OCTOBER 1993 $5.00 £2.00

STUDIO SOUND

AND BROADCAST ENGINEERING

THE INTERNATIONAL PRO AUDIO MAGAZINE

AUDIO RECORDING

Fostex D10; Cedar DeCrackler; ICE; Sony DMX-S6000

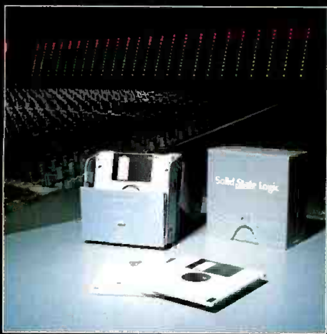
The Future of Recording

The Medium Is The Message

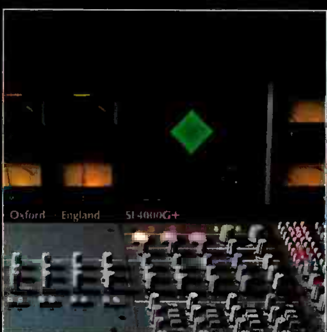
Digital Video

Bit Rate Reduction Systems

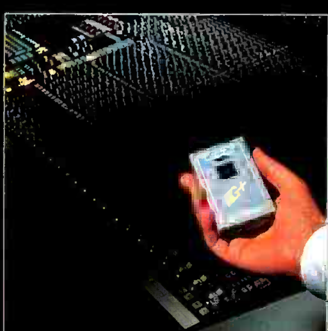
G PLUS CONSOLE SYSTEMS



3.5" disk drives allow the use of low cost, high capacity disks for mix data storage



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



Wireless talkback system uses a PCM encoded infra-red handset

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IMPROVED SPECIFICATION AT NO EXTRA COST

STUDIO SOUND
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Pebble Mill refurbishment
See page 8

"YOU WANT TO DO WHAT !?!"

So there he was, this *client*, for want of a better word, telling me, in no uncertain terms we needed a piece long ago consigned to the bin. I could see myself burning the midnight oil again, desperately trying to find this 3 second out-take from the 2000 feet on the cutting room floor.

And what about the night before! I'd mixed down a couple of nifty, if a little time-consuming crossfades, then realised I had a problem - all the edits from earlier that evening also needed crossfades to cover the gaps. Oh well. Sleep's overrated anyway! It's just something else to do in bed!

I should've listened to Jim! I'd just replaced my ageing tape deck with a gleaming new machine when he said, *"You could get a complete* SADiE *system for less than that - real-time crossfades, non-destructive editing and so fast to use, it's incredible!"*

What next! I need more tape, more time, *less grey hair......*

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*Windows 3.1 is a registered trademark of Microsoft Inc.

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un A United Newspapers publication

Buzz words and bull

New York. The 95th AES Show: *Audio in the Age of Multimedia*. It is a bold and a brave title, but I have to admit that I would be a lot happier about it if multimedia was not such a vague and abused term. Certainly, the expansion and integration of sound, vision and computer technologies is continuing apace but—and it is a big but—the 'multimedia' tag is little more than a buzz word at this stage. The trouble with buzz words is that, without a commonly agreed and acceptable definition (even a vague one), they are God's gift to purveyors of bull. Take last season's buzz: 'workstation'. Before it hit pro-audio as 'digital audio workstation', it plagued the electronic keyboard market. Korg's *M1 Music Workstation* claimed to be everything the modern keyboard player needed. But it was not too long before it became just another (very popular) keyboard instrument. In its darkest moment, the workstation tag was even hung on a stand intended to prop up an Atari *ST* computer...

Soon the bubble burst and people were moving rapidly away from the term while simultaneously refining the concept. If there is one thing that moves more rapidly than technical progress, it appears to be marketing strategies. At present, there is at least one pro-audio equipment manufacturer with a digital audio workstation in their catalogue worried about the detrimental effect other, less worthy, items bearing the same tag may be having on the perception of the term. I cannot help but sympathise.

Returning to New York, multimedia and the future, we have to accept that the pro-audio industry is changing. Advances in technology, a worldwide recession and fundamental demographic changes are all playing a part in reshaping our industry. But the worst single mistake we can collectively make is to talk about the 'destruction' of the business, as so many seem resigned to do. Is it not strange that, in a business that revels in ever-changing technology, changes in business are perceived as being so frightening? Surely the advent of integrated media opens more doors for audio technicians than it closes—sound is an important aspect of this progress, and audio people have the very best audio skills to offer. The alternative to leaving the audio to people already involved in some other aspect of the new media is for audio people to take on other aspects of these media themselves. And for audio-only facilities to address other aspects of their production—it is really only an extension of the CD plants which now manufacture CD-ROM and CD-I as well as 'good old' audio CDs. And doesn't the growing importance of multichannel sound offer audio people and facilities more than two audio channels to work on? I would call that expansion...

Of course the refining, if not defining, of terms such as 'multimedia' is certain to assist us all in making career as well as business decisions for the future. Which is where I came in.

There are also healthy signs for those of us who feel more comfortable with audio the way we have come to know it. Data reduced recordings offer genuinely high-quality audio systems and audio productions an important opportunity for the future. If consumer formats such as MD and DCC—not to mention broadcast systems— are to be reliant on compromised audio, it is sure to generate considerable pride in the uncompromised article. After all, dogged defence of purist technology is one of our best talents. Surely the ferocity of the recent debate surrounding classical recording practices is some indication that none of the parties involved is anticipating the imminent demise of audio-only recording. The recent reports on DAT tape have also proved as contentious as they have difficult, and I do not see DAT being subverted into the service of video or 'multimedia'. Certainly, *Studio Sound* attracted its share of legal threats over both issues—but then, that is not unusual for a publication that is taken seriously.

As for multimedia, let us chew it up and spit it out with decent audio. And thank the AES for helping us get on with the job. ■

Tim Goodyer

Cover: tc electronic's *M5000*

The World's Leading Professional Audio Workstation Now Has Major New Features, Dramatically Improved Performance, And Lower System Prices.

SINCE 1991, THOUSANDS OF AUDIO PROFESSIONALS HAVE COME TO depend upon the power and flexibility of Digidesign's Pro Tools™ for audio post, music, and broadcast production. Indeed, you'll find hits made with Pro Tools on just about any top movie or music chart, just about any week of the year. In many circles, Pro Tools is even considered the *de facto* standard of multichannel professional audio workstations.

Rather than let this success go to our heads, we let it go to our brains. And ears. And hands.

You see, we've been thinking. And listening. And working hard. All to build something even better. Now, it's ready, and it's called Pro Tools 2.0.

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More Than Speed and Efficiency. The benefits of Pro Tools 2.0 go far beyond the obvious. For instance, we improved Pro Tools' already acclaimed audio quality: Our new digital EQs are as effective and musical as they are clean. We've added a host of intuitive automation, autolocation and transport features. Pro Tools now has a no-wait waveform overview mode. There's complete time-stamping. Enhanced grouping. Better scrubbing. More session management options. Bigger, brighter,

Thanks to our new Apogee-clock-equipped SMPTE Slave Driver, Pro Tools 2.0 delivers ultra high-fidelity, ultra low-jitter, ultra-easy digital synchronization in playback and record, complete with varispeed.*

faster, and more accurate VU metering. Improved MIDI sequencing and control. Extensive undo commands. In fact, Pro Tools 2.0 has dozens of new features, and scores of enhancements, for audio post, music, and broadcast production applications.

You Might Call That Impressive.

you'll notice is that 2.0 combines full-featured recording, mixing, signal processing, automation, along with advanced waveform and event editing — all in one, easy-to-use, integrated program.

There is simply no other interface as fast, as powerful, as flexible, and as complete.

Are you a Pro Tools owner who has been holding your breath as you read this ad? All registered owners will be offered a 2.0 Upgrade Kit, including software, new manuals, and an instructional video. All for just $49. So breathe easily.

There's Much More To Come. We've become the industry leader by responding to the needs of the people who use digital audio. And by listening carefully to people like you, we have a clear vision of what a professional digital studio should offer.

First of all, it should offer power, flexibility, loads of features, and excellent sound quality. It should be highly cost-effective, without compromising performance or quality. It should be modular, to allow each user to have the power he or she needs, without having to pay for unnecessary features. It should also be

Lower system prices now apply to 8-channel, 12-channel, and 16-channel Pro Tools systems. Four-channel Expansion Kits now also cost less. See your Authorized Digidesign Pro Tools Dealer for details. While all Pro Tools systems include built-in "trigger sync" for playback, the optional Digidesign SMPTE Slave Driver delivers the highest-fidelity record and playback sync (call us for the details). DINR uses Digidesign's optional Sound Designer II™ software, which runs with any Pro Tools system. *A US$49 Upgrade Kit cost applies to residents of the US and Canada, and includes shipping. Internationally, the cost is just US$69, including express shipping. Pro Tools owners must be registered directly with Digidesign to be eligible to receive the Upgrade Kit.

backed up by the best customer support in the entire industry. It's no coincidence that all of this describes Pro Tools 2.0, perfectly.

A Vision To Share. We believe that a truly professional digital audio workstation should have a truly open architecture.

By "open," we mean that you should be allowed to add software-based power when you need it. (DINR™ our amazing Digidesign Intelligent Noise Reduction™ system, is the first of many forthcoming Digidesign software "plug-ins.")

By open, we also mean that your workstation should have powerful and affordable upgrade paths. (We think Pro Tools 2.0 makes this case quite elegantly.)

And by open, we actually believe that your professional digital studio should allow you to integrate hardware and software from a variety of manufacturers, and not just us. Which is why, in 1993, we are releasing the remarkable Digidesign TDM Digital Audio Bus—a 256-channel, 24-bit system which will allow you to "drop in" DSP and other cards from Lexicon, Apogee, and other leading manufacturers. No muss, no fuss. You can even automate and route your existing analog and digital gear within this new, all digital environment. (Of course, all past and present Pro Tools systems will be easily and affordably upgradable to be totally compatible with the Digidesign TDM Digital Audio Bus.

Now that's what we mean by open.

Make Pro Tools 2.0 Your Reality. Thanks for taking a few minutes to read what we had to say. Hopefully, we've helped you understand better the depth of our commitment to the people who use Pro Tools today, and will be using it tomorrow.

So while we call it Pro Tools 2.0, you might call it exactly what you need. And if so, perhaps your next call should be to us.

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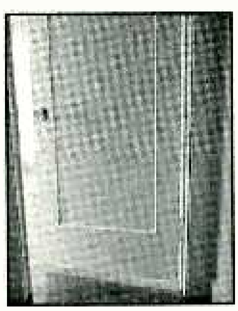
THE NEW PRO TOOLS POSTVIEW™ OPTION INTEGRATES FRAME-ACCURATE, FULL-SCREEN, FULL MOTION RANDOM-ACCESS DIGITAL VIDEO* WITH YOUR PRO TOOLS SYSTEM. THE POSTVIEW PACKAGE ALSO FEATURES MACHINE CONTROL* FOR PERFORMING CHASE LOCK OF VIDEO DECKS TO PRO TOOLS AUDIO. CALL ONE OF THE DIGIDESIGN REPRESENTATIVES LISTED BELOW FOR MORE INFORMATION.

*REQUIRES SOME ADDITIONAL THIRD-PARTY HARDWARE.

WE CALL IT PRO TOOLS 2.0



With our TDM Digital Audio Bus, Pro Tools offers your studio an open door to the future.

PRO TOOLS

For more information about Pro Tools, or about upgrading to Pro Tools 2.0, you can reach us at one of the dealers listed below:

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TEL: 041 226 3250
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TEL: 041 204 2774

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International News

In-brief

● CD-ROM firm get Alabama bound
Microinfo's CD-ROM Division has been appointed UK distributor for the full range of CD-ROM based information products and services produced by EBSCO Publishing based in Birmingham , Alabama

● AES links with *One to One* Mag
The AES has joined forces with *Studio Sound*'s sister publication *One To One* to set up an audio duplication and CD technology forum within the 96th AES Convention in Amsterdam next year. Celebrating the 30th anniversary of the audio cassette, the forum will feature a range of technical sessions presented by authoritative speakers in the business to what is likely to be the largest gathering of companies in the industry.

▼ New man takes over at AKG

Hendrik Homan, new MD at AKG Acoustics

Due to the changes of ownership with AKG, Mr Helmut Gunst will resign, with full consent of the Supervisory Board of AKG, from his posts of Chairman of the Managing Board of AKG and President of AKG Holding AG, at the end of October. Mr Hendrik Homan has been appointed Managing Director of AKG Acoustics from the beginning of September

● AMS-Neve awards to graduates
AMS-Neve have presented their annual awards to Tonmeister BMus graduates from the University of Surrey. The prize for Creative recording was awarded to 23-year-old John Hide who also received the Jacques Levy Prize for his final year project on the objective assessment of digital audio systems using perceptual models.

● Apologies to Sony Classical
We would like to make it clear that Barry Bongiovi formerly occupied the position of General manager of the Sony label studio facility at Sony Classical before he joined Touchdown and not Director of Operations as stated previously.

SoundField gets third owner in three years

AMS-Neve have sold their *SoundField* and *ST250* business to Drawmer Distribution. This follows a review by AMS-Neve of their product portfolio and the decision to focus wholly on their range of editing and mixing equipment.

AMS-Neve's MD, Mark Crabtree commented on the sale, 'The merger of AMS and Neve has obviously produced a large and varied product portfolio. It was felt that the new company was therefore not in an ideal position to give these excellent microphones the attention they deserve in terms of R&D and marketing activities.

All responsibility for product development, manufacturing, marketing and service of the *SoundField* and *ST250* microphones has been transferred to SoundField Research Ltd, a sister company of Drawmer Distribution Ltd.

Ken Giles, MD of Drawmer added, '*SoundField* is a unique product and is the only internationally acclaimed British microphone. We will continue to use the patented *SoundField* technology and are committed to manufacturing and supporting the products to the same high standards that users have enjoyed from Calrec and AMS-Neve.

SoundField Research.
Tel: 0924 201089. Fax: 0924 201618.

Steve Mac of Skratch Music in London poses in front of his new Soundtracs *Jade 48PB* console. Larking Audio installed the desk in Scratch's number two studio. As well as producing material for clients such as Lindy Layton and Nomad, Steve remixes for Right Said Fred and Kenny Thomas. His own band Under Cover were famous for their version of *Baker Street* by Gerry Rafferty

Codename NISKO

For some time Radio DRS in Zurich has planned to renew their main studios. The project was given the codename NISKO (new information and transmission).

After evaluation by Radio DRS, Studer received the complete order for the switching room, digital mixing consoles, and the installation.

Four studios are now equipped with one digital console each. The connection to the central MADI router is implemented via one pair of fibre-optic connections each. Each of these can simultaneously transmit up to 56 audio channels.

The MADI switching matrix is operated via terminals in the different studios, that is the studio engineer or the DJ on the mixing console connects his studio to the correct line and calls in audio input lines from other studios or from other remote locations.

An electronic logbook records automatically all switching states. Depending on the task, setups that are needed later can be prepared and instantaneously recalled with the push of a button. The operator can format the screen in such as way that the display optimally satisfies his requirements and that only that information is displayed which he actually needs for his production or broadcast.

Hotel Studios become SSL landmark

Landmark Studio in Yokohama, Japan, a new facility owned by the Bay City Group, has recently installed two *SL 4072 G Plus* consoles. These are the first *G Plus* consoles to be installed in Japan.

The studios are located in Landmark Tower, the tallest building in Japan (296m high), and linked to Landmark Hall for live recordings.

Mr Hiromi Kuroiwa, Studio Manager commented 'Landmark Studio chose the SSL consoles because they are very popular both in Japan and throughout the world. The *G Plus* consoles also have excellent operational features, including the 3.5-inch floppy disks which make the storage of material very easy. The sonic enhancements of *G Plus* are also appreciated.'

One of two *G Plus* consoles at Landmark

Major meeting for CD-I developers

A major compact disc interactive title engineering development seminar aimed at all sections of the software community is to take place on November 7th–9th.

The conference, sponsored by Philips Interactive Media and the European CD-I Association, is designed for existing and potential developers from a wide range of disciplines.

Visitors should include games companies; audio-visual producers; developers for corporate market applications; postproduction houses and representatives from film, television and record companies.

Contact Tel: +31 30 932 209.

Spatializer signs up with Matsushita

Spatializer Audio Labs have announced that they have signed a letter of intent with Matsushita to produce *Spatializer* 2-speaker 3-D Surround Sound audio processor ICs for use by consumer electronics companies worldwide, and will underwrite the the associated development costs.

'We are pleased to be associated with an advancement in sound reproduction as revolutionary as *Spatializer*' commented S. Teramoto, Director of Matsushita's IC division, 'the potential for this new technology is almost unlimited, and we see markets worldwide from home audio to multimedia computing.'

Production quantities of the new IC are expected to be available to manufacturers by mid-1994.

CEO Steven Gershick commented for Spatializer Labs, 'Truly effective and affordable 3-D sound is now a reality for today's audio consumer. So whether you are enjoying a new CD or televised broadcast recorded with our professional system, or listening to your favourite old album or movie through a stereo or VCR equipped with our consumer IC, *Spatializer* adds a natural and vibrant new sense of dimension, depth and clarity.'

MicroSound return for veteran

Industry veteran Armin Steiner, until recently Director of Recording for 20th Century Fox and before that engineer to over one hundred gold and platinum selling records, has returned to recording and chosen a *MicroSound* digital audio workstation as his main work tool.

Steiner commented 'Although there are many viable workstations available, only Micro Technology had the answer to two of my greatest concerns. First *MicroSound* was the only workstation that had real virtual tracks and second, unlike some of the MIDI sequencer or visual overload types of interfaces, their user interface simulated exactly how I would compose, record and edit.

Armin, his Gold records and *MicroSound*

Jean Michel Jarre tour whirls on

Jean Michel Jarre's *Chronologie* tour kicked off in France and is due to cover all of the major centres in Western and Eastern Europe by its end. As usual the production parameters are immense and in fact two separate towers are erected for sound and light and the rear of each tower provides support for the Meyer *MSL-3* delay systems, 24 per tower in the case of the concert at the Pontaise football stadium in Lusanne, Switzerland.

Chronologie tour—Jean Michel Jarre uses up more electricity

Supplying the sound was French PA company Dispatch, who supplied 42 SCV *246* compact enclosures per side (designed in conjunction with Dispatch) and distributed over four levels; three rows of 12-inch arced arrays plus a top array of six cabinets. There are also 16 Meyer *UPA-1A* speakers for nearfield coverage and a row of 24 Meyer *USW-650* subwoofers across the width of the stage at ground level. The show is mixed through a Yamaha *PM4000* with 48 channels with a stereo submix for a Sony *PCM3324A* with the various effects and additional backing tracks coming from a TAC *Bullet* 28-channel console.

According to Dispatch there is nothing special in the way of outboard equipment—just standard.

Show Previews

Exhibitions, conferences, courses

AES 95th Convention New York, New York. 7th–10th October.
Vision '93. Olympia, London. October 5th–7th.
Broadcasting Cable & Satellite India '93, Pragati Maidan, New Delhi. October 25th–28th.
EuroComNet '93, RAI Conference Centre, Amsterdam, 2nd–4th November.
LD Orlando 93, Celebrating entertainment and lighting technology, Orlando Convention Centre, November 12th—15th.
Communications and Broadcast —Turkey '93, Turkey's premier trade exhibition for the electronic communications, broadcasting equipment and programming sector. Istanbul Hilton Exhibition Centre. November 25th–28th.
Coming up in 1994
AES 96th Convention, Amsterdam, The Netherlands. February 27th–March 2nd.
Broadcast Asia '94, the third Asia-Pacific sound, film and video exhibition and conference will take place 1st–4th June, at the World Trade Centre, Singapore.
AV and Broadcast China '94, covering the broadcasting market in all of China. China Foreign Trade Centre, Guangzhou. May 16th-20th.

Contracts

● Avid and the Manchester bid
Avid Technology delivered a *Media Composer 800*, at very short notice to Stockport-based Vector TV —Manchester Olympic Bid's postproduction house—to cut the final video in support of the British bid for the 2000 Olympic Games.

● Tannoy lend a hand to King Kong
Tannoy found the biggest customer ever for its new *SuperDual* loudspeakers—Universal Studios' 38ft King Kong monster. The deal is part of Universal's strategy of helping their visitors to enjoy total involvement in the movies by using the ultimate realism. The *SuperDual* fits inside King Kong's head to be the monster's voice.

● AMS-Neve find answer at Oasis
Recently rated by producers as one of the top UK postproduction houses, Oasis Television is the first UK company to order AMS-Neve's new *Logic 3* digital mixer. Oasis Television has been assessing audio editing and mixing equipment for several months but as executive director, Tony Cloarec explains, 'Value for money, the integration of digital mixing and editing and the pure power of the system were the persuading factors.'

● You've seen the film now play...
Jurassic Park, the video game which features Dolby Surround Sound. The Dolby system is featured on the Super NES version of the game which was created by Manchester's Ocean Software.

● Frank Zappa turbos his NED
Frank Zappa has installed a significant upgrade to the Synclavier *9600* sampling, synthesis and compositional system he owns. The new package consists of six of the company's new 64Mb MegaRAM sample memory cards and a custom expansion chassis used to house 32 additional playback voices. It makes Zappa's Synclavier the largest, integrated sampling system in the world.

● Plus XXX open new studios
Plus XXX studios in Paris have now opened their £1million three-studio complex. Apart from two Neve and one SSL desk, it is the sole complex in France to be completely equipped with Genelec monitoring.

● Freddie Stars for DDA
Comedian Freddie Starr has bought a DDA *AMR24* console as the centrepiece for his private recording studio in Berkshire, UK. The system works together with an Otari *MTR-90* 24-track recorder.

BBC Radio Two's refurbished Studio 2

Pebble Mill bares all for R2

BBC Radio Two's primary music recording studio in Birmingham has been completely rebuilt by Greenwich-based Recording Architecture.

Pebble Mill studio two was stripped to the bare concrete and totally refurbished under the direction of Senior Audio Supervisor, Tim Green and Recording Architecture's Roger D'Arcy.

Tim Green commented: 'Recording Architecture created a recording space that was flexible and neutral enough to handle everything from music to speech and the other sound needs of BBC Radio.'

APRS contribute to MMC 'Music' inquiry

The APRS has been invited by the Monopolies and Mergers Commission to make a submission to its inquiry into 'All aspects of the supply of recorded music in the UK.'

The MMC had been asked, by the Director General of Fair Trading, to investigate and report the existence, or possible existence, of a monopoly situation in the supply of music recordings of all types; this goes beyond the previous hearing, by the Parliamentary Heritage Committee, which centred on the retail price of compact discs.

Given its place at the heart of the creation and supply of recorded music, the APRS responded to the Commission's invitation with a submission in several parts. These reflect the Association's character as an umbrella organisation for a number of sectors—the commercial recording studio interest, the pressing and duplication (PAD Group) interest, the manufacturers and distributors (Suppliers Group) interest, and the Re-Pro (Guild of Producers and Engineers) interest.

The Association reiterates its serious concern that there is virtually no margin for cutting the rates paid to studios, duplicators, record producers and others who service the creative and manufacturing processes, and that any move to reduce retail prices might well be counterproductive; it also draws the Commission's attention to the public interest aspect of some of the practices of the Musicians Union, and—in the record producers section—to the issue of royalties collected via the PPL.

The APRS submission is now being reviewed by the members of the Monopolies and Mergers Commission, and representatives of the Association will meet the MMC panel on the 30th September. It is understood that the panel is receiving the views of a wide range of music industry bodies, and will report on this inquiry by the end of March 1994.

APRS, 2 Windsor Square, Silver Street, Reading, Berks RG1 2TH. Tel: 0734 756218. Fax: 0734 756216.

Klipsch *KP-260*

The Klipsch Professional *KP-260* is a compact 2-way, trapezoidal loudspeaker designed for solo or cluster installation. Features include traditional Klipsch mainstays such as high efficiency, wide dynamic range and high power handling capability.

Found as standard equipment are 12 fly points, providing a wide range of mounting points. The drivers used in the *KP-260* include a die-cast aluminium 12-inch woofer and a heavy duty compression driver mounted to a 60° x 40° Tactrix Technology horn.

Klipsch Professional Products, PO Box 688, Rt. 4 Oakhaven, Hope, AR 71801, USA. Tel: +1 501 777 6751. Fax: +1 501 777 6753.
Europe: JWM Ltd. Tel: 0637 877170. Fax: 0637 850495.

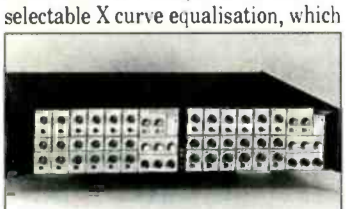
Apogee EQ

Apogee Sound have introduced the *CRQ-12* parametric equaliser, featuring what they call 'MultiMode' operation. The unit has 12 fully parametric filters, each adjustable in three ranges (from 20Hz to 20kHz), four shelving filters, and four adjustable 12db/octave filters, all configurable in three distinct modes of operation. The modes are called 6/6, 6/12 and 12/12, and restructure the internal signal routing to the unit's four output level controls.

The *CRQ-12* has a dynamic range of 115dB and distortion is less than 0.003% at +21dBu.

Apogee have also introduced its new THX Motion Picture Theatre System, *MPTS-1*. Designed by the THX division of LucasArts, the unit is the only THX system specifically intended for postproduction mixdown facilities, dubbing stages, professional screening rooms and small to medium-size cinemas.

Employing a 12-output processor, the unit features selectable screen compensation filters for each of the three main channels, and remote selectable X curve equalisation, which



Apogee's *CRQ-12* EQ

***RED 3*, a dual stereo compressor-limiter**

allows switching between cinema and video playback session.

Apogee Sound Incorporated, 1150 Industrial Ave, Petaluma, CA 94952. Tel: +1 707 778 8887. Fax: +1 707 778 6923.

RED 3 arrives

Focusrite Audio Engineering have introduced *RED 3*, a dual stereo compressor-limiter. The design is derived using the circuit ideas and the proprietary Focusrite VCA from the *ISA 130/131* dynamics processor. *RED 3* maybe switched to stereo operation when only the lower controls function over both channels, ensuring accurately matched stereo performance. In this mode an equal amount of gain reduction is applied according to the larger of the two input signals, essential in mixdown and stereo miking situations.

RED 3 features transformer balanced inputs and outputs which effectively blocks external noise pickup.

Focusrite Audio Engineering, Unit 2, Bourne End Business Centre, Cores End Road, Buckinghamshire, SL8 5AS, UK. Tel: 0628 819456. Fax: 0628 819443.

Stereo for *DMC*

Yamaha have launched the *DMC1000 STEREO* digital console, designed specifically for broadcast, film, video production and mastering applications. Major broadcasters, including the BBC, made it known that they had a requirement for a digital mixing console with dedicated stereo inputs to equip digital video editing suites and audio postproduction suites with hard disk editing systems.

DMC1000 STEREO console represents a significant progession from the stereo mode status on the standard model. Features include the ability of any channel to be configured to function as a single stereo input channel or two mono channels; all 22 inputs on the *DMC1000 STEREO* can be routed simultaneously to any or all of the eight program buses and the main stereo output bus; MS decoding facility on all of the eight main stereo input channels; full dynamic automation of all mixing console parameters includes the MS decoding.

The *DMC1000 STEREO* is available either as a physical console or as a software update package for existing users.

UK: HHB Communications, 73–75 Scrubs Lane, London. NW10 6QU. Tel: 081 960 2144.
USA: Yamaha Corp of America, 6600 Orangethorpe, Buena Park, CA 90620, USA. Tel: +1 714 522 9011. Fax: +1 714 739 2680.

Spectral Tracks

Spectral Synthesis has announced the release of *AutoTracks*, a new software utility for use with their Digital Audio Workstation products. *AutoTracks* permits autorecording and autoconforming of *CMX 340, 340A, 360* or *3600* format compatible source files. Also released is *StudioTracks 2.0*. The *2.0* system is the most powerful ever delivered by Spectral, and adds support for removable media, plus enhancing editing, mixing and patching features. All existing systems in the field are being upgraded at no charge to the owners, as part of Spectral's open upgrade path concept.

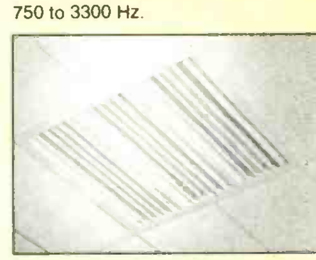
Prisma is a Digital Audio Workstation system which features 96 tracks, 12-channel real-time mixing with dedicated multiband ▶

In-brief

● Master calender display unit
The *ES-195* is a six-digit Master Calender display. The unit is designed to receive ESE serial time code and display six digits of date (Month, Day, Year) or six digits of time (Hour, minute, second). In addition to the front panel display, two different time code outputs (one ASCII and one ESE) are accessible on rear mounted connectors.
ESE Tel: +1 310 322 2136.

● Graham-Patten release video
To help users become more familiar with the operational style of the *D/ESAM 800* and *D/ESAM 400* Digital Edit Suite Audio Mixers, Graham-Patten have prepared a videocassette that demonstrates the various features and functions
G-P Systems. Tel: +1 916 273 8412.

▼ Quadratic Diffusor introduced
The Wenger Corporation has introduced a quadratic diffuser acoustical panel designed to improved sound diffusion in music rehearsal rooms and performance environments. The new panel is a one-dimensional quadratic residue diffuser that effectively diffuses sound in a frequency range of 750 to 3300 Hz.



Wenger's quadratic diffusor panel

● Shure release new mixer
Shure has announced the availability of the *FP32A* portable stereo mixer which is similar to the previous *FP32* model, yet offers over 40 new features and improvements. Self-noise has been reduced by 30dB to make the unit compatible with digital recording formats and transmission schemes; and both 48V and 12V phantom, as well as 12V(A-B) power are available to operate all types of condenser microphones. Active input controls have been added that simultaneously lower an input's volume level and increase its clipping point. The *FP32A* measures 58mm H x 161mm D x 184mm W and weighs only 1.6kg (without batteries.) **Shure GmbH. Tel: +49 7131 83221.**


BBC feedback helped in DMC Stereo's arrival. See Page 13

parametric EQ on each channel, 24-bit DSP, MIDI, SMPTE, MTC and SCSI support—all on a single PC board. The *Prisma*, under the control of Spectral's new *Prismatica* system software, takes advantage of the widely distributed *Windows* user interface.

Spectral Synthesis, 47 Lafayette Circle, Suite 308, Lafayette, CA 94549, USA.
Tel: +1 510 284 8417.
Fax: +1 510 284 8421.
Europe: Spectral Synthesis. Tel: 0442 64205.

B&K Sound Level

Bruel & Kjaer have introduced its new *Type 2236* sound level meter with a revolutionary shape and new features for measurement of noise in the workplace.

The *2236* provides facilities for an accurate assessment of noise levels and can be upgraded by addition of built-in octave filters for frequency analysis of noise sources. It can also act as a low-cost front-end for tape recorders, transfering signals via an unweighted, calibrated AC output for spectrum analysis or subjective evaluation. Internal memory stores results from 40 survey locations and up to six hours of 1-second results. A serial interface is provided for downloading results, or for direct connection to a PC.

Designed specifically for the UK market, the *Type 2236* has built in standard national and international parameters for all industrial noise measurement tasks.

Bruel & Kjær, Harrow Weald Lodge, 92 Uxbridge Road, Harrow, Middlesex. HA3 6BZ.
Tel: 081 954 2366.
Fax: 081 954 9504.

AKG add a rifle

AKG Acoustics has added the *CK 68-ULS* rifle microphone to the *C460 Series*. The mic incorporates two shotgun capsules by using a divisible interference tube.

In full length format the *CK 68-ULS* can be used for medium distance recording applications such as film and television dialogue or outdoor and on stage ambience recording. With the front tube removed the *CK 68* can be used as a short shotgun capsule suitable for recording motion picture, television and video close-ups, interviews in noisy environments, etc.

Features include switchable -10dB pre-attenuation pad and 70Hz or 150Hz 12dB/octave bass cut; ultralinear frequency response and electrical transfer characteristics; minimal distortion, low current consumption and low noise performance.

AKG have also introduced the *DSM-7* (Digital Status Monitor), a portable unit for checking all significant parameters of digital audio connections to AES-EBU and S-PDIF standards. In situations where connectors, digital standards and sampling rates vary from each other, and where cable attenuation, signal loss and confusing harnesses create further difficulty, the *DSM-7* promises to save downtime in studios, broadcast and live applications.

The unit comes complete with AKG headphones, battery pack and a carrying case.

AKG Acoustics, Brunhildegasse 1, Postfach 584, 1150 Wien, Austria. Tel: +43 1 956 51 72 42. Fax: +43 1 956 51 72 45.
UK: AKG Acoustics, Vienna Court, Lammas Road, Godalming, Surrey. GU7 1JG. Tel: 0483 425702. Fax: 0483 428967.

ScreenSound v5

SSL's v5 software for the *ScreenSound* combines a faster processor and an extended range of operational features. Jog and shuttle on the desk, Edit peel, Global edits, Autoconform and reconform of audio are some of the new additions. Also offered is an integral random access video option in the form of *VisionTrack*, providing up to one hour of random access video, with instant location of audio and picture to any frame. Reusable hard disk storage has none of the costs associated with optical media. Other features include, simultaneous sound and picture recording; ADR countdown overlays and integral machine control.

Solid State Logic, Begbroke, Oxford, OX5 1RU. Tel: 0865 842300. Fax: 0865 842118.
US: Solid State Logic Inc, 320 West 46th Street, 2nd Floor, New York, NY 10036. Tel: +1 212 315 1111. Fax: +1 212 315 0251.

3M in two format CD-R release

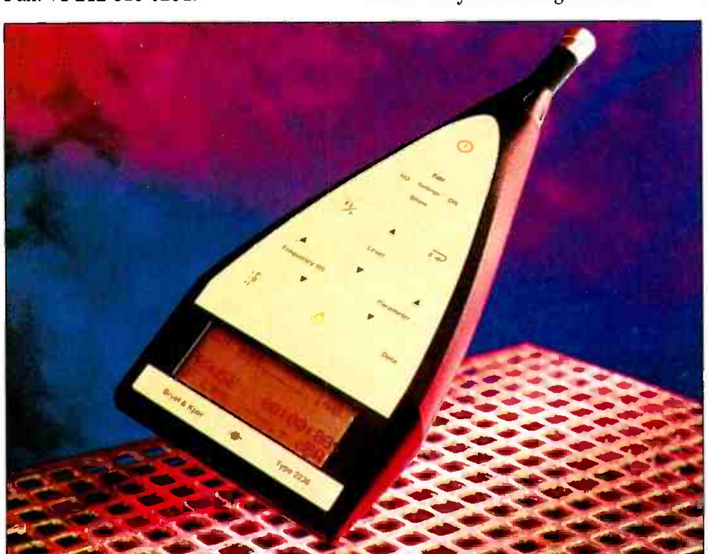
3M have announced the availability of 3M's new 63 and 74-minute format CD Recordable media. Because the two formats fully comply with the Orange Bool Part II standards, the discs will work with most manufacturer's CD Recordable drives. The discs are colour coded for easy recognition by users.

'The new-line of media demonstrates 3M's increased committment to the future of the CD-ROM industry. The growth and increased accessibility to CD-R has vast implications for application developers and information publishers, who want to prototype their CD-ROMs, for business with low-volume distribution applications, and, of course, for governmental, library and other uses that store massive quanities of data,' says Rusty Rosenberger, 3M Business Development Manager.

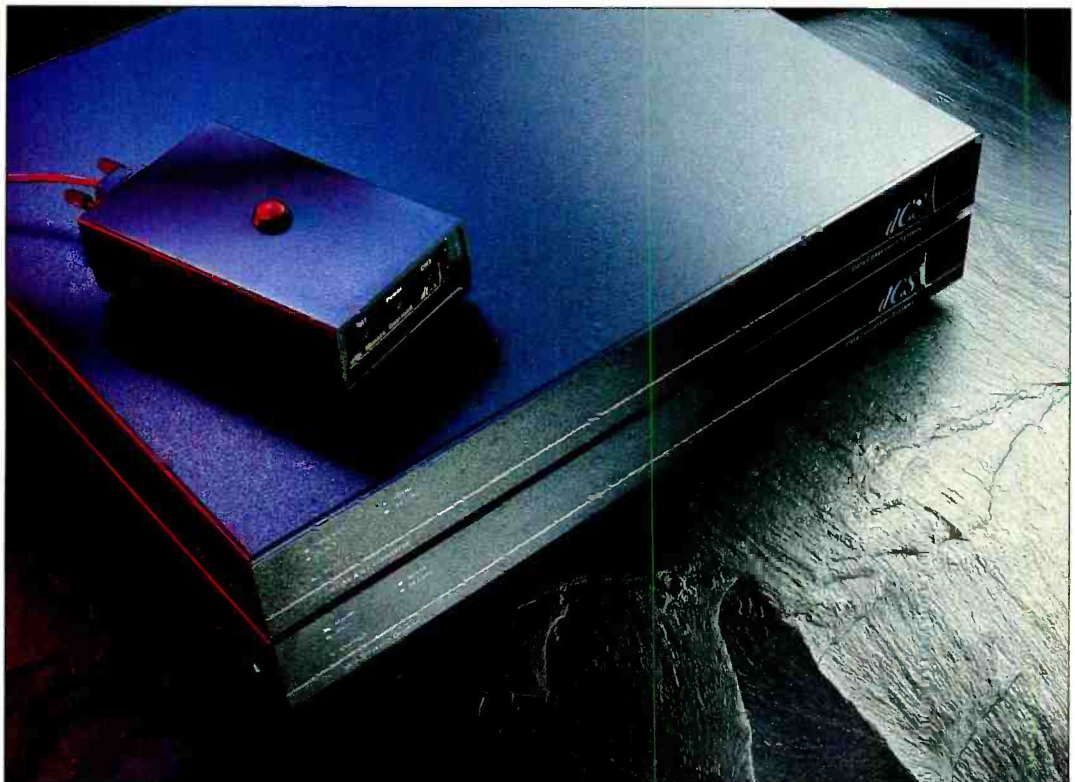
3M have also announced availability of a 5.25-inch 1.3 Gb optical disc for use on new double-density optical drives.

Working under codevelopment agreements with Hewlett-Packard and IBM AdStar, 3M have doubled the rewriteable optical disc capacity from 650Mb to 1.3Gb.

The new disc doubles storage capacity primarily through the banded-format technique to optimise the number of recording sectors per track. Capacity gains also were achieved by increasing the linear ▶



B&K revolutionise sound level meters



dCS claim first convertor to provide 88.2 and 96kHz sample frequencies

recording density while decreasing track spacing.

The new drive and media technology together offer users a 40% reduction in cost per-megabyte and a 30% improvement in the read-write transfer rate.

3M are the world's largest supplier of removeable magnetic recording media and is a leading producer of magneto-optical discs.

3M Professional AV Products, 3M Ctr, St Paul, MN 55144-1000, USA. Tel: +1 612 733 3477.
UK: 3M Data Storage Products Group, 3M United Kingdom PLC, 3M House, PO Box 1, Bracknell, RG12 1JU. Tel: 0344 858447.

dCS deliver 96 kHz

dCS have launched the *dcs 902* high speed A–D convertor and they claim its the first outboard convertor of its type to provide 88.2 and 96 kHz sample frequencies. Based on the architecture of the *dcs 900B*, conversion is achieved by a discrete dcs proprietary oversampling convertor. Gain ranging techniques are not used, avoiding unwanted effects due to noise pumping.

The convertor may be operated in either Master or Slave mode and offers selectable output word width, remote overload monitoring and double speed AES-EBU, SPDIF and SPDIF-2 digital interfaces.

Data Conversion Systems, The Jeffreys Building, St John's Innovation Park, Cowley Road, Cambridge. CB4 4WS, UK. Tel: 0223 423299. Fax: 0223 423299.

Digital Audio News

FOR THE LATEST IN DIGITAL AUDIO INFORMATION

WORLDWIDE DAT RECORDER SHORTAGE-HHB TO THE RESCUE!

In the 6 years since its launch, DAT has become established as the undisputed industry standard for two track digital recording with a wide variety of professional machines now available.

But with the major manufacturers replacing the domestic models in their DAT ranges with sonically inferior DCC or Minidisc devices, supplies of the type of suitably featured consumer DAT machines favoured for use by audio professionals have become scarce. But fear not!

As Europe's leading supplier of DAT technology, HHB Communications is well placed to scour the globe in search of the last of the "prosumer" DAT machines and ensure their continued availability - for the time being at least.

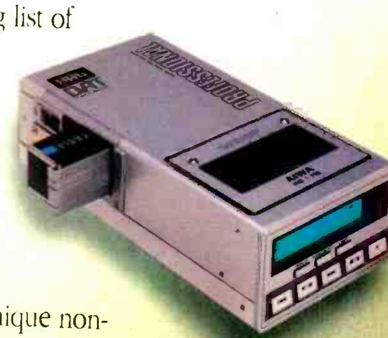
HHB SECURES EXCLUSIVE SUPPLY OF THE AIWA XD-S1100!

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

BEWARE - ALL DAT TAPES ARE NOT THE SAME

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

AIWA HHB 1 PRO: THE POWER PORTABLE



An uncompromising list of features including dry cell and rechargable battery power, balanced mic / line inputs and AES/EBU digital I/O (plus a unique non-SCMS copy prohibit free SPDIF digital I/O) make the HHB1 Pro the portable of choice.

HHB

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

CEDAR CR1

Launched at this year's APRS, the *CR1 Stereo DeCrackler* joins the *DC1 DeClicker* in a new departure for CEDAR—the production of stand-alone rackmount units containing the essential modules of their established audio restoration system. These two units (physically almost identical), can be used individually, as a pair, or in conjunction with a full-blown CEDAR system for additional processing power, the combination required depending on the job in hand.

CEDAR have identified various distinct tasks for which their systems have been used, and not surprisingly there are applications which predominantly require one particular module of the software and make less (or no) use of the rest of the system. CEDAR therefore believe there is a market for the individual modules in a convenient, easy-to-use format for those whose principal work does not require the purchase or learning curve of the complete system.

Declicking has always been a fundamental part of the CEDAR process, quite distinct from the removal of steady-state noise, and it comes as no surprise to find that other intrusions such as buzzes, hums, distortion and general crackle are treated separately with a variation on the process. The kind of faults targeted by the *DeCrackling* software include thyristor buzz and other stray mains-induced noises, certain types of distortion—the spec cites overloaded analogue inputs, overdriven microphones and digital clipping—and the degree of scratchy noise found on a vinyl LP. This last should be of interest to the CD, DCC and MD remastering market in cases where no playable original tape exists.

I approached this review with a DAT full of transcriptions from 78s, and it will now be clear that this was not ideal test material for the *CR1* alone; however, the addition of a *DC1* to remove the major clicks first allowed the *CR1* to tackle the remaining crackles. (CEDAR have a tape of samples of actual jobs they have undertaken which can be run through the CR1 to show what it can do.) The biggest initial surprise was the simplicity of the unit, and how little adjustment is needed to produce the required results. I was expecting some considerable complexity, with page upon page of options to be set and parameters to adjust, but in fact there are only three pages on the unit's clear, bright display, of which only one is needed for this operation.

The basic principle of the CEDAR system involves the splitting of the signal into two parts, one containing the offending noises and the other containing the unaffected signal. The work is then done on the first part before recombining it with the second to produce the final result. The first job, therefore, is to establish the split between these two parts, and this is done with the Detect Level controls while monitoring the part which will not be processed. When the monitored signal contains nothing but wanted signal (albeit sounding rather odd at this stage) then clearly all the crackle is being sent to the processor.

All that then remains to be done is the setting of the Threshold parameter, which determines the point at which the unit will distinguish between wanted and unwanted signals. This again is best done by ear, although in both cases clear bargraph meters and direct read-outs show the chosen values. Two decrackling algorithms are provided, with Crackle2 more extreme than Crackle1, and it is easy to switch between them to hear which is more appropriate.

That is all there is to it—two simple adjustments—and the results are quite dramatic. My own test material, badly damaged though it was, came through with all the surface mush completely removed, leaving only the steady hiss of the (surprisingly low) noise floor of the original recordings. The wanted signal, however, survived intact, with no apparent ill effects whatsoever. CEDAR's own test material, with much subtler faults, was dealt with even more easily, and it was uncanny to hear slightly peaky distortion and constantly varying mains harmonics disappear completely without side effects.

Another surprise was how difficult it is to misadjust the unit, even deliberately. At one extreme it simply does nothing, while at the other it starts to eat into the ambient information—reverberation, background noise and other low-level signals start to pump and breathe slightly. At worst, transients can be damaged—a hi-hat loses its sparkle, for instance—but this is so easily avoided by correct setting that there is no excuse for any significant degradation.

The *CR1* supports both AES-EBU (up to 24 bits) and SPDIF digital I-Os, complete with a digital output attenuator in case the internal 40-bit processing produces higher levels than were present at the input. Balanced and unbalanced analogue I-Os are also provided. MIDI and RS232 remote control facilities are incorporated, and SMPTE sockets anticipate future control upgrades.

Even though the *CR1* costs nothing like as much as a full-blown CEDAR system, you will have to be pretty serious about restoration to afford one; those who are, however, need be in no doubt that it will do the job. The whole CEDAR process seems to be chiefly associated in most people's minds with the problems of archive material, and it is interesting to see its potential for other applications, such as rescuing hum-plagued live recordings, salvaging that one magic take where something went briefly into clipping—problems most facilities will encounter from time to time which are virtually insoluble by conventional means.

CEDAR have suffered in the past from problems with their image, but this appears to be a thing of the past. Certainly products like this, delivering the goods in a simple, fast, unfussy way, should win many friends. ■

Dave Foister

UK: HHB Communications Ltd, 73–75 Scrubs Lane, London NW10 6QU. Tel: 081 960 2144. Fax: 081 960 1160.
US: Independent Audio, 295 Forest Avenue, Suite 121, Portland, Maine. 0401-2000. Tel: +1 207 773 2424. Fax: +1 207 773 2422.

Cedar *CR1*—difficult not to use well

Catching Code

Over the years, the record industry has consistently been on the look-out for a magic bullet to stop all pirating at a stroke. So far, however, no-one has been able to come up with a system that could be carried within the audio band—not be heard and not be easily defeated with simple filtering. The adoption of SCMS as practical anticopying format has been widely accepted by the record industry and while it does not affect the audio performance in any way, it works in the digital domain, and then with only limited effectiveness.

There is a growing realisation that a truly effective copy protection system sets almost self-contradictory technical requirements and is unlikely to be designed while music can be copied in the analogue domain. Consequently, the industry is turning its attention to adding coding to music tracks so that they can at least be effectively 'watermarked' with ownership details, This particular ball has been set rolling with the International Standard Recording Code which places a unique code which identifies the track into the digital sub-code.

Like SCMS, this is a purely digital implementation, and while it can be extracted when the CD is being played the information is lost the moment the audio is converted into an analogue signal.

Now the Central Research Labs —which was formally EMI's research division (responsible in the 1930s for such developments as stereo and the 405 line TV system)—have developed an effective way of placing this code within the audio track itself so that music can be automatically tracked no matter whether it is being monitored directly from a digital source or from analogue cassette, or off radio, satellite, or TV transmission.

This latter option could be used to fully computerise the logging of music delivered via electronic delivery systems or played on the radio. The information can be used for royalty payments and for automatically collecting marketing information —checking out who is playing which tracks when, to support the record company's sales effort.

The ICE (Identification Coding, Embedded) system developed by CRL takes the track identification data and splits it into data bursts of around a second duration each which are then recorded directly into the audio. Insertion is done by first cutting out a narrow frequency notch in the audio in the 2kHz area and then adding the data as an audio signal. The actual frequency of the insertion point is regularly altered making it difficult to strip the data out using a simple narrow filter. To read the data, a suitably equipped personal computer is fed the analogue audio from which it extracts the data. The data is read from several bursts which are compared to avoid erroneous readings.

The data code extracted from the track can then be compared to codes stored in a database, and the relevant song information displayed and stored to file for later analysis.

A notch system such as Copycode has previously been ruled out because of the sonic degradation created by the analogue filtering. According to Dr Nigel Johnson, Manager of CRL's Signal Processing Division, the organisation is taking the audibility of ICE very seriously. 'We had specific aims for ICE —ideally it should be able to survive over FM-AM radio, satellite and cassette as well over digital formats. It should be secure and the encoding and decoding systems should be simple to operate. But the list is headed by inaudibility—if it does not satisfy that basic requirement then the system falls down completely.'

CRL's approach deals with the audibility problems in two ways. Since the filtering can now be accomplished in the digital domain, the sonic problems associated with steep analogue filters can be overcome. But to ensure the data bursts cannot be heard, the ICE encoding system constantly evaluates the masking potential of the audio and only inserts the data as a when the potential for masking goes above a preset value.

Rather than try to persuade the industry adopt the system and then offer it for sound quality testing by 'golden ears' groups, CRL have been quietly distributing test CDs containing tracks with and without ICE data to assess the audibility of the system.

The first CD was produced using an analogue prototype which proved successful and led to the development of a fully digital version of ICE and the test CDs for the digital version displayed marginal audibility with two respondents being able to differentiate between the ICE'd and clean versions. Further investigation has highlighted the need for some software changes which are currently being implemented prior to the production of a third assessment disc.

Even if expert consumers are satisfied that ICE functions inaudibly, CRL will need ISRC to be much more widely adopted if it is to work effectively as a track logging system—unless a high proportion of the tracks have ISRC codes in the first place for music general music logging ICE will have limited use.

According to Pete Rogers of the PPL, the UK royalties collection agency, widespread adoption of ISRC may still be few years away.

'There are some practical problems with ISRC; for example the coding for back catalogue happens when the music is first transferred to digital form which may not happen in the original country. The communication throughout these large international companies has slowed its adoption down. This irony is that the companies at the leading edge of putting the ISRC codes in are in fact the smaller independents.'

Although PPL are already receiving computerised royalty data created automatically by radio stations Rogers believes that the need for powerful data collection systems will become important in the future, especially if the way music is distributed moves more towards purely electronic methods.

'The ISRC gives detail at a much greater level and there will be more need with electronic delivery— it is important to artists and it may be important when things are far more sophisticated in marketing terms.'

In the meantime CRL are looking at other applications for the system such as marking advertising for playout logging—applications that use customer-specific codes rather than waiting for ISRC to become universally adopted before the system can be adopted in the marketplace. ■

Tim Frost

Central Research Laboratories, Dawley Road, Hayes, Middx. UB3 1HH. Tel: 081 848 9779.

Roland *SRV-330* and *SDE-330*

Roland's *RSS* 3-D 2-speaker surround processing system has made a curious impression on the business, with most people being unsure how to react to it. Understandable scepticism about what it is supposed to do is not helped by its price—which precludes most of us buying a system on spec, and means there are not many around to evaluate.

Fortunately, that chance now presents itself in the form of two digital effects processors, the *SRV-330 Dimensional Space Reverb* and the *SDE-330 Dimensional Space Delay*, both of which incorporate elements of RSS technology to add depth to their effects. Perhaps surprisingly, a considerable degree of control is provided over the '3-D' elements of the various algorithms, allowing freedom for experiment.

The best access to the raw 3-D facilities is provided by the delay unit, the *SDE-330*. For instance, the Simple 3-D Delay algorithm has effectively a scaled-down single-channel RSS processor on its output, allowing full 180° control of the horizontal position (azimuth) and ±60° of vertical positioning (elevation). This makes it easy to become familiar with the possibilities, which, as anyone who has heard a Roland *RSS* demo will know, can be quite spectacular.

Horizontal movement is very impressive, with sounds easily placed outside the loudspeakers and starting to move round towards the back of the listening room. Images become very difficult to locate as the rear is approached, but if movement is sustained the illusion of rotation through the full circle is both convincing and startling. A certain amount of willingness to believe is required, and the best way to destroy the illusion is to turn your head towards the apparent source of the sound, but if you are prepared to sit back and enjoy it the effect works every time.

It is less easy to be persuaded by the vertical motion, particularly when the sound is located centre-front, where all the elevation control seems to do is alter the EQ in a 'phasy' sort of way. Put the sound way out to one side, however, and it can apparently be made to swing up in an arc over your head or down into the floor, albeit with the strange tonal alterations still evident.

Once this simple setup is mastered, the more complex algorithms become self-explanatory. Almost every algorithm on both units has two versions, with and without the 3-D prefix, and again the fullest access to 3-D control is provided by the Delay unit. This offers, for instance, a stereo delay where each channel has its own azimuth and elevation controls, an 8-tap delay with independent azimuth for each tap, and a pitch shift program with four independent delayed pitch shifters, each with azimuth and elevation adjustment.

Used sparingly, the impact of these setups can be very dramatic. For instance, a single word repeated once over your left shoulder and again over your right can really make you jump. Too much of this kind of thing can get a bit fatiguing, however, and once the novelty wears off a more subtle approach can show the true power of the system, adding depth and front-back perspective to familiar effects. At this point the *SRV-330* perhaps becomes more interesting, since the 3-D effect, while less obvious, brings real depth to the reverb programs, allowing the various halls, churches and rooms to acquire a life and space of their own rather than sitting in a line between the speakers.

Most of the algorithms have only an 'amount' control for the 3-D effects, unlike the comprehensive adjustments on the *SDE-330*, and the 3-D effects are by and large confined to the early reflection part of the program, but this is enough to give the required illusion, adding a significant extra dimension.

None of this, of course, would be of much use if the basic facilities were not up to scratch. Lest anyone assume that these units are little more than flyers for RSS, it should be made clear that even without the 3-D effects both boxes would stand up well against the competition. The reverbs are good, with simulations convincingly natural, and deliberate 'effects' powerful and flexible, and the selection of delay-chorus-flange-type algorithms covers all the possibilities. The range is illustrated by the excellent and varied factory presets, of which the *SDE* has 100 and the *SRV* a mind-boggling 300, with plenty of additional storage space for user programs.

The controls are novel and very intuitive in use. The Edit mode displays three parameters at a time, any one of which can be selected with a soft key and adjusted with the data entry wheel. The parameters scroll sideways through the window, making it much faster to find the adjustment you want than on many other systems, and in case you do not like the order in which they appear in the window, even that is user-definable.

I liked these units very much indeed. Bearing in mind the price of a full-blown *RSS* system, I expected a watered-down, factory-set imitation tacked on to some fairly basic effects, and was pleased to be proved completely wrong on both counts. Between them the *SRV-330* and the *SDE-330* provide everything you are likely to need in the way of reverb and delay, and much more besides —and behind, above and below, come to that. ■

Dave Foister

UK: Roland (UK) Ltd., Rye Close, Ancells Business Park, Fleet, Hampshire GU13 8UY. Tel: 0252 816181. Fax: 0252 812692.
US: Roland Corporation, 5759 Uplander Way, Culver City, CA 90230. Tel: +1 310 338 9974. Fax: +1 310 338 9973.

Roland's new signal processors offer to bring *RSS* into every studio

GS3V

The ultimate mix
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Killer Horns

One of the biggest problems encountered on approaching a sample CD of any type is what you would expect to find on it—and it is particularly true of a CD of horns. Single-note chromatic blasts, fine; stabs and shots, yes; but surely the beauty of a horn sample set ought to be its focus on signatures, riffs and all those other things that no amount of flattery, trickery and technique will allow you recreate convincingly in any other way. The point at which you say, 'I think we should get some real horn players in,' is the point at which such a CD should step in.

The problem with such an assembly is that most perceptions of what such a collection should constitute varies wildly beyond the realms of cliché and obvious, and my search for an out-and-out classic soul riff on *Albie Donnelly's Killer Horns* CD were in vain. But I found a lot else of interest.

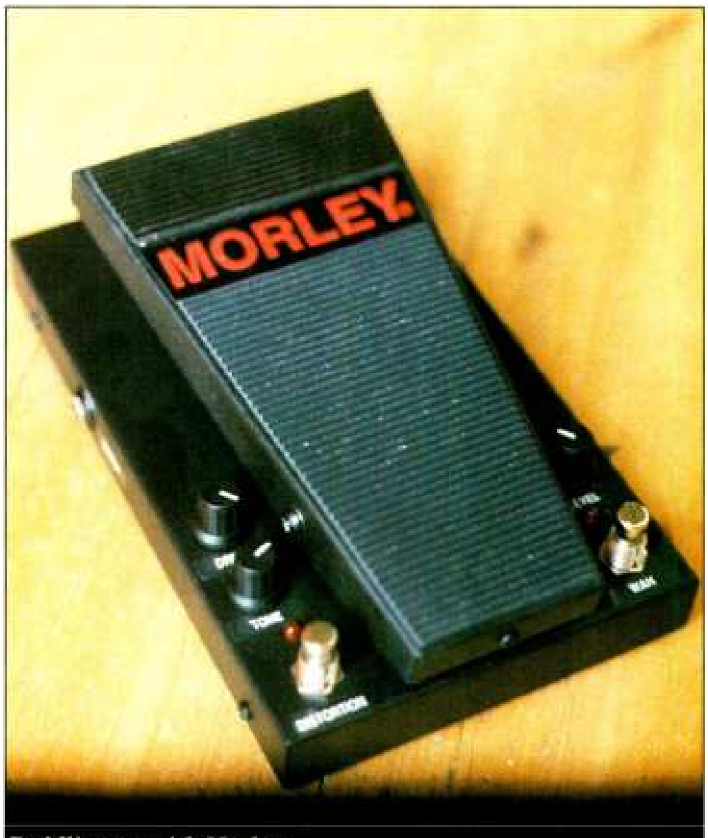
By definition it is a hard sample set to put together because it is wide territory to cover. The horn section sampled here comprises Donnelly on tenor and alto sax, Dick Hanson on Trumpet and Flugelhorn, Steve Crane on Trombone and Paul Owens on baritone sax. Between them and together they have covered the land from Georgie Fame to The Clash.

Frankly I was surprised by the amount of swing jazz (verging on Dixieland in places) on the disc not to mention some stunning pastiches of Mexican 'The bandits have raided the village so play some trumpet as the good guy rides in slowly to save everyone' styles.

Listings are comprehensive, giving keys, tempo, and a brief indication of the instruments involved. The CD kicks off with a stack of brass sections, followed by solo studies on trumpet-trombone and saxophones including chromatics. Finally, there are the inevitable 'far outs' including some techno bass riffs with real-time swept boost EQ Pass. Looping a goodly proportion of this stuff for a sustain envelope is pretty much out of the question but the variety of material and interrelation of keys means that the riffs are very usable for beat loops.

If it is stabs you want then you got them here in many different styles and textures and it is worth mentioning that Donnelly and the gang have not just gone for hernia-inducing power horns throughout but have tempered it with a wodge of mellow material that really expands the applicability of the collection. In addition to the audio CD version of *Killer Horns* there is also available a CD-ROM for Akai, *Emulator III* and Ensoniq *EPS 16+* and *ASR*. The quality is good—you can hear mechanical noise and breath at times—with a defined stereo image and a consistent instrument balance. However, there are audio artifacts in places, clicks in others, there is a bit too much stereo movement on some of the solo instruments, some of the endings are far too abrupt and gating seems to have been used between phrases at times. But the ambience is pleasant, the skill and delivery of the players is beyond doubt and the variety of styles is extremely wide.
I like *Albie Donnelly's Killer Horns*; it is usable, convincing and adaptable but just do not expect it to have everything you need.
UK: Time and Space, PO Box 306, Berkhampstead, Herts HP4 3EP. Tel: 0442 870681.

Morley



Peddling on with Morley

Here is a blast from the past. One look at a Morley pedal and the trousers flare and the soles thicken. My soles certainly thickened after only brief use of the Swell-Fuzz-Wah as the sticky-backed nonslip rubber on the pedal top dislodged and attached itself doggedly to just about everything but the pedal, and the Morley name plate, looked set to follow quickly. However, I detected only a minute increase in trouser flare in the same period of use.

But a Morley pedal without chrome? Well, this is the new model which, while it harks back to the principle of using the electro-optical system of some 25 years ago for interpreting pedal position ago now uses an LED for the light source rather than the small light bulb of old. Needless to say it has done wonders for the reliability. Four years ago Morley were bought by the reverb spring people Accutronics—now called Sound Enhancements—and their product range has improved and expanded as a consequence.

Operation is simple. To the left of the pedal are Drive and Tone pots and a footswitch with LED for Distortion. On the right is a level control for the distortion and a footswitch with LED for the Wah. When Wah is not selected the pedal acts as a Swell. In honesty the footswitches will be a too close to the base of the pedal for the size-10 boot and beyond the operational skills of the in-a-hurry brigade. But the pedal exhibits all the smoothness that Morley's reputation was founded on, because you are not pushing a cog, and because the pedal's response is not all bunched up at one end. You can also turn the pots with the side of your foot.

The distortion sound is a tad fizzy at both extremes of the Drive setting but, as it is unlikely to be used in isolation, a bit of help from the amp smooths it out. The Tone is harsh when wound up on full Drive, but is surprisingly responsive on milder distortion settings—it centres on 1.5kHz and boosts bass, and cuts top to the left and down to the right. The box is capable of a very useful 'graunch' tone with a spectacular glassiness being introduced on the Tone pot. I tried to convince myself that the basic lowest Drive setting sound was not that good, and it is not with the tone down, but open it up and it is not half bad.

The Wah is superb. Wonderful on a clean signal and totally preposterous on something hot. It is noticeably sharper and broader in character than comparable units—there is more 'vowel' to the sound and it is more expressive due in part to the excellent pedal feel and spread. I would not want to say how in line this is with the old Morleys because I honestly cannot remember, but it sort of sounds familiar.

Wahs are presently enjoying a rejuvenation in interest, and a couple of minutes with this box and the reason becomes apparent—it is so much fun. An able Swell-Fuzz-Wah is just so performance oriented and the characteristic honk you can get from setting and leaving the Wah still takes some beating in my book.
I think I would be happier to walk on stage and see one of these on the end of the lead than most multieffects units.

The Morley pedal is not cheap but as they say, 'A Morley is for life—and not just Christmas.'
Sound Enhancements, 185 Detroit Street Cary, IL60013, USA. Tel: +1 708 639 4646.
UK: Radius International, PO Box 3, Basingstoke RG24 9QA. Tel: 0256 477 222.■

Music News is compiled by Zenon Schoepe

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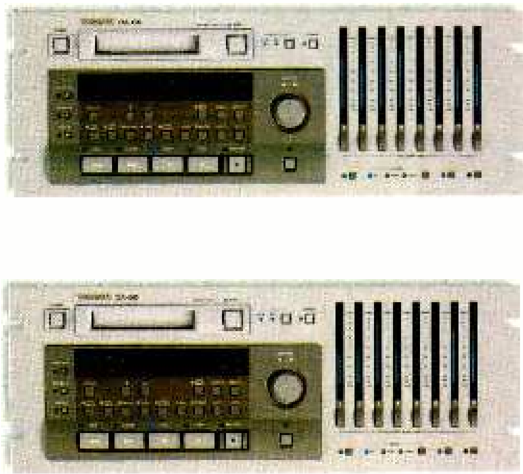

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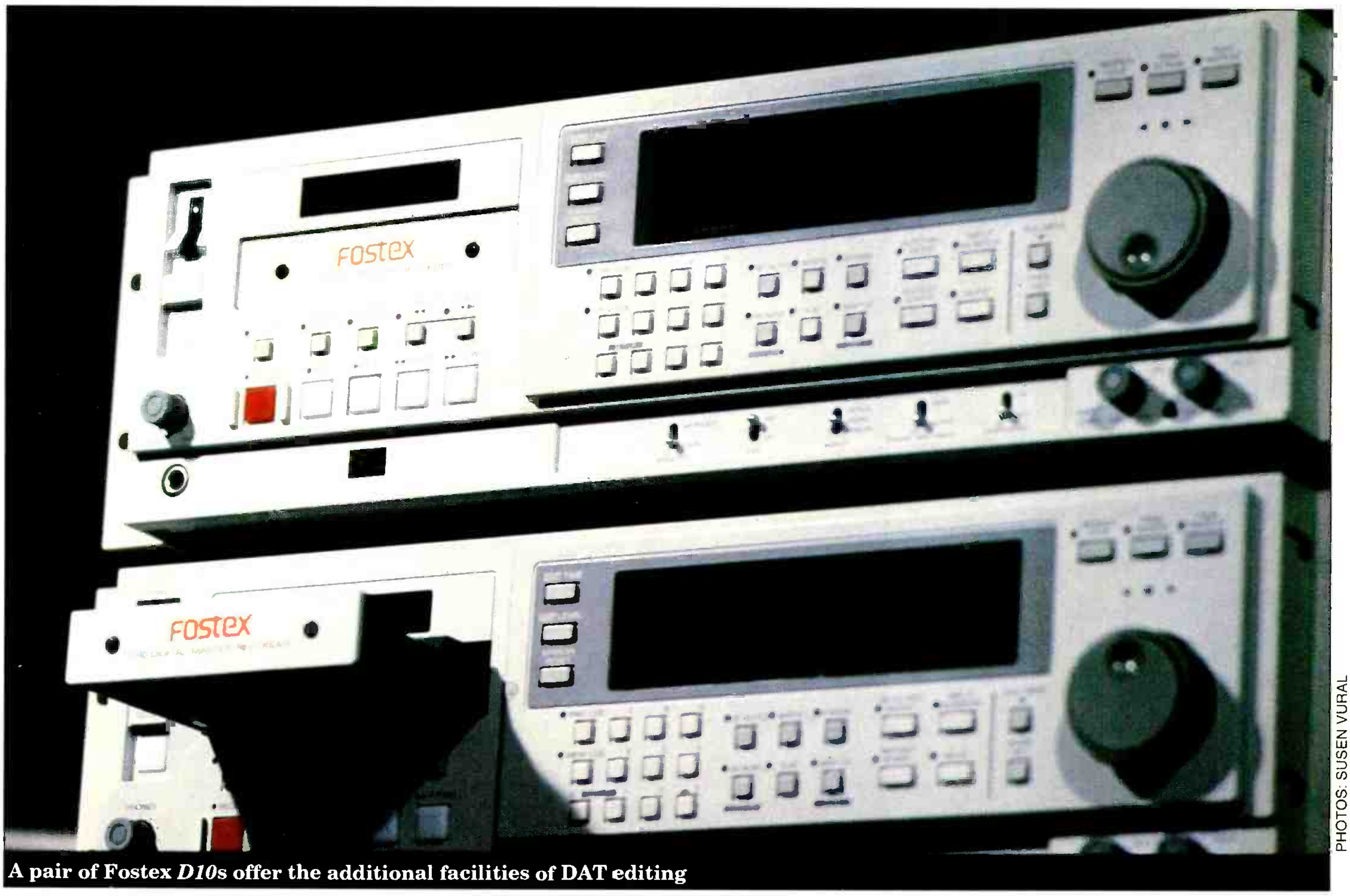
THE INTEGRATION GAME



PHOTOS: SUSEN VURAL

A pair of Fostex *D10*s offer the additional facilities of DAT editing

Jim Betteridge looks at the latest Fostex DAT recorder and discovers the basis of a cost-effective DAT editing system

The Fostex *D10* joins the company's *D20B* and *PD2* portable as the third member of the Fostex professional DAT range. Despite the implication of the model numbering, the *D10* is far from being a lower cost version of the *D20*—though it has only two heads and no time code record or sync capability, it does have a range of other facilities that set it apart from the rest of DAT-kind and put it firmly in a class of its own. These facilities centre around five seconds of onboard RAM and a wealth of locate, cue and instant start functions plus a unique ability to link with a second *D10* to allow accurate digital editing. This link is achieved using no more extra hardware than a standard MIDI lead and a method akin to 2-machine video editing.

Like the *D20*, the D10 is a 3U-high rackmounted machine. A glimpse of the rear panel reveals analogue inputs provided at +4dB (electronically balanced XLRs) and at -10dB (unbalanced phonos). An adjacent switch determines which are active. AES-EBU digital in-outs are on XLRs and there is also an optical digital input and output complying with the IEC domestic standard.

The *D10* comes with a wireless remote capable of controlling two machines and featuring most of the controls you would need to find and play preprogrammed tapes. Any serious work, including editing, requires use of the front panel. This itself is quite different from that of the *D20*. Instead of pushing the DAT tape into an open slot, VCR-style, a button marked OPEN-CLOSE brings out a cassette tray into which the tape is placed; pressing the button again causes the tray to close. The combination of a transparent panel above the tray and internal lights allows the DAT label to be read when the tray is closed—very handy.

The transport keys are reasonably chunky and are back-lit. Their operation is identical to that of the *D20*'s; STOP toggles between Stop (constantly lit) and Pause (flashing); the FFWD and REW buttons toggle between five times play speed (PLAY and WIND buttons lit) and Fast Wind—up to an impressive 250 times play speed. Tape handling, however, is very gentle—not only is it ramped carefully to and from maximum wind speed, it also automatically ramps down when approaching the end of a tape. Hitting STOP, however, brings it to an almost immediate halt.

Above the WIND buttons are the ID SEARCH buttons—forward and backward. These are incremental and ▶

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cumulative—hit one once and it will go to the next ID; hit it four times and it will proceed to the fourth ID. Alternatively, you can punch in an ID number using the 10-key pad and press the PNO LOC button to locate directly to it.

Unlike the majority of other machines, the *D10* (like the *D20* and *PD2*) supports all 799 ID numbers allowed for in the DAT spec rather than the usual 99. Clearly, this could be very useful when storing lots of short recordings such as effects or stings-jingles and so on. It is also worth noting that it does record an ID and a PNO (program number)—earlier models (including Sony's *DTC1000*) record only an ID and derive the number by counting incrementally from the top of the tape.

On display

The *D10* display is larger and more comprehensive than that of the *D20*. A button marked DISP TIME allows stepping through three time-display formats: A-Time (unusually this shows DAT frames as well as hours, minutes and seconds); R-Time (the time element of IEC standard time code); Date Pack—this gives a continuous record in years, months, days, hours and seconds of when the recording was made. This is part of the standard DAT format, although few machines currently have it implemented.

In addition to two large bargraphs (28-segment, fluorescent tube), the metering has an associated numeric display. This can be switched to show the current level of either channel one or channel two, or to hold the highest peak level registered after pushing the RESET button (shown as available headroom below 0dB), or to show PCM error rate as a percentage.

Independently of the 799 ID points the *D10* offers 99 memory locations. A location can be memorised in a one of two ways. You can punch in a memory number (0-99) via the 10-key pad and then hit the MARK-SET key (on the fly or when stationary); or you can key in memory number, followed by the desired time value, followed by hitting the MARK-SET button. Similarly, you can locate to a memory point by either entering the memory number and hitting MEM LOC or you can punch RCL MEM, key in the relevant time value and hit MEM LOC. These memory functions will relate to either A-Time or R-Time, depending on which is displayed at the time.

Reinventing the wheel

The wheel on the *D10* has two parts: the outer ring which operates in the style of a shuttle wheel (the further you turn it the faster the replay speed) and the inner part which works in a video-style jog mode. For approximate location of a position the wheel can be used in the Search Cue mode to actually move the tape against the heads, producing a low quality, interpolated audio. This allows shuttling from ½–10 times play speed, and lets you find a given point quickly, to an accuracy of a few frames. This replaces the Cue mode on the *D20* in which the tape can be wound at 5 times speed with the tape against the heads to allow approximate cueing up.

For more accurate work, the onboard RAM is used. In this mode the outer control shuttle operates to produce a normal analogue-style scrub where the pitch varies with speed. The central dial (jog) operates in the style of what Fostex are calling 'Digital Scrub'. The combination of these modes is apparently a unique Fostex invention. Although intended for the same purpose of accurately locating an edit point, this 'Digital Scrub' or Jog mode is quite different from the usual type which generally seeks to emulate the scrubbing action of rocking an analogue tape against a replay head. Instead it repeatedly loops a single DAT frame around the current position, producing a rather unpleasant digital chattering. Though initially unsettling, a few minutes of experimentation shows that it is in fact a very accurate and quick method of finding an edit point. One advantage is that you hear the programme at its real pitch making it easier to identify a particular key sound. Also, with scrub editing, the tape (real or virtual) always has to be moving to hear anything, and so you are always a jog away from where you want to be. If you are in Play mode and you hit the STOP button, you instantly have five seconds of audio in RAM up to the point at which you stopped the tape; there is no need to go through any loading process. This is because, unlike some other machines, when the *D10* is in Play it is continually updating the RAM—very slick. At this point you can shuttle the five seconds at 0.1–1 times play speed or digitally jog it down to 10ms (or less) accuracy—and now the audio has a full 20kHz bandwidth. Hitting the RAM SCRUB button while in Play stops the tape transport and gives you 2½ seconds of audio either side of the entry point with the same shuttle and jog facilities. If you get to the end of the RAM and want to go further, pressing the RAM SCRUB button again at any point reloads it from tape and recentres it at that point.

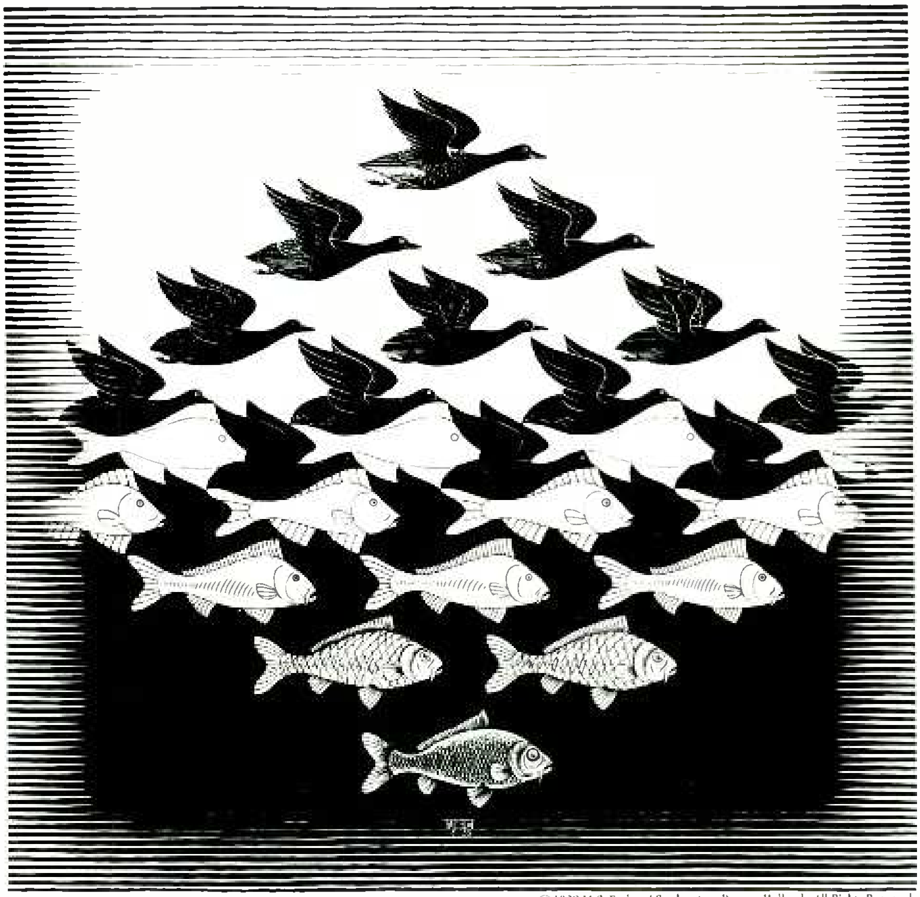
Instant start

The *D10*'s Instant Start facility plays the head of the programme from RAM while the tape gets up to speed, thereby eliminating any lag between pressing the button and hearing the sound. It too is smarter than the average memory start in that, once INSTANT START is pressed, the RAM is always kept updated when in Play mode. Thus, wherever you stop or cue up, instant start is available. Of course, if you wind to a new location or load a new tape, the RAM has to reload. This it does in real time, automatically, and its associated LED will blink until it is ready. A button marked RAM REPEAT automatically loops the first 1½ seconds from the instant start point forward. As you listen, the inner wheel allows you to adjust the start point with 1ms accuracy until it sounds right. Once you have cued, another button marked PREVIEW allows you to hear what the playback is going to sound like when you hit the PLAY button.

The writing and use of IDs is another area in which the *D10* shines. The auto-ID facility writes a new ID whenever the signal level drops below a ▶

A button marked RAM REPEAT automatically loops the first 1½ seconds from the instant start point forward. As you listen, the inner wheel allows you to adjust the start point with 1ms accuracy until it sounds right

If you think only your eyes can play tricks on you...



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Study the illustration. Are the geese becoming fish, the fish becoming geese, or perhaps both? Seasoned recording engineers will agree that your eyes *and* your ears can play tricks on you. In the studio, sometimes what you think you hear isn't there. Other times, things you don't hear at all end up on tape. And the longer you spend listening, the more likely these aural illusions will occur.

The most critical listening devices in your studio are your own ears. They evaluate the sounds that are the basis of your work, your art. If your ears are deceived, your work may fall short of its full potential. You must hear everything, and often must listen for hours on end. If your studio monitors alter sound, even slightly, you won't get an accurate representation of your work and the potential for listener fatigue is greatly increased.

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Models pictured (L-R)
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Though Fostex are giving little away, it is clear that room for growth has been provided in the *D10*, including a couple of as yet unused expansion slots on the rear panel

user-definable threshold level. When you are about to write a Start ID, the *D10* again allows you to preview the start point and adjust it with millisecond accuracy before writing it to tape. It also has an Auto Cue mode which, rather like a broadcast CD player, automatically parks itself at the start of programme when you send it to search for an ID. If Auto Cue is selected when you load a tape, it will automatically search that tape within a window of a few seconds for the nearest start of programme. This removes the fatal unreliability out of cueing up to IDs and allows the *D10* to perform cart-like applications.

GPI and editing

A standard five-pin DIN socket on the rear panel bears the legend GPI In. GPI stands for General Purpose Interface and relates to no particular standard. Its current implementation allows external control of four transport functions: Play, Stop, FFwd and Rew; this may soon be changed to replace the wind functions with ID search. It works via simple contact closure making interfacing simple—including fader start.

Next to the GPI In DIN is an identical socket marked GPI Out. Essentially this is intended to be used for editing between two machines, although there is no reason why it should not be utilised to trigger anything else looking for a contact closure. Editing between two *D10*s is very simple; the digital output of the play machine is connected to the digital input of the record machine. For both machines, Memory 0 is taken to be the In point for edit, Memory 1 to be the Out point. These points can be set with considerable accuracy using the various RAM-based techniques mentioned earlier. When the record machine reaches its In point it simultaneously drops into Record and sends out a Play command to the play machine (which is set to Instant Start). An extension of editing between two *D10*s is to trigger a different type of source machine, such as an instant start CD player or a sampler, and assemble on to a *D10*.

The *D10*'s editing capabilities were originally presented by Fostex Japan as an effective means of simple compilation editing. However, it has become apparent that it can be accurate enough to execute straightforward music edits. The smallest increment on the RAM Scrub facility on the review machine is 10ms. With reference to analogue tape, that is about 0.33in at 30ips or 0.165ins at 15ips—of course, it could be less, depending on where in the 10ms window the leading edge of the sound is. Suffice it to say that, assuming a definite event to cut to, I found it quite accurate enough to execute inaudible edits.

The execution itself could not be simpler. Having cued-up the play machine, you press its MARK SET button and that automatically loads that time into Memory 0 (notice you do not even have to select Memory 0 first). Do the same for the record machine and hit its PREVIEW button. The record machine then automatically shuttles back and plays past the edit point, triggering the play machine and switching to line-in at the crucial moment to show the effect of your edit. If you approve, you simply hit MEM LOC on the play machine, shortly followed by the RECORD button on the record machine, and the edit is written to tape. If you happen to get a *D10* on trial, do allow a good few attempts to get used to the Digital Scrub before writing it off as inaccurate. It takes a while to master, but once you get the hang of it, it is very impressive.

Expansion

Though Fostex are giving little away, it is clear that room for growth has been provided in the *D10*, including a couple of as yet unused expansion slots on the rear panel. The provision of some form of time code/video sync facility would not be unlikely, and a front panel switch marked 9P REMOTE suggests that a matching socket might be on the cards for some future time. The remote also has four buttons marked F1, F2, F3 and F4 which, the machine's manual tells us, are left open for future additions.

Conclusion

When the *D10* was first launched it had a UK price of £1,895 plus VAT, although this has been reduced to an introductory level of £1,695 plus VAT to entice those still to be convinced of the benefits of the format. For a simple non-time-code, 2-head DAT machine, it is still rather expensive. But when you realise that a pair of *D10*s offer a complete DAT editing package for under £4,000 inc VAT, it becomes rather more of a bargain. After all, most facilities require at least a couple of DAT machines, and who would not find the facility to execute simple edits extremely handy? With this in mind, it is worth noting that any other professional DAT editing system is likely to cost you around £12,000.

No one is suggesting that the *D10* could replace a sophisticated hard-disk-editing system, but for simple assembling jobs it is actually likely to be faster and certainly a lot cheaper. Obvious applications include CD compilation, voice editing and simple music editing. And for live and broadcast work its quick intuitive, cart-like operation and GPI facilities (fader start and so on) must make it very attractive. ■

UK: Fostex (UK) Ltd., Unit 1, Jackson Way, Great Western Industrial Park, Southall, Middlesex UB2 4SA. Tel: 081 893 5111. Fax: 081 893 5237.

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DAT122
HHb


DAT15
DAT122
DAT30
DAT92
DAT62
DAT48

THE FUTURE OF SOUND RECORDING

John Watkinson examines present developmental lines of technology and their likely effect on future sound recording equipment

Advancing technology has transformed audio equipment out of all recognition in the decade since the introduction of the compact disc. While analogue recording fought a valiant rearguard action, the future of audio is now, to all intents and purposes, digital. If anything, the pace of technology is accelerating, making it harder to anticipate developments. In this look at what the next decade might bring, we will abandon the crystal ball, and return to basics—like physics and human nature—to establish what products might be ahead.

One of the strengths of digital audio is that once converted from analogue signals, audio becomes data which are indistinguishable from other types of data in that they are simply a quantity of zeros and ones. The quality in data is its reliability—a measure of the proportion of bits which are in error. Even if all types of data look the same, we can draw some distinctions between the reliability required by different applications. Computer data are quite intolerant of error, whereas digital video is remarkably tolerant. Audio comes somewhere in between.

Real recording media have error rates which are a function of two main factors. The first is the recording principle employed, that is, the physics of the process. The second is the recording density we are using with the process, or the amount of data we attempt to fit in a unit area (or volume) of the medium. The error rate of the medium is adapted to the allowable error rate of the application by the use of an error correction strategy. It is the combination of the medium and the correction strategy which gives the audible quality. It does not really matter from a sound quality standpoint whether we use a grim medium with powerful correction or a wonderful medium with weak correction, although it may matter economically. Error correction circuitry costs money, but so do wonderful media. A Winchester disc cannot be removed from its drive, so only one is necessary and making it of high quality is easier than making all of the tapes which work with one recorder of high quality. As a result tapes have more powerful correction than discs. Another factor is that the cost of error correction circuitry falls with time. Thus this year's satisfactory balance between the medium quality and the correction power could be unsatisfactory next year.

More than ten years after CD, no single digital recording technology has dominated the others, leading us to the conclusion that no single one is best in all circumstances. I intend to take a good look here at why that should be, because this may reveal a fundamental pattern which will hold as a model on which the future may be based. The only factor which might invalidate the model is the sudden discovery of some new recording technique. It will be seen in the sidebar on storage technologies that digital recording only requires some parameter to be maintained in one of two states. The examples given here cover all of the families of physical processes, so something different is unlikely not least because it would have appeared by now if it were simple enough to be useful. Experience teaches us not to use the word impossible, and there is still a slim chance that something may come along and turn recording on its head.

The message and the medium

Today's digital audio recording takes place on a wide variety of media. These include RAM, magnetic and optical disc, stationary-head tape and rotary-head tape. In computerland, media have primarily been compared on three factors. The access time, the cost per bit and the transfer rate. Subsidiary considerations include exchangeability, reliability, and reaction to power loss.

Fig.1a contrasts technologies in access time terms. RAM has extremely rapid access time because it has no moving parts (except for electrical charge). Magnetic discs come next because the whole recording area is exposed to a two-dimensional access mechanism (rotation and radial address). Optical discs have the same access principle, but the pickup is heavier and slower. Tape and film come last in this race because they have to be shuttled to expose the wanted area to the pickup.

Fig.1b contrasts the cost per bit. Here magnetic tape is supreme ► because it

a) Access Time	Medium	b) Cost per bit
Nanosecond	RAM	Very High
Millisecond	Magnetic Disk	High
Seconds/Minutes	Optical Disk	Moderate
Seconds/Minutes	Stationary-Head Tape	Low
Seconds Minutes	Rotary-Head Tape	Very Low

Fig.1: There is no one best medium as it depends on the criterion

is such a simple medium to manufacture. Rotary-head tape comes top because it offers higher recording density than stationary heads allow. Magnetic disk drives need an air film between the disk surface and the head to eliminate wear so they can stay on line for years at a time. This causes a spacing loss, and limits the practical recording density. Also the precision metal disk substrate costs more to make than plastic film. These factors push up the cost per bit. Optical discs are also expensive to make because of the complex construction. Most expensive is RAM which is extremely intricate, with every bit having its own wiring inside a chip.

There we have it; the best medium on one scale is the worst on the other! Thus there is no overall best storage technology, and this will continue to be true in the future, because improvements will occur to all media in parallel until physical limits are reached.

It is worthwhile exploring these limits. The sidebar on recording densities shows how the storage density of any technology is determined by the size of the bit which can be individually created. In RAM, the size of the bit is limited by our ability to produce sufficient resolution in the photographic process which precedes the etching of the RAM structure. The same is true of the size of feature which can be resolved on an optical or magneto-optical disc. In fact, the disc has an easier job because it scans one bit at a time. The optics of a disc drive require only a very small field of view, whereas the optics needed to expose the track pattern of a chip must have a wide field of view, and this is harder to achieve. The only way to increase resolution is to use shorter wavelength light. Any progress in this direction can also be employed to increase the capacity of RAM and optical disc, maintaining the relative status quo.

In magnetic recording, the density is determined by the wavelength along the track which can be resolved by the head and the narrowest track which can be followed by the mechanism. As the area of the bit gets smaller, noise becomes a problem, and this is opposed by the adoption of higher coercivity media. The coercivity of current media is nowhere near the physical limits, but is instead limited by the availability of heads which can apply sufficiently powerful fields to media without themselves saturating.

Unlike optical recording, where the wave nature of light sets a limit on density, there is no such limit in contact magnetic recording, and magnetic recording densities in the future may well outstrip those on optical discs. If one considers bits per unit volume rather than bits per unit area, magnetic recording is already ahead. Compare the volume of a compact disc with the volume of the tape in a DAT cassette. The DAT cassette has a smaller volume but plays for longer.

when time costs money, the best recorder may be the one which dubs fastest

In the past, the sheer data rate of digital audio was a problem, and we were grateful to be able to record it at all. In the future, the inevitable increase in density offered by all media will mean that the actual recording step becomes easy and we will compare equipment using other criteria. One of these will be the transfer rate. We have become accustomed to the limitations of analogue production equipment, where real-time operation was the norm. Apart from half-speed vinyl disc mastering and cassette duplication, all dubbing was done at normal speed so that it could be monitored. With read after write and error correction, digital media can transfer data reliably without human intervention; they can be designed to monitor themselves better than we can monitor them. Consequently, there is no longer a constraint to use real-time transfer, and when time costs money, the best recorder may be the one which dubs fastest—as it is in computerland.

Although digital audio has been commonplace for a decade, the number of products which can dub faster than real time is small; this is a lesson the audio industry and its manufacturers have to a great degree not learned. The computer industry has already learned this lesson, and may make stunning audio products in the future before today's audio manufacturers have discarded the analogue tradition. In addition to media which can operate at high speed, there will also be a need for an audio interface standard for high-speed transfer between units.

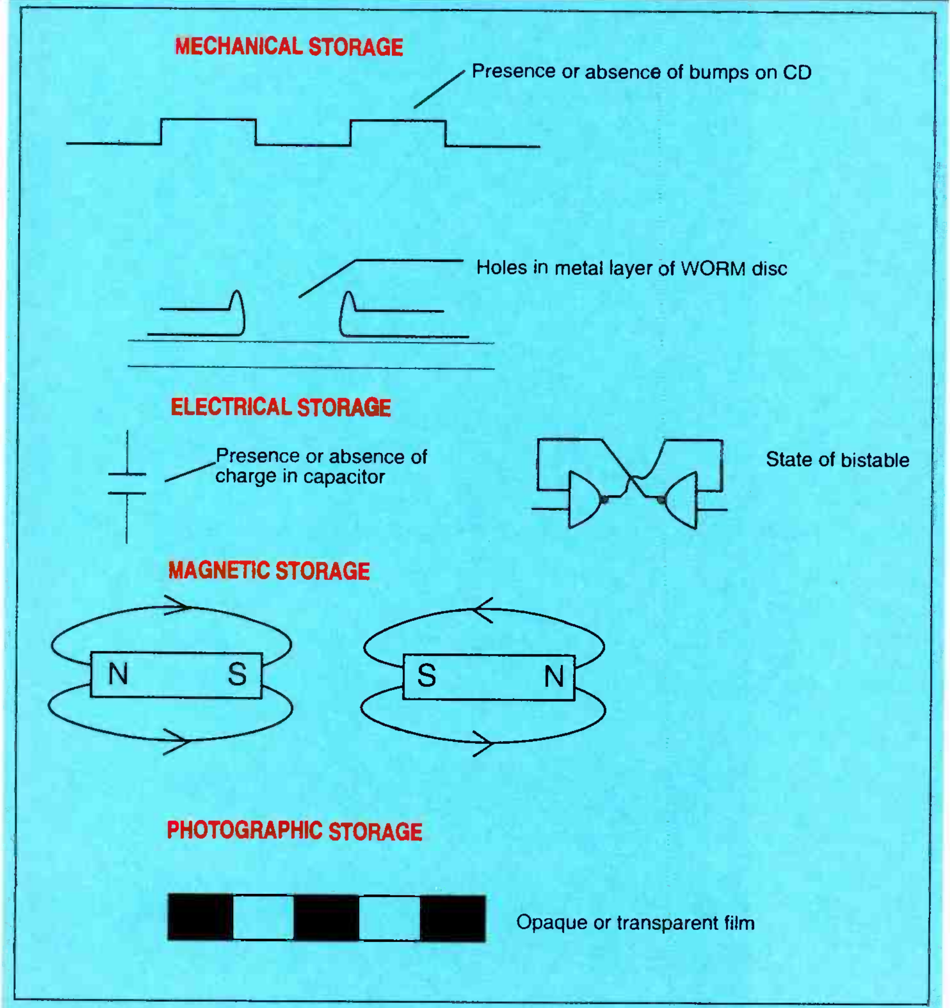
Another consequence of increased storage capacity is that data reduction will no longer appear so attractive for audio recording. It may be just a phase we are going through. Even with today's storage technology, phenomenal audio capacity is quite easy to obtain. For example the data capacity of one large size D3 digital video cassette is the equivalent of 300 hours of 16-bit stereo audio at 44.1kHz! One justification for data reduction is that it helps in faster than real-time transfer. If for the sake of argument 4:1 compression is used, the data rate is a quarter of the original. If the original data rate is maintained, the audio can be transferred at 4 times real time.

The relative merits of different storage media will not change greatly in the future, so current computerland solutions will still be applicable.

Computer age

For a long time, computers have combined storage media in real applications to extract the best of each. **Fig.2** shows a typical computer, in which data and instructions are sourced from RAM, as the rapid access time allows the fastest computation. RAM is too expensive to ▶

STORAGE TECHNOLOGIES COMPARED



We currently have mechanical storage; the presence or absence of features or holes on a carrier, electrostatic storage; the presence or absence of charge in a RAM cell, magnetic storage; the direction of a remnant magnetic field, and optical storage; the variation in opacity or contrast of a carrier. All of the main branches of physics are represented here, so it is difficult to conceive of a new technique.

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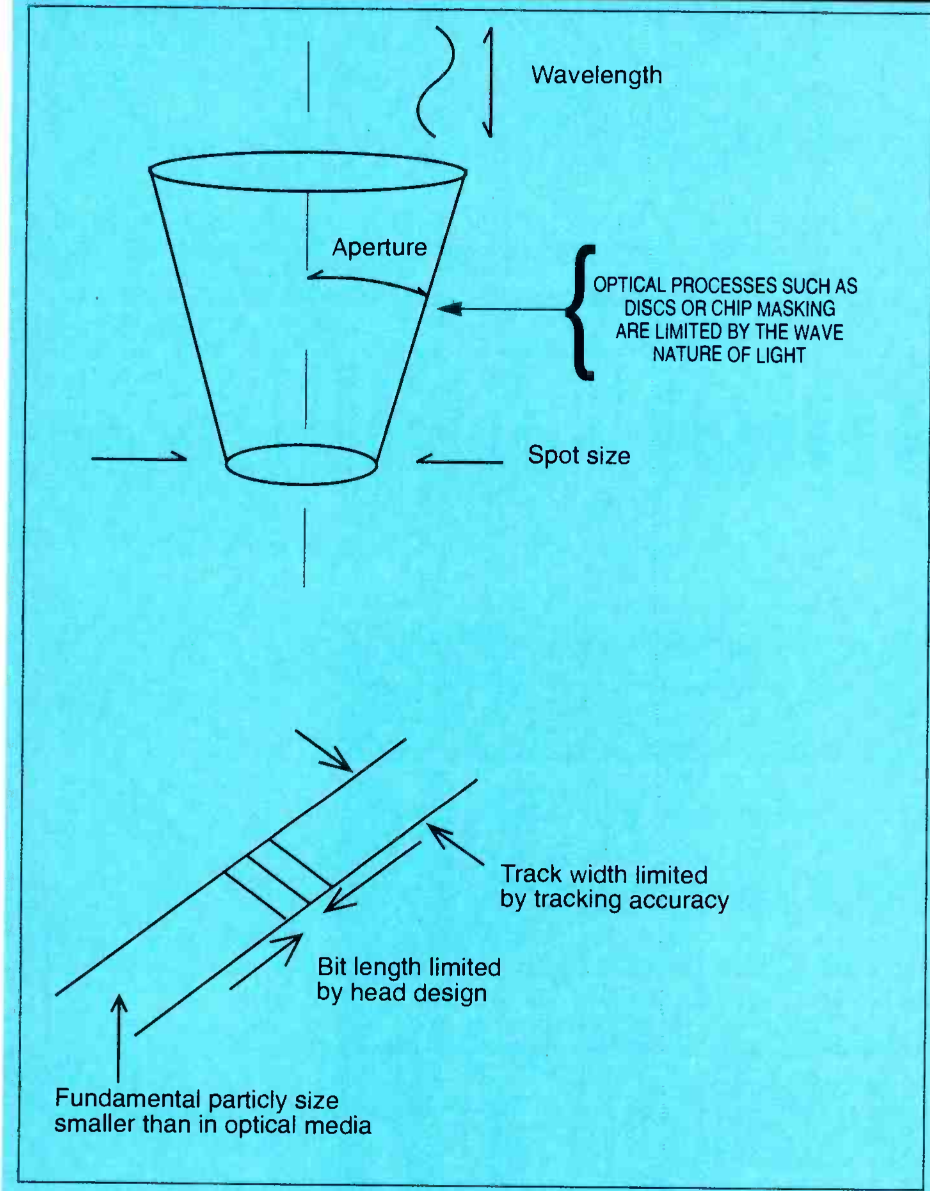
RECORDING DENSITIES COMPARED



The density is limited by how small we can make the individual bit on the medium. RAM bit cells are limited by the optics needed to replicate them. Optical discs share the same restriction shown here where the wavelength of light and the aperture of the lens determine the spot size. Driving down the wavelength puts up the cost. Magnetic recording can have bits down to molecular size, so the limit is in our ability to make heads which have a sufficiently small gap to replay short wavelengths and the availability of very accurate tracking systems to allow narrow tracks to be reproduced. In the future magnetic recording densities may advance beyond those of optically restricted media.

keep every program memory resident, hence the use of a fast access hard disk which swaps programs in and out of memory as needed. Again the disk is too expensive to archive all data files, and this is the job of the tape deck where the low cost per bit is its strength and the slow access time presents less of a problem.

Fig.3 shows how this approach can be applied to solve audio problems. The computer processor is replaced or supplanted by a DSP device (a computer optimised for signal processing rather than general calculation), but the general arrangement of RAM, disc and tape is retained. The communications ports are replaced by high speed audio interfaces. The disc drive here would use Winchester technology because it does not need to be removable as there is a tape cassette for that purpose. The disc might well use parallel transfer where each head has its own circuitry and all can move data in parallel where required. The tape deck might use stationary heads using thin film technology and narrow tracks requiring a tracking servo. The adoption of stationary heads is designed to allow operation at several speeds. Alternatively a rotary head transport may be used which has a single high data transfer rate, but which is buffered by RAM and works intermittently if a lower rate is required.

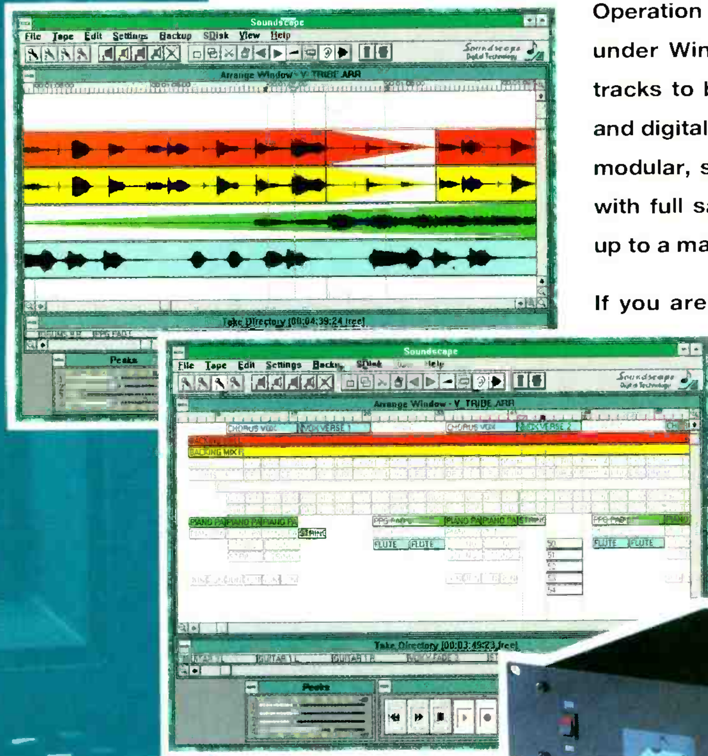
Such a general-purpose audio system is extremely flexible, but one example of its use will be given here. Consider that a classical recording is to be made using a stereo pair with another couple of spot microphones. Several takes will be made of each piece, then each piece will be mixed and edited by assembling from various takes.

Finally, the various pieces will be joined up ▶

SOUNDSCAPE

MULTI-TRACK HARD DISK RECORDER

Soundscape is a high quality 16 bit digital audio recording and editing system, and is capable of expanding your studio with 4/8/12/16 or up to 64 tracks. The system can be used in a recording/composing environment and has extensive non-destructive audio editing facilities.



Operation is from an IBM PC™ or compatible and runs under Windows 3.1™. Software allows up to 64 virtual tracks to be recorded in stereo, edited (non-destructive) and digitally mixed down to four outputs. As the system is modular, several Soundscape units can be synchronised with full sample rate accuracy and used together giving up to a maximum of 32 inputs and 64 outputs.

If you are looking for a Hard Disk recorder/editor with "Open" architecture that can be totally integrated with any Windows™ sequencer or editing package, is random access to the disk, expandable beyond 8 tracks and offers full "chase lock" synchronisation to analog/video tape machines then the next stage of the digital revolution starts here.

2U 19" rackmounted unit.

Physical tracks: 4

Sampling rate: 22.05/32/44.1/48KHz

Data format: 16 bit linear

Signal processing: 24-bit internal N.L.

Data storage: IDE hard disk 18mS, fitted in the rack unit (not supplied), size depends upon recording time required, e.g. 130MB gives 25min 45sec total @ 44.1KHz, 1gB gives 3 hours 22 minutes

2nd internal IDE drive can be fitted.

A/D conversion: 16 bit sigma-delta 64 x oversampled

D/A conversion: 18 bit sigma-delta 64 x oversampled

Synchronisation: Master or Slave, MTC with full chase lock, MIDI song pos.pointer + clock

Analogue in: 2 x RCA/cinch, unbalanced – 10dBv/+4dBv (2 tracks in)

Analogue out: 4 x RCA/cinch, unbalanced + 4dBv (4 tracks out)

Digital in: 1 x RCA/cinch, S/PDIF format (2 tracks in)

Digital out: 2 x RCA/cinch, S/PDIF format (4 tracks out)

Input S/N Ratio: > 93dB un-weighted

Output S/N Ratio: > 113dB un-weighted

Wow and Flutter: Un-measurable

Pro-Audio Option: XLR balanced Analogue inputs and outputs, AES/EBU Digital inputs and outputs (XLR)

Host Interface: IBM-AT: parallel via PC expansion plug-in card (ISA). Supports 2 x 4 track rack units.

MIDI: in, thru, out

Back-up medium: DAT-recorder with digital i/o, or via the PC (e.g. to a SCSI optical drive or any logical PC drive)

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Cardiff: + 44 (0) 222 811512

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(PRO SDT/002)

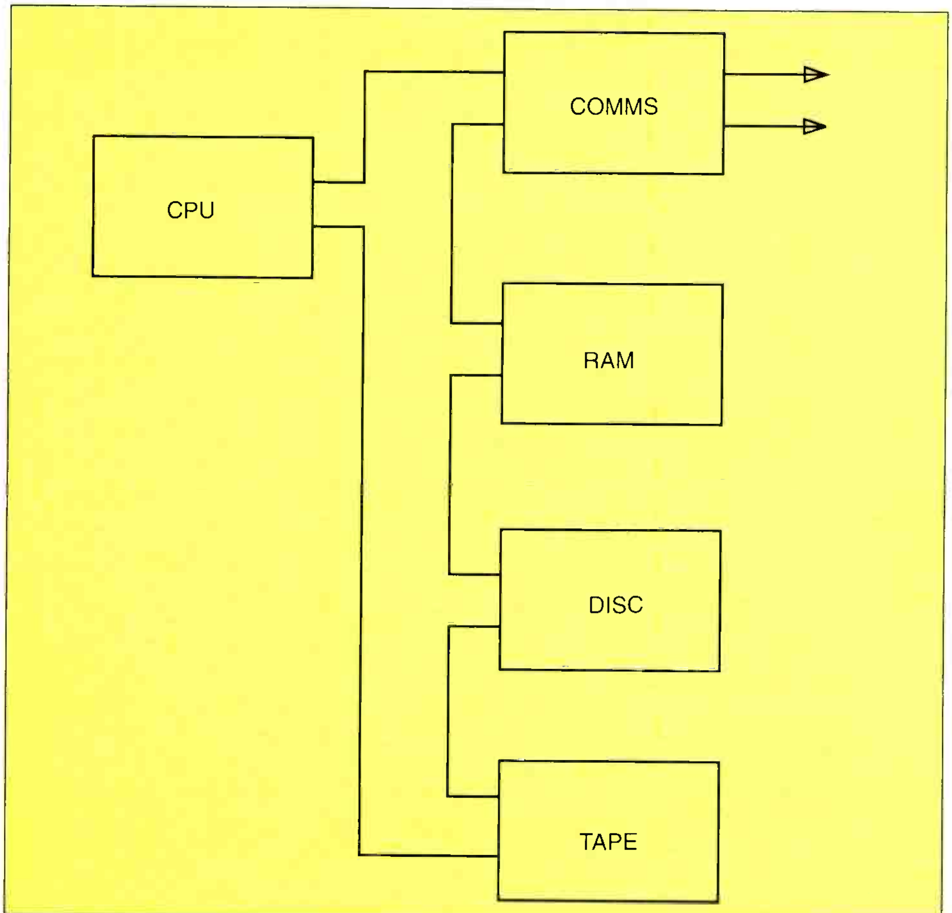

Fig.2: A typical computer mixes storage technologies to obtain the best features of each medium without excessive cost

to make an album for release. During each take, audio from the convertors is recorded in full bandwidth on the tape cassette in the second of two partitions. At the same time the audio is data reduced in the DSP, and recorded on the hard disk. At the end of each take the data reduced file from the disk is transferred to the first tape partition. Along with a complete record of the console setup. At the end of the recording session the tape contains a full-bandwidth version of everything, but at the beginning of the tape is a browsing file which contains a compressed version. The tape can be taken away and edited in a different machine, or brought back to this machine. Upon installing ▶

Within reason, digital technology can achieve almost anything, and the difficulty becomes one of knowing what to make rather than how to make it

Encore!

Vienna II

The new Vienna II gives your performance that little bit more

The new Vienna II shares the same smooth looks as its predecessor and has many of the same attributes - with up to 40 input frame size, 8 groups, aux sends and effects returns together with optional 11 x 8 matrix.

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- Improved EQ for tighter control over LF in both bell and shelving mode
- Fully featured stereo module giving higher input density
- Individual pre-post switching on all auxiliary sends
- New input stage giving wider range and allowing mic and line level signals to be connected via the XLR input
- Insert points on auxiliary outputs
- VCA Grand Master module for extended control of 8 VCA groups

Whether you are on the road or have a permanent venue to fill, the Vienna II offers unrivalled performance together with Soundcraft's impeccable pedigree at a price which may surprise you.

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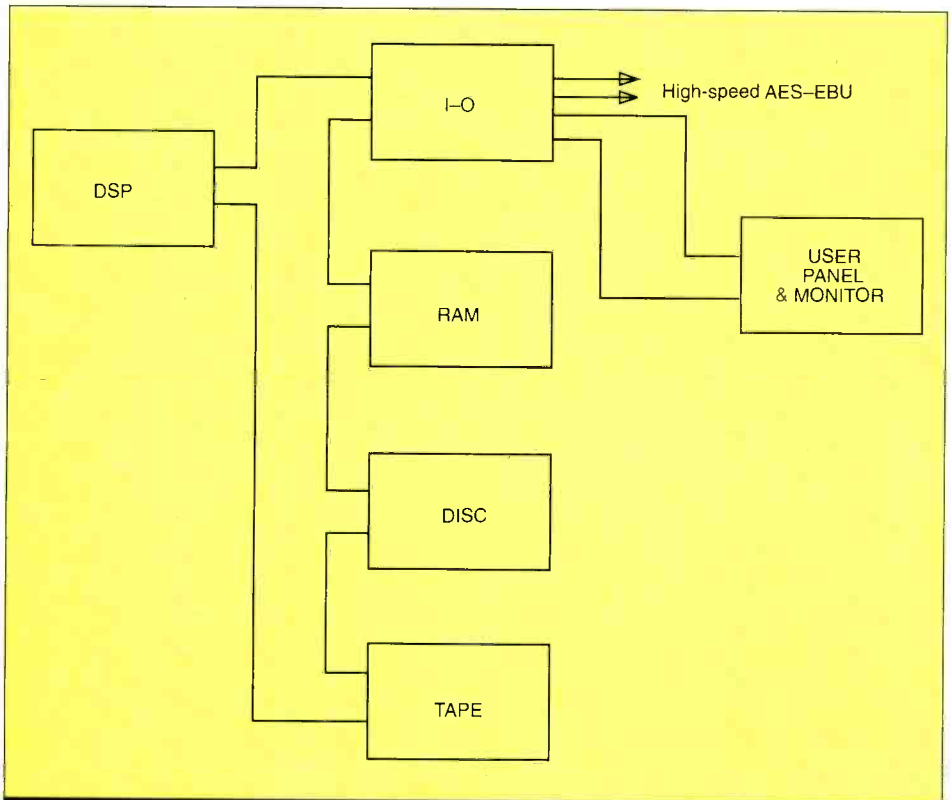

Fig.3: An audio computer combining storage technologies

the cassette, the data reduced file is transferred to the disk, and the console setup is reloaded. If 4:1 compression is used, and the tape plays at 10 times speed, this transfer occurs at 40 times speed. However, this process need not be completed before editing begins, because the disc controller supports multiple access and the user can listen to the beginning of the recording before the end is transferred. Mixdown and editing are performed using the compressed disk files, but only for the purpose of creating a console automation file and an edit decision list. When the final editing is finished, the user can leave the machine and take lunch. In his absence, the automated mix and the EDL are executed on the full bandwidth recording on the cassette and stored on disk. On their return, the user can play the full-bandwidth recording from the disk, and check that everything is as expected. After any last minute changes, the disk contents can be dumped back to a new cassette which becomes a CD master. Such a device could be assembled tomorrow from existing components.

The view of the future advanced here is that there will be more freedom because data recording will become easier and complex processing will be inexpensive. Within reason, digital technology can achieve almost anything, and the difficulty becomes one of knowing what to make rather than how to make it. This is a double-edged sword because manufacturers are unlikely to risk a radical product if the market is too conservative to understand its benefits. There is still a lot of thinking constrained by limits which were due to analogue technology and which no longer exist. Unless such thinking is liberated, then users are unlikely to take advantage of the freedom of the digital domain and they will not demand innovative products from the manufacturers. ■

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*. He is listed in *Who's Who in the World* and regularly presents papers at conventions of learned societies. ■

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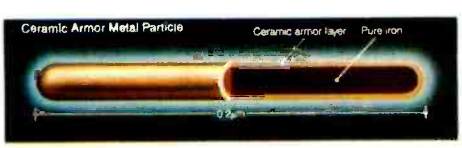

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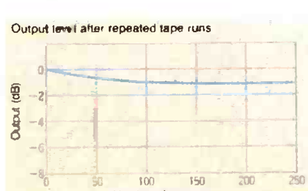

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RUNNING IN THE FAMILY

Francis Rumsey assesses the state of SSL's 'system' approach to high-end digital postproduction, including a look at the new *OmniMix*

Solid State Logic have been in the digital audio postproduction business for a number of years now, and a series of products has evolved with something of a family likeness. Common to the members of this family are a number of key technologies which are worthy of discussion because they demonstrate an interesting attitude to system integration, and evidence of a carefully planned approach to the overall development of a product line. In this article a number of these key technologies will be described, by way of showing how the products relate to each other, how they can be made to work together, and some possible directions for the future. Since all except one of the products have been reviewed individually, this article will avoid concentrating on specific product features (except the new ones), with a view to looking at the systems aspects of design.

The digital family

The eldest and smallest member of the SSL family is *ScreenSound*, an 8-channel digital editor which presents material to the user in the form of separate 'reels' or tracks, very much akin to the film editing approach. The product has basic built-in digital mixing with automation, and is used widely for editing and dubbing sound to picture in film and television work. About a year ago the second member of the family, *Scenaria*, was born, this being a larger postproduction system with more comprehensive mixing facilities, designed to offer 24 channels of simultaneous reproduction from disk in conjunction with 38 channels of digital mixing, with a 2-channel stereo main output. One of its key claims to fame is that it records and replays video using a random access disk store which is designed to hold up to one hour of PAL or NTSC video. This allows the user to cue any point in the programme very rapidly, eliminating the normal spooling and lock-up time of tape-based video machines.

OmniMix (**Fig. 1**) was launched earlier this year, due to be shipping in September, and takes the *Scenaria* concept one step further by expanding the physical frame to accommodate more faders, another monitor, and a more comprehensive control panel. It also expands the audio capabilities of *Scenaria* by adding 32 submix buses which can be configured as a hierarchical matrix for creating surround sound submixes of different programme elements such as dialogue, music and effects. A patented system called *MotionTracking* is provided on every channel for panning sources dynamically between the surround submix buses, and a number of digital effects are available internally. Since *OmniMix* is a development of *Scenaria* it is possible to upgrade a *Scenaria* by adding processing and other hardware and software.

The family therefore provides the means of editing right through to surround sound mixdown, and the higher-end products are clearly orientated towards the film world (where large numbers of channels and surround sound are commonly used), as well as towards the growing television market (where surround sound productions are on the increase). The random access picture playback has also proved attractive to commercials houses where fast turnaround of productions is the norm. ►

Fig.1: *OmniMix* was launched this year and takes the Scenaria concept one step further

Key technologies

The products outlined above rely on various common elements, each of which is relatively independent. The larger products are constructed out of many of the building blocks which also work with *ScreenSound*, and indeed the *ScreenSound* editor is a functional part of *Scenaria* and *OmniMix*. Networking is an important factor in the system design, although the network is not primarily used for carrying audio data, and the concept of a central pool of storage resources is also paramount. Mark Yonge, SSL's Digital Products Manager, is adamant that the future of such systems is in multiuser operations, and much of the work that has gone into these products is designed to ensure that material can be passed between different production stages or workstations with the minimum of organisational confusion.

Disk drives

Audio is stored on SCSI disk drives in all of these systems, and each disk drive stores data for eight channels of audio. *ScreenSound*, therefore, accesses one disk drive at a time, whereas *Scenaria* and *OmniMix* are capable of dealing with three drives to serve 24 audio source channels. The file format and edit information are common to all systems, making it possible for a disk written using one to be read by another, and for access to a disk to be transferred to another user with *SoundNet*.

SoundNet

SoundNet is not really a network in the conventional sense of the word, but an assignment switcher for SCSI storage resources such as magnetic disks, optical discs and tape cartridges. A large number of different storage resources may be connected to this switcher, and the user may assign one or more of the disks to his system (either one or three, depending on the system) provided that no one else is using them. In this sense the storage resources are not truly shared, since only one person at a time can access each one, but the advantage is that once a user has a disk it is always available at maximum bandwidth for as long as required. Furthermore, it helps to keep projects organised, since there would be a whole new collection of system management issues to deal with if resources were truly 'multiuser'.

Projects, desks, reels and sounds

A common format of sound file lies at the bottom level of a tree structure which is used to organise and keep track of projects (**Fig.2**). Sound files are organised into 'reels' (individual edited tracks of audio), and the reels which make up an 8-channel block, plus their associated automation and setup

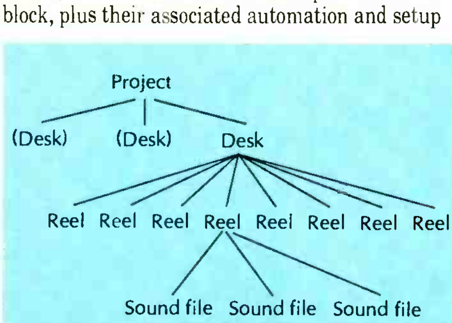

Fig.2: Organisation of sound files

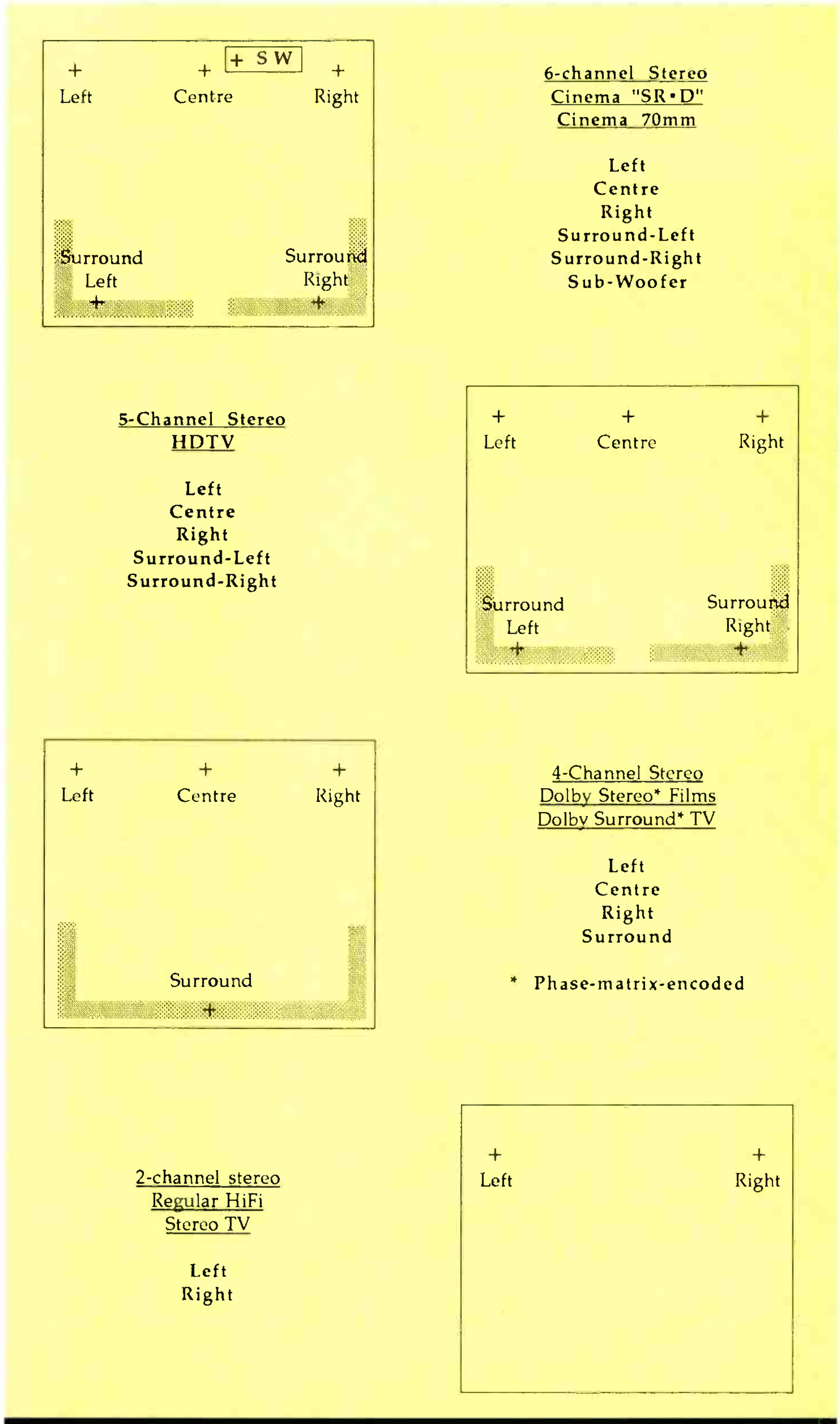

Fig.3: A hierarchy of stereo formats

data are described by a 'desk' file. When a desk file is recalled in *ScreenSound* it reinstates an editing project on the screen, and finds all the sound files and reel information which contributed to the edit. The desk file is where the tree stops for *ScreenSound*, but *Scenaria* and *OmniMix* go one step further in adding a 'project' description. The Project Manager screen contains information coordinating a complete large-scale project, including information about channel routing, automation data, clip histories and so on. It allows the user to store complete system setups for *Scenaria* and *OmniMix*, making it possible to recall each stored revision for any aspect of the production. This can be particularly useful when keeping track of a large production, where, for example, the director may wish to make changes at a late stage in the postproduction process. Provided that clip histories, edits and mixes have been stored, it will be relatively easy to go back to the original source material which made up the tracks concerned and re-edit, whereas in conventional film production it is likely that the sound may have gone through a number of premix stages. A project refers to a number of desk files, each describing eight reels. ►

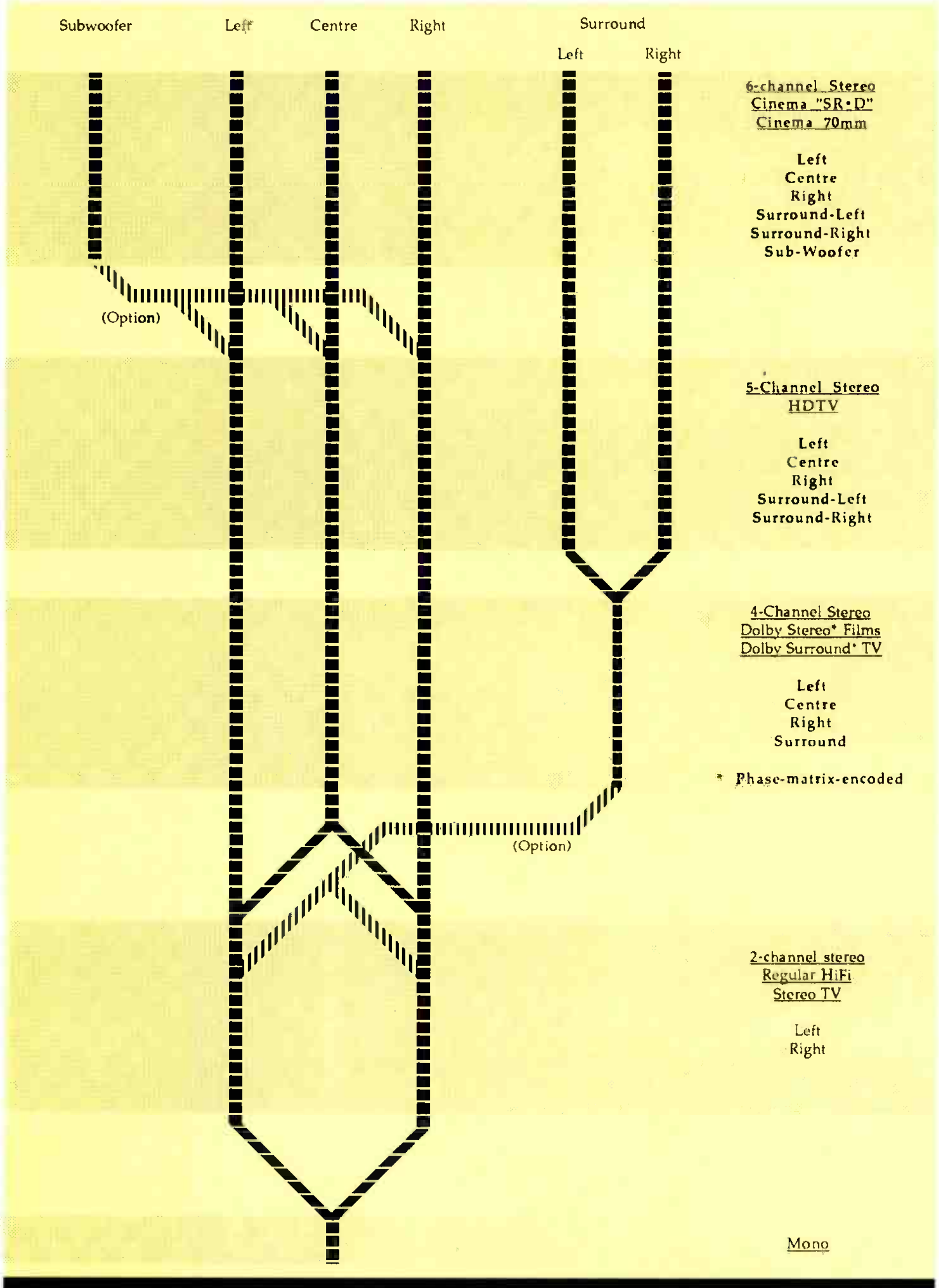

Fig.4: Hierarchical format reduction

Ethernet

While some digital audio system manufacturers are harnessing high-speed networks to carry audio, SSL use the relatively slow *Ethernet* as a control network linking all of the functional processors in a system. For example, in *OmniMix*, the *Ethernet* runs between the control panels, main processor, optical disc drive, *SoundNet*, *ScreenSound*, *VisionTrack*, patchbays, and so on, as a means of passing control commands. Each device has a unique *Ethernet* address.

The only instance in which *Ethernet* is used to carry audio is when a local magneto-optical (M-O) disc drive (housed near the user, rather than in the central machine room) is used to audition a sound clip. Users may keep a handy library of optical discs and listen to single mono files for auditioning purposes only. In order to use a clip from the M-O disc it must be transferred over Ethernet onto the active hard disk. SSL connect the SCSI M-O disc onto the Ethernet using a proprietary interface, fetchingly called *SCSINet*, which passes conventional *Ethernet* data packets at a rate sufficient to handle a single audio channel. Enough buffering is used in the main processor to ensure an unbroken audio output even if other traffic is present on the network.

An example of system integration using Ethernet and *SoundNet* can be seen in *Scenaria* and *OmniMix*, where an 8-channel editor is built into the system as a standard feature. It is really nothing other than a separate *ScreenSound* processor which resides on the *Ethernet*, and which has SCSI access to the same disk drives as the main *Scenaria* processor via *SoundNet*.
During postproduction any group of eight tracks which forms part of the mix can be passed to the *ScreenSound* for editing, at which point the hard disk on which that material is stored is automatically reassigned to the editor. A desk file describes the part of the project contained on that disk. Control is assigned to the editor by swiping the graphics tablet pen in a particular direction, at which point the master-slave configuration of *Ethernet* control is altered so that the pen now controls the editor. When the material has been re-edited, a swipe of the pen back to the main *Scenaria* processor results in another change of the SCSI assignment to remove the disk from the editor, and the newly edited material is now available on the original *Scenaria* channels.

One multiuser installation, Post Perfect in New York (a commercials production house), has three *Scenarias* on one *Ethernet*, and SSL are watching this installation with interest to determine whether the increased traffic on the network results in the need for some form of network zoning or other management. One channel of audio requires just under 1Mbit/s of bandwidth, and *Ethernet* offers 10Mbit/s, although can often in reality only carry perhaps a third of this because of overheads in the packet structure and other housekeeping matters. If three users were auditioning audio files from their local M-O discs at the same time, together with a sprinkling of control packets, it is conceivable that one could be running fairly close to maximum load on the network. Network zoning and devices such as concentrators, though, have been used in computer networks for many years to regulate traffic in certain sections of the network, and to confine packets for particular addresses to particular zones, in order to keep the data rate within workable bounds. Such an approach could also be applied in large audio installations.

VisionTrack and *VisionCue*

VisionTrack is SSL's random access picture store, and along with its patented *VisionCue* system it provides storage of PAL or NTSC video from RGB analogue inputs, with almost instant access to any cut point in the picture. Rather than buying digital video compression technology from a third party, SSL developed a process in house which, one must assume, offers modest data reduction whilst retaining good picture quality. The quality is roughly as good as U-matic video, and offers reasonable performance in freeze-frame, which was one of the main design aims, since many editors need to be able to sync sound material to lip movements in slow motion or freeze-frame.
The video material is stored on a disk pack which appears to be connected to the processor using three SCSI interfaces (giving some indication of the data rate required). The company is prepared to say almost nothing about the video technology at the moment.

VisionCue analyses the video data during recording and draws a plot which graphically describes the positions of the picture cuts. Thus picture cut locations are not imported from a video EDL but determined by what is happening in the picture itself. The system looks for rapid changes in the composition of the picture and plots a small spike on a time line to indicate this. The user may therefore cue to any picture cut very quickly.

Surround sound and *MotionTracking*

One of the main things that *OmniMix* brings with it is a large number of additional submix buses to facilitate complex surround sound mixing.
The additional 32 buses can be configured using SSL's so-called Hierarchical Submix Matrix (HSM), which is a means of arranging the buses into say 4 or 5-channel groups for different surround formats. There are all sorts of surround formats in existence for use with picture, as illustrated in the example shown in **Fig.3**, and to arrive at the lower level formats (**Fig.4**) various combinations of the channels are made by matrixing them. With 32 submix buses one could easily have music, ▶

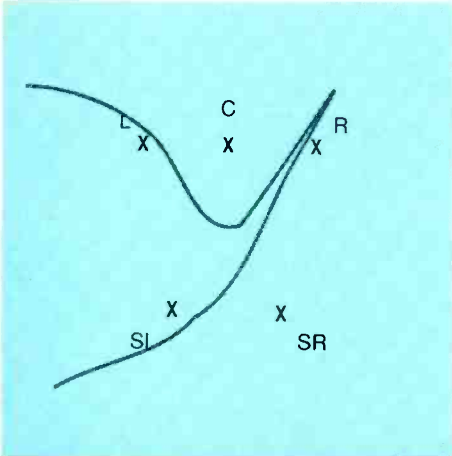

Fig.5a: Motion tracking

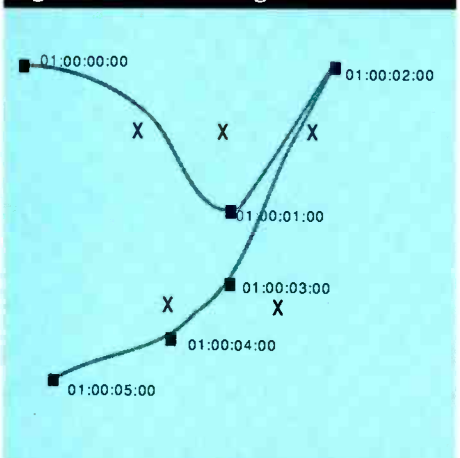

Fig.5b: Keyframes

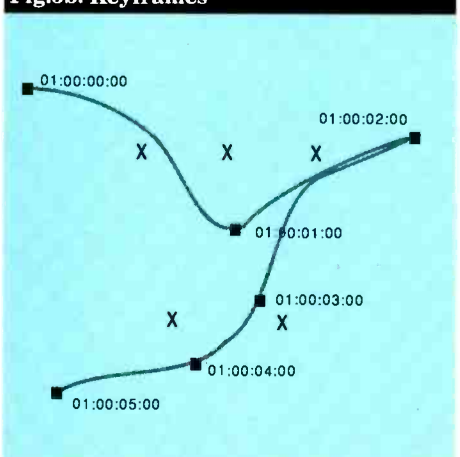

Fig.5c: Modify keyframes

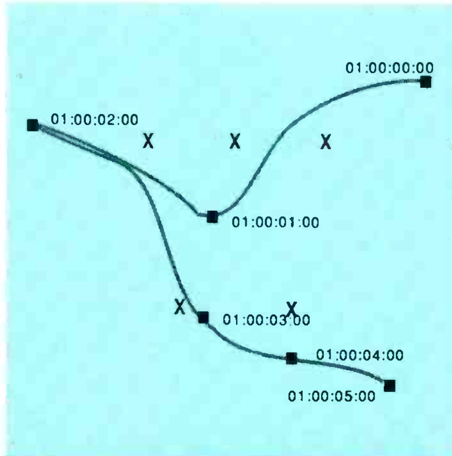

Fig.5d: Flip

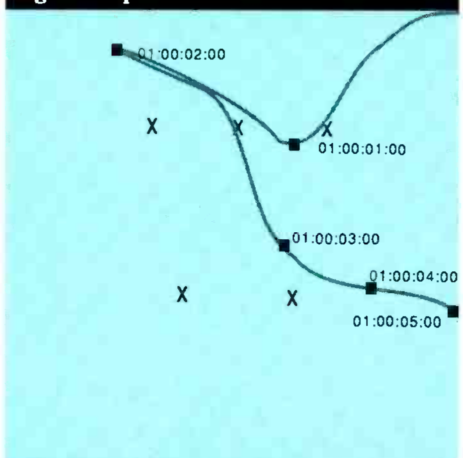

Fig.5e: Move

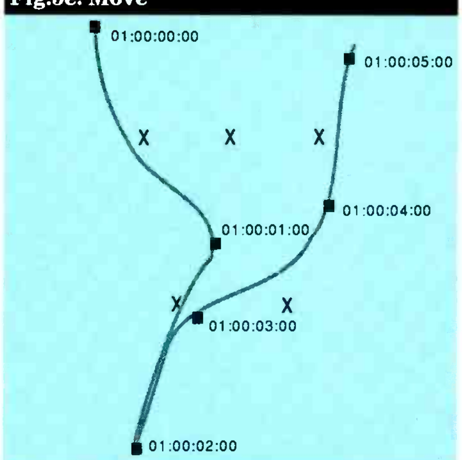

Fig.5f: Rotate

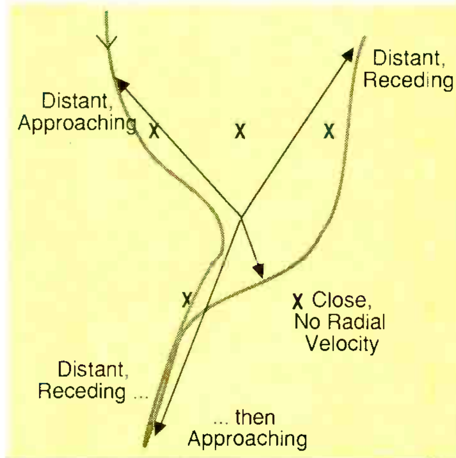

Fig.6: Radial Proximity

dialogue and effects submixes all in 5-channel surround, plus an overall 5-channel mix, a stereo reduction and a number of effects returns.

To date, surround sound panning has either meant using a combination of panpots for left-right and front-back, or very expensive joystick panners (of which even large film desks may have only one or two). *MotionTracking* basically adds a dynamically automated surround panner to every source channel in *OmniMix*, configured for the surround format which is currently set up by the HSM. The panner is no longer a knob but is depicted on one of the screens as a 'room' with loudspeakers in certain places. Using the pen and tablet, a path may be traced on the room plan in real time as the audio is playing, allowing the user to hear the spatial effect of the move. As the path is traced it is displayed visually (**Fig.5a**, depicting a 5-channel format in which a source is panned from left, forward into the centre, then right, then over the head to back left). As shown in **Fig.5**, keyframes may be shown on the curve to indicate the time-code points at which a certain position is reached, and these may be altered manually so as to ensure that a source reaches a certain position by a certain time. Points can also be dragged, rather as in a computer drawing package to modify the shape of the path (**Fig.5c**). **Figs.5d**, **Figs.5e** and **Figs.5f** show various transformations that may be effected to turn the path around or move it, and it is also possible to copy and paste a path to another desk channel with a delay, so that one sound effect could be made to follow the same path slightly later, say.

The built-in effects processor of *OmniMix* offers a variety of reverbs and delays which have been considered important by clients, as well as a unique means of simulating the Doppler shift which occurs when sounds move towards a listener and then away. As shown in **Fig.6**, by calculating the relative velocity of the source with relation to a listener (based on the panning path) it is possible to introduce a pitch shift digitally to simulate the appropriate Doppler shift. This could be exaggerated if necessary to give a more extreme effect than the position would suggest.

Other effects like this may be possible in time, such as EQ and level changes to simulate distance or closeness, and doubtless the users will provide sufficient feedback to fuel this development.

Conclusion

It is interesting to look at what a 'high-end' company such as SSL are doing with digital postproduction. There are an inordinate number of systems at the lower end of the market which are often based around PCs, and these offer incredibly good value for money, but it is hard to imagine dubbing a large scale multichannel film production using a *Macintosh* and a mouse. SSL are right that surround sound mixing will be an important feature of the coming decade, in television as well as in film, and they are probably also right that multiuser installations will be attractive in the sorts of markets at which they are aiming. The company is therefore strongly placed to take a good slice of the market, particularly if it can persuade a notoriously Luddite band of film editors that they can use such technology while retaining their creative skills, and also, most importantly, while keeping track of the immensely complex sequence of events involved in large productions. If, with digital postproduction, an editor cannot go back to the rack and find the physical pieces of sound film which he put there two days ago, he will need a suitable electronic means of assuring him that he has not lost things (the film editor's recurring nightmare!). It is therefore not just bells and whistles which will sell digital products to such people, but the feeling of security that they engender, and the system's capacity for organising large projects in an understandable manner, together with reliable backup storage. ■

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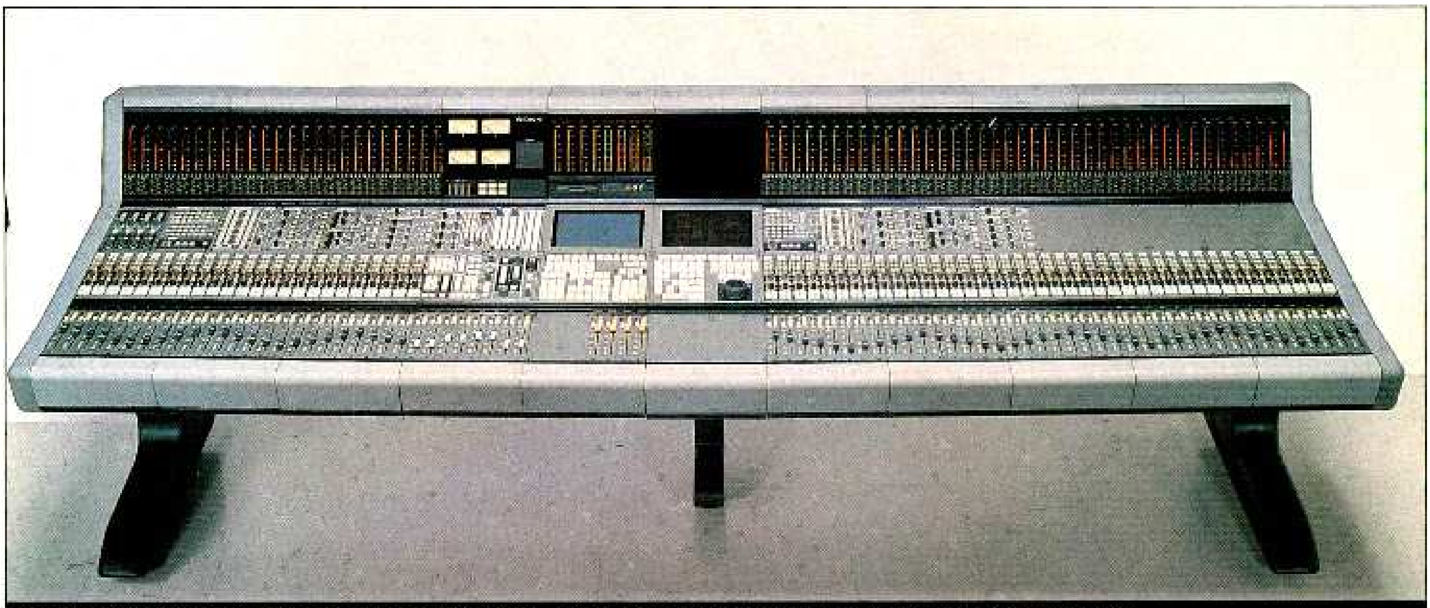


The *DMX-6000* incorporates in-line architecture, assignable processing panels and definable controls

Incorporating a digital signal path into a mixer has caused Sony to present it for postpro work rather than music recording. Patrick Stapley talks desks and delays

Scheduled for commercial release in September, Sony's all digital *DMX-S6000* console heads a family of new mixing desks aimed at the video post and broadcast sectors.

The *DMX-S6000* has been designed as an all-round video postproduction console, with additional production applications—although Sony do not recommend its use for music overdubbing (see later). The console has already been shown at the APRS and Berlin AES shows, and a first order has been placed by Sony Classics in Germany to equip a remix room.

A design philosophy which has played a significant part throughout the console's development, has been to retain familiarity with analogue operation, and to make the desk as intuitive and uncomplicated as possible. 'Central to the conception of the *DMX-S6000* has been the understanding that digital solutions are largely futile if they are encumbered by a complicated interface that only deters those who stand to benefit,' comments Sony Broadcast International Audio Product Manager, Andrew Hingley. The results is a hybrid design, combining in-line architecture, assignable processing panels, and some definable controls.

The console comes in four frame sizes, providing 24, 32, 48 and 64-channel configurations. The smallest version measures 1390 x 980 x 1010mm (WHD), while the largest is 2670mm wide; the remote processor rack is a compact 20U-high in each case. Depending on frame size, up to 56 output buses are available, which will interface directly to different DASH multitrack-digital audio workstation configurations. For example the *DMX-S6032* (32 outputs) will interface to a *PCM3324S* via SDIF2 (I-O), and to an 8-channel disk editor via AES-EBU (I-O); the *DMX-S6064* will interface directly to a *PCM3348* and an 80-channel digital audio workstation without duplicating sends. All channel inputs are AES-EBU.

The console locks to video reference (NTSC, PAL or HDVS), word clock or AES-EBU sync. It can also regenerate word clock, and will follow varipitch playback from a word synced multitrack.

The *DMX-S6000* is equipped with four programme buses which support operation in stereo, dual stereo, quad, and four channel surround (3-1). Dual mono outputs or a stereo submix can also be sourced from any combination of programme outputs. Monitoring may be in mono, stereo or quad, and monitor mixing is available. Sony propose to offer an 8-bus upgrade in a year's time to support surround formats such as Dolby *SR-D*, HDTV, and their own 8-channel SDDS system; the upgrade will be retrofittable to existing consoles with a minimum of disruption to the control surface.

A standard control surface can loosely be divided into three main areas: the I-O section, the master facilities, and the automation-setup control area.

The I-O section as mentioned adopts the in-line approach with large and small faders, and includes dedicated 'hard' controls for key functions like solos, mutes and programme routing. Each channel or monitor path has access to five assignable control panels —EQ, dynamic processing, pan and track routing, auxiliary sends, and input and signal path configuration. These panels are placed above the in line section, and in the case of the 48 and 64-channel consoles, two sets are provided for ease of access. ▶

As with conventional in-line designs, signal processing elements are placed either in the channel or the monitor path, but can, of course, be split so that EQ is in the channel and dynamics in the monitor, for example.

Processing and busing

Processing is via dedicated function, serial DSP chips that are attached to pairs of channels. This arrangement provides sufficient power to fun all functions on all channels; there is no need to 'ration' power, or allot processing. In fact there is enough processing power available to permit simultaneous DSP in both paths, and Sony are currently developing software to allow this in the future.

EQ is 4-band swept with a total range from 31Hz to 17.4kHZ (±15dB). Each band has a three step Q (0.7, 1.4, and 2.5), and the HF and LF bands are switchable between Peak and Shelf. In addition, variable low- and high-pass filters are included operating at 12dB/octave.

The dynamics panel offers limiter-compressor and expander-gate sections each with gain reduction metering and key inputs. The limiter-compressor may also be accessed by each of the four programme outputs.

Both EQ and dynamics panels include an ALT facility permitting an alternative setting to be compared with existing values. The equaliser, filters and dynamics can be sequenced in any order and switched in-out from the Path Panel which provides a clear indication of configuration via a matrix of selector keys. Additional confirmation of channel and monitor processing selection is permanently displayed at the base of each bargraph meter by a group of colour LEDs. Every channel also has a 4-character alphanumeric scribble strip which can display source, signal status, and grouping information.

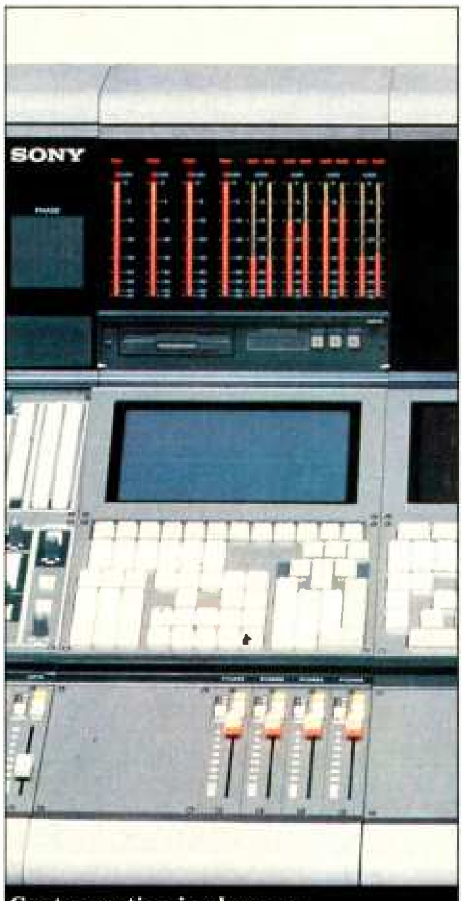
Eight auxiliary buses are provided that are switchable in pairs to operate in stereo; all sends have on-off and pre-post switching. The panning section responds to the four master modes described earlier (Stereo, 2 Stereo, 2-2 Quad, and 3-1 surround), and offers Left-Right, Front-Back, and Divergence controls—currently under development is a joystick panner.

Channel and monitor inputs are selectable from channel, track send or track return, while large and small faders may be sourced from the channel path, the monitor path or the other fader. This flexibility allows the console to be set up in a variety of ways for recording, track bounce, mix down (with all inputs and track returns, or track sends used as additional auxes), and broadcast (simultaneous live mix and multitrack recording).

Analogue inputs are controlled from a mic-line panel placed in the assignable control area. One panel is standard and controls four inputs; A–D convertors are not supplied as standard.

The third type of control in the I-O section are the definable controls positioned below each small fader. These are two rotary encoders each with an associated switch, which can control a variety of functions globally allocated to them from a central selector panel—such as, pairs of aux sends, EQ bands, panning, input trims and so on. The facility offers the user an alternative and more traditional method of operation, but is obviously limited by the number of controls.

Central facilities include aux masters, monitor mode and source selection, studio playback source, meter source and mode, oscillator controls and routing, talkback-reverse talkback, control room monitoring, global switching, assignable bus trim, and so on.



Centre section in close-up

Automation

The automation-setup section incorporate a gas plasma monitor and keyboard controlling both moving fader and snapshot automation along with set up functions for console sample rate (44.056kHz, 44.1kHz, 47.952kHz, and 48kHz), channel delay (up to 300ms per channel), insertion point assignment (16 inserts available as standard), group assignment (to eight centrally placed automated group master faders), channel copy (function selectable), and self-diagnostic circuitry.

Moving fader automation has been adapted from a system originally developed for Sony's *MXP-5000* digitally controlled analogue console, which was only available in Japan. It functions for all long throw faders and their associated mutes, having a 12-bit resolution and ½-frame sampling rate. The system operates with Read, Write, Update, Trim and Manual modes, and includes a Match function that returns the fader to its previous level at a programmable rate. The system also has a Slow mode, which allows data to be written-updated while the VTR is operating in slow motion, even down to frame increments. Full off-line facilities are currently in development.

The rest of the console and interfaced external equipment are governed by a Cue-Scene-Event-MIDI static automation system which has also evolved from the *MXP-5000*. Cues are time-code values (triggers), Scenes are console snapshots. Events are GPI closures, and MIDI refers to programme change. Up to 999 Cues can trigger up to 256 Scenes, 999 Events and 256 programme changes; this information forms an Audio EDL (Edit Decision List) and can be saved along with moving fader-setup data to the consoles floppy disk.

Full dynamic automation on all console functions is something Sony are considering and will largely be determined by customer response. For the moment though they are pursuing the snapshot route, and will soon be offering crossfades between snapshots.

An inherent problem with all digital consoles is the delay introduced by convertors and processing. Although processing delay in the *DMX-S6000* remains relatively small at 600µs, the delay from a 20-bit A–D convertor is typically 2ms. Tests carried out in Japan have shown that some musicians, particularly percussionists, are sensitive to delays as small as 1.5ms during overdubs—amazing when one considers that a microphone placed 1 metre from a source, delays the signal by 3ms. For this reason Sony are not specifying the console for use in music recording studios until more thorough research has been conducted in this area. Other digital console manufacturers take note!

Internal processing delays are compensated for automatically—for example channels without EQ are delayed to match those with EQ. The problem, however, still exists when interfacing external equipment. While this can be dealt with by adding compensatory delay channel by channel, a more elegant solution would be to utilise a scanning system that measures external delays and adjusts the console accordingly. Although hypothetical at the moment, it is a function the desk is perfectly capable of and one that Sony are examining.

Options

Two main options available for the console are those of machine control, and a sampler-shifter module. The machine controller looks very similar to the automation-setup controller, having the same type of screen and a keyboard panel. Up to six machines can be controlled via Sony 9-pin (non 9-pin machines will require a 9-pin to parallel interface); the system will read EDL data and can thus use edit points for locating, auto drop-ins and so on.

The sampler-shifter provides a ten-second memory for every track return into the console, enabling onboard track slipping operations. It also provides up to 80 seconds of stereo sampling, and has a built-in harmoniser. A possible future addition could be reverb processing.

The UK price for the console will be in the region of £140K for the *DMX-S6024* and £240K for the *DMX-S6064*. ■

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

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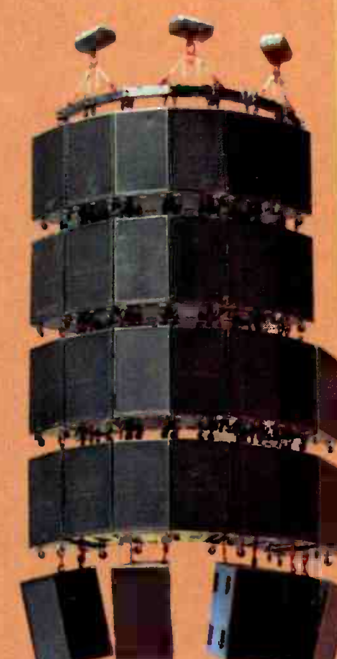


Man on the tube—Vic Keary

THE CHISWICK TUBE

Patrick Stapley visits Chiswick Reach Studio in London and discovers a shrine to valve recording—and the man who worships there

Vic Keary makes no apologies for belonging to the old school: 'It's not that I'm anti new technology, far from it,' he says, 'I simply believe the old stuff sounds a lot better.' The 'old stuff' Keary refers to is a collection of vintage valve equipment—some dating back as far as the 1940s—which he has recently installed into a small West London studio. It is ironic, if not somewhat eccentric, that as the rest of the world strives toward the all digital signal path, Keary and a few like him are busy resurrecting the tube and piecing together all valve studios. However, judging from the response so far, it is an anachronism that works and to put it bluntly, pulls in the punters.

Keary built his first studio in the mid 1950s above a cow-shed in Hampshire which was, 'fine unless you recorded during milking time'. His professional career began at Lansdowne Studios, which he joined at the beginning of the 1960s just as the legendary Joe Meek left to start his own studio. Keary stayed at Lansdowne for two-and-a-half years, gaining valuable experience as both a recording engineer and maintenance man. He then worked for a short period at Rush Electronics, leaving to set up a studio in London's West End called Maximum Sound, and for this he built a console based around the EMI *REDD* range. Due to lease problems, Maximum Sound was forced to move to new premises in South London's Old Kent Road where it expanded from 2-track, to 3-track and finally to 4-track, at which point the studio was bought by Manfred Mann and renamed The Workhouse.

Keary's next venture took him from South to North London where he opened Chalk Farm Studios in 1968, installing his valve desk and a 8-track tape machine. The studio quickly became a favourite with reggae artists such as Dandy Livingston (then Trojan's chief producer) who made many of his records there including Bob and Marcia's single *Young, Gifted And Black*.

'I still think that record sounds good,' reflects Keary, 'considering the only outboard gear we had at the time was a single spring reverb. There were no delay facilities, and we created the distinctive delay on the strings by mixing together the sync and replay heads from the 8-track.'


Details of the custom console

With the arrival of 16-track in 1971, Keary sold his original console to a studio in Blackpool and built himself a larger version (still all-valve). Chalk Farm ran for a total of 14 years offering both studio and cutting facilities, but as Reggae's popularity began to wane, so too did bookings, and finally in 1982 the studio closed leaving behind it a disillusioned and disenchanted Keary.

'It was the start of drum machines and everyone becoming very technical about things,' he recalls. I have to admit that a lot of the fun went out of music for me then—I'd always enjoyed recording music with feel and the buzz you get from people playing together live.'

Through the rest of the 1980s, Keary pursued a number of freelance activities including studio maintenance, antique dealing, disc cutting, removals, and location recording. But it was not until the beginning of the 1990s that he once again became involved in setting up a recording studio.

'It all came about due to a burglary. I'd got to know this chap in Commercial Road, who'd built a studio mainly for his own use. One night he was broken into and all the equipment was literally stripped-out. To cut a long story short, I agreed to install some of my gear (not the console) providing the facility was run commercially. I must say that it surprised me just how strong the demand for the old gear was, and a lot of people said to me you really ought to take this a stage further and put together an all-valve studio.'

Inspired by the reaction and threatened with further lease problems, Keary once again gathered his equipment and moved on; this time to a converted brewery just off the river in Chiswick, West London. The building had been home to various studios in the past, most recently Parsifal, but the space Keary chose had previously been occupied by a small 16-track called The Works. To keep a link with Commercial Road, the new studio retained the same initials and was christened Chiswick Reach.

The facility has an 18 x 12ft control room, a 18ft square studio area with concrete drum riser, and a workshop-office which is destined to become an additional recording room or lounge. So far nothing major has been done either structurally or cosmetically to the original space, other than carpeting the control room (floor and walls), and hanging reflective, hardboard panels in the studio to create a slightly ▶

more live acoustic. The studio was originally designed by a commercial soundproofing company, and has a certain utilitarian feel about it—perforated aluminium sheeting covers a thick layer of mineral wool on both walls and ceiling, giving the impression of a cold storage room.

'The room is incredibly dead, and has amazing separation. People simply don't believe you can get any separation in a room this size, but I've just finished recording an album with the Blues Band (a six-piece) and the separation on that is remarkable. On that album we started off with the room quite live but it was all wrong for their kind of Chicago blues style. Instead we went for that Chess sound, which is essentially very dead with reverb added, and so removed all the reflective panels which produced a much more authentic sound.'

To equip the studio, Keary brought together all the bits and pieces he had collected over the years, including the valve console (28-input; 8-bus; 16-monitor) from Chalk Farm which he spent considerable time refurbishing.

'I spent much longer than I'd anticipated completely overhauling the console and improving on the original design' comments Keary. 'I also fitted some original EMI valve line amps that I'd acquired which sound incredibly smooth and take and awful lot of level.

One of Keary's prize possessions is Joe Meek's 'Black Box'.

'It was actually made for Joe by Racal Electronics in the 1960s. In those days Racal were a tiny outfit based in a small workshop in Camberley, and Joe approached them to make him some equipment. He basically gave them the design for an Altec compressor, which they copied for him, and also an equaliser which was based around a Pultec but with extra flexibility. Only one of these equalisers was ever made and we have it here—it's and excellent and quite unique piece of gear. It introduces a little bit of phase shift and second harmonic distortion, but this is one of the things that gives it a very characteristic sound which is especially good on vocals. I know that Joe regularly used it to lift vocals rather than turning up the level—he would boost at 3kHz which pushes the voice right up front in the track.'

Another antique is a Sean Davies valve limiter made in the 1960s for IBC Studios where Keary claims it was used on early Who tracks such as *My Generation*. Slightly out of place is a Fairchild *Conax* treble limiter, originally used for disc cutting but now functioning as a studio de-esser. Keary is also working on his own designs for valve compressors.

'When I was working for Rush Electronics in 1964, I designed some valve compressors two of which we sold to Pete Townsend—he still uses them, and they're working perfectly with the original valves. I've used that basic design but added some more modern components, but of course kept it all valve. I hope to go into commercial production with it later this year.'

With all this original equipment, availability of the valves themselves becomes of considerable importance.

'It's never usually a problem,' claims Keary. 'There's a company just up the road that imports a lot from places like Eastern Germany, Russia and even Serbia where they're still being produced in large quantities. The only time it's difficult —and this is very rare—is when you have a microphone like a Neumann *U47* where the valves have been obsolete for years. However, it is possible to convert the mic to work with a different valve and adapt the power supply.

'Valve life depends entirely on the valve—for example I've got a couple of mics that have pretty well been used on a daily basis since 1968 and still have the original valve. Some of the valves in the desk should be replaced every 18 months or so, but an input valve like an EF86 will go on for years—I've actually got some in the desk that are 20 years old and still going strong. I periodically test all the valves in the console and replace them as necessary.'

The selection of vintage mics at Chiswick Reach include a Neumann valve *KM54* and two original *U67*s, a couple of Fi-Cord *FC1200*s (made by Calrec), an RCA *1001* ribbon which came from the London Palladium and was used there during the war, two STC *4038*s, plus a couple of STC *4021* ball and biscuits, and an old Tannoy ribbon. There are also a collection of more modern microphones from AKG, Shure, Beyer, etc, and at the top of the studio's shopping list is the Microtech ►

Geffel *92S* valve microphones.

Other valve gear includes four Levers Rich graphic equalisers, an Associated Electronic Engineering graphic, and an EMT plate. The studio also offers a range of non-valve equipment such as Drawmer *DS 201* B Gates, Klark Teknik *DN332* Graphic, Bel *BD80*, Eventide *949 Harmonizer*, two Alesis *Microverbs*, the esoteric Ursa Major *Space Station*, Digitech *DSP128*, Cyclosonic panner, Gain Brains, Keepex and so on. Monitoring is via JBL *4502*s powered by Crown *DC300A* amps and nearfield JBL *Controls* are powered by a Leak *TL25 Plus*. Also available is an Atari *ST* with *Cubase*, and a Fostex *4030* synchroniser for linking together multitrack machines.

Tape machines include a 3M *M79* 24-track and a Brenell *Mini 8* , both made in the 1970s, and a Levers Rich *E242* valve 2-track machine circa 1967. Keary usually works at 15ips and uses a high-level tape.

'I like to run my machines at 15ips because you get a better bottom end at that speed. I use 3M *966* high output tape on both multitrack and 2-track machines, you can really load it without any problems and I find that I don't need to use noise reduction.

'I've also got a DAT machine, but I don't particularly like it. We've done numerous comparisons between the Levers Rich and DAT and the valve machine wins every time. I mixed a recent album to both machines and at the end I got the band to do a blind comparison—to their surprise they all preferred the Levers Rich and found the DAT very brittle. They also couldn't believe the machine was running at 15ips because it was so quiet. Also long as you record with plenty of level, noise is never a problem—the machine itself is very quiet and the valves give out tremendous signal level without any distortion going to tape.'

Connected to the Levers Rich machine is an ancient varispeed box which has been nicknamed 'The Blue Meany'.

'There were a few of these made to varispeed machines with synchronous motors. In fact this one was made for the Rolling Stones and installed at Olympic Studios where it was used on albums like *Satanic Majesty's Request*. When I bought it, I was unaware that there was a slight design fault—it runs at an incredibly high voltage (up to 800V) which has a tendency to leak to the outer case, and if you touch it, you can get one hell of a belt. The problem is that you can't earth it because that shorts the whole thing out and blows transformers. We treat it with the greatest of respect, and try to keep our fingers firmly on the plastic control knob!'

'I've also got a Cadey 1-track machine which was made by a guy called Steve Wadey about 20 years ago. It's a very peculiar design in that the record electronics are valve, while the replay and monitor amps are transistor and sound bloody awful. I'm in the process of replacing all the transistor circuitry with valves means that it will soon be possible to record here without touching a transistor anywhere between microphone and tape.'

The studio also offers a 100-year-old Bechstein piano (not valve), a Hammond *M102* organ with Leslie, and a Sound City *120* valve amp with a 4 x 12 cabinet.

So apart from producing an authentic vintage sound, and cocking a snook at digital, what does Keary regard to be the benefits of the all valve studio?

'In a nutshell, it sounds more realistic,' comes Keary's confident reply. 'There's a weird thing that happens working with valve equipment that I've never experienced with any other setup. The first time it happened to me was at Lansdowne—I was in the control room when somebody spoke to me, I turned round to answer them only to discover they were actually in the studio. The sound was so real that it fooled me. Exactly the same thing happened here shortly after we'd opened; I literally thought that the person talking was sitting next to me—it's an unnerving experience, but a great indication of how natural and true valves sound.'

At rates set at about £250 a day, Chiswick Reach has has been very busy—in fact so busy that Keary has just taken on a full time engineer-maintenance man who, just like the boss, is totally crazy about valves. ■

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B&W's *801*—seen here at Abbey Road—is one example of a hi-fi speaker that has become a pro-audio reference

What are the differences between studio monitors and hi-fi speakers and how do they relate to the pro studio? Philip Newell compares 'home and away'

To the uninitiated, it must seem strange that the loudspeakers commonly found in recording studios are rarely found in the homes of anyone outside the recording industry. It must seem equally strange that monitoring loudspeakers are often put down by hi-fi reviewers as lacking that 'something' that endears loudspeakers to the hi-fi world. How can professionals monitor on devices not necessarily representative what the end result will be heard on?

The answer is that pro monitors do represent the domestic situation, but it takes experienced ears to interpret the representation. If this sounds like mumbo jumbo to protect the cognoscenti, all I can say is that I wish it were, for it would make life easy. The truth is not so simple, as we shall see.

What is right?

In the absence of an accepted reference in hi-fi or pro audio, hi-fi loudspeakers are used in studios for final mixdown. In 1991, I wrote an article in which I looked at the different requirements of the worlds of film, broadcast, classical and rock music recording and a complementary article outlining how the personal priorities of different designers could lead to the production of some very different monitors, despite ostensibly aiming for the same goal: objectivity. An objective 'truth' however becomes a nebulous concept when music is created from entirely electronic sources, where no 'real' sound or acoustic existed, and when no complete performance of the music is heard until mixdown.

If, for the time being, we ignore the concept of a listening 'reality', a monitor loudspeaker is required to highlight potential problems, indicate realistically the frequency balance of the recorded sound, and be capable of revealing distortions of both linear and nonlinear natures. Certainly during the ▶

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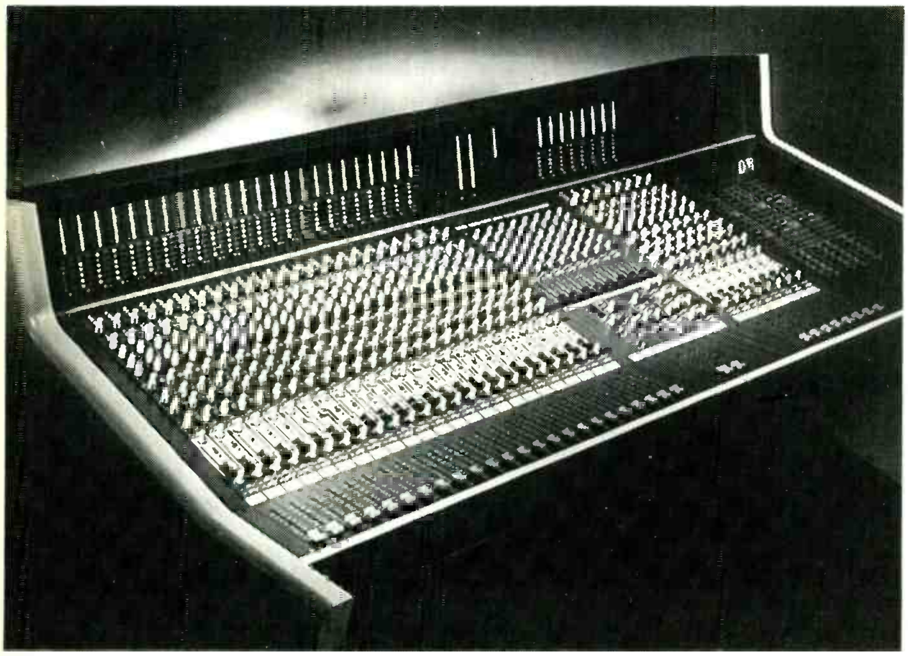

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mixdown stage, 'neutral' monitors should be used, enabling signal processing and equalisation to be applied to the programme free from the masking of monitoring coloration.

These capabilities, although seemingly reasonable, are by no means easy to achieve in any single loudspeaker. If we take the points one by one, each may appear to be the characteristics claimed by the manufacturer of every hi-fi loudspeaker, but in practice they are rarely achieved in any but the most advanced units.

What makes loudspeaker design so infuriating is that written specifications mean so little in terms of perceived subjective qualities. Even a universally accepted specification of tight tolerance would produce a range of monitor performances as wide as the range of manufacturers. The reverse can be even more surprising: wide specification differences produce similar perceived responses under different conditions.

Alignment and loudness

If we take three reputable studio designers (one of whom is myself) specifying their own monitor systems, three very different philosophies emerge. One designer aims to make pressure amplitude responses in the room as linear as possible when octave averaged, (*not* $^1/_3$-octave equalised). Another produces responses around 5dB up at 100Hz with a high frequency response extending well beyond 20kHz. Conversely, I have produced large monitors about 4dB down at 100Hz and 12kHz. There are valid reasons for each of these choices of parameters as optimum for a main monitor system, especially when related to human beings and their

If we take the points one by one, each may appear to be the characteristics claimed by the manufacturer of every hi-fi loudspeaker, but in practice they are rarely achieved in any but the most advanced units

listening experiences.

In the history of research into hearing, Fletcher & Munson produced a classic set of response plots (know as Fletcher-Munson curves). The human aural perception system (the ear and brain), is highly nonlinear—an absolute necessity considering that the difference in power between the quietest perceivable sound and the threshold of pain is 1,000,000,000,000:1. (A billion to one in British English; a trillion to one in America.)

Unfortunately for the loudspeaker designer, this response does not apply equally across the entire range of audible frequencies—hence the Fletcher-Munson curves of equal loudness by frequency. As expected from the log nature of the system, a doubling of loudness is generally perceived from an increase of 10dB, which at 1kHz is well borne out by these curves. Effectively, each curve represents the sound pressure necessary to either double or halve the perceived loudness, as compared to the curves below or above respectively.

From the 90dB and 100dB curves, we can determine that sounds at 100Hz, 1kHz and 6kz will be perceived by the average person at equal levels of loudness. (These are averaged; no two individuals perceive exactly the same responses, save perhaps for identical twins). Under such circumstances, changing the volume control setting from whatever position necessary to produce 90dB to that required to produce 100dB, would, in turn, produce a general doubling of the perceived loudness of most signals present in the audible frequency range. If we back down the volume control to produce 70dB, the 1kHz region will halve by each 10dB reduction, therefore 30dB down on 100dB would produce $^1/_2$ x $^1/_2$ x $^1/_2$ or $^1/_8$ of the subjective loudness at 100dB. At 100Hz, the story is not quite the same. If we follow the 70dB horizontal line to 100Hz, the 70dB and 100Hz crossing point intersect, the curve which would be between the 50dB and 60dB curves. In other words, the 100Hz frequencies would approach the loudness of a 55dB sound in the 1kHz region, so the $^1/_8$ loudness at 1kHz compared to the 100dB level, would be perceived only at about $^1/_{20}$ of the perceived level at 100dB, when listening to the 100Hz sounds. At lower levels and lower frequencies the discrepancies become even greater.

Operational obstacles

In 'domestic' studios, where neighbours live close by and where sound isolation is poor, a general monitoring level of 80dB would be found to be quite usual. In a purpose-designed commercial studio with good sound isolation, 100dB may be a more common reference level. Following a 50Hz line in the Fletcher-Munson curves, we find the 80dB and 100dB equal loudness curves are separated not by 20dB as at 1kHz, but by only about 15dB. Subjectively, the 50Hz region of the frequency spectrum would be raised by 5dB relative to the 1kHz section as the volume control is advanced from 80dB–100dB.

The 'loudness' switch found on some hi-fi systems attempts to compensate for this unevenness of human perception by raising the highs and lows, gradually, as the overall volume level is reduced, and vice versa. There are two circumstances which make this process problematical, however. The first relates to the ear's capacity for 'autocompensation', taking the disproportionalities into their perceptive stride as normal, the second one relates to loudspeaker sensitivity, and distance of the listener from the loudspeakers.

Where amplifiers are purchased as separate items, manufacturers can only guess at the likely sensitivity of the loudspeakers with which they will be used. A guess is reasonable in the domestic world, where the bass and treble controls are usually available to adjust to taste whatever remains to be modified, listening distances are typically a couple of metres and sensitivities around 90dB. But in the world of studio monitors,

> When the additional problems of individual human autocompensations are taken into account, loudness contouring ceases to be a realistic solution

enormous sensitivity differences exist side by side; for example, an ATC *SCM10* nearfield monitor has a quoted sensitivity of around 80dB at one metre for 1W input. The UREI *815* has a sensitivity of 103dB at one metre, for the same input. Each time we double the input power, we increase the output (within the linear response region of the system) by 3dB. Therefore, if 1W into the *SCM10* produces 80dB at one metre; 2W (double) produces 83dB; 4W (double) 86dB; 8W, 89dB, 16W 92dB; 32W 95dB; 64W 98dB; 128W 101dB and 256W, 104dB. 200W would thus produce around 103dB at 1 metre, the same output as would be achieved at the same distance by 1W into the UREI *815*. Distance from the loudspeaker also affects subjective loudness. When the additional problems of individual human autocompensations are taken into account, loudness contouring ceases to be a realistic solution.

Unusual settings

There is no simple solution to these problems, which is why the three designers are pursuing different paths. The designer producing the flat pressure responses, makes *only* very expensive studios, in which he expects thoroughly trained professional staff to be responsible for making their own judgments on how to use the systems.

The designer producing systems with 5dB lift at 100Hz asserts that people use the high monitor levels (100dB–120dB) in order to achieve higher subjective punch but, which is not necessarily required when listening to the end result. Consequently, he concludes that if he provides the bass lift, the punch will exist at lower levels, so can be achieved with less wear and tear on the ear. My experience is that people (including myself) turn up the wick from time to time, come what may. My tendency has been to produce large monitor systems which probably will be used at high levels, with reductions in their responses below 100Hz and above 10kHz which to me, at high levels, relate more closely to the perceived responses of smaller systems which, within their range, are more linear in the 80dB–90dB region.

It can also be seen from the Fletcher-Munson contours, that the high frequencies decrease in perceived loudness as the level drops. Although by no means as pronounced as the sensitivity reduction at low frequencies, the 120dB range of hearing sensitivity is compressed into a 110dB range above 6kHz or 8kHz. This again means ▶

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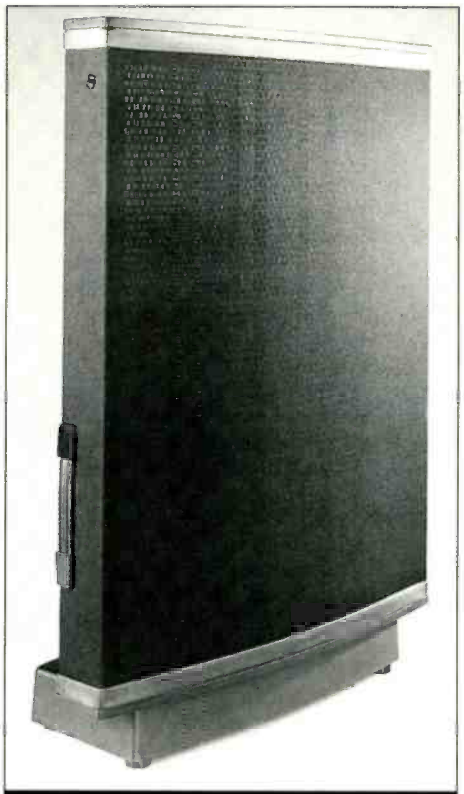
that as the level is reduced, we perceive less top. Conversely, as the level is increased, we will hear more top. My approach of reduced high and low-frequency response when compared to many nearfield monitors, at the same perceived level when sat at a mixing console, will not necessarily correspond in tonal balance. In fact, when comparing any two loudspeaker systems for tonal balance, it is essential that they should be referenced at the same sound pressure level at the listening position, otherwise the Fletcher-Munson curves dictate that their perceived responses will be modified. If the curves are curves of equal loudness, then between 70–80dB a doubling of loudness would occur at 1kHz; each adjacent curve representing what is required to double or halve perceived loudness. However, if we look at the response at 50Hz, the 70dB and 80dB curves are only about 3dB or 4dB apart, which means that in that range of low frequencies, only a 3dB or 4dB increase is necessary to double the loudness. If we are listening to identical pairs of loudspeakers under circumstances where one is 10dB louder than the other, then relatively, the louder pair will appear to have more top and bottom, and will not simply appear to have the same tonal balance and greater volume.

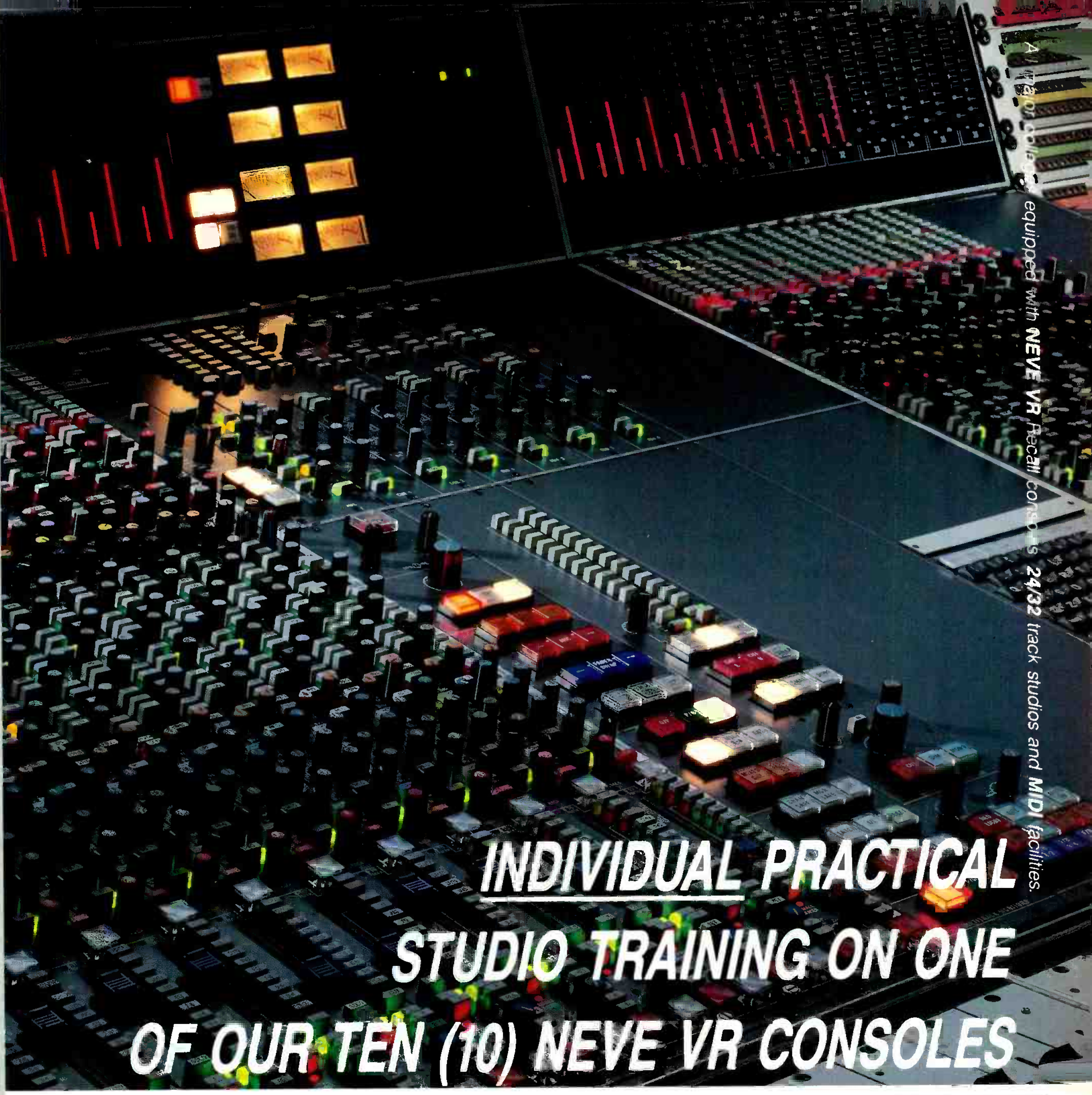
As mentioned before, different people have different abilities to psychologically compensate for these differences, but to a greater or lesser degree, they are there for all of us. All too frequently, people are still to be found comparing small and large monitor systems in unreasonable circumstances. Much of today's recording is in the hands of people who have learned their trade by trial and error, as opposed to formal training. I believe that this is behind much of the frequent rejection of large studio monitor systems by many current recordists.



Quad's *ESL-63* began life as a domestic hi-fi electrostatic design

Let us take the example of a high quality large monitor system being compared at equal level with a near-field system, possibly of hi-fi origin. Almost certainly, the large systems will have an extended low-frequency responses as compared to the small one. As you cannot generate any significant level of very low frequencies from a small cabinet and baffle, this must inevitably be the case. At the top end, as tweeters need to be small for the generation and efficient radiation of high frequencies, the small systems are at no disadvantage. Subjectively, less bass can be interpreted as more top, as the brain often relates to the top as a percentage of the overall tonal balance. A loudspeaker which is slightly deficient in top may be perceived as bright if it is even more lacking in bass. Earlier, we discussed three different approaches of three different designers to the concept of tonal balances for large systems. If we now consider the systems with essentially flat pressure responses from 20Hz–18kHz, then when compared to a conventional nearfield system (both at 80dB), we would probably hear almost two additional octaves of low frequencies on the large ones. As a secondary effect, this extra power of the low frequencies may be further interpreted as being short on top, an idea which may be further reinforced by the very low distortions exhibited by such high quality systems at low levels. Very low distortion can also suggest less top to the untrained ear, which is a principle used in reverse by aural exciters, some of which introduce small quantities of harmonics to give the sensation of more brightness, without actually turning up the top.

When the above large system is used at high level, a significant lift is felt at both ends of the spectrum. At no level is there any true correlation between the large and small systems, but as that particular designer insists, 'Why should there be?' In his view, the large system, especially when used in an acoustically dead control room, is relating to the signal which is being sent to the ▶



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recording medium. The system uses full range monitors of high definition, low distortion and good transient accuracy. This is to monitor what is actually happening, and not attempting to double as a domestic reference (the nearfield system is for that). When monitoring on a large system, one is listening to the whole frequency range, and at levels which can reveal underlying problems which may not be revealed on smaller systems.

When people rely on small monitors, they tend to use general low frequency equalisation for its effect on the 60Hz to 200Hz region which they can hear—yet often what it is going on at 30Hz or 40Hz is unacceptable when replayed on a good quality system—professional or domestic.

Human aspects of design criteria

In the majority of instances, studio designers are called in to advise and to recommend to a client what they consider to be the most suitable solution to a problem. The second designer referred to, makes more compromises to human frailty and market realities. His belief is that people tend to use high monitoring levels in order to achieve an exciting punch, possibly to lift the spirits of the musicians during recording, or possibly from time to time or to see how the music might sound on a large loudspeaker system. This designer believes that higher monitoring levels are unnecessary if the response of the large system is raised by about 5dB around 100Hz. To my ears, these monitors sound like large hi-fi systems with the bass turned up. Compared to nearfield systems at around 80dB, the bass is heavy—at levels over 100dB, the bass is truly thunderous. At 100dB levels, the top also

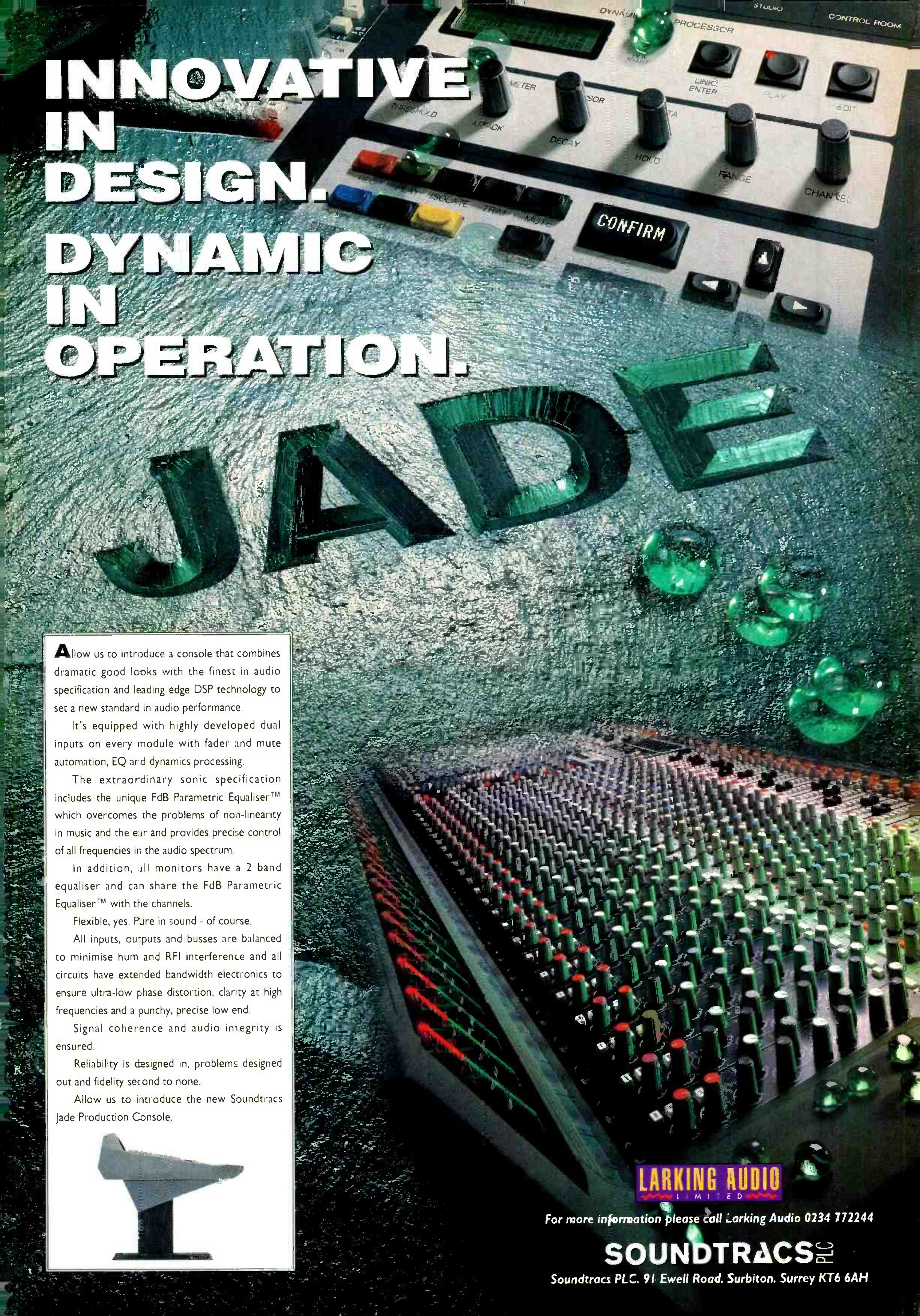
> Large monitors are not there just to sound good; in many instances good quality hi-fi loudspeakers sound more pleasing than do many monitors

becomes excessive. But the designer realises this and takes it into account, making high-level listening representative of circumstances elsewhere. Large monitors give a punch at lower levels, and the designer usually uses a nearfield system of the same family for a domestic level reference, often in further conjunction with the ubiquitous *NS10s*.

My current thinking is entirely the reverse of the above, with bass shelving around 4dB down below 100Hz and continuing down into the 20Hz–30Hz region. The top is also rolled off above 7kHz–8kHz, though again, extending out to 22kHz. My experience behind a mixing console tells me that large monitors are used loud—it is the nature of the beast. I am also currently using control rooms which are quite acoustically dead from the monitoring direction. As a result of this, the room does not add any significant characteristic coloration to the main monitors, so the more 'direct' sound of nearfield systems is close in character to the general nonambient sound of the larger systems. When these large systems are used at the expected high levels of 100dB plus, the ear's own nonlinearities help to emphasise the bass and treble regions in such a way that lows and highs 'missing' from the shelving and roll-off are filled in as would be expected, giving an overall tonal balance akin to nearfield systems running at around 80dB. When the two systems are both compared at the 80dB level, the shelving at lower frequencies is partly compensated for by the extension of the bass to lower frequencies. In other words, though there is less of it, the extended frequency range renders an average level of low frequencies similar to the perceived average from smaller systems. The advantage here is that, as long as monitoring is taking place above 70dB, the full frequency range will still be monitored.

Low-level limitations

The 0dB Fletcher-Munson curve represents the threshold of hearing. This threshold at 30Hz is over 60dB higher than that at 1kHz. Even more sensitive is the region between 3kHz and 4kHz —a fact commonly attributed to evolutionary survival processes. Hence, if one tries to monitor a level of 55dB one would not be able to hear below around 50Hz, though 3kHz could be reduced a further 60dB and still be audible: such are the frustrations of having ears. If we could hear ▶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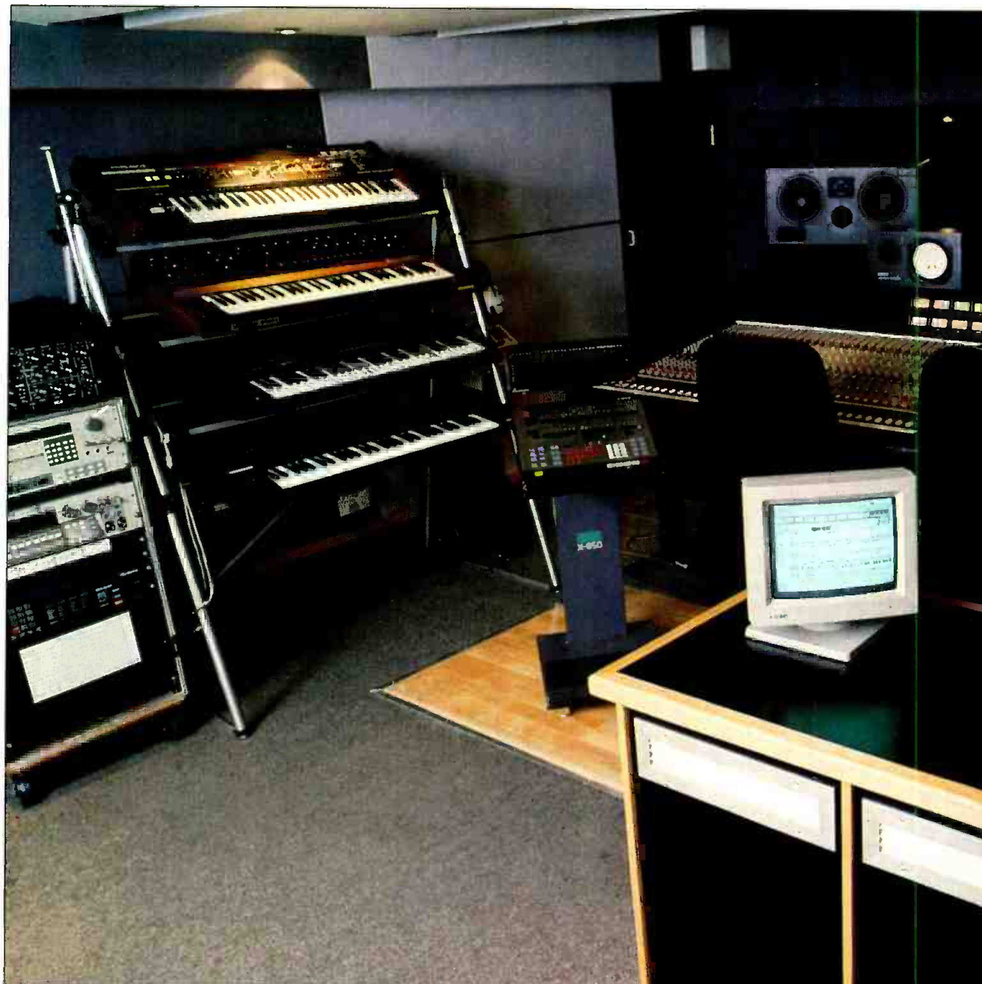


Yamaha's hi-fi *NS10* design has become a worldwide close-field studio reference

10dB at 20Hz, we would probably find our senses swamped by wind noise, distant motor cars, air conditioning plant...

Returning to the case of the large monitors with roll-offs, a high frequency roll-off is an old, unwritten, industry standard. When monitoring at high levels, it has long been considered that a lot of top tends to be tiring and, consequently, to dull the senses. There are various opinions as to the reason why, but the upshot is that if having linear top end induces engineers and producers in to making mixes which may be short of top, it is safer to risk a top-heavy mix by dulling the monitors. A little extra top on a final mix is rarely as objectionable as a disappointingly dull mix.

Conclusions

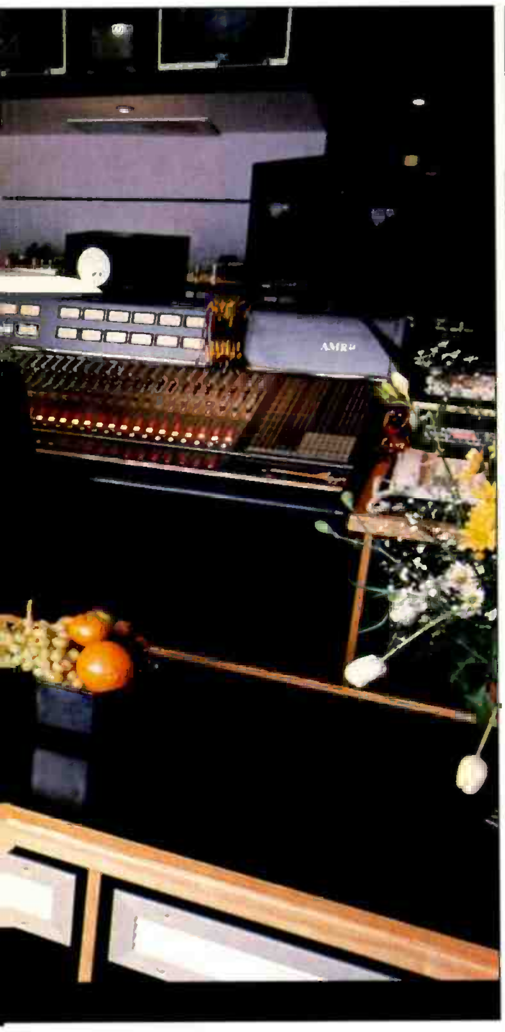
These three concepts are remarkable in their degree of difference, yet the three designers involved have probably, between them, worked on the monitoring of around a thousand studios around the world. Clearly all three have arrived at different conclusions based on their own experience. It may seem absurd that there can exist a 10dB difference in the 100Hz responses between the two extremes of the three, yet all are successful in their own right. They all have their followers, and all appear in studios which produce recordings of renowned quality. It is a question of interpretation—the interpretation of what the large monitors are telling you.

In this article, I have purely discussed one aspect of loudspeaker performance, axial frequency response, and this is only one small aspect of the whole story. Large monitor systems are not just big hi-fi systems, they are tools which, in the right hands, can allow an insight into what is really being recorded, an insight to a much greater depth than can be achieved by using small systems alone.

Many who claim to be unable to 'get on with' large monitor systems are frequently revealing much, not about the inadequacies of the systems, but about themselves. Frequently, they do not like the large systems because those systems tell them things which either they do not want to hear, or which they are too inexperienced to interpret. There are also those who say that they mix on certain small monitors which they feel are representative of the majority of the home systems, but without referring to larger systems, they are short changing the people who pay big money to listen to music at home on a better-than-average system.

Another problem however, is that if designers do not widely disseminate the details of how they intend a system to be used, then the movement of staff from studio to studio can lead to confusion —particularly where somebody has been accustomed to one system philosophy and then moves to another with an entirely different philosophy. Again, experienced persons adjust to a system by listening to a good recording which is well known to them, but reason and logic are not the main pillars of this industry.

The crux of the issue is that the endless requests to make all large monitors sound like *NS10*s at all levels and on all music is neither possible or desirable. If the large system were to sound like *NS10*s but with more bottom, then there would be no need for *NS10*s in a good room. Merely switching filters into the main monitor circuits to mimic the response of the *NS10*s would,



by definition, render the two systems indistinguishable.

Large monitors are not there just to sound good; in many instances it must be accepted that good quality hi-fi loudspeakers sound more pleasing on a greater range of music than do many monitors. But they are designed to sound pleasing on a wide range of music, where Monitors are designed to be as revealing as possible of as wide a range of potential problems as possible. Hence, few monitors find their way into domestic environments. Obviously, the price of good monitors also takes them out of range of the pockets of most hi-fi enthusiasts, as does their size. Wide responses, high damage tolerance, and high quality do not come cheaply. Hi-fi loudspeakers for the domestic market, and monitor system for professional use are distinctly different animals. The two types of reference *must* be used in sensible conjunction with each other, as each one has its strengths and weaknesses.

As can be seen from the preceding discussion, even the 'experts' cannot agree on one 'best' approach. One reason for this is that the problems involved are so complex in their interrelationship, and like many complex nonlinear systems, even minimally different input data will produce seemingly unrelated results. One cannot, however, use this as an excuse for dispensing with large systems. It is not a comfortable problem, but it must be addressed; three ways of doing this are described above. I cannot say which is the best; probably nobody can. People have preferences, but the broad spread of acceptance suggests that in the right hands, each system is workable by those with the skills and experience to use it. ■

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Martin Polon

Data reduction: the gateway to practical digital audio (and video) or the undermining of pro-audio standards?

Mention digital audio data reduction around audio practitioners today and you elicit opinions ranging from approval to absolute disgust. One audio guru went so far as to suggest reduction was like 'putting your music and anything else you hold near and dear into a food blender and hoping for the best!' A studio chief engineer felt that reduction is 'the audio equivalent of buying *filet mignon* and putting it through a grinder, adding bread crumbs and an egg, baking for about an hour in an hot oven and serving it to your guests. It is not *filet mignon* anymore —it is meatloaf. More positive comments range from restrained approval to wild enthusiasm.

But the 'great truth' about reduction at this point in time is that it must exist if we are to record, playback or transmit audio in the digital domain and fit it into storage formats or into transmission 'pipelines' that are size limited.

Let us face the facts: digital audio is an extraordinarily needy format to be stored or sent as computer bits and bytes. It takes about 10Mb of storage space to record one minute of 2-track digital audio information (assuming a 16-bit sampling rate and uncompressed information).

To get some measure of what storage capacity is all about, consider this: the average personal computer performs all of its tasks and stores all of the data pertinent to its use and all of the hundreds of files created by its user on an 80Mb hard disk. Some users do not fill an 80Mb disk in two years of business usage. Yet a hard disk of this capacity would only store eight minutes of unreduced 2-channel digital audio. That means any storage format with less than the (approximately) 66Mb capacity of the compact disc cannot store one hour of playback of 2-channel digital audio. Enter reductive coding to extend the capacity of the consumer audio formats—DCC (digital compact cassette) and the 130Mb capacity MiniDisc —to the required one hour capacity.

However, the need for reduction has entered the lexicon of the professional project and mainstream studio as well. Reduction systems are being used for hard disk storage, digital audio editing, original recording, broadcast 'cart' playback, transmission between broadcast studios and transmitters, between recording studios for postproduction and for remote studio artist insertion on album projects.

Not all reduction is the same, however. When a computer uses compression software to 'double' hard disk capacity, it 'squeezes' the redundant information present in data storage formats. On retrieval and playback, the data is restored. Unless there is a systems crash with associated difficulty in restoring the complete file structure, the recorded information will more or less retain its complete integrity.

Many of the digital recording formats presently being proposed for audio use rely on data reduction via psychoacoustic coding algorithms. These digital filtering schemes remove those portions of the audio signal that are theoretically inaudible to the listener. In recent tests, any number of listening panels and professional users have tentatively found many of the coding schemes used by themselves with a single pass to vary from 'acceptable' to 'very good'. However, the use of several different kinds of coding systems in a 'cascade', as might happen in broadcast production and transmission chains, or during recording studio project work, has caused a great deal of alarm among audio professionals.

Among the red flags raised in testing recorded audio processed with several reduction systems, or in beta testing of new schemes and equipment in complex audio environments, the following have caused the most concern: fragile, almost brittle transient distortion and 'spreading' or 'smearing' of high-frequency information; loss of integrity of the stereo image, including 'swishing' left to right or right to left; intermodulation distortion on certain material and low frequency 'flutter' on some sustained passages (this is clearly frequency dependent); increased perceived distortion at low playback levels and-or through small speakers with questionable linearity; the presence of 'beats', 'clicks' and 'chirps' in the recorded programme material; and the insertion of an 'invisible curtain' between the listener and the music, reducing the perceived impact of digital program dynamics.

In all cases noted above by many separate testers and with a variable infinity of equipment options and hook-ups, the recognition of impairment is aural. The human ear remains one of the most useful tools to explore the impact of reduction systems since the nonlinear characteristics of psychoacoustic coding algorithms renders conventional linear audio test equipment less than useful, at this stage of state-of-the-art test equipment development.

To model the number of potentially signal-degrading variables in everyday audio usage of reduction systems, requires testing various combinations of coding schemes and other non-coding equipment reaching nearly to the infinite. Some possible areas of concern would be any interaction or any combination of the following: single or repeated passes through the same compression coding system; sequential passes through several different coding systems; the use of equalisation and analogue signal processing before using a reduced digital recording scheme; the use of equalisation and analogue signal processing after using a compressed digital recording scheme; the impact of digital delay and signal phase shifting in a signal chain utilising coding schemes; D–A conversion and the obverse analogue to digital process; especially when the conversions occur repeatedly in the signal chain utilising compressive coding for transmission; D–A conversion and the obverse A–D process (especially when the conversions occur repeatedly in and around the signal compressed digital audio recorder); audio delivered for project insertion or other studio usage via satellite, T1 telephone company data carrier, or other transmission carriers where compression coding schemes are used to provide extended bandwidth; the use in completed mixes of various psychoacoustic perceptive enhancements that add warmth or 3-D imaging (and so on) when psychoacoustic coding is also employed.

Problems with compressed sound could be magnified for the majority of the music consuming public

Now, none of this is to say that the use of reduction itself or in any of the above combinations will necessarily produce unacceptable results. There have been some very convincing demonstrations and applications of the technology—especially in transmission and with allied technologies such as film. But there do seem to be two flies in the proverbial ointment. Firstly, there has been a very large body of published and anecdotal reports of problems similar to the guidelines suggested above when there is more than one coding pass, multiple coding schemes or interaction with other signal processing. Secondly, most if not all of the satisfied users of reduced audio are professionals who have characteristically monitored the coded sound with professional monitors of large dimension driven by appropriately large amplifiers.

It is very important to recognise that the nonlinearity of lesser speakers seems to accentuate the possible degradations of cascaded coding errors, could shift the perceived effect of the spectral masking and accentuate the effects of interactions with other audio signal processing equipment in the studio chain. The public listen to recorded audio via a range of small and frequently inexpensive listening devices over 75% of the time. Even when a music consumer actually owns and uses a high quality home music system in the living room, most of his or her listening time will still be spent in a car or bedroom with small speakers and power limited amplifiers. The problems with compressed sound could therefore be *de facto* magnified for the majority of the music consuming public.

What has to be done by the audio industry at this time is to develop a reliable and scientific inquiry into all of the variables of interaction for the various formats. It is vital that we identify any kind of 'invisible' interactions that could render many of the standards of the pro-audio industry defunct. It may be that the audio business needs a coding standard that will ensure compatibility between all equipment which does not exhibit any noticeable degradation or artifact creation in the audio signal. Or it may be that if we fully identify the current problems with the various coding and reduction schemes, we may be able to create a clear agenda for their use . But to allow the incompatibilities to mount, is to court disaster. ■

Conceptual error?

Dear sir, I am disappointed in *Studio Sound* for the publication of the 'Microphone Preamplification' article (*Studio Sound*, April 1993) by FM Acoustics' Manuel Huber.

Although this 'article' never mentions the mic preamp that FM Acoustics manufactures, it is obviously a promotional piece for it. The article should have been treated as an advertisement and labelled as such. I suppose you paid him for this 3-page ad!

A curious side issue is this: considering the fact that my company has manufactured a mic preamp since 1987 called the *M-1*, it is an interesting coincidence that the new FM Acoustics mic preamp is called the *ClassAmp M-1*.

Mr Huber lists many alleged shortcomings of transformers, then concludes that 'Transformers must therefore be avoided in a precision microphone-transducer preamplifier.' Apparently he is not familiar with Jensen Transformers; particularly their best mic-input and line-output models, the *JT-16-B* and *JT-11-BM* respectively. The Hardy *M-1* mic preamp uses the *JT-16-B* and *JT-11-BM* transformers, and has established itself as one of the finest preamps available. Some of the alleged 'major disadvantages' that Mr Huber describes are insignificant, as follows:

'Limited dynamic range': since the noise floor of the *M-1* mic preamp is -129dBu (0dBu=0.557V, 20–20kHz bandwidth, unweighed, 150Ω source termination), and the maximum output level is +25dBu, the total dynamic range is 154dB.

The *JT-16-B* input transformer can handle input levels as high as +8dBu at 20Hz and +12dBu at 30Hz and above.

'Phase errors at low and high frequencies': the *JT-16-B* has <1° of deviation from linear phase at 20kHz. The *JT-11-BM* has <0.5° of deviation at 20kHz. They each have <2° of deviation from linear phase at 20kHz.

'Frequency limitations at low frequencies': referenced to 1kHz, the frequency response of the *JT-16-B* is -0.08dB at 20Hz.

'Limited common mode rejection': the *JT-16-B* has a CMRR of >140dB at 50/60Hz and >80dB at 10kHz. Transformers can typically handle several hundred volts of common mode voltage, limited only by the breakdown voltage of the windings. Transformerless inputs, however, generally cannot handle common mode voltages beyond the power supply voltage of the circuitry, typically ±15 to ±24V DC. 'All transformers are designed to work optimally only when loaded with a specific load resistance': the *JT-16-B* input transformer does require a specific load, but the load is provided by a termination network that is carefully selected and built into the mic preamp. The *JT-11-BM* output transformer can easily handle any load of >600Ω.

Mr Huber claims that '...shocking errors in frequency response and rise time as well as overshoot and ringing can occur'. He provides graphs to substantiate his claims, stating that they present a '...typical high quality audio transformer...', yet he fails to specify the exact model of transformer and does not provide any circuit details regarding the drive circuitry that immediately precedes the transformer. Without this information, his claims are meaningless.

Figure 2 in particular is quite suspicious. Notice that the second square wave has about 150mV of negative offset compared to the first square wave. The ringing at the leading edges is different too. Even a terrible transformer would be expected to behave the same on successive waves of a repetitive waveform, so something is goofy with **Figure 2**! I won't even speculate as to why the notes on the graph say that the input frequency is 998.3Hz and the output frequency is 890.3Hz.

I hope that *Studio Sound* will avoid such unmitigated crap in the future.

John W Hardy, President of The John Hardy Company, Evanson, Illinois, USA.

Manuel Huber's article 'Conceptual Errors in Microphone Preamplification' prompted critical comment from various parties. The above letter is an example; it has been edited for publication as has Manuel Huber's reply.

Manuel Huber replies

Dear sir, thank you for giving me the opportunity to reply to the above letter(s).

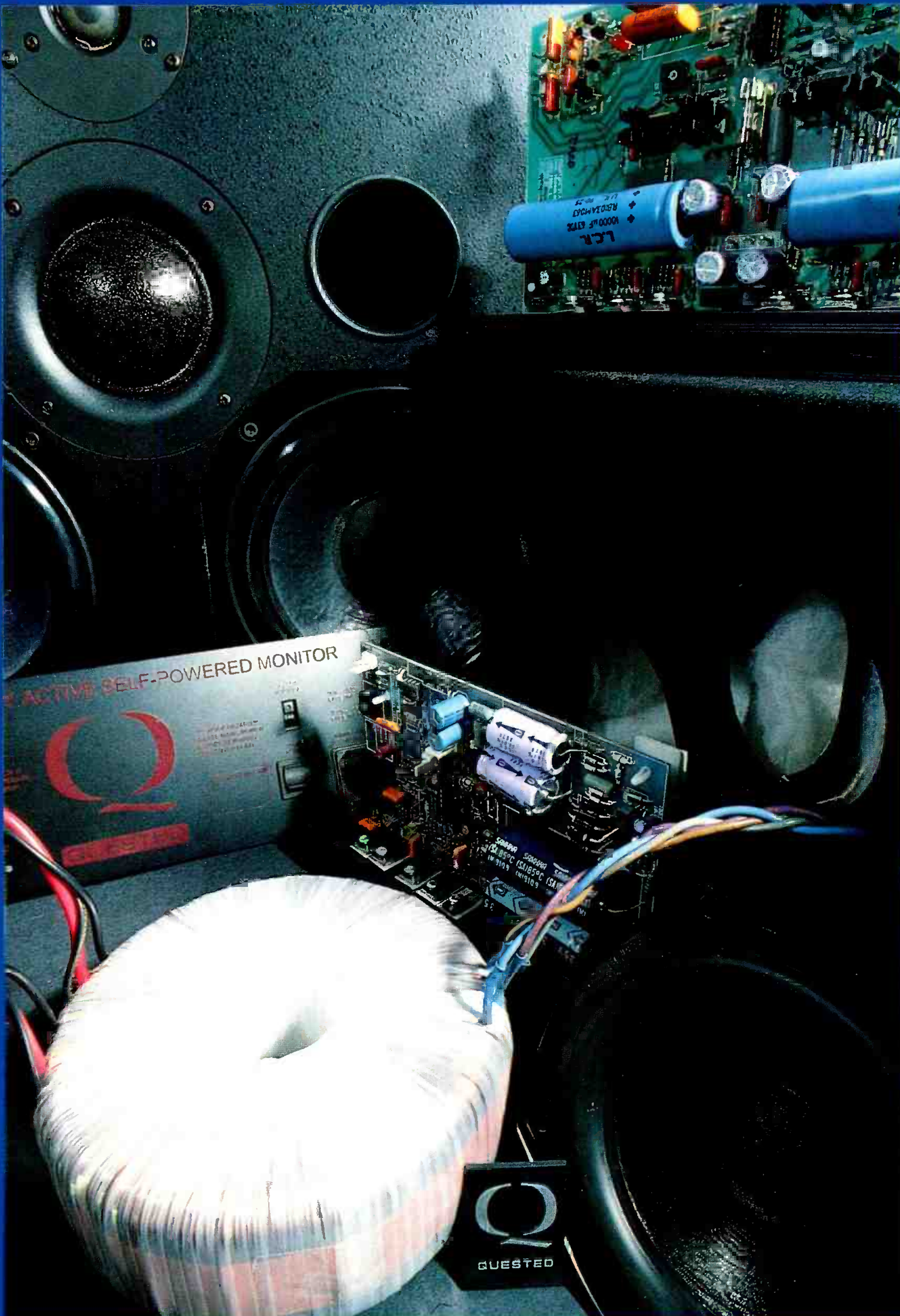
It seems that these gentlemen did not read the title of the article which reads 'Conceptual Errors in Mic Preamplification.' Intended to shed a little light on some of the limitations of current concepts and technology, the idea was to provoke some thoughts on how things could be done if enough attention was paid to relevant facts. The article is based on years of R&D, stimulated by comments from renowned engineers on the limitations of current preamplifiers in recording situations.

In the interests of limiting 'free PR', both Tim Goodyer and myself took care that FM Acoustics was not mentioned in the article and that names of any actual products were avoided.

The background: last winter FM Acoustics asked some of the world's leading engineers, producers and musicians to try beta samples of microphone-transducer preamplifier named *ClassAmp M-1*. As this was not available at the time the article appeared, Mr Hardy should theoretically not have known about it. Could it be that Mr Hardy felt in some way threatened by rumours about the performance of the *ClassAmp M-1*?

As was made clear in my article, there are some applications where transformers are unavoidable (for example wherever galvanic isolation is required, for instance, in PA systems, mic splitters, etc.), so there will still be demand for transformers and transformer-based preamplifiers for years to come. However, in situations that require the most accurate signal preamplification—as in a recording situation—transformers are *not* a requirement. The fact that, in the last 10 years, so many manufacturers of high quality equipment have moved away from transformers confirms this. If Mr Hardy really believes that the 'disadvantages of transformers are insignificant', it may be time for him to start considering the actual reports ▶

THE HEART OF THE MATTER...



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Identity crisis: does the *ADAT* warrant professional support? See 'Professional problem'

from professional users.

If transformers are used to create certain coloration, this is acceptable from an artistic standpoint. One can sympathise with producers who use transformers to create 'warmth' or 'body' that sometimes—for various reasons—is lacking in certain recordings. However, it is the mixdown stage and not at the first amplification stage where such colorations should be added. I am not alone in holding the opinion that the entire signal path to the recorder should be as neutral as possible.

Mr Whitlock's [President, Jensen Transformers—letter not reprinted] statement that 'all the other benefits such as shielding, long line driving can be achieved by any well-designed preamplifier no matter where it is situated' shows that he did not really understand the advantages of having the preamplifier as close to the microphone as possible. If the mic preamplifier is located in, or close to, the mixing desk there are no long lines to be driven (as the recorder is usually only a few metres away). In such a situation, it is the microphone that has to drive the long lines, not the preamplifier. This is exactly the situation that one should try to avoid.

Trying to play down the advantages of having the mic preamplifier near the microphone by simply mentioning some arbitrary dB attenuation levels is simplistic, especially for someone who claims to 'make the best that money can buy.' Nor are the specifications mentioned by Messrs Whitlock & Hardy really meaningful. They are (intentionally?) avoiding the conditions under which these data are obtained: for example a distortion of 0.036% at 20kHz may be a fine figure for a transformer, but at which levels and with what load was it obtained? It is not unknown that in transformers, higher levels give higher distortion.

The 'suspicious **Fig.2**' can be explained—perhaps by Mr Hardy's unfamiliarity with the Neutrik *A1*. As can be seen, the Neutrik displays the 'Input' frequency (the input to the analyser) and the 'Output' frequency (the output of the analyser's internal generator). Because of limited speed of the Neutrik's generator, an external square wave generator was used. The 'Output' frequency on the Neutrik's display indicates the internal generator's frequency which, in this case, is of no importance as it was not used.

The slight 'droop' of the square wave is a characteristic of the Neutrik's display. Looking at the same square wave on a scope, there is no droop or offset.

Mr Hardy can rest assured that we investigate the limitations of measuring instruments and ▶

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Disregarding the motivation behind these letters, what is disconcerting is that manufacturers who seem to think of themselves as 'leaders' maintain such a narrow-minded attitude, apparently succumbing to the 'you hear what you measure' syndrome. It is the listening experience that matters. This business is about audio.

Manuel Huber, FM Acoustics, Switzerland

Professional problem

Dear sir, I was recently involved in what was probably the biggest live recording ever to have taken place in Portugal. The recording of six bands at the Alvalade Stadium were made on 40-track digital for TV, video and possible live albums; specialists were brought in from many countries—in total around 100 people. The tape machines used were five slaved *ADAT*s plus one *ADAT* for backup and stereo feeds.

My first encounter with *ADAT*s and a *BRC* was another TV recording—the whole system was a great surprise in terms of performance. (My son is presently working with a Sony *2234A* and neither of us feel the *ADAT*s' signal is inferior.) Consequently, I had no hesitation in agreeing to use the *ADAT*s for the Alvalade recordings. The company making the recordings owned three machines and another three were hired in. The whole thing was an exciting proposition.

It is possible to record directly to unformatted tapes but I had previously found that they can require up to a dozen attempts to get the things started. Also, during formatting, a tape would occasionally stop for no apparent reason. I therefore considered it safer to preformat my tapes; I calculated that about 130 cassettes would be required, which would take a daunting 30 hours to format. In reality, I spent a couple of days at my hotel lying in the sunshine and changing cassettes every 45 minutes. Two such 16-hour days were quite a pleasant experience—after I had all the machines functioning.

I set the equipment up in my hotel room and found one of the machines refused to format; at first it showed 'Error 2' and sometimes 'Error 5' then it indicated that the write protect tab had been removed from the tapes. The same tapes formatted perfectly in another machine. Further investigation showed that the rogue machine refused to record even on formatted tape. It was 9pm, and there would have been nobody in the Portuguese importer's offices but it was lunchtime in California, so I called Alesis direct.

I explained my problem and was first told to contact my local dealer. I explained the situation and was asked to hold; three or four minutes later I was informed that it was not company policy to divulge the meaning of error messages to users. I explained the scale of the show and the necessity of getting the machine running as quickly as possible. After another period on hold, I was apologetically told that company policy specifically denied me access to the information. I blew my top and went on to suggest that there was nothing further to say except that they would read about this, along with people in over 100 other countries, in *Studio Sound*.

'Don't hang up...don't hang up...' was the reply. Exactly what followed I can't say except that I was 'pointed in the right direction' and within an hour, had all three machines formatting splendidly. Alesis' butt was saved by the ability of one person to see reason.

I do have a certain amount of sympathy with Alesis' dilemma; they have created a machine with a truly professional performance at a price which makes it accessible to amateurs and semiprofessionals. Certainly, many *ADAT*s will find their way into the hands of nontechnical, though professional, musicians. Yet for a system to be professional, it needs more than the word 'professional' printed on the tape loading door. A professional recording system needs professional backup and service, without it, confidence in the system will not be maintained for long by pro users.

As a professional, and especially when involved in the recording of large and costly events, I *must* have support when it is needed. In the past I have used professional setups like The Manor Mobile, fitted with professional equipment with professional support. Without doubt, the signal path through the *ADAT*s is equal to that of the most professional setups, but dare I use equipment which is priced so low as to be accessible to the domestic user if I am going to be treated as a domestic user when I need pro support? I know I would receive 100% backup from Sony, but then Sony don't have have to deal with too many unknowledgable *3324A* users.

As a further result of my discussions with Alesis, I was told that the '301' software installed in some *ADAT*s I was using should not be used with the *BRC*. I was also told to always use the machine with the oldest version software as the master in any synchronised group. The studio that supplied the first three *ADAT*s and the *BRC* I was using had no knowledge of this.Thanks to one helpful person I am now also aware of a potential problem which could seriously jeopardise an expensive live recording. Simply snipping a wire and insulating it will disable a troublesome piece of 'idiot-proofing' which is totally unnecessary for the pro user (though it may save the bacon of a moron).

This has been a salutary lesson for me; it has caused me to consider many thing, and to that one helpful person associated with Alesis I send my respect and thanks and I suggest the management at Alesis do the same as, without such people, I am not sure the word 'professional' should grace your machine. Since this incident, it has come to light that Alesis are not alone in their predicament—there is a general problem with equipment with 'pro' performance also aimed at the domestic market and for which pro backup is not available. 'Please return to your local dealer' is *not* a professional answer to a problem.

Philip Newell, Lisbon, Portugal. ►

In the post

Dear sir, having worked as a production mixer on a freelance basis for half of my 30-year sound career, I am well acquainted with the observations in your May 'Production Sound' article.

Improvements in the technology of typical location equipment has been slow until recent times but ultimately the results committed to tape rely on the skill of the location mixer, and the constraints that exist any location.

The quality of release and display formats have improved so much, Dolby *SR*, Hi-fi VHS and Beta, Videodisc and NICAM, exhibit our results with greater fidelity, but these developments rarely encourage producers to translate this into direct help for the location sound department.

Having spent weeks recording dialogue, for a TV miniseries, on locations where the noise level of waves on a nearby coral reef was around 90dBA, I am convinced that either a 'deaf ear' was turned to the problem, or that it was presumed, without reference to me, that I could handle it.

An obvious reality is that the hungry television systems of the world need more and more programmes and now with tighter budgets the department to squeeze is the sound department, it's simple, tell them that good results are needed and leave them to it!

Any two-person department is hard pressed when dealing with the tasks of: applying radio mics, boom operating, paging radio mic receivers, sound mixing, operating a recorder, as well as operating a playback machine and loudspeaker or a second boom. Yet for complex setups involving extra cameras, sufficient camera crew are available or flown in for the occasion.

Yes, the suggestion of an overall sound 'minder' is a good one particularly as the awareness of a director, regarding sound, is often eclipsed by the many other decisions and pressures that he or she faces during shooting.

Even the introduction of new technology poses a hurdle. Having imported a Fostex *PD-2* recorder earlier this year, I am now faced with attitudes of 'we can't afford that', rather than positive responses that applaud the better sound quality and the time code advantages that relate well to AVID editing systems and modern postproduction techniques in general.

I have worked on three totally postsynchronised feature films and although a result of controlled quality can be produced with actors performances modified if needed, the depth of detail and feeling of a location sound track is difficult to totally recreate, even when time and budget is available.

In three weeks time I start work on a North America series to be shot on the South Island of New Zealand. The show shoots for five months—a long time to be away from home! I asked, 'How much time have I for preproduction?'.

The answer was short and delivered with an air of munificence, 'One day, and that includes your travel down here!'

In this business you soon realise that good sound is not noticed, but bad sound is, and that the most creative part of location sound recording, is writing out your invoice.

Fond wishes to all at *Studio Sound* and thanks for a balanced, informative, sensible, turned on magazine.

Mike Westgate, Sound Ltd, Auckland, New Zealand.

Cut to fit

Dear sir, looking through the June 1993 issue of *Studio Sound* over breakfast, I read again with interest your Editorial ('The Main Event'), about the 'honesty' of modern recordings in the light of current technology's ability to edit and otherwise manipulate musical material.

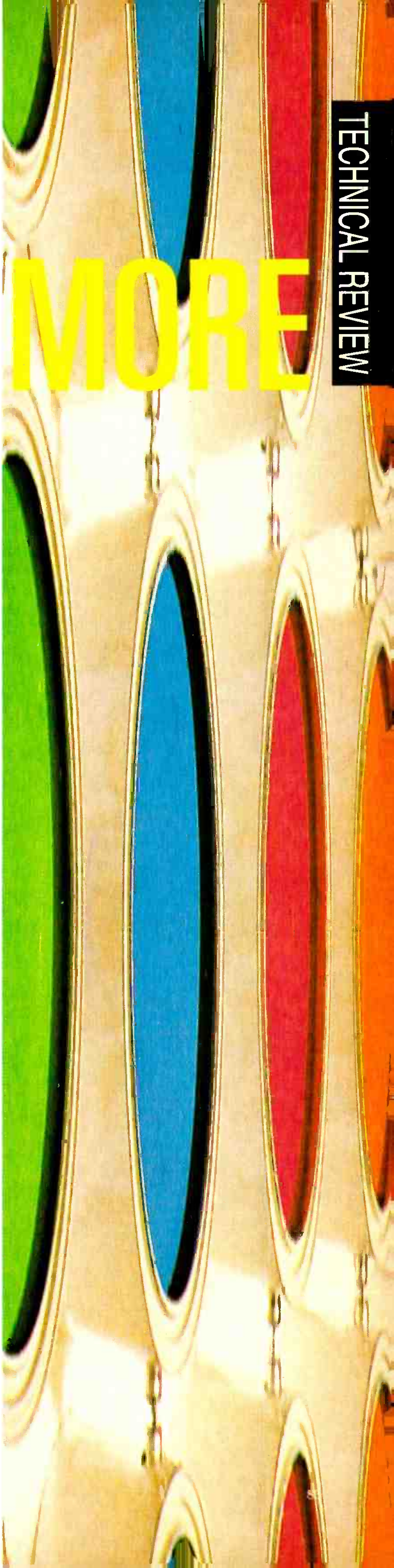
I suppose that one's viewpoint will depend to some extent on the kind of music one works with but as someone who is mostly involved in the recording and editing of classical music, I cannot help but be drawn to sympathise with your last statement that, '...the only rational conclusion to be drawn is that the purpose of the majority of recording sessions is now to deceive the listener rather than present them with a genuine record of a musical event'. I might, however, substitute 'editing sessions' for 'recording sessions'.

What prompts me to write is that I have nearly finished editing (I hope) an album I recorded for a local amateur choir. According to the play list from my *Sound Tools* editing system, there are 244 edits in a 64-minute file. This averages out to an edit every 15 seconds. One 2-minute piece has 25 edits in it. The raw data with extra bits (clones for pitch shifts—never tell an amateur producer-director that you can do this because it rarely works!) is over 100 minutes long. Perhaps you cannot hear the individual edits, but the cumulative effect of so much editing must rob the music of some of its life, even if the end product is 'technically' superior.

While I acknowledge the need to edit in order to eliminate minor imperfections an inevitable slips which might go unnoticed in a live performance, I find myself agreeing more and more with the folks at Nimbus Records who, when I visited them about ten years ago, stated: 'We will edit to save a performance, not to create one.' In other words, get your act together before you come in to record —don't expect us to patch it up afterwards. In a past issue of *Studio Sound*, Decca were quoted as saying that there are 100 hours of editing in many classical albums. No doubt the increased time expected on CD as against LP and the cost of hall rental, musicians' fees and so on can make it difficult to allow enough time for extra rehearsal, but perhaps we are risking less honest recordings in our quest for technical perfection. If the performer ends up sounding better on disc than in real life, surely there is something wrong.

In closing, I am reminded of a story about a famous pianist who spent many days recording and even more days editing. During the final playback the producer turned to him and said, 'Don't you wish you could play like that?'

Stuart Tarbuck, Mobile Audio Recording Vancouver, Canada. ■



LESS IS MORE

Beginning with an examination of digital video data reduction, Francis Rumsey introduces a series of articles addressing technology surrounding audio

Audio is no longer an island, but a component in the larger field of information or media communications. Although the audio engineer's work will remain principally with sound, there is an increasing need to be aware of what may best be called peripheral technologies—such as video and computer systems—since many jobs and products today incorporate elements of these alongside audio. Here we begin an occasional series dedicated to demystifying these peripheral technologies, aimed specifically at those whose primary place is in audio and thus who may not be familiar with the principles involved. The intention is that audio professionals may become better informed about issues which may affect their work, especially as the need grows for greater diversity in the profession.

The first article in the series deals with a topic of increasing importance: that of bit-rate reduction (BRR) for digital video. It precedes an article which will examine the most recent digital video recording formats and their audio capabilities, since a number of these formats use video BRR.

Considerable coverage has been given in the last few years to the principles and applications of digital audio BRR, and parallel to these developments have been similar ones dealing with digital video. Video BRR is important for many of the same reasons as audio BRR: greater economy in storage and transmission; easing of mechanical constraints in recorder design; efficient use of bandwidth for broadcasting; and ease of use with consumer media and personal computers. In fact many of the same principles apply to video BRR as to audio, since both rely on the removal of redundant information from the signal, to reduce the data rate with as little perceived effect on quality as possible. If the reader understands some of what has been said in previous articles on audio BRR, then many of these concepts will be transferable to the subject of video—it being simply necessary to review what is meant by such terms as frequency when applied to pictures.

Although it is not intended to cover the principles of video and its digitisation in detail, a short summary will be given before going on to look at ways in which the bit rate may be reduced.

Picture this

A video picture, such as might be viewed on a domestic television, is made up of horizontal lines, created by a spot scanning the screen from top to bottom. Each screen full of lines represents a still frame, and television systems often use a method of interlacing two sets of lines (each called a field) to make up one frame, presenting one field with gaps between the lines followed by the other filling in the missing lines to reduce flicker (**Fig.1**). The field rate is thus twice the frame rate. There are 625 lines per frame in current European systems and 525 lines in America and Japan. The frame rate is the number of complete still frames per second making up the moving picture, being 25Hz in Europe and 29.97Hz in countries such as America and Japan. This is called 'conventional definition television' or CDTV. In computer systems, images may be scanned and stored at a number of different rates. High definition television (HDTV) doubles the number of lines in the picture in order to increase the resolution.

In audio, the term frequency relates to the pitch of a sound, being the repetition rate of the waveform. The human ear is most sensitive to ▶

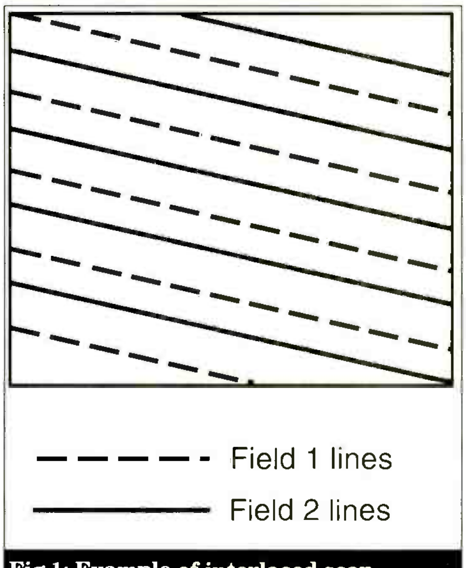
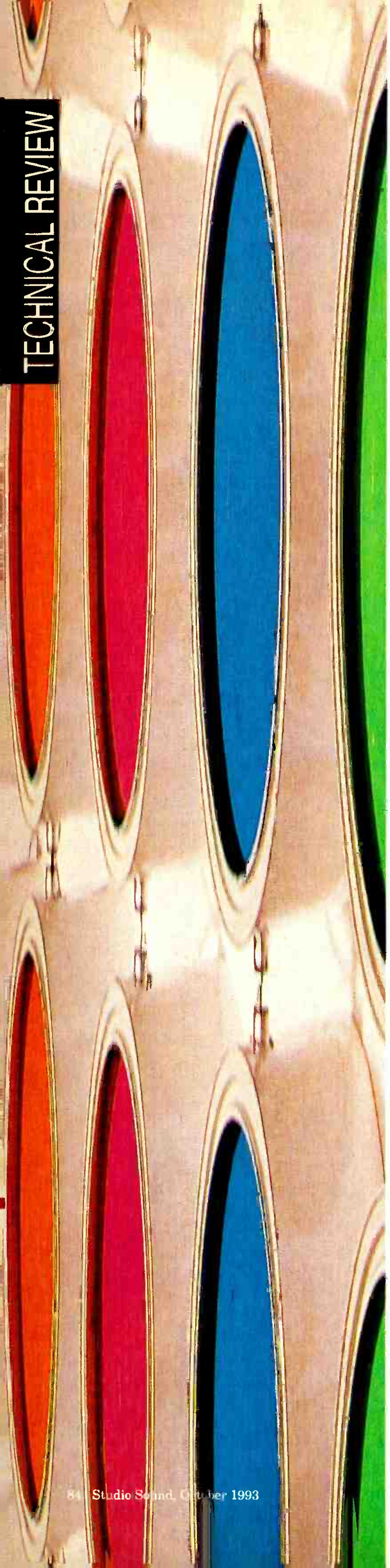


Fig.1: Example of interlaced scan



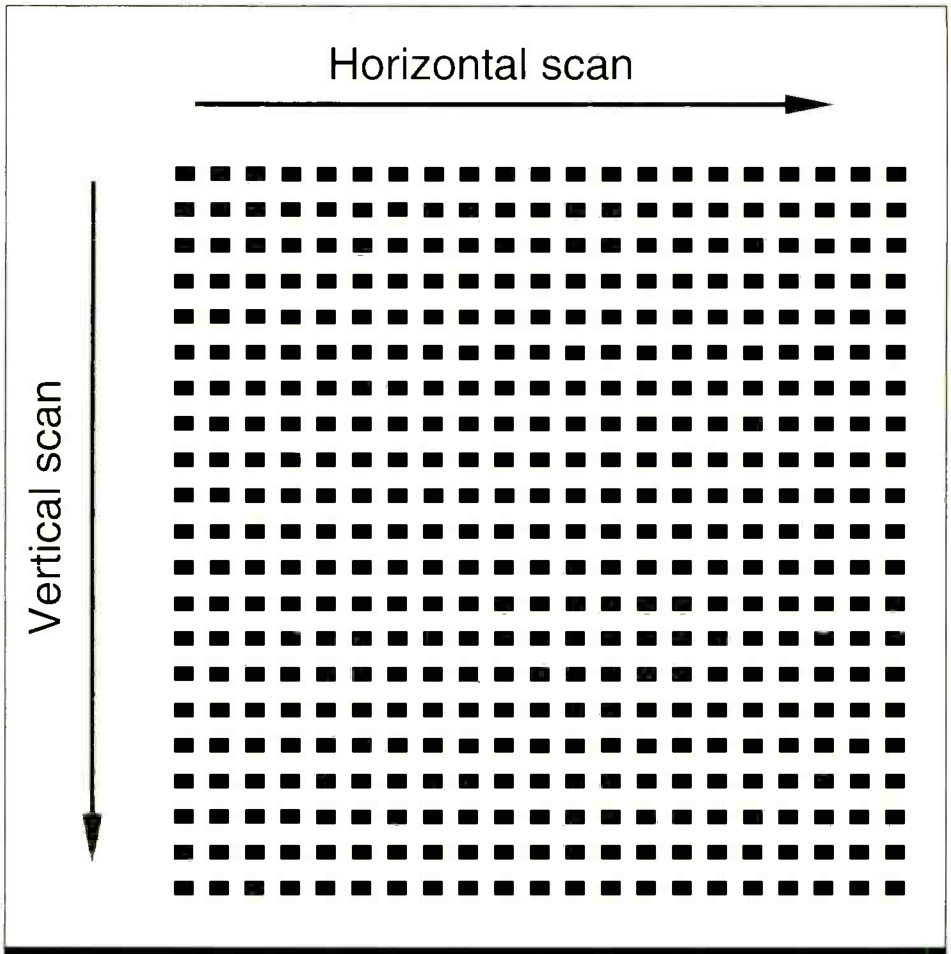


Fig.2: A picture can be considered as an array of pixels or picture elements with a horizontal and vertical dimension. Spatial frequency refers to the rate of change of an image in either direction

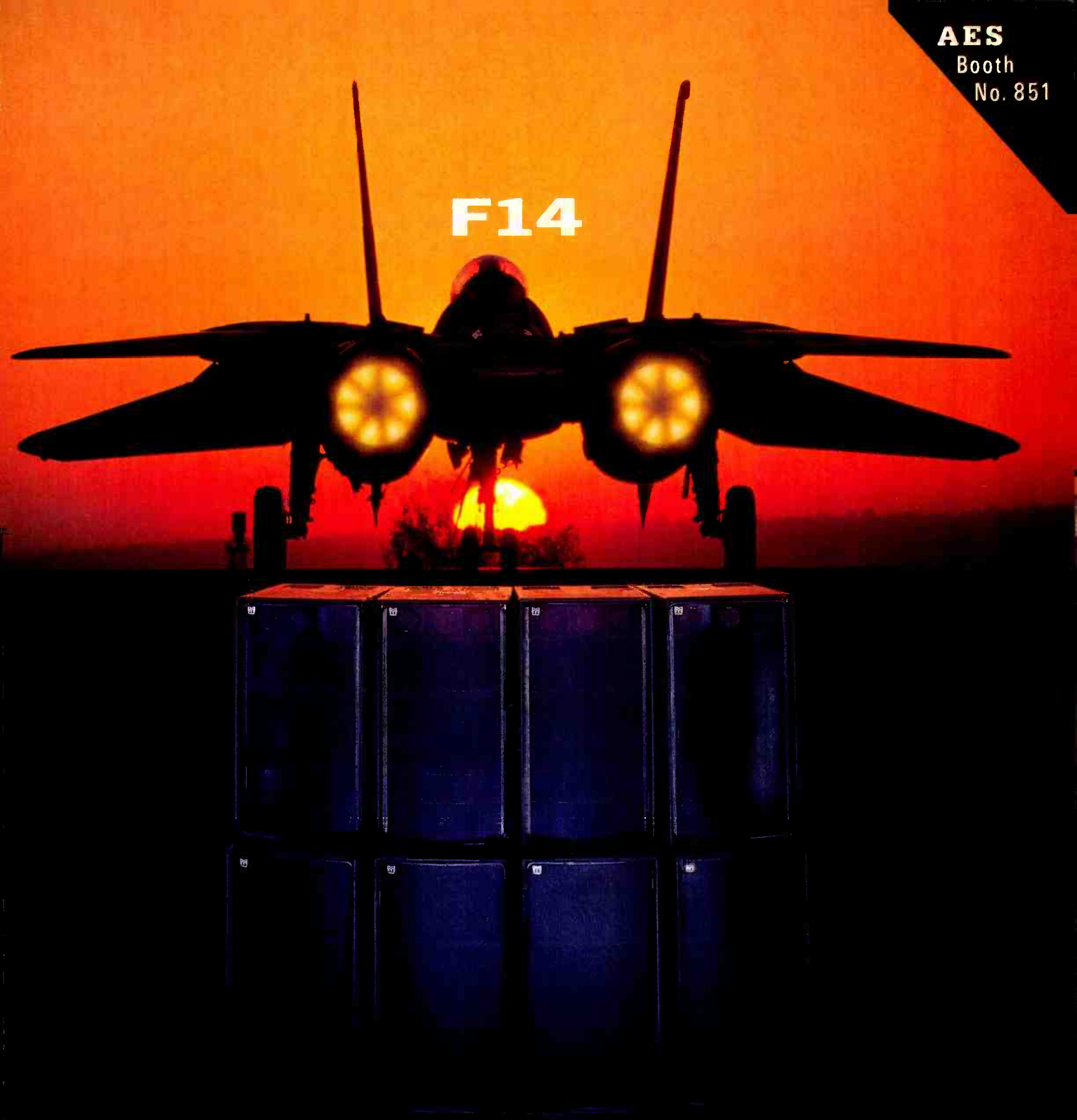
middle frequencies around the 3–4kHz region, and becomes increasingly less sensitive towards both extremes of the spectrum. If the high frequency response of an audio signal is reduced it becomes less 'bright' sounding, and more 'muffled'. In video, the term frequency relates to the rate of change of visual information, but there are two dimensions in a still frame since the picture may be considered as an array of points or pixels (**Fig.2**) in the horizontal and vertical dimensions, thus the concept exists of 'spatial frequency'. There is also the concept of 'temporal frequency' which relates to the way in which the image changes over successive frames, as time passes. A high spatial frequency may be imagined as the fine detail in a picture, where picture elements change from light to dark in close succession, whereas a low frequency may be considered as a coarser pattern (**Fig.3**). The human eye is less sensitive to high frequency visual information than to low frequencies.

In analogue video, the horizontal dimension (a line) may be considered as a continuum of graduated shades from light to dark (in the monochrome sense), but the vertical dimension is effectively sampled (although not quantised) by the line structure. Digital video superimposes a sampling pattern on the horizontal dimension also, allocating a binary value to each pixel in the matrix, depending on its brightness. To represent colour, three video signals are normally required —red, green and blue (RGB video)—from which other colours and white may be formulated, but these are often combined in such a way as to produce a luminance (brightness) signal (Y) made up of an appropriate combination of R, G and B, and two 'colour difference' signals, namely R–Y (Cr) and B–Y (Cb), from which R, G and B may be reconstituted by matrixing them with Y. Cr and Cb can get away with a considerably lower bandwidth than Y, because the eye is much less sensitive to reduced colour bandwidth than it is to reduced luminance bandwidth. This colour difference format is often called 'component video' when referring to recording and transmission systems.

The term 'composite video' refers to a system in which the colour difference components have been modulated onto a subcarrier and combined with the luminance signal, creating one waveform in which the colour information is spectrally interleaved with the luminance.

Digital video

The CCIR 601 standard determined a basic sampling frequency for component video of ▶



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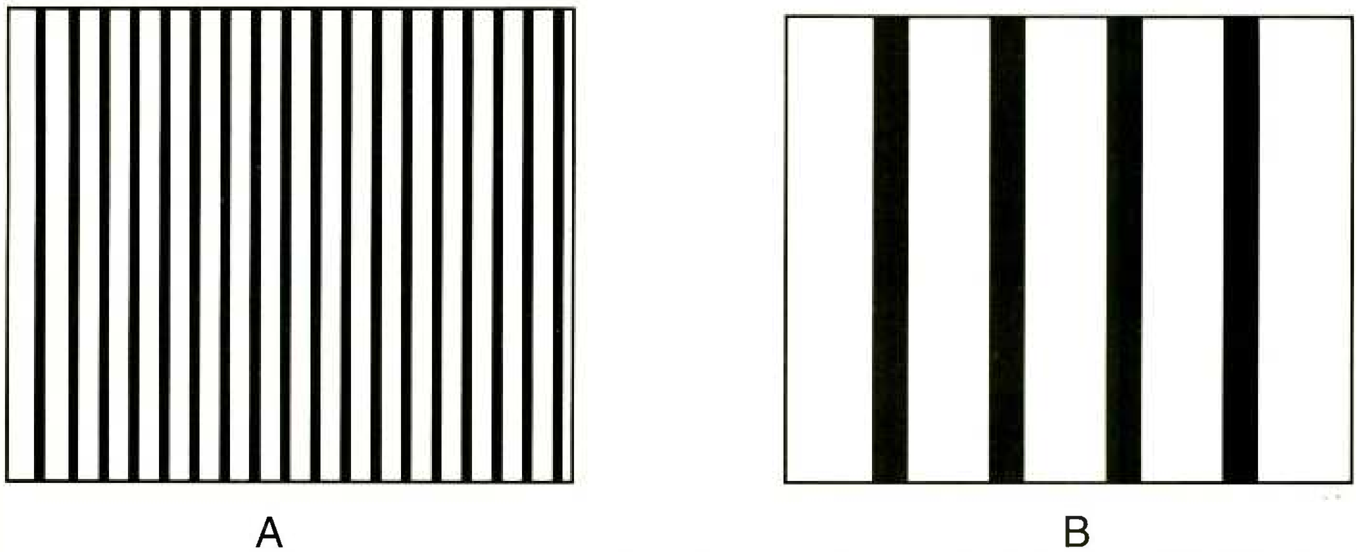

Fig.3: Image A has a higher horizontal frequency than image B

3.375MHz which can be used in various multiples for luminance and colour signals. The so-called 4:2:2 format for video sampling uses four times 3.375MHz (13.5MHz) for the luminance component, and twice it for the colour difference signals (reduced bandwidth). The same sampling frequency is used for both 525 lines/59.94Hz TV systems as for 625 lines/50Hz formats, since it is a multiple of the line frequency of both, thus simplifying the design of machines designed to work with both formats.

Composite video is normally sampled at a different rate of four times the colour subcarrier frequency, thus resulting in different rates for 525-line and 625-line video. This is due to the need to decode the colour information digitally, and simplifies the design of the required filters.

Typically, digital video samples are quantised to either 8 or 10-bit resolution—considerably lower than the 16 to 20-bit resolution used in audio. This is because the signal-to-noise ratio required for high video quality is not as great as that required for audio.

The bit rate which therefore results from a 4:2:2 component digital video signal of either standard, sampled to 10-bit accuracy, is therefore 270Mbit/s. For composite video it is rather lower: 152Mbit/s for 625lines/50Hz. For HDTV, the bit rate for studio quality pictures is over 1000Mbit/s (1Gbit/s). Thus the data rate of digital video exceeds that of digital audio by between 200 and over 1000 times.

Reducing the bit rate

There is good reason to consider reducing this bit rate for certain applications because of the economies which may result, and also because of the need to transmit digital video (either CDTV or HDTV) in a limited bandspace. There is also a growing interest in using digital video in consumer products such as the various CD formats, and there is great potential in having digital video at a bit rate low enough to be handled and stored by personal computers. As with audio data reduction, video data reduction is a business where virtually any compression factor is possible—but it rather depends on what you are prepared to put up with in terms of final quality!

To give a few examples: using MPEG BRR (bit rate reduction) it is now possible to represent what the interactive media industry calls full motion video (FMV) at a bit rate as low as 1.2Mbit/s, which is very similar to the rate required for full resolution audio. This represents a compression factor of over 100 times from the full composite data rate for PAL TV, and it is used to store conventional definition video onto CD-I for domestic distribution. Granted, the picture quality is only passable, but it stands comparison with VHS video.

It has also been shown that HDTV signals may be reduced in bit rate to around 70Mbit/s with acceptably high picture quality for the majority of viewers, and tests have shown that it would be possible to carry digital HDTV in a conventional 8MHz broadcast channel (as used for current analogue TV broadcasts) using a suitable modulation method, which is indeed an achievement.

Although the MPEG group has standardised the very low bit rate versions of picture compression which will be used by the consumer CD industry and others, it is still working on the algorithms which may be used to code high-quality conventional definition TV pictures at rates between 2 and 10Mbit/s, and the BBC have ▶

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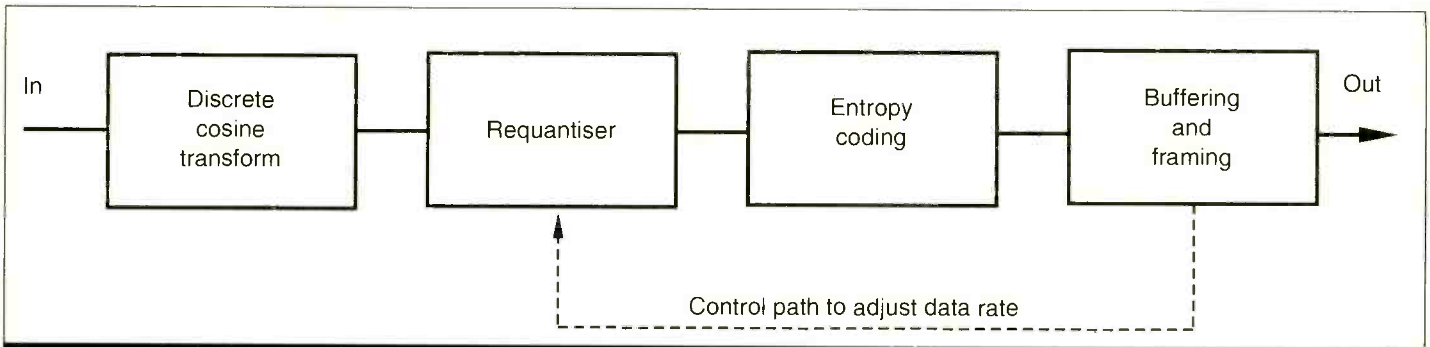

Fig.4: Block diagram of a single video BRR encoder

recently shown HDTV pictures compressed using a similar algorithm at a rate of 25Mbit/s. Even at such a low rate, the picture quality was in no way unwatchable, but showed clear artifacts of the BRR process.

So how is this rate reduction achieved and what is the difference between MPEG and JPEG? The following will give a basic overview.

Video BRR principles

A block diagram of a simple coder for reducing the bit rate of digital video is shown in **Fig.4**. It shows only some of the techniques described below, and indicates the possibility for introducing some control feedback to the quantiser in order to keep the output bit rate constant.

The DCT

Most approaches to video BRR divide the picture up into groups of pixels, changing it from a line structure into a block structure (**Fig.5**). A common block size is 8 x 8 pixels, although systems using other possibilities such as 16 x 16 or 8 x 4 are in evidence. This block structure may often be seen in still pictures which have been data reduced if they are magnified considerably, or if a very high compression factor has been used. The sample values corresponding to the pixels represent their amplitudes (or brightness in a monochrome picture). The block is then subjected to a discrete cosine transform (DCT) as in many audio BRR systems, which results in a frequency domain representation of the block, whose sample values now represent the relative strengths of different spatial frequencies in the block, both horizontal and vertical. This is an efficient approach to BRR in itself since many of the spatial frequency components will be near zero in the typical picture block, with only a few of significant amplitude which will be stored or transmitted.

To give an example, a block with a finely detailed black and white pattern would result in a high frequency component at the pattern repetition frequency after the DCT process. Most other components would be zero. Applying an inverse DCT would restore the original spatial amplitudes of the pixels. If one were to remove the zero or near zero samples in the frequency domain, then perform the inverse DCT, the reconstituted ▶

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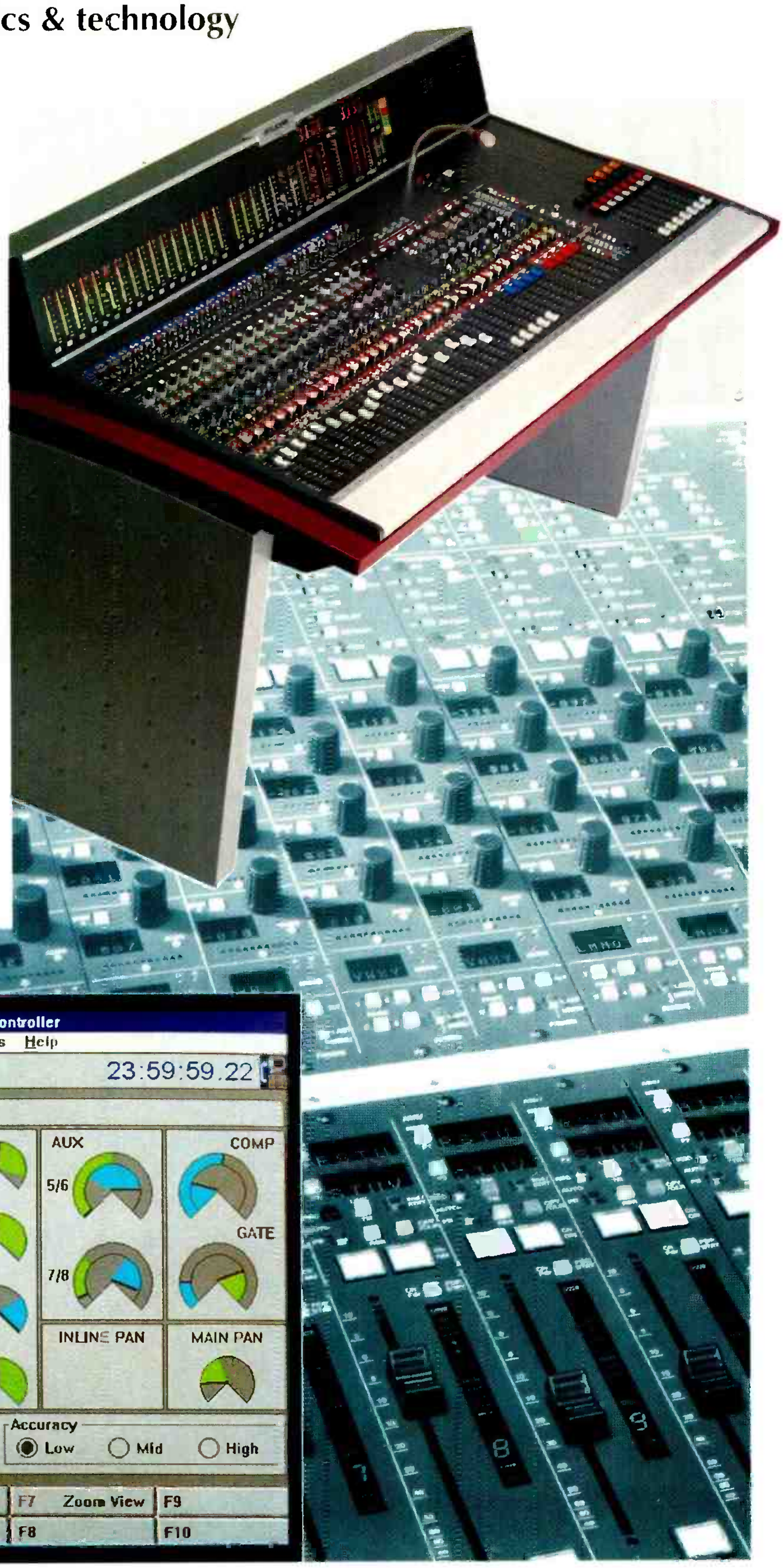

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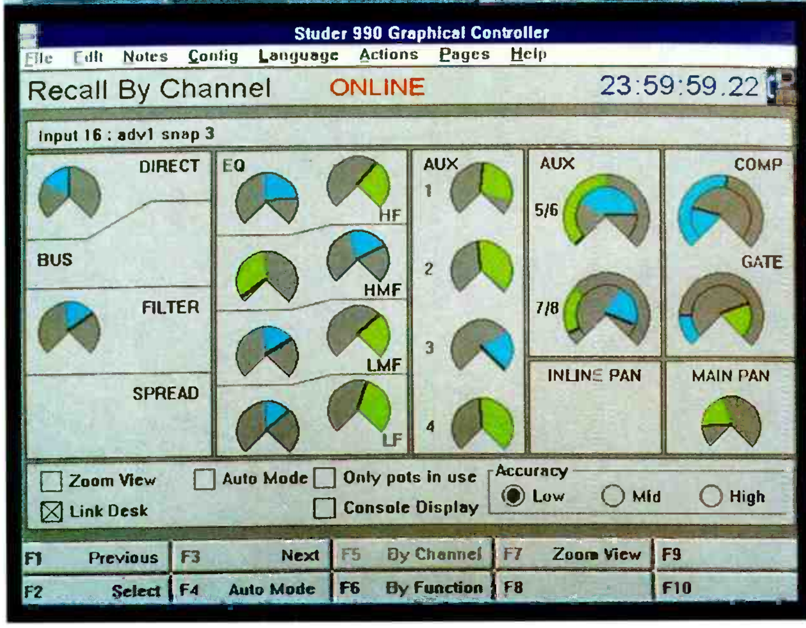

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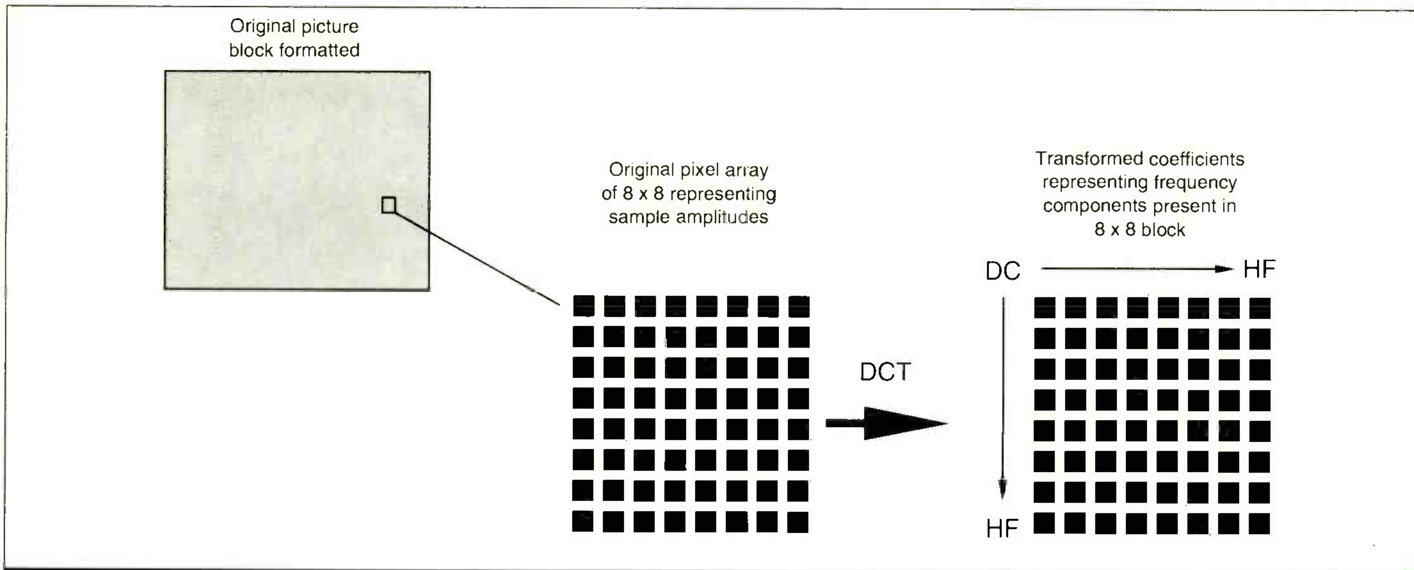


Fig.5: Block formatting and DCT results in an array of coefficients representing spectral components

picture would probably look almost identical to the original. Even without any requantisation this technique can result in a significant reduction in the amount of information required to represent a block.

Psychovisual model

Since the eye is less sensitive to high frequency information and to colour information it is possible to allow more noise in both these regions than in the low frequency luminance information. Thus the spectral coefficients produced by the DCT may be weighted and requantised according to a table which is based on the visibility of noise. These weighting and quantisation parameters may be varied to control the average bit rate of the audio signal, which would otherwise vary considerably from block to block.

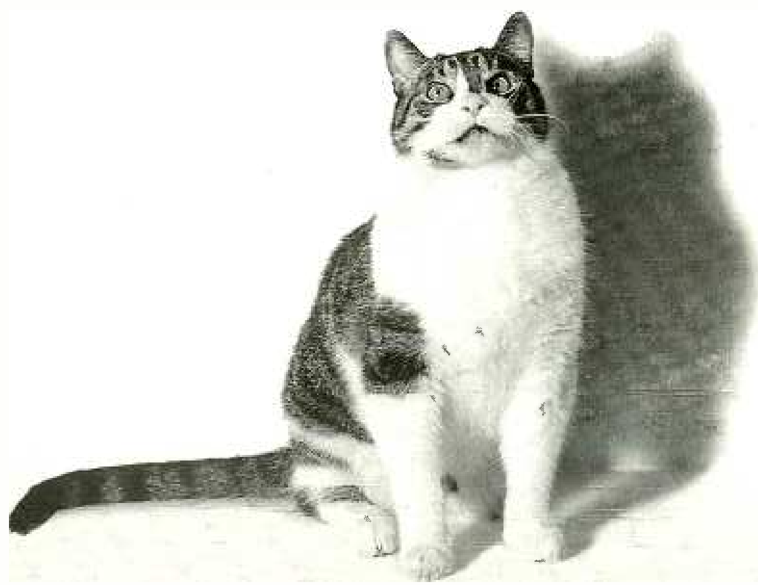
Entropy coding

The spectral coefficients, thus requantised, may be subjected to a variable length or Huffmann coding process which assigns a binary code of a certain length to each sample. Based on an analysis of which bit patterns occur most often in typical TV pictures it is possible to assign the shortest codes to the most common patterns, and the longest to the least common, thus introducing a further saving in data rate.

Frame coding

In addition to the aforementioned methods, it is also possible to achieve even greater reductions in the bit rate by analysing the temporal aspects of the moving picture, attempting to predict how the picture changes over a number of frames. This is called interframe coding. When coding is simply performed within the field or frame itself, the process is called intrafield or intraframe coding. Which of these methods is appropriate depends on the type of data reduction process in use, the application, and often there is some analysis of which will provide the greatest reduction in bit rate after variable length coding, allowing the system to switch to the most suitable 'on the fly'.

The interframe mode relies on attempting to predict the sample values in the block, based on the same block or a motion-compensated version of it in the previous frame, performing the DCT only on the error or difference between the predicted ▶



and the actual sample values. Basing the prediction on the same (co-sited) block in the previous frame will only be useful if the picture is static. If it is moving then it will be necessary to predict the direction in which a block (representing a picture element) is moving, by making searches in each direction and comparing the current frame to the previous frame, attempting to find the best match with the current frame's block. A displacement vector can then be determined, and transmitted along with the coded picture coefficients, so that it may be used by the decoder to form a similar motion-compensated prediction.

Such motion compensated interfield coding requires that a simple intrafield mode is used every so often for each block, in order that prediction errors do not accumulate. In some computer-based systems, such as Apple's *QuickTime* video compressor, a similar approach is used, called 'frame differencing', whereby only the differences between successive frames are coded, it being possible to set the distance between 'key frames' which are full frames rather than difference coefficients. The less frequently these 'key frames' are stored, the poorer the picture quality. Interframe coding does not always result in savings in the bit rate, since not all images move in a predictable way, and backgrounds may be confusing, and in such cases intraframe coding is more appropriate.

> The Moving Picture Experts Group have still to finalise algorithms for higher bit rate CDTV and for HDTV

Skip field

A technique also used in some nonlinear video editors involves coding only every other field of the picture, repeating the first field with a small vertical shift to create the second field of each frame on replay. This clearly halves the amount of data to be stored but also halves the vertical resolution of the picture. It can be adequate for deriving a picture for off-line editing, but would not be used for broadcast quality video.

Buffering

All of these techniques applied to a typical video signal will tend to result in a bit rate which varies considerably with time unless something is done to keep it constant. In computer still-picture storage applications it may not matter that the disk space taken up by one picture is different from another, but in most transmission and storage applications it is necessary to have a constant bit rate. This may be achieved by buffering the output of the coder using memory from which data is clocked out at a constant rate. The fullness of the buffer can be used to control the requantiser or coefficient weighting earlier in the chain, adjusting the accuracy of requantisation of the DCT coefficients to suit the required bit rate at that instant. Alternatively a form of forward prediction may be used to estimate the entropy of the final coded signal, and this used to control the quantiser to maintain a constant data rate.

MPEG or JPEG?

As mentioned above, the Moving Picture Experts Group have standardised algorithms for very low bit rate picture coding, for consumer applications, but have still to finalise algorithms for higher bit rate CDTV and for HDTV. MPEG-style coding may involve prediction and frame difference techniques such as those described above, and thus is most appropriate for transmission or replay of video programme material where the picture sequence ▶

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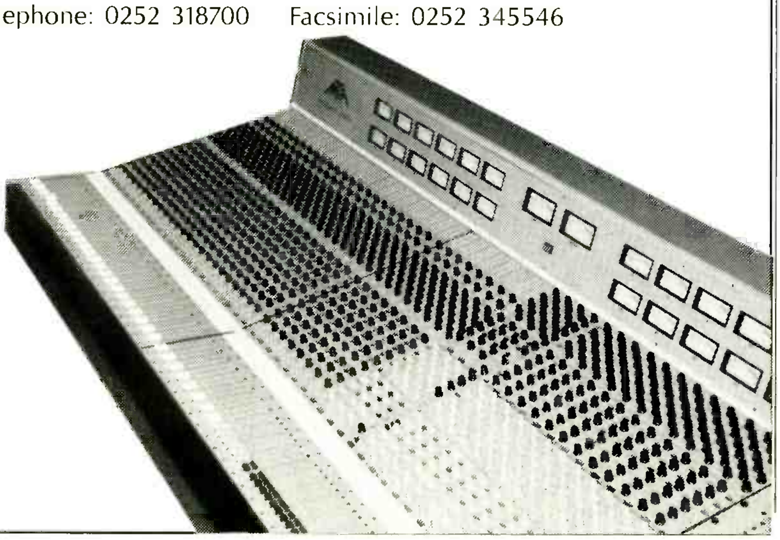
is presented in a time continuum and in the forward direction, as opposed to studio recording or storage where slow motion, shuttle, reverse and still frame modes may be required. It is possible that a programme coded using MPEG-style techniques would give poor performance in some of these modes when compared with the JPEG (Joint Picture Experts Group) algorithm which was defined slightly earlier and relies entirely on intraframe coding methods.

JPEG was intended originally for high quality still frame pictures, and is used widely in computer graphics and multimedia applications for compressing 24-bit still images. It is possible to trade off compression factor against picture quality, and the compression process can either be carried out relatively slowly using an off-the-shelf software package, or quickly using dedicated hardware such as the C-Cube JPEG compression chip, which is fast enough to compress images at FMV frame rates for computer applications. JPEG-like algorithms are used in professional digital video recording systems such as Sony's new *Digital Betacam*, (at a relatively mild compression factor of 2:1) because of the performance in the operational modes mentioned above. JPEG is also used in off-line nonlinear video editors such as Avid's *MediaComposer*.



> The introduction of cheap digital video compression for desktop computers is resulting in a sea-change

The future

As with audio BRR, digital video is now entering a phase where acceptable trade-offs must be determined between bit rate and quality. The degree of reduction will be determined by the application, and there will probably be few situations in which full-bandwidth pictures will be either stored or transmitted. This will be especially true once HDTV becomes more of a reality, since BRR is really the only way of ensuring cost-effective links, transmission and recording.

The introduction of cheap digital video compression for desktop computers is resulting in a sea-change in the way that video programmes are edited, since the cost of purchasing the equipment is dropping very quickly compared with high-end products. Editing decisions are increasingly made off-line using such equipment, using on-line equipment to assemble the final programme, but the time is coming very shortly when broadcast-quality pictures (whatever that means these days) will be available from nonlinear equipment, using BRR, possibly obviating the need for a separate on-line assembly of the programme.

It is also interesting to notice the increasing number of products integrating compressed digital video and linear digital audio editing, offering the ultimate cost-effective postproduction station. Clearly, though, none of this cheap desktop video technology will instantly make anyone who can purchase an Apple *Macintosh* into a world-beating video editor and audio dubbing mixer, just as DTP did not automatically turn everyone into world-beating typesetters but for those with the necessary skills, the introduction of digital compression technology is certainly lowering the costs involved in many areas of programme production. ■

Dr FRANCIS RUMSEY is Chairman of the British Section of AES, and a lecturer on Surrey University's Tonmeister degree course in Music and Sound Recording. He is the author of numerous conference and convention papers for AES, the Institute of Acoustics and the Royal TV Society, and six books on audio technology including *Digital Audio Operations* and *MIDI Systems and Control*

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Confusion in consumer electronics and the many talents of MiniDisc

To subvert John Donne, 'no industry is an island'—so if the consumer electronics companies can spark enthusiasm for new audio and video formats, consumers will buy more music and movie software. And if the consumers spend more money, things look up for the studio business.
On the other hand, if the new formats fail, then people will be more likely to spend their spare cash on holidays or video games.

The recent International Audio and Video Fair in Berlin—traditional launch pad for new electronics in Europe—left a clear and depressing impression: the people selling these new formats are selling boxes. They do not understand the technology or the key issues like compatibility. Worse still, they do not see how their ignorance is creating confusion which encourages the buying public to 'wait and see'.

Repeatedly at the Berlin IFA, large companies held press conferences without having anyone on hand to answer technical questions. Take interactive multimedia (the great white hope for home entertainment): if you set aside the shoot-'em-up games platforms sold, or promised for sale, by Sega, Nintendo, Commodore and Atari, the field narrows to CD-I, developed by Philips and supposedly backed by Panasonic, and 3DO, developed in California and most definitely backed by Panasonic. CD-I is already on sale and 3DO is promised for sale this winter.

The two systems are wholly incompatible, although both players will play audio CDs, CD+G discs (that have simple graphics buried in the audio subcodes) and Photo CDs.

There is now a digital video compression standard, MPEG-1, for putting 74 minutes of full-motion video (FMV) on a CD. This winter, Philips will start selling a plug-in MPEG adaptor which allows a CD–I player play back FMV discs. 3DO promises a similar adaptor.

Philips, Panasonic, Sony and JVC recently agreed a new White Book standard for Video CD, or Digital Video CD with MPEG FMV, but without the control codes which a CD–I player uses to provide full interactivity. Video CD is a 'linear' format, like a video tape—you play it and watch it from start to finish.

Video CD follows the CD-ROM XA 'Bridge' standard, like a Photo CD. This allows a Video CD to play on a personal computer equipped with CD-ROM drive and an FMV decoder, or a new generation of Video CD player which is broadly similar to an audio-only CD player but has a built-in FMV decoder.

Because the CD is a ROM XA disc, it has a Yellow Book data flag in the bit stream. This flag will (or should) mute all the outputs of an audio-only CD player to prevent speaker damage. This kills the idea of using an add-on FMV decoder to make Video CDs play back on a conventional CD player equipped with a digital output.

Nimbus have suggested that this problem can be solved very simply, by making the Video CD follow the Red Book standard—not have a data flag—but this would stop the Video CD playing on a CD–I player or ROM drive, as both are designed to treat and Red Book disc as a music disc. These machines would try to decode the FMV data on an unflagged Video CD as audio data—and fail.

Also, the White Book now specifies control codes for VCR-like functions such as fast search and freeze-frame. Consequently, the FMV decoder must feed control signals back to the CD player drive. There is no input for these control signals on any existing audio-only CD player.

Words cannot describe the confusion which has swept the audio and video world over this scenario. And it all stems from the clumsy wording of a joint statement put out in late June by Philips, JVC, Sony and Matsushita (parent of Panasonic). This statement on the White Book promised playback of Video CDs on 'modified CD players (with a digital data output) with an add-on Video CD box'.

What was meant was that there will be a future generation of audio CD players, built slightly differently, to play either Red Book audio CDs or Yellow Book Video CDs when connected by a digital output to a video decoder. What people very reasonably understood it to say was that Video CDs will play on existing Red Book audio-only CD players that already have a digital output and which may be connected to an add-on video decoder.

Confusion has piled on top of confusion. Just as we thought everything would finally be clarified by statements to be made in Berlin, the same four companies put out another statement which repeated the error, this time without even the word 'modified':

It read: 'Video CD discs can be played on... CD players (with a digital data output) with an add-on Video CD adaptor.'

The bottom line, which the four largest electronics companies in the world seem incapable of drawing is that there will be at least two types of FMV CD. One will be a CD–I/FMV disc which plays back with full interactivity on CD–I players when equipped with an FMV decoder. These discs will not play on other players. The other version will be a White Book linear digital video CD; this will have very limited interactivity but will play on CD–I players with FMV adaptors, on personal computers with CD-ROM drives and FMV adaptors and on a new generation of Linear Video CD player which will work with both Red Book audio CDs and White Book Video CDs.

At the Berlin International Audio and Video Fair, Sony announced that they will soon be offering recordable MiniDisc as an alternative to NAB carts. This makes good sense because radio stations are already using solid state carts and floppy disk carts. MD would beat both on counts of cost-effectiveness and playing time—with at least an hour on a single MD disc which retails in the UK consumer market for around £10. Likewise, JVC now suggest that MD would make the ideal tool for recording dictation or conferences, giving rapid and indexed access.

It is unclear yet whether Cart MD will use the same ATRAC reduction system as consumer MiniDisc, or whether it will use an upgraded system. In Berlin, Sony continued to plug the now-standard company line that the sound quality available from consumer MD 'approaches CD'. SO MD is not a threat to CD but is quite good enough for listening to music on the move. Consumer MD quality would thus be well suited to broadcast carts.

But Sony may wish to create a new standard to help keep the price of cart discs higher than that of consumer discs, much as Kodak split the writable CD market to keep the price of audio CD blanks above that of *Photo CD* blanks. In both cases, polymer dye write-once *Photo CDs* and magneto-optical MDs, the low priced option is a lost leader.

A panel of Sony top brass in Berlin included Norio Ohga, Sony's president, Michael Schulhof, Head of Sony Music and Ron Sommer, Head of Sony's operation in Europe.

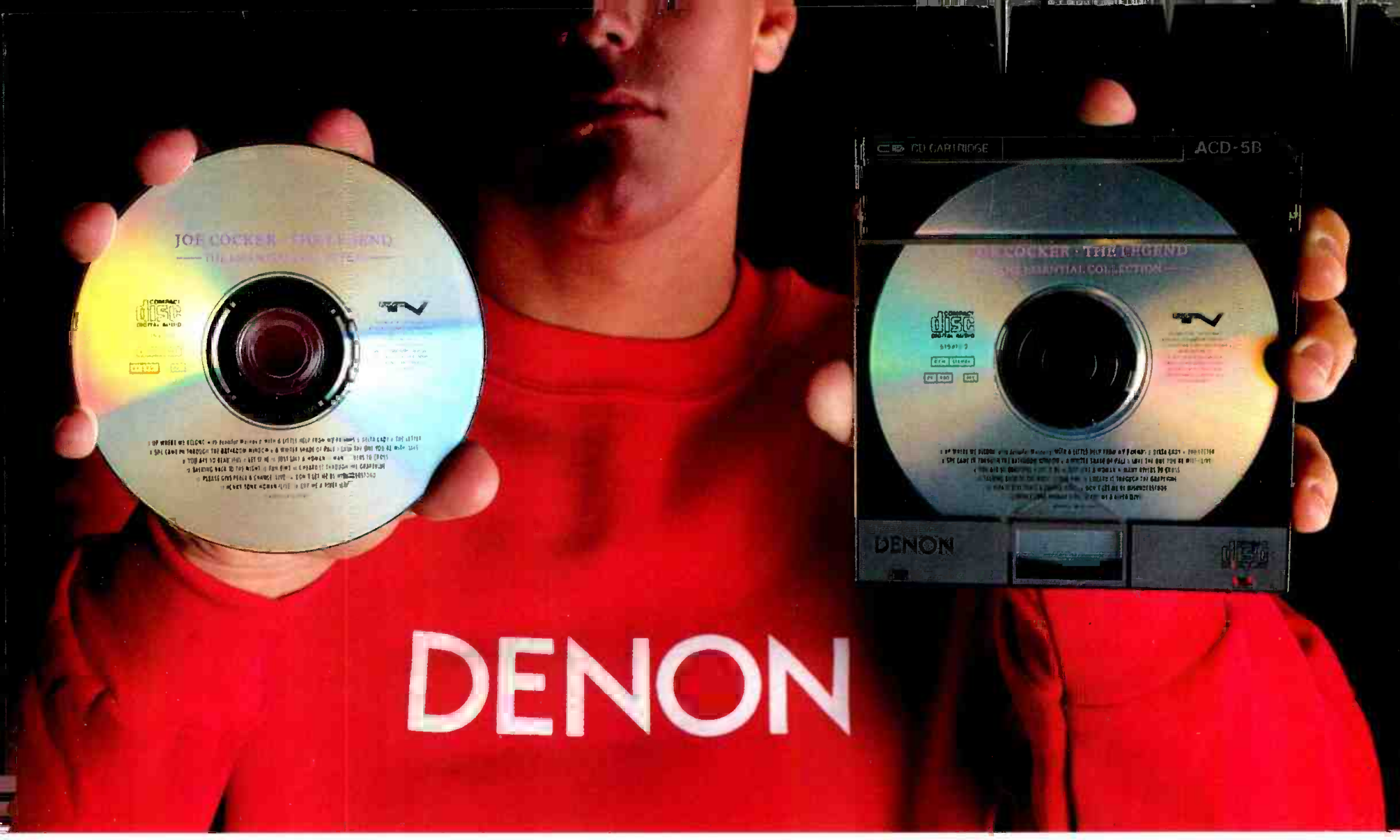
'The real future of tape is definitely a disc,' said Sommer. Schulhof then screened a lengthy selection of video music clips. The sound was very nasty—well below MD quality. But Schulhof proudly announced, 'you have just witnessed an amazing example of the power of technology... digital quality sound... the optical disc experience'.

Had Sony sync'd up an MD player with time-coded video tape? If so, what on earth went wrong? And why no apology for the awful quality?

In fact, the nasty sound was coming from the analogue track of a *Betacam SP* videotape.

If Messrs Ohga & Schulhof want support for their efforts to reshape the future of tape as MiniDisc, it would help to show a little more interest in sound quality. ■

ILLUSTRATION: CARL FLINT



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