some means of selecting the combination of buses heard from each loudspeaker. This, although simply stated, is not a simple problem. Techniques for editing multichannel audio with video have been fairly well defined, but the use of the various tracks has not. Until a limited number of commonly accepted practices further reduce the large number of possible applications, ease of operation will have to be traded against maximum flexibility.

Conclusion

Video postproduction suites using the new generation of VTRs need to accommodate large numbers of analogue and digital audio signals. An audio mixer designed for use in such an environment must be small, flexible, intuitive in operation, and fully controllable from the editing system. It must also provide all the benefits of digital recording, where applicable, while allowing analogue signals to be freely intermixed.

A hardware architecture based on a high-speed DSPs, and an operational philosophy using virtual, physical and logical machines, has been shown to fulfil these exacting requirements. The concepts described here have been developed in a fully operational audio mixing system that is now in routine use in video editing suites throughout the world. ■

MICHAEL D. PATTEN is Vice President of Engineering and co-founder of Graham-Patten Systems, based in Grass Valley, California. GPS manufactures the D/ESAM series of digital edit suite audio mixers and accessories.



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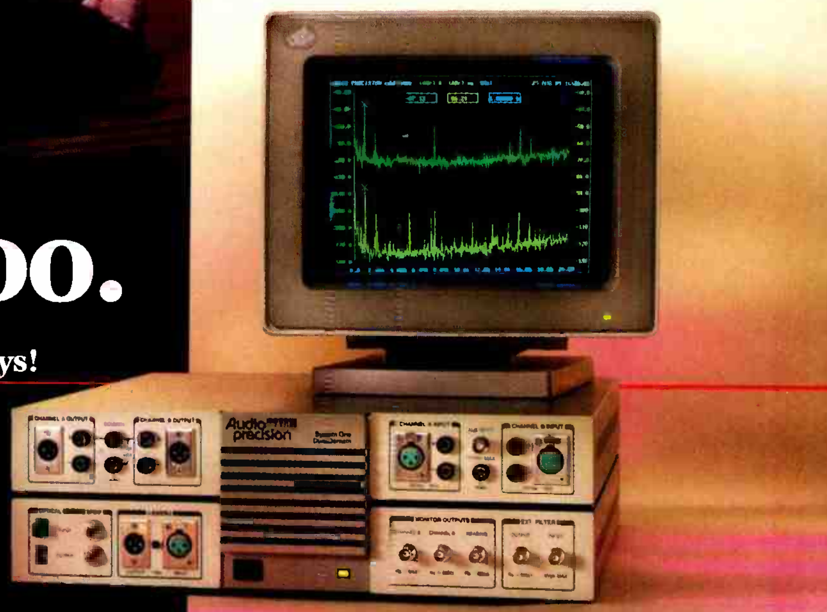

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Dithering about

Dear sir, Sony's David Smith gets his facts precisely backwards in the sidebar to his article '20-Bit: Fact or Fantasy?', (December 1992). Because of all the hoopla surrounding the introduction of Sony's Super Bit Mapping (SBM) proprietary wordlength reduction procedure, I feel compelled to elaborate upon Daniel Weiss' comments in the same issue (page 66), to clarify the relationship of SBM to psychoacoustically-based dither noise shaping, such as used for the past three years in Harmonia Mundi Acustica's redithering module for their *BW102* mastering system.

The facts are as follows: the Harmonia Mundi Acustica module uses a 50th-order shaper to shape the noise floor to mimic the ear's low-level noise sensitivity, achieving an 18dB reduction in the noise floor around 4kHz (the ear's most sensitive area) after full triangular probability density function dither has been applied. This dithered psychoacoustically-based noise shaper achieves a perceived 3-bit reduction in the noise floor, making it possible to produce 16-bit CDs with 19-bit apparent signal-to-noise ratio. Moreover, the full dither ensures that there can to be no low-level requantisation artifacts at all. This noise shaper is based upon published work.[1,2,3]

The SBM procedure as implemented is an undithered noise shaper whose noise floor does not really mimic the ear's response, and achieves only a 9dB noise floor reduction around 4kHz.[4] Being undithered, low-level requantisation artifacts will not be banished. It will achieve less than a 2-bit apparent noise-floor reduction when requantising to the 16-bit CD format.

All noise shapers can lower the noise power density in one portion of the baseband only raising it elsewhere in the baseband. Both the above schemes lower the noise power below 15kHz, where it is most audible, at the expense of raising it above 15kHz where audibility is less. The Harmonia Mundi noise shaper boosts the noise power density by 26dB at high frequencies, while the SBM noise shaper, which provides less reduction around 4kHz, boosts the high-frequency noise power density by only 19dB. These two noise power spectral densities are shown in the accompanying figure.

Stanley P. Lipshitz, Audio Research Group, University of Waterloo, Ontario, Canada

References:

1 M.A. Gerzon and P.G. craven, *Optimal Noise Shaping and Dither of Digital Signals*, Paper presented at 87th AES Convention, New York, 1989 (preprint 2822).

2 S.P. Lipshitz, J. Vanderkooy and R.A. Wannamaker, *Minimally Audible Noise Shaping*, J. Audio Eng. Soc., vol. 39 (1991 Nov.).

3 R.A. Wannamaker, *Psychoacoustically Optimal Noise Shaping*, J. Audio Eng. Soc., vol 4 (1992 July/Aug.).

4 M. Akune, R.M. Heddle, and K. Akagiti, *Super Bit Mapping: Psychoacoustically Optimised Digital Recording*, Paper presented at 93rd AES Convention, San Francisco,1992 (preprint 3371).

David Smith replies

Dear sir, thank you for your comments concerning the noise shaping section of the sidebar that accompanied our technical presentation of the recording of *Don Carlo*. Your explanation of the differences between the algorithms implemented in the Harmonia Mundi module and Super Bit Mapping with respect to noise floor shapes and the application of dither were not only clear but very much appreciated.

In writing the sidebar I went to extremes to avoid a factual presentation opting instead for a short string of generalisations. At no point in the discussion did the words 'Harmonia Mundi', 'Super Bit Mapping', 'dither', or 'shape of the noise floor' appear. You therefore accuse me of confusing facts that I did not present. I would, however, like to apologise for not stating that your 50th order dithered noise shaper's noise floor is psychoacoustically weighted.

While we are on the subject, Sony's Super Bit Mapping is not patented as I stated, but the phrase and the logo are registered trademarks of Sony. I apologise a second time. To conclude, we consider it an honour to have been published in the pinnacle of professional audio magazines, *Studio Sound*, and I subjectively feel that your point could have been made more diplomatically.

Since you have shaken the bottle and removed your thumb from the top, I would like to raise three points relating to this most contentious of subjects.

There are at present no less than five different redithering devices commercially available with all of them here at Sony Classical under study and evaluation. Of course, all of our CD releases are redithered to 16-bit precision using Super Bit Mapping, however this does not preclude our listening to and studying other algorithms.

Your use of the word 'hoopla' surrounding the introduction of Super Bit Mapping raises two related issues. I would note that the literature explaining this technique has existed since 1989, that Sony Classical's use of second order noise shaping predates the publication of the seminal paper by Gerzon & Craven, and that both your module and Sony's device have existed in one form or another since 1990 even though little attention was paid to promoting their use. It was only in 1992 that Sony began to promote the application of the technique, this promotion being responsible for the entry into the market of three additional pieces of hardware that serve a similar purpose. I would secondarily note that Sony's promotional efforts were directed at the consumer end of the market and not the professional end. 20-bit recording and noise shaped redithering are logical companions in an effort to refine the sonic characteristics of the CD. The fact that some of the professional community mistook this publicity as being directed at it combined with a rush to market by three additional manufacturers of noise shaping hardware, resulted in a case of professional hypochondriacal 'hoopla'. One would think that a campaign that makes note of a little-known signal processing application might be welcome but alas, this is not the case.

I feel compelled to raise my last point in the form of conjecture that is based on our listening so far. Comparing these devices with a single DAC and 20-bit recordings done with two or three microphones has lead us to believe that not only do they sound different, but that the algorithms themselves vary in efficacy with respect to program content and recording quality. Another point concerns the boosting of the power density of the noise of high frequencies in as much as some program material renders this boost clearly audible. The delicate question of the trade off between the apparent improvement in signal to noise ratio and this increase in high frequency noise power is one I am neither empowered nor qualified to address, especially in this forum.

Thank you again for your comments.

David Smith, Director of Recording Operations, Sony Classical, New York, USA

Conversion factor

Dear sir, I believe that the article in the January issue of *Studio Sound* on the Lexicon *20/20* A–D converter may confuse your readers. In this article, and in associated advertising, this product is promoted to be something it is clearly not —a 20-bit converter. Lexicon equate their product's dynamic range specifications to an effective number of digital bits. What slides by is the fact that this particular specification was traded at the expense of other parameters, such as resolution and constant noise floor.

At normal operating levels, Lexicon's 20-bit converter is putting out only 18 meaningful bits (just 17 bits if you use their effective bits measure). According to the designer, in the 20-bit mode, their product makes use of two A–D converters on each channel. The conversion task is shared between one converter for the high-level inputs and switches over to the other when input drops below a threshold.

An easy way to visualise this configuration is to imagine an audio input split and fed into two mixing console inputs. One module would be set up for line level and the other for microphone level. Because the same analogue input is driving both modules, line-level inputs will be handled well by the line module while the mic module will overload. When the input drops to lower levels, the microphone level module will stop clipping and function. If we were to add a piece of gear to cleanly select between the two module outputs, we could choose the mic output when it was not clipping and the line output when the level was higher. Because we are effectively switching gain, the noise floor appears to be very low when the input audio drops below the changeover threshold. The penalty is that the noise floor is now dependent on input signal level and will move up and down accordingly. The quality of the seam when the outputs switch can also grossly affect performance.

If we were to add an A–D converter to the output of each channel and digitally select the appropriate converter, we have a system similar to Lexicon's approach. The lower level converter is normally overdriven, with normal level inputs ►

making its digital output unusable. The digital output is now only as good as the convertor handling the higher levels. Lexicon's choice of A–D converter is a good device made by Analog Devices. According to the manufacturer, this delivers dynamic range of not much above 100dB and Total Harmonic Distortion plus Noise (THD+N) in the low 90s. Analog Devices sell this as an 18-bit component. For normal level inputs, the *20/20*'s dynamic range can only be around 100dB with a maximum of 18-bit resolution, which is a far cry from the 120dB of a 20-bit converter.

Making an inaudible digital seam between the high-level and low-level converters is an enormous challenge due to gain errors, DC offset, noise modulations and so on. The noise floor of the *20/20* is signal dependent—when the input level drops, the noise floor will, at one point, plunge by 12dB.

A point is also made in the article regarding digital dynamics processing. Analogue dynamics processing before conversion preserves the precious bits lost in digital dynamics processing. Because the *20/20* has only 17 bits (ENOB) for typical input levels, throwing away just one of those bits for headroom or dynamics puts it back at the 16-bit level. It appears that the author is referring to Apogee's popular Soft Limit/Soft Saturate analogue dynamics processing functions and the *20/20*'s emulation of these functions. It is very difficult to digitally emulate the complex myriad of parameters involved in analogue processing with any precision. It is the same reason digital emulation of transformers and vacuum tubes has not replaced the originals.

Not only does the 'bit budget' get eaten away by digital dynamics processing but the various dither methods available on the *20/20* chew into the remaining bits (if you measure using the ENOB method), putting competitive products on a more level playing field than Lexicon woud have its potential users believe.

Bruce Jackson, Director of Engineering, Apogee Electronics Corp, Santa Monica, California, USA.

Jim MacArthur replies

Dear sir, Mr Jackson's comments point out the necessity for more agreement within the high-end audio community about specifications and their value. However, I would like to clarify some of the issues that he raises.

The whole issue of effective bits versus resolution is confusing and, frankly, a poor indicator of a converter's performance. It is for this reason that I maintain that the Dynamic Range and THD+N measurements are far more accurate and useful specs. As was mentioned in my article, the typical dynamic range of the *20/20* AD is 112dB, which translates to an effective 18.3 bits—not 20 and not 17 as Mr Jackson states. However, to answer Mr Jackson's charge that the *20/20* AD is not a 20-bit converter, if you force the *20/20 AD*'s LSB to 0, the dynamic range drops by roughly 1dB, indicating that there is, in fact, audio in the 20th bit. Thus, the *20/20* AD is a 20-bit converter. The *20/20* AD indeed uses two converters operating at two different levels to achieve its high dynamic range. Mr Jackson's analogy of two mixing consoles is accurate, provided you add the ability to seamlessly crossfade between the two signals, rather than simply selecting the most appropriate one. The converters must also be able to remove gain and offset errors to improve the quality of the splice. Lexicon's DSP, which has been highly regarded for its accuracy for well over a decade, has been significantly refined in the *20/20* AD to handle these tasks. In the *20/20* AD we can keep these artifacts (and all other forms of THD+N) below 0.004% for all input levels from full scale to -24dBBFS.

'Normal Input Levels' fall below -12dBFS quite often. It is during these quite moments that the *20/20* AD really shines. The question of whether the improved low-level performance is worth the vestigial noise pumping during the loud moments depends, to my mind, on the programme material being recorded, and on the tastes of the engineer.

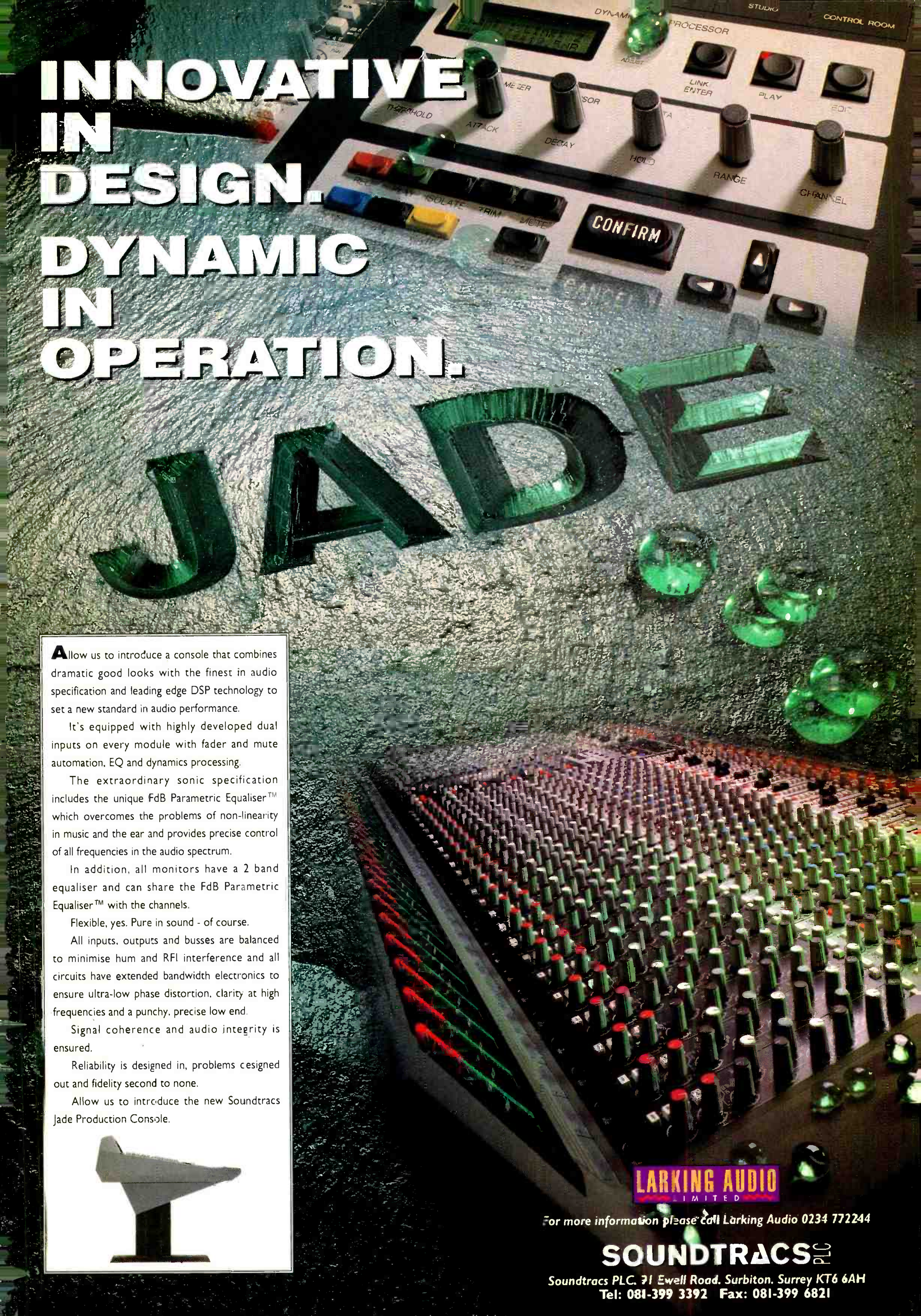
I agree that if one wants to add a soft knee to a converter's transfer function, doing it in the analogue domain (as Apogee did with the *AD-500*) makes far more sense than waiting until after the conversion. However, the *20/20* AD's compressor includes the ability to look ahead and to recover slowly from transients, two features which are very difficult to implement in the analogue domain. The *20/20* AD and the *AD-500* respond to overloads in radically different ways, and once again I would leave it to the engineer to choose the sound that is appropriate to the programme material.

Finally, I would add that the whole point of dither is to 'chew into the remaining bits'. Naturally, dither can be disabled when recording from the *20/20* AD to 20-bit media.

Lexicon invites Mr Jackson, a renowned mixer in his own right, to take a test drive of the *20/20*. In fact, we would suggest that anyone interest in serious digital recording should listen carefully to the *20/20* (as well as other A–D converters). The bottom line is audio quality, and we are confident that the *20/20* is simply the best sounding converter on the market today.

James MacArthur, Design Engineer, Lexicon Inc, Massachusetts, USA.

Letters should be addressed to: The Editor, Studio Sound Magazine, Spotlight Publications, Ludgate House, 245 Blackfriars Road, London SE1 9UR



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Back in action—Malcolm Toft and his new *Series 980* console

TOFT RETURNS

Malcolm Toft is back. The man who founded Trident Audio in the early 1970s and ran the company through to the late 1980s, is once again designing consoles. The new company, Malcolm Toft Associates (MTA) includes two ex-Trident personnel, Don Whitney and Warren McCulloch, plus freelancer John Oram who designed the electronics for many of Trident's consoles. MTA's first product is the *Series 980*, an affordable, analogue, multitrack recording console which bears the unmistakable hallmarks of Toft design.

Malcolm Toft began his career as a recording engineer at CBS Studios, later joining Trident Studios when it opened in 1968. There he worked with many legendary artists including David Bowie, James Taylor, Elton John, T-Rex and The Beatles (remixing 'Hey Jude'—one of the few singles not recorded at Abbey Road). He was soon promoted to studio manager, but feeling dissatisfied with an office job, Toft channelled his creativity into a new area: console design.

'The whole thing began because we needed a new console to fit into a reasonably confined space. We'd been to see all the console manufacturers but we couldn't seem to get a console with the kind of features we wanted. One of the things we'd ask for was EQ on the monitors—this was an anathema to manufacturers at the time who believed monitors should remain strictly flat. We came away from meetings feeling nobody really understood what we wanted, and certainly no one had any comprehension of why we wanted it—these people were technicians not engineers.

'As an engineer, I'd always been interested in what went on behind the front panel; my philosophy being a bit like a racing driver's: you get the best performance out of a car, if you understand something about the way it works. We eventually decided that the only way we would get what we wanted was to build it ourselves. So I agreed to take a year out from running the studio, and with the help of a very capable maintenance department, began putting a desk together.'

Trident's reputation as a studio was such that a proprietary console was a very exciting prospect, and only a short time after work started on the project Toft began finding interested parties.

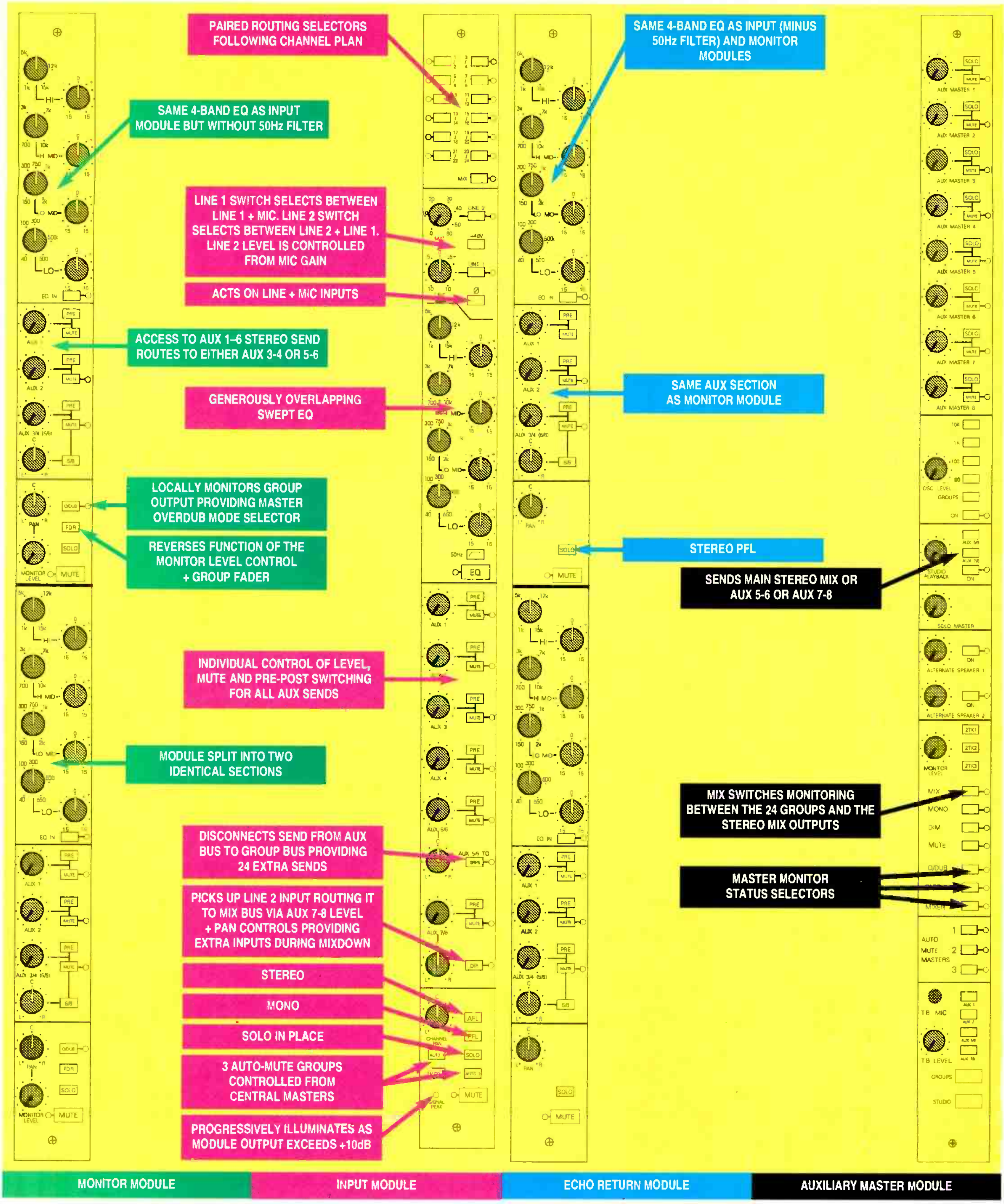
'News of what we were doing spread fast, and people like Gus Dudgeon and Tony Visconti were really keen to see the drawings. This led onto one of Roy Thomas Baker's friends, Dave Grinstead, asking if we could build a console for Chipping Norton Studios which he was setting up with the Vernon brothers.

'It was extraordinary, before we knew it we were getting orders for consoles. I went back to Trident's management and said, look not only can we build our own desks but it appears we can sell them too! So Trident Audio was set up, but don't think anyone at the time envisaged that it would grow the way it did.'

These were pioneering days, and consoles were built, perhaps for the first time, from a recording engineer's perspective. Toft and his team displayed a healthy disregard for convention, adopting the principle that if it sounds good or makes the job easier, do it.

'When we designed our first consoles we knew nothing about phase relationships and so on. If you actually measure them, they have horrendous phase shift and very bad overshoot and ringing on the mic transformers, but because we'd listened to the way the thing sounded it didn't matter. You put that old *A Range* console on a brass instrument or a vocal and it comes alive, there's tremendous warmth and richness. That sound was created through painstaking listening tests by the engineers at Trident; people like Ken Scott, Roy Thomas Baker, Barry Sheffield and myself would get together and literally change components until we got sounds we were happy with. It was something you couldn't possibly quantify technically, and quite honestly if we'd been able to measure things accurately we probably would never have got these kind of results. I think this is a problem with some of today's designs—too ▶

Malcolm Toft returns to console design with a new company and a new desk. Patrick Stapley talks to the man about both



much attention is paid to clinical precision and transparency. This empirical approach to design is something I still adhere to very strongly and have applied to the *980* console.

'I'm also a believer in continuity of design and performance. One of the things that worried us at Trident when we went from discrete circuitry to ICs was that the overall sound would be different, and we put a lot of effort into ensuring there was very little audible change. I think it comes down to a cultural thing within the company—your products tend to sound the same because of an underlying approach to the way signal flows are designed and put together. For example I've always maintained that every electronic block should go in and out at unity gain and have a headroom of +24dB. All the products I've been responsible for have this kind of continuity, so obviously there will be strong similarities between the new MTA consoles and the original Trident desks'

There are also definite similarities between the formative days of Trident Audio and the setting up of MTA. In just the same way that word circulated about the development of the first Trident console, so whispers spread about Toft's new venture. Before either a product existed or a company had been formed, orders began appearing based purely upon Malcolm Toft's reputation.

'When I started thinking about getting back into consoles and the kind of product to design, I did some computer drawings of layouts and ▶

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facilities mainly for my own interest. I got a call from David Michaels, who had worked as American distributor for Trident in the early days, and he asked me to send over the drawings. I faxed the stuff to him, and after that he was on the phone solidly asking questions about operation, pricing and so on. Ten days after receiving those rough drawings, he rang me to say that he had a firm order with a deposit from a studio in Florida. I said, that's impossible I haven't even set the company up yet! A month later, a second order came through and so it's gone on—we've not got more than 15 dealers working for us in the States.

'The response has been absolutely staggering and it's extremely reassuring to have that kind of faith shown in you after being our of the business for three years. The only sad thing is that the reaction hasn't been the same in the UK—the Americans have always been our best customers, and I have to say that if we sell any consoles in the UK at all it will be a bonus.'

MTA will be producing a family of consoles, but Toft was keen that the first desk should be traditional, 'work-horse' analogue design offering good value for money. Having researched the market and identified a hole around the £25K price range, he decided to build a console that would be, in his view, a natural successor to the discontinued Trident *Series 80*.

'I wanted to play safe, and produce a console that I felt would be easily marketable—I make no apologies for the similarities with the *Series 80*—it was a very successful desk and hundreds were sold. We've taken the basics of that original design and added the extra facilities that people were asking for such as extra aux sends, 4-band EQ on the monitor, extra mute groups and so on.

'The console is, of course, a split monitor design which I've always felt to be more flexible and simpler to operate than in-line designs. I'm a great believe in simplicity: if a console has been well thought-out it should be possible for an engineer to sit down and find his way around the controls without having to ask questions. Using another motoring analogy, its like me handing you my car keys—although you've never driven my car before, it wouldn't be a problem because the car's been well designed and facilities intuitively laid out.'

At present Toft and his small team are working from his Hampshire home, but manufacturing is being handled by a Kent-based company.

'MTA is effectively a design and marketing company which subcontracts its manufacturing requirements. There are many companies set up now that provide full manufacturing facilities, with all the necessary infrastructure for full purchasing, metal working, assembly and so on. We're using a company called Electrocue, which

> 'I wanted to play safe, and produce a console that I felt would be easily marketable—I make no apologies for the similarities with the *Series 80...*'

also manufacture lighting control consoles, and the results have been absolutely excellent. It means I don't have the worries of running a factory with all the overheads and staffing requirements that entails, and at this stage that's pretty essential. However, if the business does take off in a big way, we will probably be forced to open our own premises just to deal with shear volume. We'll just have to wait and see what the future holds.

The *980* console

The standard *980* console is made up from the following modules: 32 input modules, 12 dual-section monitor modules, 3 dual-section echo return modules, and a single auxiliary master module. Thus the standard console can offer a total of 62 channels during mixdown all with 4-band EQ and auxiliary sends. The 8ft x 3½ft frame also contains a horizontally laid out, printed circuit mounted patchbay (Re-An bantam jacks), and a meter bridge housing mechanical VU meters (bargraphs and phase meter can be supplied as an option). Faders have been kept separate from modules for retrofit of automation systems—the first two consoles to be shipped were fitted with *Optifile* and *Flying Faders*.

Mechanical construction is based on a chassis of welded steel boxes, and the aluminium alloy modules are fitted with gold plated connectors which plug directly into a frame-mounted motherboard. All I-Os are via EDAC connectors.

The layout and operation of the desk is extremely straightforward, helped by the continuity between modules—inputs, monitors and echo returns all have the same 4-band EQ, and monitors and echo returns share the same aux send arrangement. The layout of the dual monitors has been clarified by colouring the control knobs of the upper section black and the lower grey—thus distinguishing clearly between 1–12 and 13–24. Throughout the console extensive use is made of LEDs, which again helps to clarify and simplify operation.

The input module can select one of three sources: Mic, Line 1 and Line 2. Both Mic and Line 1 inputs have individual gain controls, but if Line 2 is selected the mic gain will pad down to provide a ±20dB line level control. Alternatively the second line input can be used during mixdown by pressing the DIRECT button next to Aux 7-8. This causes Aux 7-8 to source the Line 2 signal disconnecting its send from the aux bus to the stereo mix bus via the aux level and pan controls. The facility enables the standard console to cater for up to 32 extra inputs during a mix.

The console has eight aux (4 mono and 2 stereo). Individual controls with pre-post and mute switching are provided for all aux sends on the input module. The monitors and echo returns can each send to six aux buses through individual sends to Aux 1 and 2, plus a switchable stereo send to either Aux 3 and 4, or 5 and 6.

An additional auxiliary feature incorporated on the input modules, is the ability to route Aux 5 and 6 to the group buses to provide an extra 24 sends during remix.

Three SOLO buttons are included in the input module providing AFL, PFL and Solo In Place. Solo operation can be either latching or nonlatching depending on customer preference. An AFL-type solo is used for monitors and echo returns, and overall level control for both PFL and AFL circuits can be adjusted from the auxiliary master module. Also controlled from here are the three Auto-Mute groups which can be set up from each input module.

Monitor levels are controlled from pots rather than faders, but a fader reverse facility is included that swaps the function of these pots with the long throw group faders directly below the modules.

Local OVERDUB buttons are provided which operate in conjunction with master monitoring status switching: MIXER, TAPE and OVERDUB. MIXER and TAPE globally switch the monitor paths to group or tape while disabling local override. OVERDUB globally switches to tape monitoring but allows local selection of group outputs.

The master module contains master level, mute and solo control for each aux bus; a 4-frequency oscillator; studio playback sourced either from the stereo bus or one of the two stereo auxs; two alternative loudspeaker selectors with independent level control; three external machine monitor source selectors; MONITOR DIM, MUTE and MONO buttons; and an integral electret talkback mic that can be routed to individual auxs, the 24 group-and-stereo output buses, or studio speakers.

The *980* console is predominantly being aimed at the multitrack music recording sector, but postproduction is another area where it will have applications, and MTA are currently looking at various options for this market. An option that will be available shortly is a stereo input module.

Although the standard console is 32-input, there are no real physical constraints on size and the largest *980* built so far is a 48-input. To date, ten consoles have been sold—all to American customers. ■

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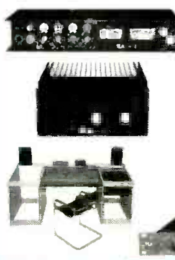

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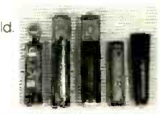

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PLAY IT AGAIN SAM

Japan's Sam Corporation has been in the news of late, ever since it bought out Steve Flood's Master Rock Studios in London—home of the only other Rupert Neve Focusrite *Forte* outside of Electric Lady Studios in New York. Master Rock has thrived from the association, with the Japanese connection providing a steady stream of Japanese work for the Kilburn-based studio and Sam Corporation has gained considerable prestige from the link.

Sam Corporation Tokyo is based in a five-storey building in the Shinjuku-ku region of the city and houses a Neve *VR60-Flying Fader*-equipped recording studio, a radio studio and production division, a keyboard studio and offices for the Corporation, the studio and the performers signed to Sam Artists. Sam Corporation also has premises in Kobe where company President Masao Suzuki first started making programmes for radio stations and this work is continued in Tokyo with programmes prepared for Kobe Kiss FM, Yokohama FM and Tokyo FM as well as Japanese language programmes for foreign broadcasters.

The room used for the production of these radio programmes is based around a Soundcraft *200B* desk with Technics CD players and turntables, Sony *DTC 1000ES* DAT machines, Otari *BTR10* open reels, and Sony and Pioneer cassette decks. As anywhere in Japan, space is used efficiently with the equipment surrounding a central booth for voice-overs.

In line with Sam's self-sufficiency, the building also boasts a small MIDI-based commercial studio used almost exclusively ▶

In the wake of the Japanese Sam Corporation's acquisition of British's Master Rock, Zenon Schoepe visits the Corporation's Tokyo studio to talk kit, cash and culture

for the production of jingles and commercials for advertising companies and radio stations. Two booths feed a Soundtracs *MRX 32:8:16* and a Fostex *G16* in a $15m^2$ control room peppered with keyboards and modules throughout including Akai samplers, E-mu *Emax SE* and *E-III*, Roland *D550*, Korg *M1*, Oberheim *Matrix 1000* and Yamaha *DX7IIFD* running from Atari computer-based sequencers.

Property is extortionately expensive in Tokyo and Sam's real estate investment must be made to earn its keep. Consequently the building was as busy at 9pm as it would be expected to be during daylight hours—as the world's daylight shifts Sam Tokyo has offices in London and New York to liaise with.

However, working through the night is commonplace in recording studios throughout the world and Studio Sam was working late at the time of the visit. The studio had recently played host to the actor Yuzo Kayama and Hitoshi Ueki, Yume, Toru Kazama and Isako Washio —names that may not be familiar to Westerners but represent the up and coming hopefuls on the Japanese scene and is very much in line with the facility's position as a middle market recording venue.

Middle market means Sam goes out for around £170 per hour compared to the £400 per hour that can be paid in other places and the absolute bottom cut-off point for a Neve *VR60*-equipped room of around £100 per hour. For the money Studio Sam offers a $33m^2$ control room and $32m^2$ live area with a separate machine room housing the essential Sony *3348* plus an Otari *MTR100A* with Dolby *A* and *SR* on a commercial basis to all including Sam artists. Other machines include Studer *A820* with ¼ and ½-inch blocks and an *A812* with Gauss and JBL horned *SAM 1* custom monitors powered by Amcron amps in a room designed and constructed by JVC—yet another little-known string to this Japanese giant's bow it seems.

While claiming that Japanese engineers 'expect large consoles with lots of inputs to handle a Sony *3348* multitrack and the newest auxiliary equipment', Sam Corporation Producer Tsuneo Kawagiri comments that competition on price among studios is not unheard of but mostly at the lower end of the market.

'There is a little,' he adds, 'but, it is not nearly as bad as it has become in Britain, for example. This is because we don't have to, because there is enough work to go around. We can sell on service.

'I don't think you can expect to see loads of stunningly well-equipped studios opening in Japan in response to the market's health but, at the moment, nobody is going bankrupt,' says Kawagiri. 'But there will be growth and when the standard rises across the board, which it must eventually do, perhaps then we will start to see some of the smaller studios starting to suffer.'

Much of Sam Studios' work is handed down from the major record labels whose artist roster far exceeds the capacity of their in-house recording studios. However, Kawagiri adds that Sam is particularly keen on involving itself with young talent even though most Japanese artists share the prospect of only limited global penetration and will rely almost exclusively on the Japanese domestic market due most notably to the language barrier. These circumstances make the Japanese recording industry's health all the more surprising.

Bearing this in mind, what are the benefits from the Japanese point of view of a tie-up with a Western studio like Master Rock?

'Many Western engineers are more talented than the ones we have in Japan. We have a totally different background and culture and it is more difficult for our engineers to work with foreign clients,' says Kawagiri. 'The solution is for foreign clients and engineers to come here and use our facilities or work with our artists or for our artists to travel abroad.'

Kawagiri also believes that Japanese engineers will benefit from the interchange of ideas between Western and Eastern engineers and that was what the buying of Master Rock was all about—putting the Sam Corporation in a very strong position among Japanese studios.

'Big record companies are divided across many territories and could draw on all the influences open to them, but they don't. Sam Corporation, while relatively small, is unique in this respect,' Kawagiri says.

An enduring memory of Japan is the degree of loyalty of studios to each other—it is difficult to illicit a bad word from a studio about is competitors, something that is far easier to provoke in Europe, for example. Given this united front I asked Kawagiri if the Sam tie-up with a Western studio was a popular decision in Japan.

'For something like this to happen is very rare,' he replies and when pressed about whether the act was deemed a little precocious, merely adds 'They are surprised'.

However, Kawagiri believes that the implications of such an East-West link have not been lost on other Japanese studios which, in his opinion, are now considerably more receptive to the idea. Any takers? ■

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Zenon Schoepe is a freelance pro audio journalist with a long history of international magazine editorial who contributes regularly to leading international titles. His experience of audio is wide, and he continues his activities as songwriter, musician and sound engineer. Zenon also runs his own well-equipped private recording facility.

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DEADLY EMBRACE

Ins and outs of synchronising digital audio in a video environment. Chris Meyer dances to the music of time

'Synchronisation' used to mean getting two or more multitrack tape decks to agree to be at the same place at the same time. Maybe they took a bit longer than you would like to get there, and if you forgot the golden rule and spread a multi-mic recording across decks you got a bit of phasing, but for the most part it worked.

A decade ago MIDI entered our lives, and with it a couple more ways to synchronise these new virtual recorders called drum machines and sequencers to our decks. But in time we mastered that as well.

Then came the double-whammy: the increasing call to synchronise multitrack audio to video, and the advent of digital audio recording. Sometimes the fool things refused to even go into Play; other times, they would play alright but slowly drift out of time the further they get along the tape... It's not voodoo, it's not ill temperament, it's just that these new friends have rules—very strict rules—that we have to know and follow. I will try to explain those rules and to follow (or creatively break) them in practice.

Clockwork audio

To understand why it is difficult to synchronise digital audio, first we should understand the nature of analogue versus digital signals.

Normally, sound is the fluctuation of either air or electrons and magnetic fields (when we record and play back) at a rate somewhere between 20 and 20,000 cycles per second. When we record these vibrations, we use a mic to create electrical signals that fluctuate in a similar pattern, and these patterns are recorded on a piece of rusty cellophane. Reverse the process and we have good old vibrating air again. Digitising audio changes this process considerably. Instead of keeping sound as a continuously varying signal, the audio's level is sampled periodically with the resulting number being recorded along instead. The sample rate is typically 48,000 times a second for DATs, 44,100 times a second for CDs, and occasionally 32,000 times a second for broadcast applications. When you want to play it back, you have to recover these numbers and convert them to an electrical signal at exactly the same rate.

And therein lies the rub—you have to worry about that sample rate throughout the entire process of working with digital audio. If you use different sample rates, at best it will be as if you nudged the varispeed knob (the result being off-pitch and off-speed playback). Worse still, if you intend to pass this digital signal onto another device or mix it with other digital signals: everyone must agree on exactly what that rate is and exactly line up each sample in one signal with one corresponding sample in the other signal, or else the entire process falls apart—sometimes rather noisily.

The enforcer

When you connect one digital audio device to one other, usually none of the above problems arise. The common AES-EBU digital audio interface transfers the signal in such a way that the receiving device can figure out the sample rate and each sample arrives at the same time it is decoding the numbers themselves. As long as it uses this derived sample clock, there are no obvious synchronisation problems.

In theory, the sample rate clock is always rock-steady, and digital audio devices usually have good clock sources built in. In reality, a derived sample clock may actually have a bit of jitter (digital wow and flutter), which will cause errors in the way the audio is reconstructed that are similar to those that would occur if you had not measured it accurately in the first place. (This is why some digital audio devices mysteriously sound better when running on internal clock as opposed to being slaved to another device.)

Again, a far worse situation is where you have more than one signal flying about. For example, if you are trying to record two digital audio devices—such as a pair of DAT players, or the outputs of two digital signal processors—to a digital audio multitrack recorder, you have to decide which one the multitrack should slave to. If the other input is running slightly off-rate, the result is clicks and pops, or in extreme cases, even muting, where individual samples do not line up.

If you have two devices, such as a digital signal processor and a digital mixer, connected in a loop, you run into the interesting scenario of the send running from mixer to the processor, making it a sample rate slave to the mixer, and then running the output back to the return of the mixer making it a slave to the processor. The result is a 'deadly embrace' where each one depends on the other to move forward, and quite often the result is every one shutting down.

There are three solutions. The first is just to use analogue audio to pass the signals around (negating the entire purpose of this exercise). The second is to use a sample rate converter to reclock the errant signal. Sample rate converters or analogue connections are required when running a system with mixed sample rates, such as mastering from a 48kHz DAT recording to a 44.1kHz CD.

The third (and preferred) solution is to set up a master sample rate clock, and have all other devices slave to that clock. With one master, there can be no argument about who is supposed to run the show. In an audio-only studio, this means looking for a separate word clock (typically on a BNC connector) or DARS (Digital Audio Reference Signal, which is an AES-EBU with no audio—just clock) input to ensure you have a way to slave each device.

The second clock

In many ways video is just like digital audio; you need to follow a master clock or nothing will work. The problem is, video runs at a different rate than audio, and in matching together these two rates is where the *real* aggravation begins.

Sight, like sound, is normally a continuous phenomena. Like digital audio, film and video also 'sample' images discretely and store them as individual frames. Play them back in the same order at the same speed, and the original image is more or less reconstructed in a fashion that fools our eyes into thinking they saw the original. These 'sample rates' are 24 frames per second (fps) for most film, 25fps for PAL and SECAM video, 29.97fps for NTSC colour video, 30fps for NTSC ▶

> It's not voodoo, it's not ill temperament, it's just that these new friends have rules—very strict rules—that we have to know and follow

MIDI SYNC: CLOCKS AND MTC

Tape decks and hard disk recorders are not the only devices we have to worry about synchronising these days—there are drum machines and MIDI sequencers as well. Not only do these have their own kinds of sync, they also have more than one way to synchronise.

In the very earliest days, drum machines were slaved to tape by recording their equivalent of a click track. This click track ran at the smallest timing increment the drum box could deal with (typically a 96th or 384th note). There were two main problems with this system: you had to determine the tempo map and any changes before you recorded it, since each click represented a hard increment of time; also, you always had to start the tape from the top to make sure you were in sync, since each click had no way of telling you where it occurred in the song or how far it was from the downbeat of a given measure.

MIDI Clocks in their earliest form were almost an exact copy of this: one was sent per 96th note of the tempo map, with the main embellishment being the addition of Start, Stop, and Continue messages to let a slaved device know the master had indeed stopped or started. Several boxes subsequently appeared that could convert back and forth between ordinary drum machine clicks and MIDI clocks.

The secret weapon of MIDI clocking, however, is the Song Position Pointer (SSP). This is a location message that tells you precisely how many 16th notes (semiquavers) you are from the top of the song. Follow it with a Continue message, and subsequent clocks make a lot more sense—they are increments from the location related from the SPP. More or less in parallel with this came Smart FSK, which was a more advanced drum machine click that also encoded the number of the beat along with the beat itself. But it still locked you into a tempo once you had printed it, and required a translation between it and the more common studio synchronisation standard of SMPTE time code.

Then came MIDI Time Code (MTC); believe me—I wrote it. Several years ago I was interested in creating sound effects for picture, and wished there was some inexpensive way to feed SMPTE code into a drum machine or sampler so they could more directly slave to video. MIDI is not fast enough to send a message for every bit of a normal longitudinal SMPTE stream, but it is more than fast enough to carry a message per frame of video (which is what SMPTE actually counts off). Sound effects editors told me they felt they needed at least half-frame timing resolution to get the feel right; sound effects synchronised directly to film are often slipped by sprocket hole increments (which occur four per frame).

It was decided that MTC should send a new message every quarter-frame to deliver this resolution. I came up with a fairly clunky way to do this and Gerry Lester (formerly Chief Engineer of Adams-Smith who now works for TimeLine) came up with a more elegant method that encoded one digit of the absolute SMPTE time inside each message along with a number telling you which one of the eight digits this one was. There are also additional flags that tell the receiver what the counting method is: 24fps, 25fps, 30fps drop-frame, or 30fps non-drop. MTC does not say explicitly what the frame *rate* is; that is to be inferred from how fast you actually receive the messages. Unfortunately, many MTC devices (including one I designed several years ago) assume that a counting method of 30fps non-drop means a frame rate of 30, when it may be actually be sent at 29.97fps (the rate of NTSC colour video). This little detail has caused no end of trouble for MIDI musicians trying to work in audio-for-video situations.

MTC is often tagged as an inaccurate, semipro version of SMPTE, but this is not entirely fair. Since it is transmitted serially instead of read off of tape, it is like the more sophisticated Vertical Interval Time Code (VITC) flavour of SMPTE in that you can pause the tape or shuttle it fast and still be able to read an accurate time. It only sends a message every quarter-frame compared to 80 bits per second for longitudinal SMPTE, which initially makes subframe resolution appear limited, but finer increments can be interpolated between these messages (indeed, SMPTE readers often only look at whole frame edges and interpolate 100ths of frames between these themselves). Also, if sent properly, MTC actually has less timing jitter than called for in the SMPTE specification.

However, most MTC generators are not that diligent in their timing. Merge MTC with other MIDI messages, and a quarter-frame could get nudged by up to several milliseconds, having anything from an inaudible to drastic effect on the timing. MTC is also rarely resolved to follow video sync (another source of semipro grief). For these reasons, it is never used as a master with which to slave audio or video or recorders of any stripe. It is also often misused in situations where a drum-machine-like click track would actually be better, since different devices have different mathematical errors in converting from absolute SMPTE-like time to the relative time of tempos and beats. But as a way to get SMPTE-like timing into a MIDI-capable device, it remains exactly what the doctor ordered. ■

black-and-white video.

Video is also like digital audio in that all video signals must be precisely synchronised and running at exactly the same rate in order to mix or edit any two of them together. When video and digital audio are to be combined, then some relationship between the two must be established and adhered to, or else the two will lose their sync—and people lose their minds, if not their jobs or lives.

The most important thing to remember is that a precise number of digital audio samples must fit into the length of each video frame, and under most circumstances, this number must never change. You can not assume, say, the video deck is running at *exactly* at 48,000.0000 samples per second, and therefore everything will automatically work out—you must precisely slave the audio rate to the sample rate.

Video studios are already used to the idea of having one master clock that every piece of equipment must slave to. It is commonly called 'house sync' although it is also known as 'video sync' or 'black burst' (that latter being a video signal with no visual information; just timing —like DARS is to audio). If you want to introduce a digital audio device into such a studio, try to make sure it has one of these connections and can slave its sample rate clock to that video clock. This is the first recourse to making sure everyone stays on the same page.

Failing this, you might still have the option of using a SMPTE time code input. SMPTE actually came from the video world, and if on a video tape it is usually running at the same rate as the video signal itself. (There are times when some lower form of life will record SMPTE time code that has not been resolved—slaved to—the video rate onto a video tape. Unresolved time code is the worst kind of lie when you are trying to make studio work. If your digital audio recorder lacks a video sync input, make sure it resolves its sample rate to follow the SMPTE rate exactly, and you will still be sitting pretty. However, note that SMPTE tends to be a less stable signal than video sync, which can raise the possibility of reduced audio quality as mentioned above.

Not all digital audio recorders with SMPTE inputs can slave their sample rates without additional hardware, many will remember the SMPTE time they started recording, but will use their own internal sample rate clock to actually record the audio. Even more devious, some will record unresolved, but then play back resolved —and when those two sample rates do not match exactly, you get drift rather than airtight sync.

From this, we can derive two rules of synchronising digital audio to video: make sure all devices and time codes are slaved to the same master clock (preferably video sync), and make sure you play back under the exact same conditions under which you record.

Breaking the system

What these two rules do not take into account is what happens when someone changes operating conditions such as the video or audio sample rate somewhere along the line.

Those who work strictly in PAL or SECAM video have it relatively easy: The video rate is almost always the same. Follow the two golden rules, and things will work out most of the time. There is the problem of when film is dubbed to PAL or SECAM video by merely speeding the frame rate up from 24fps to 25fps; your slaved digital audio recorder may not be able to track the 4.2% increase in speed, and if allowed to run on its own clock will fall drastically out of sync with the dubbed video. In such a situation, transfer to an analogue deck capable of following the shift might be the only solution.

Far worse are the horrors of NTSC video and the two rates of 29.97 and 30fps. the simplest mistake to make is when a music studio uses a frame rate of 30fps either to create a project on ►

> ...note that SMPTE tends to be a less stable signal than video sync, which can raise the possibility of reduced audio quality...

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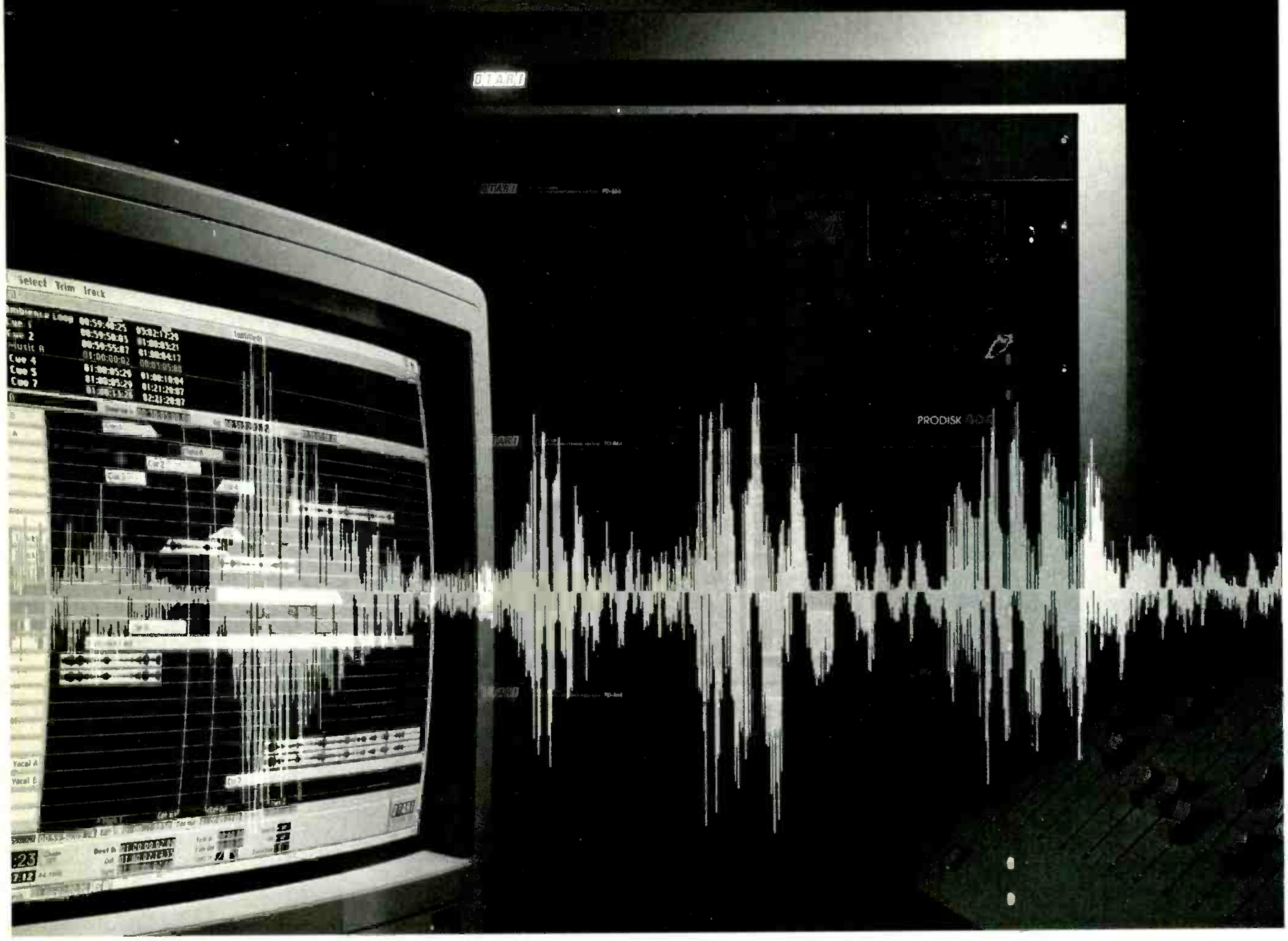

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SYNCING ANALOGUE TAPE DECKS

So how do good old-fashioned analogue decks sync up? And do any of the same issues apply when synchronising to video?

The equivalent of a sample rate clock in an analogue tape deck is a tachometer control pulse sent to the capstan motor. This pulse usually happens at a frequency of 9,600Hz (in countries with a 60Hz AC mains frequency) or 8,000Hz (in those with 50Hz mains). Changing the frequency of this pulse changes the speed of the tape deck. That is why Varispeed was often referred to as VSO (Variable Speed Oscillator): what actually changed was the capstan oscillator's tach pulse frequency.

A tape deck synchroniser takes over control of the capstan's frequency by remotely feeding it this pulse. It receives SMPTE time code from a track on the tape, and compares the speed of that code to a reference such as the master deck's time code. If it finds the slave's time code drifting a bit ahead, it will slow down the capstan frequency to compensate (and vice versa). This has the same end effect as changing a digital audio device's sample rate.

Whenever we mention changing the sample rate to follow some timing kink in the road when slaved to video, in essence the same thing must be done to the capstan tach pulse in the analogue deck as well. For example, if an analogue Nagra tape deck was resolved to a film camera's pilot tone on location, and then the film was slowed down by 0.1% to transfer it to NTSC video or sped up by 4.2% to transfer it to PAL or SECAM, the Nagra must also be slowed down or sped up by the same amount to keep everything in time.

This is the goal of synchronisation: keeping everyone at the same relative pace. The real difference when dealing with digital audio is that, not only may you drift out of sync, you might shut down altogether. ■

a digital audio recorder or to stripe a time code track on analogue tape: when later slaved to 29.97fps video, things get slowed down and fall out of sync. Remember the second rule: always record in the same circumstances under which you intend to play back.

More subtle is adding film to the equation. When film is transferred to NTSC video, its frame rate is actually slowed down to 23.976fps so that film frames can be better matched to video frames. If audio was recorded on location when the film was shot, the chances are it that was resolved to a 50Hz or 60Hz pilot tone coming from a camera shooting at 24fps. Say a DAT recorder sampling at 48kHz was resolved to this signal—if you took the same tape and resolved it to still play back at 48kHz when slaved to a video signal at 29.97fps, the audio will now actually be running too fast since the visuals had been slowed down by 0.1%.

The old solution was to also slow down the DAT player by 0.1%, so it was actually working at 47.952kHz when slaved. Of course, after we finally figured that out, someone changed the rules again: The new digital video formats such as D1 match a sample rate of 48kHz (not 47.952) to 29.97fps video rate, and they simply will not accept any deviation. You might be tempted to just transfer via analogue or a digital sample rate converter, but this will do nothing to change the speed of the audio itself—it still will be playing back slowly. Instead, on-location recordings should be made at a speeded-up sample rate of 48.048kHz when slaved to a camera's pilot tone in order to be properly pulled down to 48kHz when slaved to

on-location recordings should be made at a speeded-up sample rate of 48.048kHz

video later. (The same rules apply to 44.1kHz recordings as well.)

There's more: not only does NTSC video have different frame rates for colour and black-and-white video, it also has two different counting systems: drop-frame and non-drop-frame. If you simply number successive frames 0–29 within each second, you would drift out of time when compared to the clock on the wall. To make up for this, every minute (except for even exact 10s of minutes, such as 0, 10, 20 and so on) two frame numbers are skipped (dropped) to get time code back in line with the real-world clock.

Many audio systems assume that if you are using a drop-frame counting system, then your time code or video sync must be running at 29.97fps, and if you are not, then it was really running at 30fps. However, many people keep the non-drop system when they stripe SMPTE even though they are actually running at 29.97fps. If you tell a digital audio recorder you are using non-drop-frame code, it might assume the video is running at 30fps and resolve its sample rate with that assumption in mind; when you actually feed it video sync or SMPTE at 29.97fps, the sample rate is again slow.

After the first two rules have been observed, it is this last puzzle that causes most other audio-video synchronisation problems today. It can most often be solving by lying to the audio device and telling it that you are using drop-frame time code (even if you are not), which will then set it up to slave properly to 29.97fps.

Plan ahead: work backwards

When trying to synchronise digital audio to video, the best thing you can do is first decide exactly where you want to end up in the relation between the two (for example 48kHz with 29.97fps video). Work back, keeping the ratio between audio and video rates intact no matter what. Then record at the rate you end up with, always resolved to some master clock that is also locked to the video.

Since not every system today can take every possible combination into account, there might be times when you have to be creative about lying to your equipment—or even have do the transfer through analogue or a sample rate converter. But once you have this road mapped out, you *will* make it to your final destination intact. ■

CHRIS MEYER is Chief Engineer at Roland R&D in Los Angeles. He is also a member of the MIDI Manufacturers' Association and has written various aspects of the MIDI specification

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Martin Polon

What are advancing technology and changing priorities doing to our industry bodies? Has the Audio Engineering Society become the Audio Elite Society?

The question that reverberates about the US pro-audio industry at the moment is whether the AES has become the Audio Elite Society rather than the Audio Engineering Society?

For those asking the question, the evidence takes several forms. First and foremost is the attitude expressed by many that the AES has become too diverse an organisation and strayed from its goal of serving audio engineers. Within the Society, some members—including those in leadership roles—have felt that the steady erosion of the engineering base damages the Society's efforts to achieve its main purpose of disseminating of technical information.

The Society, as stated in every copy of the *Journal of the AES*, is organised 'for the purpose of: uniting persons performing professional services in the audio engineering field and its allied arts; collecting, collating, and disseminating scientific knowledge in the field of audio engineering and its allied arts; advancing such science in both theoretical and practical applications; and preparing, publishing and distributing literature and periodicals relative to the foregoing purposes and policies'. Throughout the 1960s the membership was greater than 80% technically trained, with the remaining 20% or less being more applications orientated. A survey taken during the first three months of 1993 indicated that those proportions have roughly reversed. For the technically trained among the 6,000 or so AES members in the US—and among the 75,000 professional audio practitioners in the US—less than 20% hold engineering-specific or acoustics-specific technical degrees.

A further validation of the elitism charge could be taken from the disparity between AES memberships and industry practitioners, AES having less than 9% of the total number of individuals employed in audio.

Curiously, the number of university-trained audio engineers and-or physicists in America who enter audio engineering has dwindled to a trickle from the thousands who graduated 40 years previously. The emphasis on computer sciences and defence related technologies in funding from the computer industry, the defence sector, aerospace companies and the Federal government created an irresistible attraction at America's universities and colleges that swept the youthful but relatively impoverished discipline of audio engineering out of technology curriculums in the 1970s and early 1980s.

Coupled with the shift of more than 60% of domestic audio manufacturing offshore during the last 30 years, the pool of audio engineers trained and at work in the US audio industry has shrunk significantly. With so many of the consumer, personal and project, professional and industrial audio products sold today designed and-or manufactured offshore, the number of engineering jobs in basic and applied research, product design and so on has declined by nearly 75%. What has risen, especially in the United States and in the United Kingdom is the number of university and college trained music and-or sound recording practitioners who own, operate, sell or install audio equipment and facilities. These graduates of what are essentially labelled as music programmes, undergo rigorous preparation in virtually all of the traditional areas of audio engineering in addition to being versed in musical performance and in critical listening skills.

Yet the AES retains an original membership framework that relegates those labelled less technically proficient into the category of 'associate member'. In the past, this has meant that graduates of 'regular' engineering programmes have received greater recognition than those bearing the tag of 'musician' on their diplomas. After all, the instructions accompanying the membership application clearly state that a member 'may be any person who: is active in audio engineering; has an academic degree or its equivalent in scientific or professional experience in the field of audio engineering and its allied arts; and is familiar with the application of engineering principles'. It probably is appropriate to mention that engineering graduates who focus on the audio industry take a negative pay differential (read pay cut) of 10% to 25% below that of their computer industry-employed brethren.

Further complicating the issue of just who is a member of the Audio Engineering Society is the fact that the Society, once more-or-less a strictly American organisation registered in the State of New York, has become an international organisation with nearly half of its total membership outside the borders of the US. The foreign memberships tend to have greater numbers of full members in the regional sections but that does not mean that the process of selection is free from the same abuse as in the US. A member of an section within the European Community commented on his history with the AES, 'After several years as an associate, I felt it was time to request full member status. I did what was required for paperwork and heard nothing for several months. I inquired of the individual responsible at a meeting and was told my "qualifications were not sufficient". Over the next three years, I tried again several times. Each time the outcome was the same. Then I was made redundant at my company and promptly offered a new position at a more prestigious firm. After several months, the Managing Director suggested that I try again for member status. It was quite remarkable to see my change in grade sail right through, despite the fact that nothing had changed in my professional demeanour. Coincidence? Perhaps.'

A strictly American organisation has become an international organisation

None of this is to say that the AES is a great deal different from other technical societies founded during the same era. The AES bylaws were designed for a simpler time and a more homogeneous technical population. They have not held up over nearly 50 years but what else has? The answer to the problem of membership 'double standards' may be to consider the solution to a similar problem opted for by the Board of Governors of the SMPTE. With a similar membership demographic, bylaw context and organisational structure, what the SMPTE has done is to call a special election to vote on changing the group's bylaws to eliminate the grade of associate membership. In addition, the election is to remove requirements for active (or full) membership that measured eligibility in terms of time spent in the industry and of chronological age. Also earmarked for the ballot is the removal of the requirements for sponsorship by active members. What the simplified SMPTE structure would consist of is a simple test of qualifying employment, educational status or achievement that would, in the Society's words, 'encourage and enable a wider group of members to participate in the activities of the Society.'

Now, it is clear that the AES, which is closest in many characteristics to SMPTE, should consider a separate but similar course of action. The elimination of associate member status can serve to open the doors of the AES to the creative elements of an audio industry that more or less are excluded now. It can build a bridge to the year 2000 and widen the scope of the AES at a time when the entry-level studio musician has emerged at the practitioner and project level more or less triumphant. The fact is that there are virtually no recognised engineering programmes offering a course specifically in audio engineering while there are over 200 institutions offering courses of varying lengths in audio as related to music and technology. Even the bulk of the European graduates operating as audio practitioners owe their success to the Tonmeister programmes; which have more in common with many of the music technology programmes in the US than with pure engineering. Nobody is faulting the efforts to keep the AES' standards high, one is only asking that the reality of the marketplace be recognised. ■

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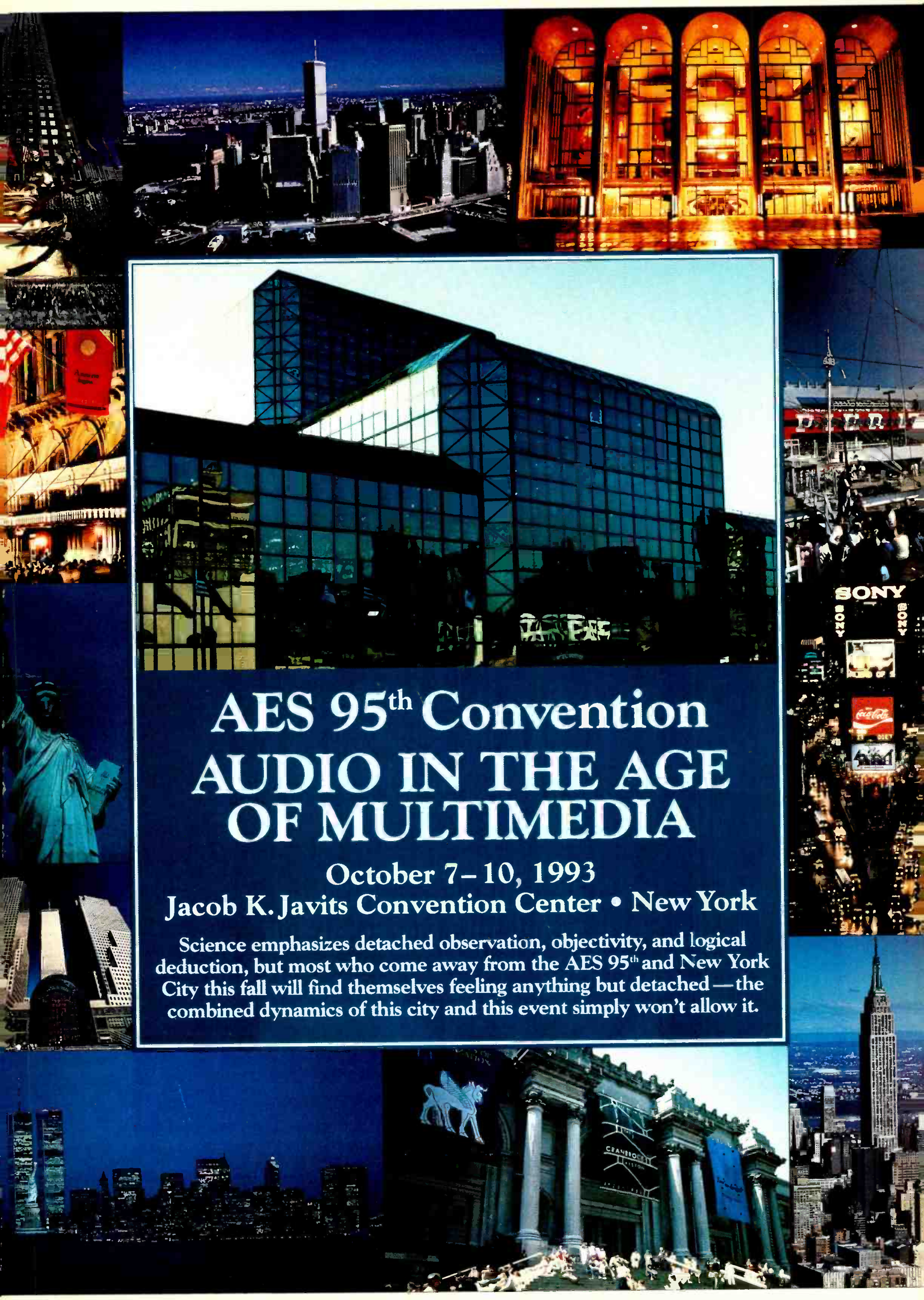
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ERRORS IN RDAT

The publication of the first part of *Studio Sound*'s DAT test has raised many questions regarding the format and the testing. John Watkinson casts an impartial eye over the results and draws some conclusions about their significance for the DAT user

The comparison of error rates in RDAT tapes made by Sam Wise ('The *Studio Sound* Dat Tape Test', May 1993, *Studio Sound*) showed quite a disparity between brands. I have been asked to cast an impartial eye over the tests and the results and, via a detailed treatment of the error correction strategy of RDAT, draw some conclusions about the significance of the tests for the user. Digital audio is no more than an alternative way of storing or conveying the waveform of air pressure or velocity at a microphone. Instead of trying to vary some continuous parameter like the transverse velocity of a groove or the strength of magnetic flux as a function of time, the magnitude of a series of whole numbers changes. Whole numbers are advantageous for recording because they are more robust than variable waveforms. Timebase error can be removed by buffering, so there is no wow and flutter. A whole number is either right or wrong, it cannot be off colour, and so error correction is possible. A corrected whole number is indistinguishable from the original.

The advantage for audio is that if the data from the ADC are subsequently supplied bit-for-bit accurate to a DAC, then the sound quality of the system depends only on the quality of the converters, and not on any attribute of the channel through which the data passed. It does not matter whether the channel was error free or whether it made errors which were fully corrected, the system will sound the same (unless it has a design defect, such as the error correction chip power supply interfering with the converter).

The whole reason for digital audio is that we can focus all of our efforts on designing excellent sounding quality; but the data channel in between does not have a sound quality; it only has a DAT reliability. We instinctively want that reliability to be 100%, but in the real world 100% reliability does not exist (that is the only thing that statistics can actually guarantee). In practice there is a residual error rate (the rate after correction) in digital audio which is sufficiently infrequent that most people are not affected most of the time. We are lucky because the acceptable residual error rate in computer data is something like a thousand times less. Thus digital audio recorded on a computer data recorder is likely to be somewhat overspecified, whereas it is not a good idea to put computer data on an unmodified digital audio recorder.

The residual error rate is the combination of the raw error distribution of the medium-head combination and the power of the error correction system. If the error correction is made more powerful, the medium-head can be worse, but the system performance is unchanged. When powerful error correction is cheap, as it is today, thanks to LSI chips, it makes sense to reduce tape consumption by making the tracks narrower and the linear density higher. RDAT was originally intended as a consumer product, and so it went further in this direction than it might have done had it been an *ab initio* professional use device.

RDAT error correction

The error correction strategy of RDAT is extremely powerful[1] and uses product codes to combat the combination of random and burst errors which are seen on typical tapes. Random errors are due to noise, either from the tape or externally induced, whereas burst errors are due to medium defects, such as pinholes in the coating, or head contact problems caused by an imperfect transport, contamination or ceasing. In the event that ►

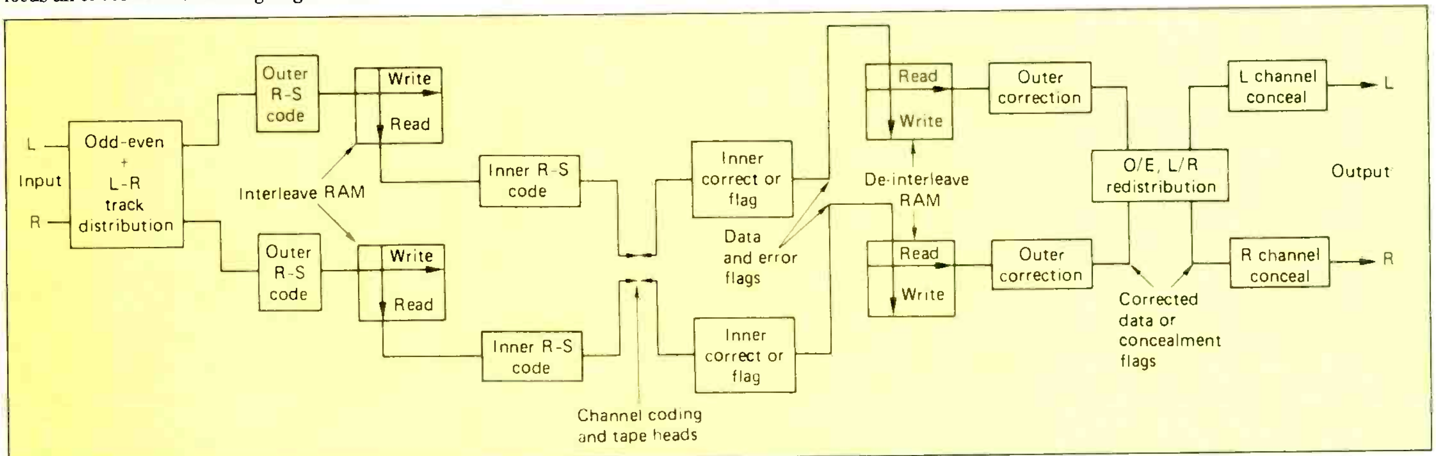


Fig.1: The error protection strategy of RDAT. To allow concealment on replay, an odd-even, left-right track distribution is used. Outer codes are generated on RAM rows, inner codes on columns. On replay, inner codes correct random errors, flag burst errors. Flags pass through de-interleaved RAM to outer codes which use them as erasure pointers. Uncorrected errors can be concealed after redistribution to real-time sequence

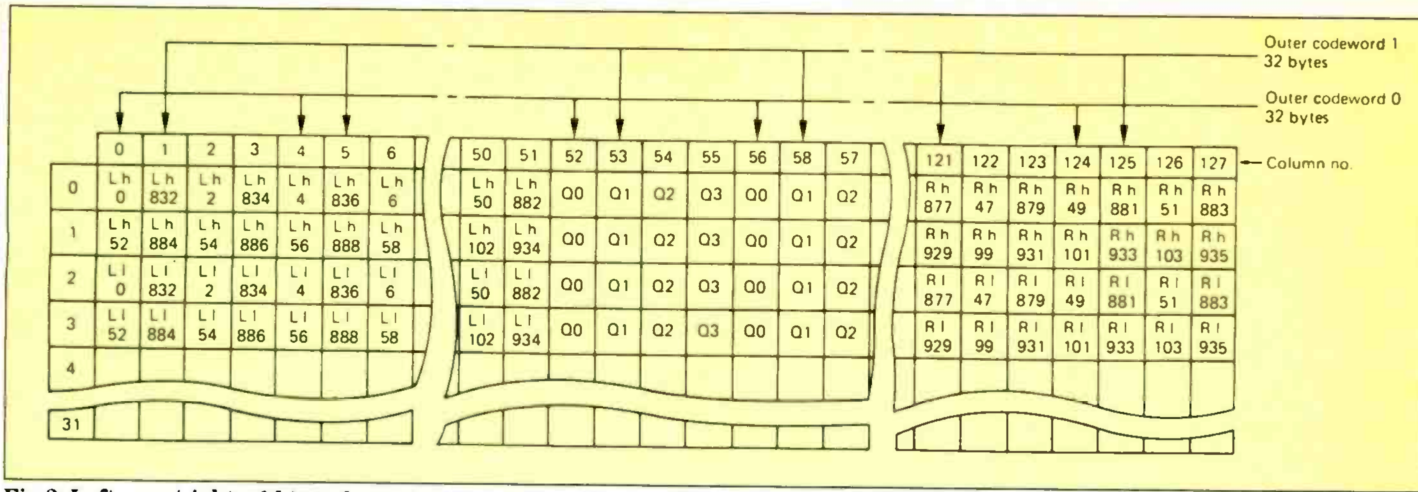


Fig.2: Left even/right odd interleave memory. Incoming samples are split into high bit (h) and low bit (l), and written across the memory rows using first the even columns from L0–830 and R 833–1,439. For 44.1kHz working, the number of samples is reduced from 1,440 to 1,323, and fewer locations are filled

errors become sufficiently numerous that correction is impossible, the system falls back on concealment rather than total failure.

Fig.1 shows a block diagram of the RDAT error handling system which uses product codes formed by producing Reed-Solomon codewords at right angles across memory arrays. The audio samples corresponding to one head revolution are routed to a pair of 4kbit RAMs, one for each track, which have 128 columns of 32 bits each. RDAT works with eight-bit symbols, and so each sample is divided into high bit and low bit and occupies two locations in RAM. **Fig.2** shows only one of the two RAMs. Incoming samples are written across the memory in rows, leaving an area in the centre, 24 bits wide. Each row of data in the RAM is used as the input to the Reed-Solomon encoder for the outer code. The encoder starts at the left-hand column, and then takes a bit from every fourth column, finishing at column 124 with a total of 26 bits. Six bits of redundancy are calculated to make a 32-bit outer codeword. The redundant bits are placed at the top of columns 52,56, 60, etc. The encoder then makes a second pass through the memory, starting in the second column and taking a bit from every fourth column finishing at column 125. A further six bits of redundancy are calculated and put into the top of columns 53, 57, 61, and so on. This process is performed four times for each row in the memory.

The inner codewords are encoded when the memory is read in columns. **Fig. 3** shows that, starting at top left, bits from the sixteen even-numbered rows of the first column, and from the first twelve even-numbered rows of the second column, are assembled and fed to the inner encoder. This produces four bits of redundancy which when added to the 28 bits of data makes a 32-bit inner codeword. The second inner code is assembled by making a second pass through the first two columns of the memory to read the samples on odd-numbered rows. Four bits of redundancy are added to these data also. The redundancy in each case is placed at the bottom of the second column. Each column of memory can be accommodated in one sync block on tape. The effect is that adjacent symbols in a sync block are not in the same codeword. The process then repeats down the next two columns in the memory and so on until 128 sync blocks have been written to the tape. This uses up the PCM audio sector of one track. The contents of the second RAM are written to the other track of the frame.

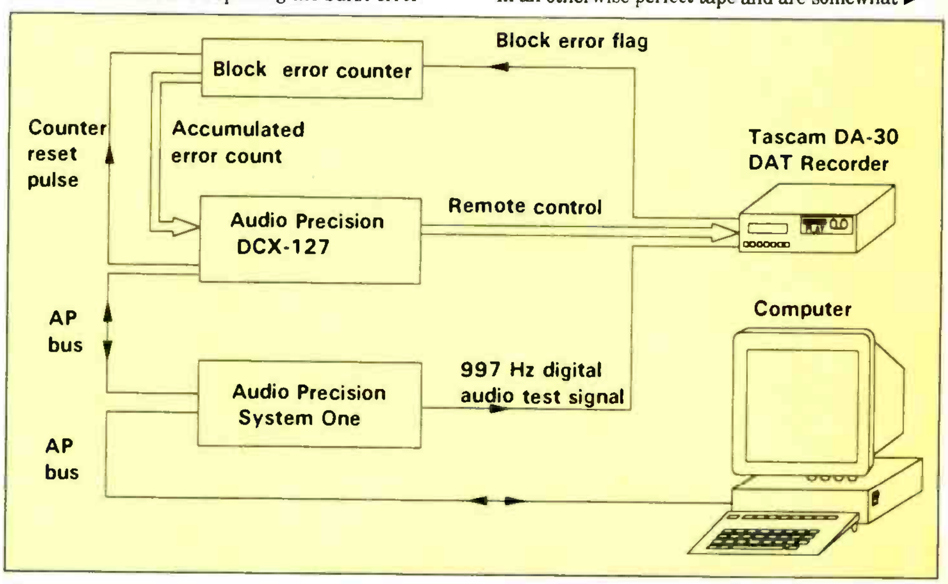
Upon replay, the sync blocks will suffer from a combination of random errors and burst errors. The effect of interleaving is that the burst errors will be converted to many single-symbol errors in different outer codewords.

As there are four bits of redundancy in each inner codeword, a theoretical maximum of two bits can be corrected. The probability of miscorrection in the inner code is minute for a single-bit error, because all four syndromes will agree on the nature of the error, but the probability of miscorrection on a double-bit error is much higher. The inner code logic is exposed to random noise during dropout conditions, and the probability of noise producing what appears to be only a two-symbol error is too great. If more than one bit is in error in an inner code all bits are flagged bad as they enter the de-interleave memory and the outer correction must be used. The interleave of the inner codes over two sync blocks is necessary because of the use of a group code. In the 8/10 code, a single mispositioned transition will change one 10 (channel) bit group into another, potentially corrupting up to eight data bits. A small disturbance at the boundary between two groups could corrupt up to sixteen bits which would be beyond the correcting power of the inner code. By interleaving two inner codes into two sync blocks, the worst case of a disturbance at the boundary of two groups is to produce a single-symbol error in two different inner codes. The inner code interleave halves the error propagation of the group code, which increases the chances of random errors being corrected by the inner codes instead of impairing the burst-error correction of the outer codes.

After de-interleave, any uncorrectable inner codewords will show up as single-bit errors in many different outer codewords accompanied by error flags. To guard against miscorrections in the inner codes, the outer code will calculate syndromes even if no error flags are detected, the outer code will correct them even though they were due because the outer code has three-bit detecting and correcting power which is never used to the full. If more than two bits are in error in the outer codeword, the correcting process uses the error flags from the inner code to correct up to six bits in error.

Fig. 4 shows the correcting power of RDAT graphically. Owing to the four-way interleave of the outer code, four entire sync blocks can be destroyed, but only one bit will be corrupted in a given outer codeword. As an outer (C2) codeword can correct up to six bits in error using flags from the inner (C1) code, it follows that a burst error of up to 24 sync blocks could be corrected. This corresponds to a length of track of 2.64mm, containing 6,336 bits, and is more than enough to cover the tenting effect caused by a particle of debris lifting the tape away from the tape head. The track angle of 6.35° means that a scratch about 0.3mm wide along the tape will cause damage of this length in every track. This limit is also reached by a transverse area of damage 2.6 mm long.

These figures assume a concentrated single error in an otherwise perfect tape and are somewhat ▶



The RDAT tape block error measurement test rig used in DAT testing

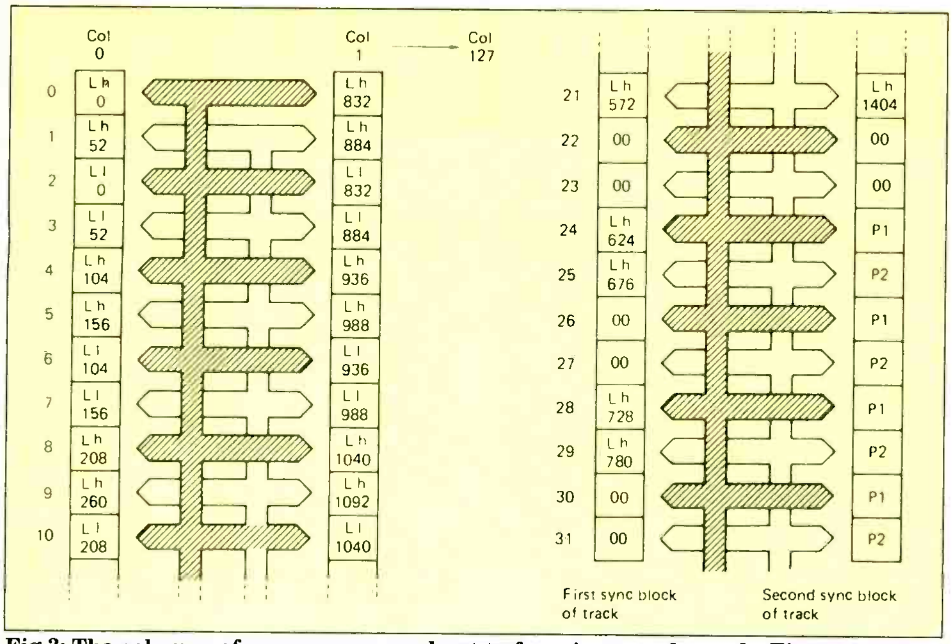


Fig.3: The columns of memory are read out to form inner codewords. First, even bits from the first two columns. As there are 128 columns, there will be 128 sync blocks in one audio segment

hypothetical. In practice the conditions will be somewhat different. As the error correction is so powerful, a single pinhole in the tape coating is easily dealt with. In fact it is mechanical problems such as misregistration which are a greater source of difficulty. If a sync block is read during mistracking, the signal will be noisy and it is more likely to suffer errors such as failing to register a flux change. Since a single flux change in error causes a symbol error due to the use of group coding, two or more flux changes missed in an inner code is beyond the power of that code and the entire code must be flagged bad in the replay RAM. As the correction limit in a single outer code is reached when 6 of sync blocks crossing that outer code need correction, it follows that in the worst case as few as 12 missed flux changes in the PCM sector of a track could cause that limit to be reached, although they would have to be critically positioned along the track by a sadist, and in real life this critical positioned will not occur. For reference the PCM sector contains 128 sync blocks of 32 bits each and each bit requires 10 channel-bit periods to be recorded. Thus around 40,000 channel bits create the waveform recorded in an audio sector, and in the absolute worst case, losing as few as 12 of them will bring us to the brink of concealment. Another worst case condition is where an error occurs in reading the sync pattern at the beginning of the sync blocks. If this pattern cannot be read, it is impossible to deserialise the block, and so a small error results in the loss of 32 bits. If this were to happen in six critically selected sync blocks on one track, again the correction limit would be reached. Mercifully these worst case critical patterns are so unlikely to occur in practice that we can neglect them. However, it does illustrate that it is in the nature of magnetic recording that the amount of data corrupted will be considerably greater than the actual number of flux changes wrongly read from the tape. Just knowing the rate at which a tape has flux change errors is not very helpful; it is their distribution which has the most effect in conjunction with the characteristics of the error correction strategy. Thus at the end of the day measuring the rate at which the error correction system of an actual RDAT performs corrections is a sensible and valid test to make. Comparisons with DDS are not relevant to RDAT because even though the tape is the same, DDS has a completely different error control strategy because of the reduced tolerance to residual errors in computer data.

The inner code of RDAT is designed to correct small random errors and it is implicit in the design of the format that these will occur. We expect them for the same reason we expect a noise floor on analogue tape. Thus the event which we should be worried about is not the occurrence of infrequent random errors, but the occurrence of errors which cause the correction power of the system to be exceeded. This will generally be caused by physical medium defects, such as missing magnetic coating, or by mechanical problems such as tape weave or transport alignment[2].

The tests

The design of the tests seems to leave little to chance. The channel code of RDAT shows a slight pattern sensitivity, such that some bit patterns are slightly more error-prone than others. The use of nonrelated audio frequency of 997Hz is a good way of ensuring that there is no preponderance of a specific code in the data and so there should be no bias from this source, particularly as this is a comparative test.

The Tascam *DA30* uses a Hitachi error correction chip which has an error flag output pin which produces a pulse every time a sync block plays back and is found *not* to contain a codeword. Thus the generation of the flag simply indicates the existence of disparity between the original data and the replayed data. It does not tell us how big the disparity was. It could have been as little as one bit, or considerably larger. **Fig.5** shows the timing of the flag bit with respect to the scanner rotation. Using the scanner waveforms, it is possible to gate flags only from the subcode, only from the audio or to select only one head. Sam Wise's test did not do this, but integrated errors from the subcode and the PCM areas of both heads. For a comparative test I see nothing wrong with this approach. The use of a control tape to verify the consistent performance of the transport and test rig throughout the tests is an essential step without which the results could be open to challenge. The spread of the control tape tests is one indicator of the accuracy of the remaining tests. However, the spread of the control tape and other potential measurement errors is such that some of the ranking in Tables 1 and 2 cannot be justified. In any statistical exercise, ranking should not be used if the difference between ranks is similar to the experimental accuracy. Some of the results were very close, especially the final error rate in **Table 2** for the Apogee, 3M and TDK tapes. I would not have ranked these six, seven, and eight, but would have placed them equal last. Similarly I would have placed the HHB and Maxell consumer tapes equal first in **Table 2**.

Having said that, the control tape test showed a spread which was much less than the overall spread

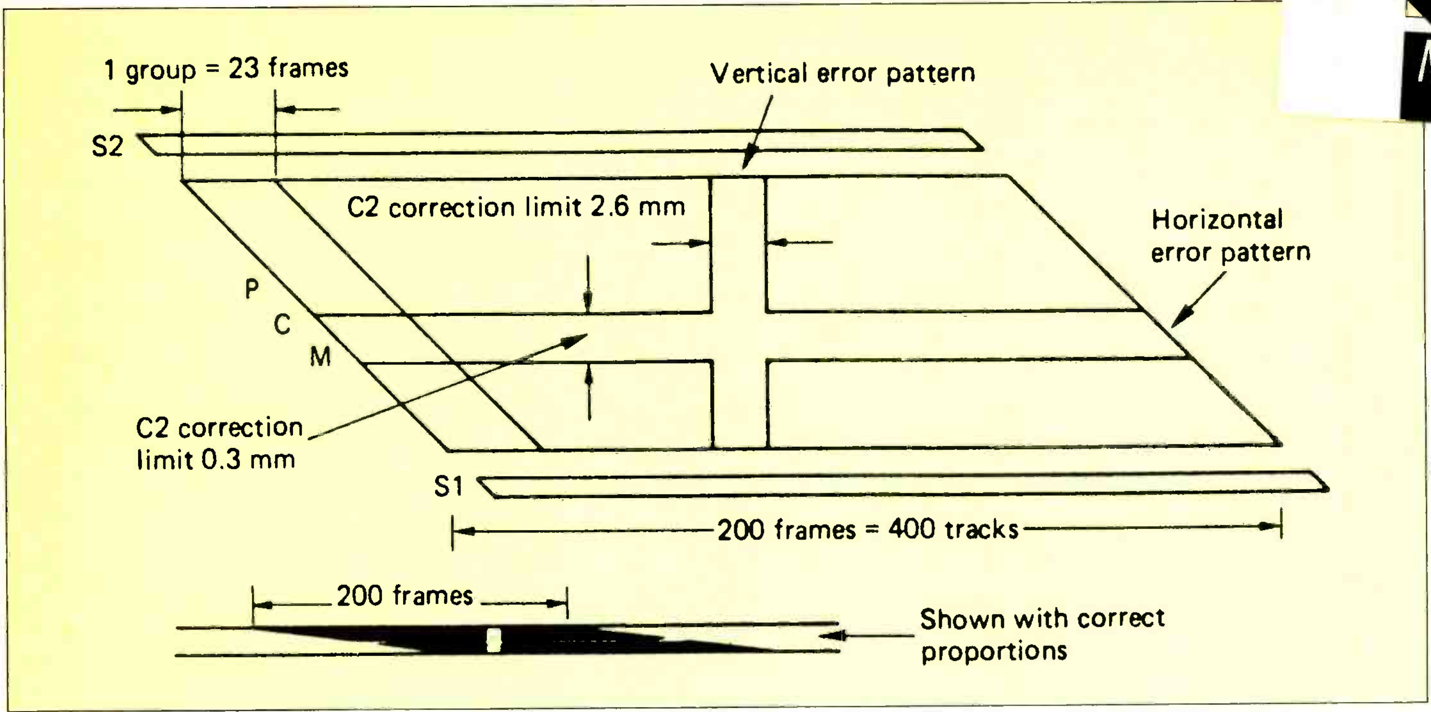


Fig.4: The correcting power of the C1, C2 codes of RDAT. The maximum correctable burst error length is 792 symbols (6,336 bits). Since the linear recording density of DAT is 61,000 bits/inch, this is equal to 2.64mm along a helical track. The horizontal correctable error width is 2.64 x cos 6.35° ≈ 2.6mm. The physical defects are shown conceptually in the upper diagram, and to scale in the lower diagram

from best result to worst, so I must conclude that the overall results of the test were statistically significant in that different brands of tape really do vary in error rate. Note that I did not say the reason for this is known or that it indicates cause for alarm.

What the results mean

The tests ran for a minute of tape, and in that time 4,000 tape tracks were played containing half a million sync blocks. In the worst result shown in **Table 2**, around 15,000 error flags were measured. If the errors were uniformly distributed, this figure would correspond to almost four flags per tape track, or about one sync block in 35 in error. This is well within the error correcting power of the RDAT code, and so there is no cause for jumping out of high buildings. On the strength of these results alone I could not refuse to use any of the tapes tested for professional recording purposes. I do, however, await with interest the results of the accelerated life tests, as I feel that these are more significant. It would have been nice to see if any of the tapes contained bursts of error flags at some point within their length as this would indicate a potentially serious dropout. Performing such a test would, however take an extremely long time.

One aspect of the test results which shows clearly is the periodicity in the error rate which results in variations in slope of the graphs. In one of the graphs there is a strong periodcity of about 13 seconds. This corresponds to one revolution of the supply side-hub near the beginning of a 90 or 120-minute tape and suggests that the tape pack does not turn evenly, but that the friction varies cyclically. If this is not the case, then there is a periodic change in the characteristics of the tape along its length. It would be interesting to repeat the tests at different places on the tape to see if the periodicity in the errors changes with the supply hub speed. I find these periodicities more significant than the spread of errors between brands.

The use of flangeless hubs and liner sheets in the RDAT cassette is a sign of its origin as a consumer format. Although this construction is cheaper and allows a slightly smaller cassette, its mechanical properties are inferior. There is a wear mechanism which is absent in a spooled cassette, and the friction can never to be constant. This causes variations in tape tension which are anathema to a rotary head transport. Tension variations cause tracking errors and the thickness of the air film between the tape and the rotating head changes, varying the head contact force. Professional cassettes such as in the DVTR formats contain proper spools and, like an open reel machine, the tape does not touch the flanges. It may be possible to design a professional RDAT cassette with single flanged spools like a U-matic which remains compatible with existing decks.

What concerns me about these periodic error rate variations is that these were observed on new media. The friction liner construction of the DAT cassette can only cause this to get worse. With repeated winding, the liner sheet wears, and with ageing the lubricants leach out of the tape and liner. The cassette itself may warp with time. The result is that the tape itself is fine, but it cannot be played because tension variations cause tracking errors which cause ▶

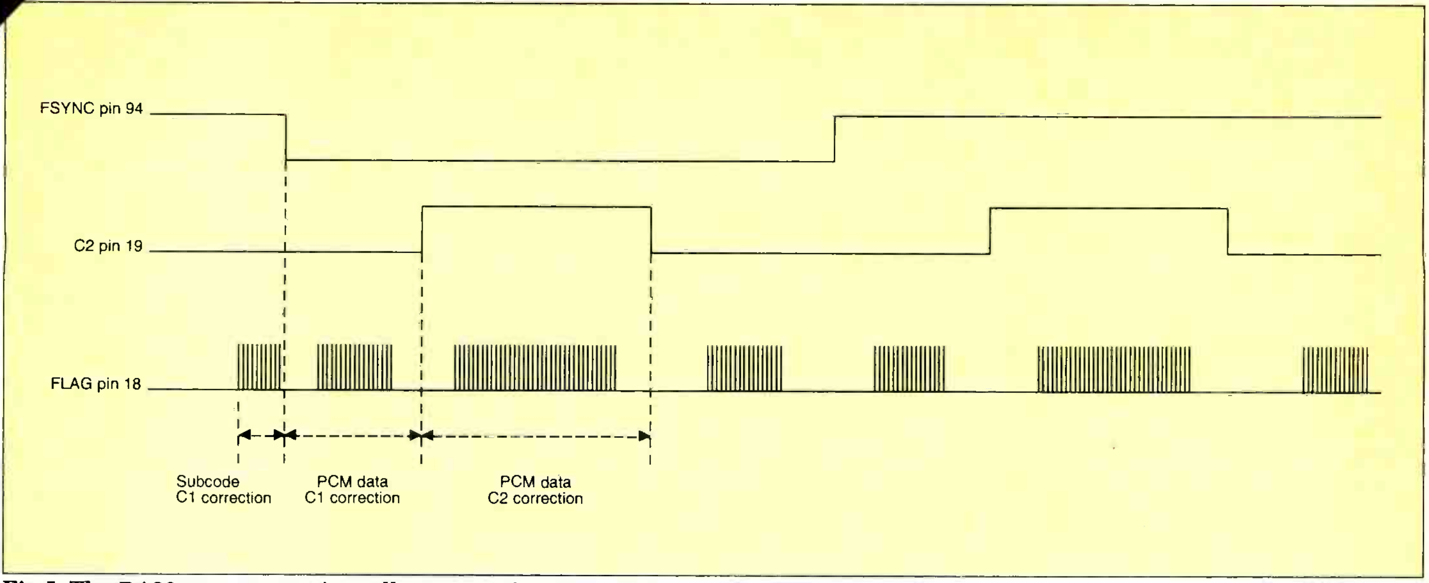


Fig.5: The *DA30* error-test points allow errors in selected areas of the tape to be gated. A maximum of 128 flags per audio block and 16 flags per subcode block can be present

excessive errors. I have seen plenty of old Compact Cassettes which suffered appalling wow and flutter which were rendered quite playable by fitting the tape in a new shell. This process may become necessary in the future when DAT cassettes start to get old.

Another factor which is of significance is the disparity between the rate at which contamination builds up with different tapes. Regular cleaning of RDAT transports is probably a more important factor than the variation in error rates in the brands tests. It is also in the user's control. How often do you clean your machine?

I am once again reminded of a fundamental truth of tape recording which is that the magnetic characteristics of media are easy to obtain in comparison to the required mechanical properties. Sam Wise has done us a service by making this truth more widely appreciated. ■

Thanks are due to TEAC UK Limited for details of the *DA30* error flag.

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1. Watkinson, J.R. *RDAT*, Focal Press, ISBN 0 240 51306 1 (1991).
2. Odaka, Tan and Vermeulen, *Designing a Data Storage Format for Digital Audio Tape (DAT)*, Hewlett-Packard (1988).

JOHN WATKINSON is an independent consultant in digital audio, video and data technology. He is the author of seven books on the subject including the definitive *The Art of Digital Audio*. John is a fellow of the AES, is listed in *Who's Who in the World* and regularly presents papers at conventions of learned societies.

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Barry Fox

Launching into vibration and the aftershock of marketing campaigns

What a lot of difference a little peace makes. With fewer military contracts to work on, Hughes Aircraft in California have teamed up with General Motors to make electric cars. Hughes also plan to use their own satellites to broadcast pay-per-view movies.

To publicise the venture, Hughes recently showed a hundred European journalists behind the previously closed doors of its satellite production factory in Los Angeles. Asked only not to touch, we got within touching distance of several dozen half-built spacecraft.

My eye was caught by a test facility which simulates launch conditions. The finished satellite is put on a giant shaker table which simulates the violent vibration it will undergo during lift-off. The table is actually just one enormous loudspeaker, driven by 300kW audio amplifiers which are fed with computer-generated audio signals which sweep between 5Hz and 2kHz. Forces of 300,000lb rattle the table backwards and forwards by at least an inch either way.

The man in charge has been there long enough to remember how Howard Hughes would come round and remember everybody's name. He also recalled how, one Christmas, one of the workers got the holiday spirit, unplugged the computer drive, hooked up a gramophone and played carols through the shaker table. The sound filled the entire factory.

What I would really like to hear now is Stephen Court's CD recording of the Chieftain tank gun played through the Hughes satellite tester.

I can picture the scene: the recording engineers working for Deutsche Grammophon in Hanover are quietly continuing their quest for improved recording quality. They do a deal with Yamaha to buy some new mixing desks, then build some new low-noise head-amps for their Schoeps microphones.

Then they find that noise and interference is getting into the long leads that run from the head amps to digital converters ahead of the mixing desk. The leads are long because DG record in large concert halls where the only place to put the mixing desk is in a distant robing room or crypt. The halls electrical circuits and lighting are 'dirty', and radiate click, pop and buzz signals.

So DG's engineers do the obvious thing. They put the digital converters as close as possible to the microphones, and run digital signals through the long feeds back to the control desk. It all makes sound sense, and pushes the sound one step closer to unachievable perfection.

But now, Deutsche Grammophon's marketing people get in on the act. They cook up a name, 4D, for 'a new concept in sound recording (which) allows the Tonmeister (a fancy name for Balance Engineer) unprecedented control over the recording process'. Then they publish a glossy mess of overlapping brochures that do more to confuse the simple proposition than explain it. This is very likely because the people writing the brochures do not understand what they are writing about.

For mixdown and mastering, Deutsche Grammophon's engineers have been borrowing from pioneering work done by sister company Decca at West Hampstead over the last ten years or more. Decca's work has not been publicised much because Tony Griffiths reckons it is better to get on with his job than spend time and money on patenting something so he can tell his competitors about it.

But Deutsche Grammophon now see Sony scoring publicity for Super Bit Mapping, a fancy name for reshaping the noise in a digital signal before mastering a 20-bit digital recording into 16-bit CD format. So DG's Publicity People cook up some fancy terms. Let us call our dither system Authentic Bit Imaging, they say, or High Bit. They produce a lot more bumph, which does not explain how ABI-High Bit works, invite opinion makers for lunch in Germany and tell people like me who question how the system works, 'Just listen to this CD, which uses the system'.

Meanwhile Deutsche Grammophon's engineers have found that they can improve the sound of their 4D recordings, by introducing delays into the digital feeds to compensate for the delays when one instrument is picked up by different microphones. The engineers also find that they can clean up the delays that have been locked into the multitrack tapes made at pre-4D multi-miked recordings. The trick here is to look at the recording logs and work out how far apart the microphones were spaced. This tells the delays, so remixing with mirror-image delays, compensates for the time errors locked on tape.

DG's Publicity people now pounce again, christening the clean up technique, Original Image Bit processing or OIB (no connection with Polygram's Obie Oberstein).

The system is being first used for DG's rerelease of 20 CDs and DCCs of mid-1980s material to mark the 85th anniversary of the birth of Herbert Von Karajan. The 20-piece set has been dubbed, *Karajan Gold*, and those with good ears and access to both original and reprocessed recordings reckon the improvement is marked.

Strangely, no-one seems to take into account the obvious, and perhaps far more significant effect, of remixing these tapes with a completely new generation of mastering equipment.

In Germany the public (brought up on a diet of Grundig instruction books and hi-fi and video equipment with knobs for functions you have never even dreamed of) may swallow Deutsche Grammophon's publicity. But the British punter, already baffled by talk of Super Bit Mapping, must surely wonder what on earth Deutsche Grammophon mean by a 4D Audio Recording, made with Original Image Bit processing, and transferred to disc by the High-Bit system of Authentic Bit Imaging.

Despite promises such as those from Klaus Hiemann, Head of Deutsche Grammophon's Recording Centre in Hanover, I never did get a coherent answer to my question, what exactly is ABI and how does it differ from what Decca, Sony *et al* are already doing?

The best I could get, after a lot of nagging, was that, 'ABI is a requantising system, 24-bit to 16-bit, which uses non-subtracted dither with noise-shaped error feedback and is based in principles established by Stanley Lipshitz'.

So now you know—or rather you don't. And although they have made no such claims so far, Deutsche Grammophon should know that if I spot any adverts with unsubstantiated claims, for example for +20-bit recording, I shall simply lodge a complaint with the Advertising Standards Authority, which will then push Deutsche Grammophon into being a whole lot more specific about where the bits are coming from.

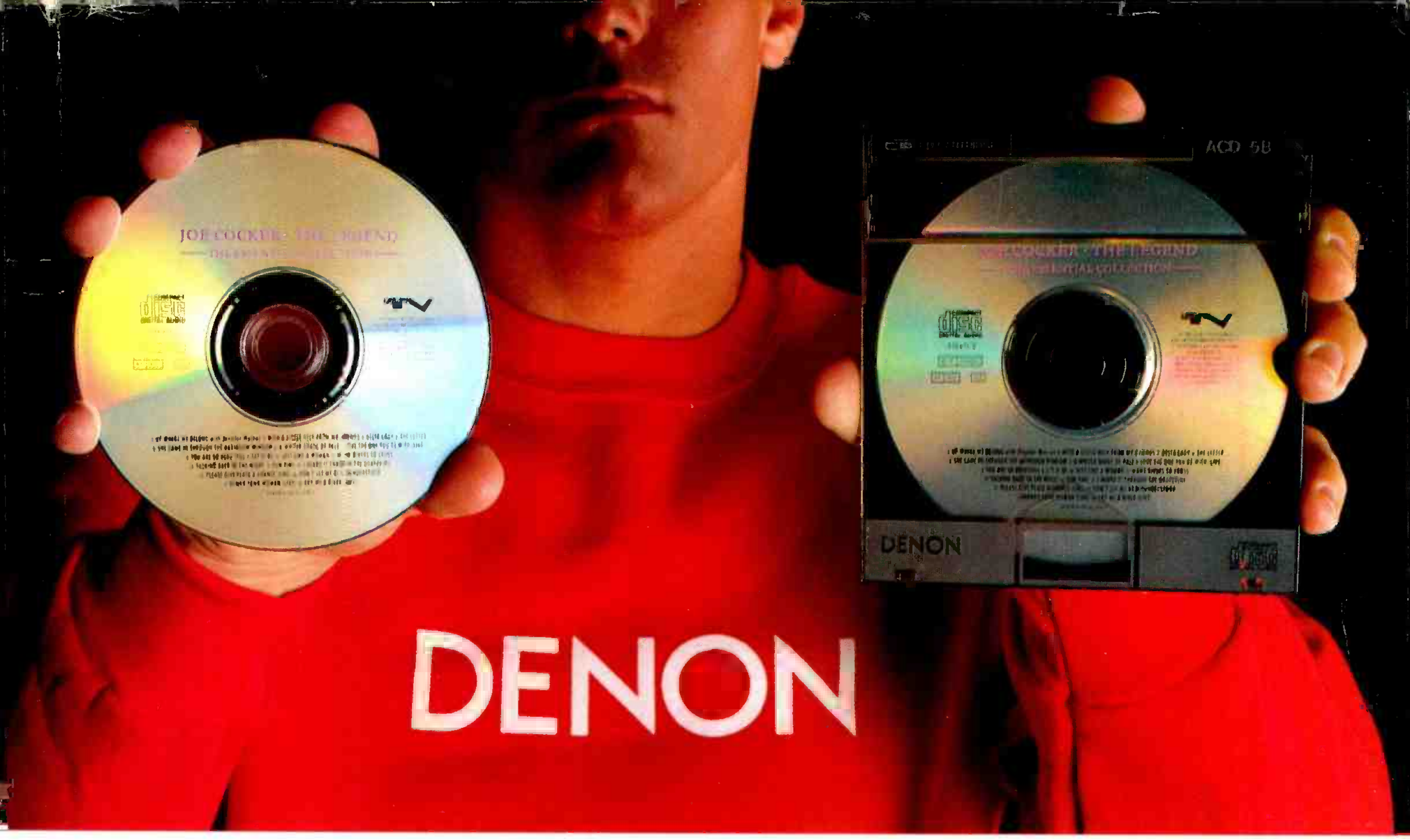
Why don't DG cut out all the electronic 'friggery' needed to compensate for time delays between microphones, and just use a single stereo pair of microphones?

'Not possible', say DG's engineers. They have only ever found one hall where it was possible to get a good sound from a single stereo pair, without first spending an expensive day moving the musicians round and asking them to play quieter or louder. That venue was the Kingsway Hall in London, and it is no longer available for recording.

A final thought: Tony Griffiths warns that Decca spotted early on that there was a hidden trap in using dither during postproduction. If the signal is copied many times and dither, whether noise-shaped or not, is added, then there will be an overall increased in noise level. So Decca's dither system does not operate when signals are copied at unity gain, during editing. Griffiths believes that Decca are unique in not adding noise when gain is unity. If Deutsche Grammophon now borrow this idea, will we see another new name?

For the record, I have now had to ask DG's people to stop talking at me about 4D. Nor do I have time to go to Hanover for yet more ear-bashing. But if DG want now to stage another seminar in the UK, but this time with someone present to answer technical questions, I will do my best to attend. Better still, if Deutsche Grammophon belatedly put together a coherent written explanation of the system, I will gladly read it. ■

ILLUSTRATION: CARL FLINT



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