
AUGUST 1993
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STUDIO SOUND
AND BROADCAST ENGINEERING
THE INTERNATIONAL PRO AUDIO MAGAZINE
MONITORING
Monitoring Low-Spec Signal Chains; Waveguide Design
TANNOY
Test and Measurement
Conventions In Testing: Are We Seeing What We Hear?
Chipping In
Integrated Sample Rate Conversion
INTERNATIONAL STUDIO DIRECTORY
SYSTEM 12

STUDIO SOUND

AND BROADCAST ENGINEERING

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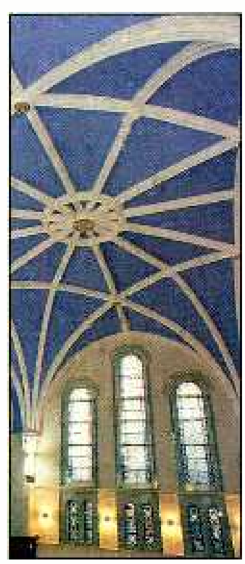

George Martin's acoustic revelry at Lyndhurst Hall. See page 39

CHOOSING THE RIGHT COMPUTER BASED AUDIO EDITOR CAN BE A NIGHTMARE

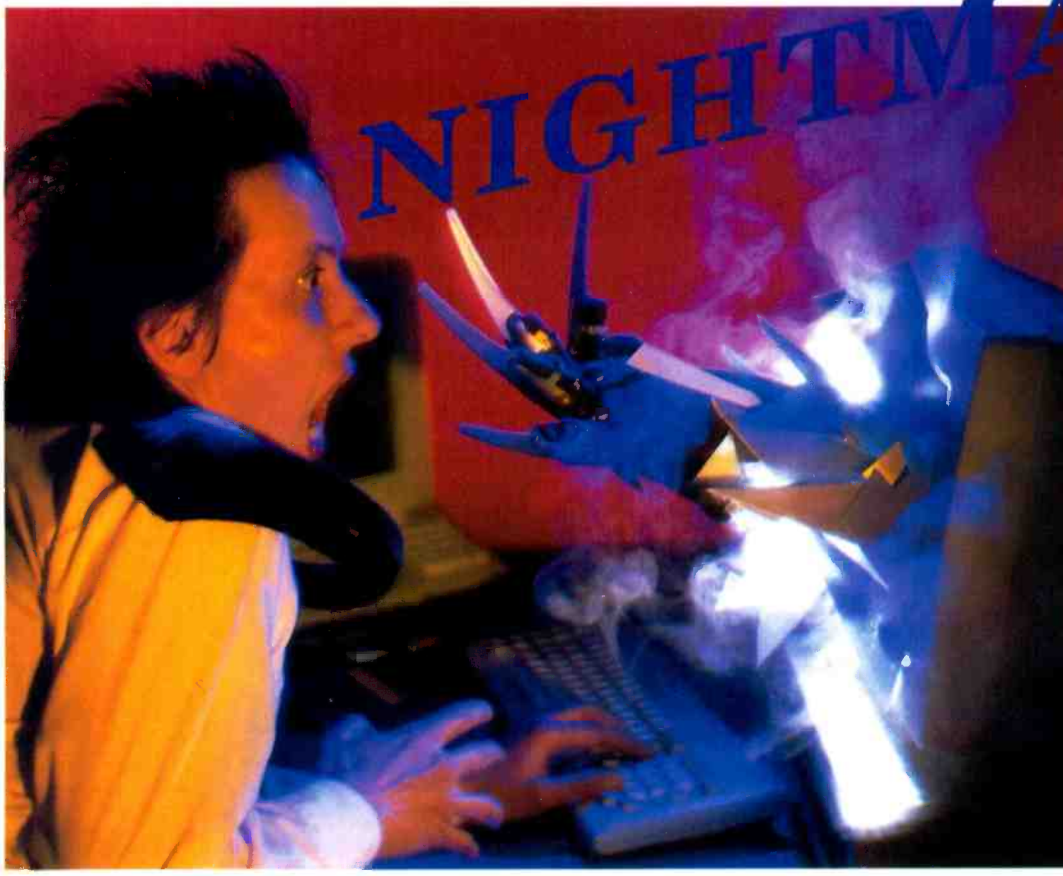

Photo Ewing-Reeson Photography

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Studio Audio & Video Ltd
The Old School,
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Cambridge
CB6 3LD
U.K.
TEL: (+44) 353 648888
FAX: (+44) 353 648867

For a list of overseas distributers contact Studio Audio

USA & CANADA:

Studio Audio Digital Equipment Inc.
P.O Box 4392
Ann Arbor
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EDITORIAL
Editor: Tim Goodyer
Assistant Editor: Julian Mitchell
Production Editor: Peter Stanbury
Secretary: Mary Walsh
Consultant: Sam Wise
Columnists: Barry Fox; Martin Polon
Regular Contributors: James Betteridge; Ben Duncan; Terry Nelson; Dave Foister; Francis Rumsey; Yasmin Hashmi; Zenon Schoepe; Mike Lethby; Patrick Stapley

ADVERTISEMENTS
Executive Ad Manager: Steve Grice
Deputy Ad Manager: Peter Turberfield
Advertisement Production: Carmen Herbert
Secretary: Lianne Davey

CIRCULATION
Circulation and Development Manager: Colin Enderson
Controlled Circulation Manager: Maria Udy

Director: Doug Shuard
Publisher: Steve Haysom

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In praise of hero worship

When a particularly well-respected recording engineer died, his endless enthusiasm, meticulous working methods and skill with a soldering iron naturally ensured him a place in the Kingdom of Heaven. On his arrival at St Peter's Gate he was met by St Peter himself who presented him with an Access All Areas pass and told him that a party was being held in his honour—and that everybody who was no longer anybody was there.

The party was in full swing as the engineer entered a room the likes of which defied the imagination of any living acoustic designer. And the band kicking up a storm boasted a lineup more impressive than any charity concert. 'Is that really Keith Moon on the drums,' the engineer inquired of St Peter, 'and Charles Mingus on bass?'

'Of course,' came the reply.

'That brass section,' he continued in disbelief, 'Charlie Parker, 'Trane, Lee Morgan, Miles. . .' St Peter nodded, happily.

'And Hendrix and Clapton on guitars—but Clapton's still alive, isn't he?'

'Ah', responded St Peter, sagely, 'that's not Eric Clapton, that's God—he just thinks he's Clapton.'

So it's an old joke, and it has been a while since 'Clapton is God' graced a wall near me, but it does make a point—that the music business, like certain others, has its heroes and heroines. Or has it?

Certainly, the music industry has thrived on the suspension of disbelief between the famous and the fanatic—what greater motivating factor could there be than the desire of a young guitarist to want to *be* Jimi Hendrix, Allan Holdsworth, Jennifer Batten or Joe Satriani, a young drummer to want to *be* Vinnie Colaiuta, Sheila E or Dennis Chambers, or a young keyboard player to want to *be* Gerry Lee Lewis, Keith Emerson or Geof Downes? Yet such 'heroes' are becoming rather thin on the ground. Significantly, it is in the area of hi-tech music making that they have thinned out most quickly. But it would be wrong simply to point the finger at technology as having replaced the visible skills of musicianship (and showmanship) with the 'invisible' skills readily associated with sequencing and sampling technology—the policies of certain producers are also to blame.

Take the empire built by British producer Pete Waterman on his mix-'n'-match pop factory: songs are provided by partners Mike Stock and Matt Aitken, and fronted by a succession of eagerly cooperative (though sometimes talented) young singers. The regular practice of musical collectives on both sides of the Atlantic using 'featured' vocalists fares little better in this respect, while those who have sampled vocals from old records and passed them off—if only visually—as the work of another performer could not have done more to devalue the role of the genuine artist. Never has Warhol's '15 minutes of fame' seemed so real, or so unattractive.

If popular music is to continue to serve our collective cultures and the music industry, perhaps we should restore our regard—and that of the public—for the talents which helped establish it.

As Elgar so lyrically put it: 'We are the makers of music, we are the dreamers of dreams'. ■

Tim Goodyer

Cover: Tannoy *System 12* monitors

CS2000

Digital Control Studio System

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

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

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

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

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

Euphonix

The European Office
56 Albion Road
Pitstone, Near Leighton Buzzard
Bedfordshire, LU7 9AY
England
+44 (0)296 661748

Worldwide Sales
Los Angeles CA (818) 766-1666

International News

In-brief

▲ **Bongiovi goes to Touchdown**
Barry Bongiovi has joined Munich-based Touchdown studios from Director of Operations at Sony Classical. Prior to that he spent 12 years at The Power Station in New York. He will oversee Touchdown's management in Munich and the new studio in Portugal

● **First ever Nigerian SR setup**
Based in Lagos, Nigeria, Mastersound is the first professional sound reinforcement company to be set up in this part of Africa. Chief Chris Okunowo, company chairman and originator of the project, sees the operation renting sound equipment and services throughout Nigeria, Ghana, Cameroon, Ivory Coast and Senegal.
Mastersound Tel: +234 1 263 2849

● **Soundcraft and live musicians**
Soundcraft are running a course focusing on live sound engineering. The main body of the course will be aimed at those who already have a basic knowledge of sound engineering. In addition there is a pre-course session on Friday aimed at beginners. The dates are 17th,18th, & 19th September.
Soundcraft Tel: 0707 665000

● **Technology Expo a celebration**
Expo '93, a two-week exhibition sponsored by London's Capital Radio and Pepsi, promised a glimpse of the future in the form of an amusement park of the tomorrow with Virtual technology, simulators, 3-D cinemas and advanced arcade rides

ISDN to become world telephone standard

Tom Kobayashi, President of EDnet, the Entertainment Digital Network, has stated that the ISDN (Integrated Services Digital Network) is becoming the world digital telephone standard.

'The US and Canada have pockets of ISDN in the big cities but not nationwide. The problem is that even though the digital telephone is becoming standardised, the equipment, such as the audio codecs are not standard.

'There is Dolby *AC-2* series, APT, and the MUSICAM codecs have different algorithms and they don't talk to each other. EDnet has been able to transcode some codecs using a bridge between them.'

EDnet was borne out of LucasArts Entertainment Company, when they were experimenting with the *T1* digital communication protocol to transmit telephone calls, computer data, and high quality digital audio between Skywalker Sound's two facilities located over 400 miles apart.

Their latest project was a five-hour ADR session between London and Hollywood to record the lines of Ben Kingsley and Robert Stephens for the motion picture *Search for Bobby Fischer* in sync with picture.

Previously, producer Phil Ramone asked EDnet to record Gloria Estefan's vocal with her backup singers in Los Angeles. EDnet setup a link between Miami-based Crescent Moon studios and The Village Recorders in Santa Monica.
EDnet. Tel: +1 415 274 8800. Fax: +1 415 274 8801.

Both studios at Paul Weijenberg audio & videopostprodukties in Amsterdam have been completely refurbished. Studio 1 is now fitted with a D&R *Triton 48* with automation and Studio 2 (pictured above) is now fitted with a Korg *Soundlink* digital audio workstation. The studio is one of the only commercial studios in the Netherlands that has an audio landline to NOB Broadcast in Hilversum

Spatializer surrounds music and post

The *Pro Spatializer* 3-D audio production system is proving popular with music and postproduction customers. Recent buyers have been Grammy-winning Producer David Foster and Tom Maydeck, President of Monterey Post, who had been using the prototype for three years. Maydeck 'Now with the *Pro Spatializer*'s multichannel joystick, I'll be able to move effects and dialogue beyond the front sound field of left-centre-right, and outside of the speaker sources, even behind the listener.'

Spatializer was invented by audio industry veteran Steve Desper.

Engineer Dave Reitzas, producer David Foster and the *Spatializer*

DOUG posted by voice

Pomann Sound Productions in New York is doing complete sound design and audio postproduction for Nickelodeon's television show *DOUG*. The Emmy and Ace nominated series is now in its third season and will be posting season four on the Autumn.

DOUG is backed by an innovative soundtrack; extensive sound effects from Pomann's in-house library and the unique mouth sound effects of voice impressario Fred Newman. All elements are produced on the studio's two SSL *ScreenSound* digital audio workstations.

Pomann Sound Production's penthouse studio was designed and installed by Ira Kemp of Cylinder Systems. Review examples of the soundtrack are available.
2, West 46th Street, The Penthouse, New York City, NY 10036. Tel: +1 212 869 4161.

Mega AMEK

Amek have bought part of the shareholding of their German dealer, Mega Audio. The company will be AMEK Deutschland GmbH.

Mega Audio was founded in 1989 by Burkhard Elsner, Peter Foeller and Uwe Grundei principally to sell and distribute AMEK in Germany. Mega quickly evolved into a broad based studio supply company for the German market.

The medium term plan is to establish AMEK as a distribution house in Germany, which traditionally has had few successful national (as opposed to regional) distributors. Working with the AMEK management team, Elsner and Grundei will seek product lines to import into the territory.
AMEK Deutschland GmbH. Tel: +49 (0) 6721 26 36.

Sony open Australia's largest mastering studio

Sony Music Australia have opened one of the largest and most advanced mastering studios in Sydney's Western suburb of Huntingwood.

Four acoustically treated isolated studios each perform different mastering tasks. One complete digital studio consists of the Sony *DAE3000* editor, *DMR4000* U-Matics, *1630* processor, *DAT7050*, and *SDP1000* 32-bit digital processor.

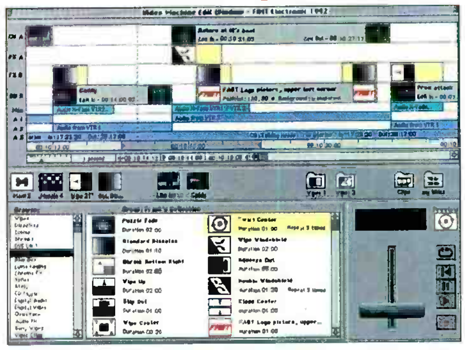
Studio 2 is similarly equipped but has analogue outboard and Dolby and dbx N/R. Studio 3 is a cassette running master suite with *DMR2000*, *1610* processor, MCI and Studer 1/4-inch and 1/2-inch machines.

Studio 4 houses the Sonic Solutions systems and Super Bit Mapping. A fifth studio consists of *DAE1100*, *1630*, *DMR4000*, for basic editing, but the room is used more as a dub room with DAT players, cassette decks and video recorders.

All studios are tie-lined and use Australian Monitor amps and custom built Landmark nearfield and main monitors. The studios form part of the CD and cassette manufacturing plant.

8-channel audio for desktop video

The Video Machine is a complete Desktop Video studio consisting of a 16-bit AT bus card and VM-Studio software. It includes an edit control unit for A-B roll operation with control functions for three video recorders, two frame synchronisers, 200 digital video effects, a video printer driver and an 8-channel audio mixer.



The *Video Machine*—dull name but promising audio

The Machine can read and generate time code (VITC) and write EDLs (CMX, Grass Valley and Sony). The VCRs are controlled via LAN-C and Edit Control.

PAL, NTSC and SECAM sources can be used as inputs; output is in PAL and NTSC and Video Machine accepts Composite and Y-C signals. **Magnifeye. Tel: 071 221 8024. Fax: 071 792 3449.**

MSL-5 debuts at Montreux

The 27th Montreux Jazz Festival (July 2nd–17th), produced by Quincy Jones and Claude Nobs, was the launchpad for Meyer Sound's latest loudspeaker—the *MSL-5*. A new venue, Montreux's Convention Centre, hosted headliners including Robert Plant, New Order, Chaka Khan, Ute Lemper, Stephane Grapelli and Stanley Clarke.

PHOTO: MIKE LETHBY

Meyer, an official sponsor, has designed and supplied the festival's main auditorium audio for some years. They also supplied the New Q's and 'Off Festival' stages.

AudioLease provided Midas *XL3* consoles, control and monitors, while Jurgen Dudda Audio installed the PA and supplied QSC *1250* and *4000* amplifiers.

For the central 1800-seat Stravinsky Hall, designed for classical music, head sound engineer Chris Ridgway, Meyer's Mark Johnson and acoustic consultant Jim Cousins specified heavy drapes to beat reverberation. Meyer's *SIM II* acoustic analyser system fine-tuned the audio systems.

Commented Johnson: 'It was the perfect opportunity to use the *MSL-5* and other recent developments—the *DS-2* and *MSL-2A*.'

The biamplified *MSL-5* is an arrayable high power unit. A 60° array pair delivers 100dB continuous SPL at 100 feet. Condensing the *MSL-10*'s high-Q characteristics into a 1.4m tall box with two 12-inch LF and three 2-inch HF drivers, it is an important addition to Meyer's range.

Mike Lethby

Show Report

New *Macs* chase AV market

At the recent MACWORLD Exhibition in Boston, Apple unveiled a new generation of computers that have audio and video work firmly in mind.

The *Mac Quadra 840AV* and *Centris 660AV* both use the Motorola 68040 microprocessor and also the 55MHz AT&T 3210 DSP. The DSP will handle specialised tasks and real time data—including speech, audio modem, telephone and fax signals. Third-party developers will be able to tap into its power to provide performance enhancements.

The new AV models feature a comprehensive video and graphics architecture, allowing full-motion video from sources such as VCRs, camcorders and laserdiscs, plus the digitsation and capture of single frames as pictures or video sequences. For video-in, all major standards are supported—NTSC, PAL and SECAM—and composite and S-video ports are provided. The new systems also support 16-bit stereo audio input and output at various sample rates.

In the US Apple is bundling ApplePhone, a screen-based phone application; FusionRecorder, an application that records video and 16-bit audio; and ES F2F, LAN-based video conferencing application by the Electronic Studio.

Contracts

● EMI upgrade to Aussie's
Australia's largest recording complex, EMI Studios 301, has re-equipped and upgraded all its control rooms, main and nearfield monitoring with Australian Monitor *1K2* and *AM1600* amplifiers.

● First ever *3348* in Middle East
The first ever Sony *PCM 3348* DASH recorder sold to a middle eastern customer has been bought by Kuwait Radio, supplied by Eastlake Audio.

● Peter Gabriel's Radio Station
Garwood's *Radio Station*, the unique in-ear monitoring system, has been in use with the current Peter Gabriel tour in the UK and now in the US.

● Japanese go for the *Scenaria*
Sales of the SSL *Scenaria* system in Japan include NHK; Sanwa Video; Omnibus; YTV and Imagica.

● Calrec OB desk to YTTV
The Yorkshire Tyne Tees Television Group has recently taken delivery of a Calrec 36-channel, four-group, *Compact* mixing console for their new Unit 5 outside broadcast vehicle.

● BJG install Dynaudio digitals
Bunk, Junk and Genius Recording in London have become the second UK customer for the digital version of the Dynaudio Acoustic *M4* monitors.

● International sales for D-ESAM
More than 250 Graham-Patten D-ESAM digital edit suite audio mixers are now in routine use throughout the world. Recent orders include The Mill, London; Wharf Cable, Hong Kong; and AAV in Melbourne, Australia.

● Recording school buy 3 *Jades*
The famous audio training college, School of Audio Engineering recently installed three Soundtracs *Jade 32* patchbay consoles.

▼ Ampex tape help Brit students
Ampex have donated *Grand Master 456* Studio Mastering tape to the British Record Industry Trust Performing Arts and Technology School in South London.

Ampex's Tom Gittins(sixth from left) and Brit's Tony Clark(sixth from right) surrounded by Brit students

Two for Dynaudio

DynaudioAcoustics have launched the *PPM3* and *C3* monitors. The *C3* is a 3-way extended version of the *C2* Classical reference monitor, with a 12-inch bass driver in an infinite baffle cabinet. It is designed primarily for highly critical applications requiring medium monitoring levels, and offers a frequency response of 35Hz–20kHz with low distortion. The *PPM3* is the big brother to the *M1* nearfield monitor and combines 2 x 7-inch bass drivers in a reflex cabinet with a high sensitivity soft dome tweeter.
The Studio, 1 Book Mews, Flitcroft Street, London. WC2H 8DJ. Tel: 071 379 7600. Fax: 071 497 8737.

***C3* and *PPM3*—new from DynaudioAcoustic**

Total's PCM status meter

Totalsystems have launched a unique product in the *SDU-1*. This is a Status Display Unit that connects to the status port of the Sony *PCM 1610* and *1630* processors, and will display all of the status information given out by the processor; Parity, CRC, Averages, Holds, Mutes, Sampling Frequency, Emphasis. Also launched is the *PM-3* Phase Correlation Meter.
Totalsystems, 59 Hatch Lane, Old Basing, Hants. RG24 0EB, UK. Tel. and Fax: 0256 54786.

Totalsystems *SDU-1*

New *LA100* —Old Price

Lindos Electronics have launched a new version of the *LA100* audio analyser system comprising the *LA101* synthesised oscillator and the *LA102* measuring set.

New features include a high contrast LCD with backlight; improved software and a new manual. Up to 100 predefined automatic test sequences can be included in the oscillator for testing a wide range of systems, including tape machines, mixing consoles, loudspeakers, lines and links, meters and filters.

A new menu system allows the user to browse through the banks of predefined test sequences or even program their own. The new software also allows frequency curves to be shown relative to a reference response stored in a memory.

Despite these improvements Lindos have kept the *LA100* price the same. Software updates are available free of charge to existing users. Old units may be returned to the factory for a complete upgrade.
Lindos, Saddlemakers Lane, Melton, Woodbridge, Suffolk. IP12 1PP, UK. Tel: 0394 385156.

Bag-End Wood

Bag-End loudspeaker systems have announced it will produce its *TA12* and *TA12-J* range in an oak veneer cabinet model. Their original loudspeakers were in hand-rubbed oil walnut plywood cabinets covered with black carpeting, but in recent years they have got more requests for a home and-or studio version, particularly of the *TA12* and *TA12JR* lines.

The *TA12JR* is a time-aligned, wide-range system which is slightly smaller and lighter than its big brother, the *TA12*.
Bag-End Loudspeakers Systems, PO Box 488, Barrington, IL 60011, USA. Tel: +1 708 382 4550. Fax: +1 708 382 4551.

Rack PCs

GAS Electronic Systems are marketing a range of PC compatible computers that are 19-inch rack-mountable. The GAS PC fits into 5U of rack space and has mouse and keyboard ports front and rear. The range starts with a 486SX machine running at 25MHz going up to a 486DX running at 66MHz. All units are upgradeable. The units come with keyboard, DOS 6, colour SVGA monitor and drive card.
GAS Electronic Systems, 16 St Alfege Passage, Roan Street, London. SE10 9JS. Tel: 081 858 9444. Fax: 081 858 1033

Kevlar from Polydax

The Polydax speaker Corporation, a subsidiary of Audax Industries in France, have introduced a complete range of woven Kevlar cone mid-bass speakers. The range consists of the HT100KO (4-in); HT130KO (5¼-in); HT170KO (6½-in); and HT210KO (8-in). The Kevlar material contributes to the overall quickness and precision of the transient response of each model. Design features include high loss rubber surrounds; large magnet structures (20oz); and high temperature voice-coils wound on aluminium voice-coils.
Polydax Speaker Corporation, 10 Upton Drive, Wilmington, MA 01887, USA. Tel: +1 508 658 0700.

In-brief

● PortaMezzo removable media
Grey Matter Response has introduced *PortaMezzo*, a new line of removable storage and archival solutions for digital media studios. *PortaMezzo* takes the high-capacity hard-disk drives used in their *Mezzo* product and encase them in rugged, individual housings that can be transported. Grey Matter Response claim their drives are the only ones rated for multitrack digital recording and have enough capacity for the memory-intensive needs of digital media. **Grey Matter Response. Tel: +1 408 423 9361**

● DigiDesign inspired by drive
DigiDesign have announced the compatibility of the Alphatronix *Inspire II* optical drive for *ProTools* workstations. The *Inspire* drive allows the *ProTools* user to record and playback four tracks of digital audio in real time.
DigiDesign Tel: +1 415 327 0777

● Sony set new standards
Sony Broadcast & Professional UK has announced that its Technical Services Group has achieved registration to BS5750 Part 2 : 1987/ISO 9002 : 1987/EN29002 : 1987. They are now one of the few companies in the audio-visual industry to attain this internationally recognised standard.

● Filmtech's new TV-Film mixer
Filmtech TFE have announced the new *LSP4* portable audio mixer, designed for use on film and television. The *LSP4* claims to be the first fully mobile mixer with four outputs. In addition to the main stereo output each of the four inputs has a direct balanced output enabling use with the *Nagra D*

● Korg's brochure on CD-ROM
Korg have produced a multimedia brochure for the music industry, one of the first of its kind. A 'mixed mode' demonstration CD combining normal audio tracks and CD-ROM tracks for Apple *Mac* computers.The CD-ROM demonstrates Korg's new *X3* music workstation. **Korg Tel: 081 427 5377**

PRODUCTS

Sony's *WMD-DTI* DAT player or is it the *TCD-DP10*?

World's smallest DAT player?

Sony have released details of the smallest ever DAT player, set for launch in September. The *WMD-DTI* was seen earlier this year in Japan and features a mechanical loading mechanism which allows the size of the machine to be dramatically reduced; its not much bigger than the smallest conventional walkman. The supplied headphones feature full remote control, with an LCD window which displays operational mode as well as information recorded on the DAT tape. The D–A system is single bit and the machine operates from dry cell batteries with four hours possible from two AA size batteries.
Sony United Kingdom Ltd, Sony House, South Street, Staines, Middlx. Tel: 0784 467284. Fax: 0784 467107

JBL 4400 Upgrade

JBL have upgraded their *4400 Series* studio monitors into the *4400A* after a major component upgrade.

Both LF tranducers and HF Dome radiators have been upgraded to offer up to an extra 5Hz–3kHz respectively, plus the units are now finished in a matte grey laminate with charcoal coloured grilles.

The *Series* is now: *4408A*—a 2-way compact monitor for smaller studios featuring a 200mm LF felted cone and a 25mm HF Pure Titanium dome. *4410A*—a 3-way system designed as a vertical array featuring a 250mm LF Aquaplas Laminate Cone, 125mm Mid-range cone and 25mm HF pure titanium dome. *4412A*—for applications needing maximum low frequency output from a bookshelf sized monitor, this 3-way system tightly clusters its transducer complement for accurate close proximity listening.
Harman Audio. Tel: 081 207 5050.

Voyager Sound nears completion

A recent announcement that Voyager Sound Inc has been granted a landmark patent brings the completion of The *Voyager Sound System* a bit nearer. The patent allows the system to use graphic icons to control multiple mixing parameters and effects in an audio environment.

The *Voyager Sound System* employs a user-definable elliptical work surface in which icons representing either single sound elements or groups of sound elements are located. The relative positions and form of the icons represent the parameters controlling the sound elements. For example, icons close to the centre of the elliptical work surface represent loud sounds in the output. Icons further away from the'Listener point of view' represent attenuated sound. Multiple sound elements or submixes can be defined by a single icon allowing the engineer to build intricate sound productions. Multiple icons and their dynamics can be linked to either SMPTE or MIDI time code. The entire state of the system can be stored.
Voyager Sound Inc, 95 Newton Street, Weston, MA 02193, USA. Tel: +1 617 893 2574.

Eminence Pro

Eminence Speakers (Europe), official distributors of American Eminence loudspeakers, are using this year's PLASA show as a platform for the UK launch of their new range of high power drivers dubbed the *Pro Series*.

The *Series* features cast alloy chassis from 10in to 18in and includes four-inch voice-coil drivers with a power handling up to 600W.

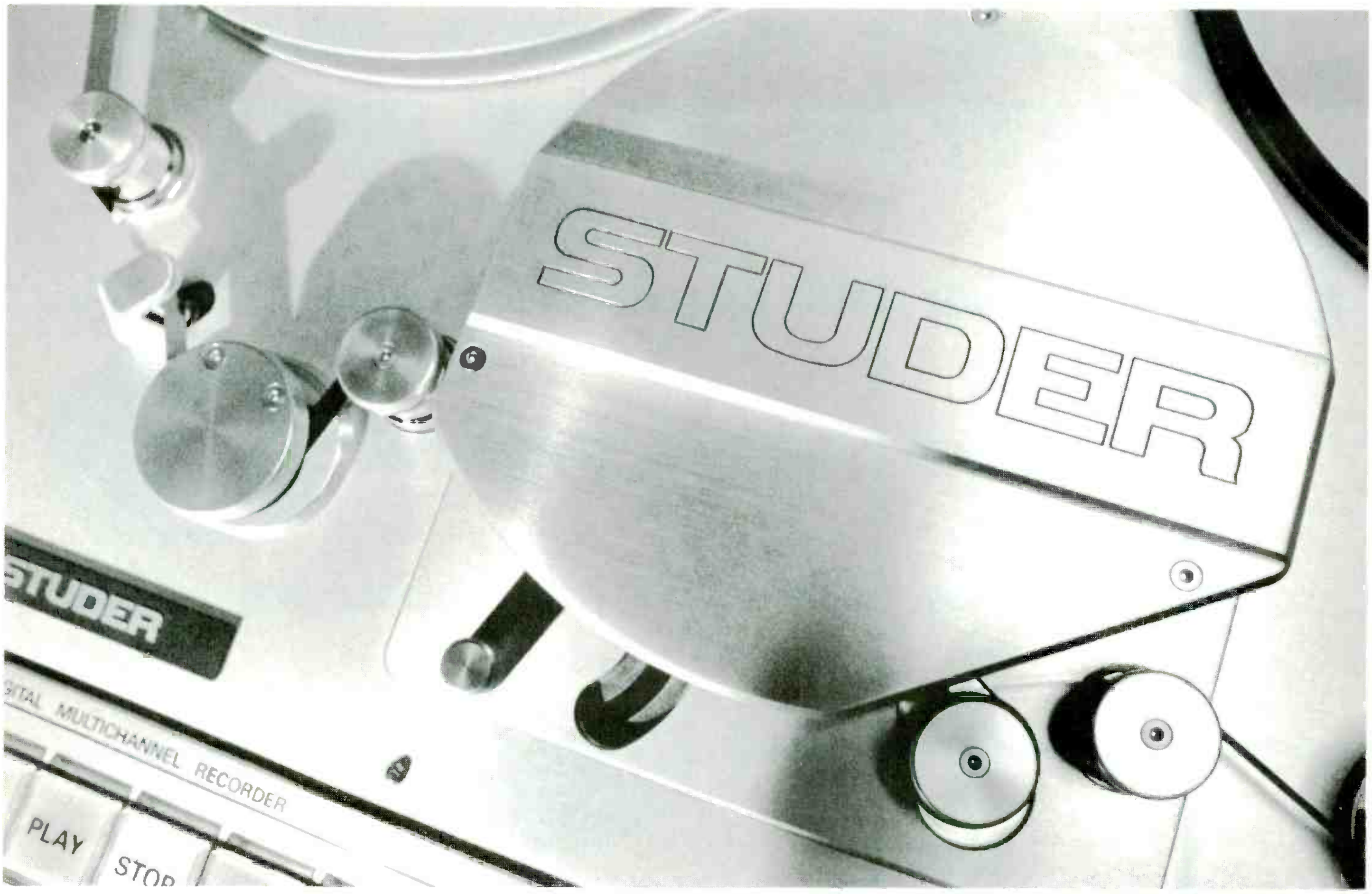
Eminence are now supplying their drivers to an increasing number of cabinet manufacturers across the world. They'll be on Stand A124 at the Plasa Show, 12th September, Earls Court 2, London.
Eminence Speakers, 381 Barnsley Road, Wakefield, West Yorkshire, UK. WF2 6BA. Tel: 0924 279297. Fax: 0924 263998.

CR-1 De-Crackler

The *CR-1* De-crackler from Cedar Audio is a 2U rackmount device uses Cedar's 'Split and Recombine' signal processing technology.

Main features are Crackle removal; buzz removal and distortion reduction. Porky's Mastering in London have been the first to order the *CR-1*.
UK: HHB Communications, 73-75 Scrubs Lane, London. NW10 6QU. Tel: 081 960 2144. Fax: 081 960 1160.

Cedar's De-Crackler *CR-1*



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Dynacord *DRP15*

While Dynacord have had a rather low-key tradition in outboard effects, the excellent *DRP20* is an indication of what the company can achieve. Units like the *DRP15* from manufacturers who do not major in effects units are always worth investigation as they often approach the subject from a different perspective—this is particularly the case with regard to algorithms and hence the sound.

All front panel buttons are arranged sensibly and accessibly around a continuous pot which can additionally be pressed to confirm actions and kept depressed to speed scrolling though parameter values, for example. Connection is via independent jacks for the stereo in and out with the level of both inputs controlled on a single front panel pot alongside a peak hold bargraph level meter.

Patches are selected on the dial and confirmed with a push or by separate UP-DOWN buttons followed by a dial push. There are 100 factory presets and 128 user locations.

Fundamental to every patch is the effect configuration which is displayed via six LEDs corresponding to effect types to the right of the 2 x 16 LCD. There are ten configurations to choose from, selected with the dial after pressing the CONFIGURATION button—I will tell you now that the ones you will be interested in are the ones that concern themselves with reverb. Playing along for a while, the configurations include: direct only (which curiously only serves as a means of storing an output level); high quality reverb; high quality modulation; pitch shifter; delay; delay and reverb in parallel; pitch shifter into paralleled delay and reverb; multieffect combining modulation, delay and reverb; instrument effect combining distortion, low-pass filter, modulation, delay and reverb; and a dual delay. All configurations have LF and HF shelving EQ except for the direct and the less said about the instrument configuration the better, the distortion circuitry sucks and is completely inappropriate on the unit.

Pressing the EDIT button enters Programming mode with repeated presses taking you through the effect blocks within the chosen configuration. Travel through the pages is via the SELECT buttons, with data entry on the dial—all familiar and perfectly simple. Getting back to reverb, the amount of room for user interference is considerable and dictated by the reverb type selected. Choosing the 'All parameters' type gives access to ten parameters while there are also 'easy' versions of room, chamber, hall, extra large hall, church, tunnel and plate which are stripped down and optimised versions with half as many adjustable parameters for those who just want to get on with it.

The sort of things you get to deal with in the 'all parameters' reverb are room size, reverb time, damping, reflection: reverb ratio, reflection type (ten types) and six kinds of reverb cluster. In pretty much all cases adjustments clearly alter the sound especially as they are accompanied by a fair amount of glitching as the unit recomputes. The modulation section only offers five parameters to play with but creates wide variety through the selection of nine modulation effect characters—from choruses and flangers down to a nasty and usable analogue-style phaser.

Similar principles are applied to the editing of other configurations, and there is very little more to say about the *DRP15* operationally apart from the fact that its MIDI control capability includes patch mapping and real-time control.

It is not that the rest of the *DRP15* performs badly in the delay and modulation departments it just that the quality of the reverb is stunning— certainly above what you would (perhaps wrongly) expect. With 128 user locations on board, the unit can be lit to the gills with a diversity of ambiences that will mean you will never need to go near the factory instrument configurations again. Reverbs have the unusual quality of a well-controlled bottom end by default mixed with a shimmering top end and a flat mid-range. This means you can use more than you would be used to without clouding the signal and overcooking the LF.

Special mention has to be made of the short sharp ambiences—the true test of a processor no matter how smooth and impressive it may sound on longer decays. Here again the *DRP15* is a revelation in the way it presents complexity in an uncluttered way without loss of definition. It has a habit of enhancing stereo which, while maybe not correct, is certainly pleasant. There are good rooms, an excellent chamber (whatever that is) and first-class hall approximations. Some of the halls are what I can only describe as good general 'wash' programs in that they faithfully maintain the balance between vocals and the backing and I would attribute this to the realism of the programme algorithms involved.

There has got to be a downside: audio connections really ought to be balanced. Changes made at editing on turning the dial are not always instantaneous, and switching basic reverb algorithms here lags noticeably. It also mutes and pauses when asked to change patches, which is really not on these days.

The three back-panel footswitch sockets, the presence of a distortion effect and inputs that can be switched to accept instrument inputs tells me that Dynacord did not really know what they wanted to do with the *DRP15* and consequently threw the net high and wide.

Fortunately for them, and perhaps also despite them, the *DRP15* has found its own balance. It has all worked out rather nicely and the unit has aligned itself quite naturally as a reverb processor of the deluxe type. It is not difficult to use, has no pretensions to fashion but delivers superb reverb treatments that are distinctly different.

The *DRP15* is one of those unlikely units that we must be grateful for whenever we chance upon them. The price will fade in proportion to how open to being impressed you will allow yourself to be. Check it out.

Zenon Scheope

Dynacord Electronic GmbH, Siemenstrasse 41–43, D-8440 Strubing, Germany.
Tel: +1 9421 3103.
UK: Shuttlesound, 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX.
Tel: 081 640 9600. Fax: 081 640 0106.
US: Dynacord Electronics Inc, PO Box 26038, Philadelphia, Penn 19128. Tel: +1 215 482 4992.

Deceptively fine reverb from a modest box

New Korg workstations

In traditional vein the Korg X3 music workstation pulses with 340 multisounds

Korg have launched the *i3* synth, which the company claims to take the concept of the workstation a step further with the introduction of an interactive element. The *i3* has a card slot which accepts 'Style' cards allowing arrangement to be created of a particular musical type in addition to 64 onboard Styles. The system amounts to a rather sophisticated form of real-time auto accompaniment with each Style having four variations, 48 chord variations, two intros, two endings and two fills. In performance, the Styles react to a player's chord changes and key, can be manipulated live and do not require General MIDI libraries to be loaded from disk.

The *i3* uses Korg's Advanced Integrated Synthesis system and has 32 voices, 32 oscillators and dynamic voice allocation with 6Mb of 364 waves and 164 drum sounds, plus 64 user programs and 128 GM programs. It comes with two digital multieffects processors and a 16-track 32,000-note standard MIDI File compatible sequencer with ten songs. The keyboard is 61-note with aftertouch and can be zoned for chord scanning in Styles for upper, lower and the full keyboard. Shipments are expected in September (UK price is £2199 inclusive of VAT). The *i2* is expected later (for around £2750) with an increase in cost attributed to an increased memory.

In a slightly more traditional vein, the *X3* workstation uses the AI² synthesis system of the established *01/W* and comes loaded with 6Mb of waves in 340 multisounds. New samples for the synth include piano, organ and strings plus guitar, bass and ethnic instruments controllable by VDF and VDA plus two stereo multieffects processors with a palette of 47 different effects types.

The *X3* stores 200 programs and 200 combinations internally, with 200 combinations added on a RAM card and there are 128 GM programs, one GM drum program and seven user drum patches. Each drum can be programmed for key range, tuning, level, decay, exclusive group, pan and effects send. The *X3* (at £1399 inc VAT in the UK) it is unusual in its price bracket in having a disk drive plus a 32,000-note 16-track sequencer.

Of special interest to players taken by the idea of the *X3* is Korg's release of a mixed-mode demonstration CD for the keyboard which combines audio tracks playable on a standard CD player and CD-ROM ability for Apple *Mac* computers. The latter is effectively an interactive catalogue for the synth which allows the user to explore different areas of the instrument including the program screens. While listening to the music from CD-ROM the viewer can interact with the programme and call up on-screen notes on how the *X3* was programmed for a selected portion of the music track.

Korg could well be starting something here, as this method of product presentation has obvious benefits over the traditional video cassette, audio cassette or brochure marketing approach. The CD-ROM is available free of charge from Korg in the UK.

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

P&G go MIDI

Fader specialists Penny & Giles have released details of their *MM16* MIDI controller box, which is, interestingly, said by the company to be the first in a new range of digital studio related hardware.

The box comes as a rackmount or stand-alone unit and features 16 key switches, 24 push buttons, MIDI Machine Control transport keys, a dial and 16 faders all of which are programmable. Most importantly, the faders are P&G's endless belt type —the translucent caterpillar track devices—which incorporate an LED ladder in their base.

Anyone who has used MIDI hardware controllers with faders will already be familiar with their single biggest disadvantage: because the faders do not move and won't move at the price, nulling fader position against continuous controller data is a bit of a nightmare hit-or-miss affair at best. The P&G continuous belt is able to display incoming data values on its LED ladder which can then be merged, modified and replaced by fader movement. This promises to be a very big leap in real-time MIDI control simply because it will allow the visualisation of data on the controller itself, and thus the manipulation of sound modules and effects units should become more of a repeatable pleasure. The fact that so many assignable switches are thrown in with the faders takes the *MM16* way beyond the scope of simple MIDI data mixing.

The unit arranges the storage of complete assignments of the faders and switches in 50 patches with 100 snapshots of these patches containing MIDI data values. Memory can be extended though a RAM card slot.

Penny & Giles, Blackwood, Gwent NP2 2YD. Tel: 0495 228000.

ART Express

ART have upgraded the really rather good *SGX2000* tube guitar preamp-multieffects unit to *Express* status. Outwardly everything looks much as before but inwardly it has had a severe tweaking.

There are more than 450 new presets and up to 20 effect types can be used simultaneously—and that involves choosing from a portfolio that includes more than 70 different effects. There are new routing, stacking and combining methods including being able to stack all three preamp channels together. Finally, the unit has new reverb and effects algorithms and two octaves of multi-interval pitch shifting.

Still on the ART case, the company has branched into guitar amps with the *Attack Module* series. These are unusual in employing moulded carbon fibre composites in the cabinets and while the literature makes much of the 'virtually vibration free' nature of these over traditional wood enclosures the big benefit must be stressed as being low weight rather than mechanical efficiency.

The range of combos offers tube or solid state preamp stages in 100 Watts and 80 Watts respectively, each coupled with a single 12-inch, twin 8-inch or twin 6.5-inch speakers. Features include clean and overdrive channels, stereo effects loop, effects loop bypass, chorus and reverb, ART's *Quad S* surround simulator and a buffered and equalised direct output for recording.

ART, 215 Tremont Street, New York 14608 USA. Tel: +1 716 436 2720.
UK: Harman Audio, Borehamwood Industrial Park, Rowley Lane, Borehamwood, Herts WD6 5PZ. Tel: 081 207 5050.

Music News is compiled by Zenon Schoepe

MONITORMANIA

Ilpo Martikainen and Nick Zacharov discuss Genelec's research into directionality and waveguide design for high and midfrequency studio monitoring

In attempting to specify the absolute performance of a loudspeaker, the most important parameter in describing the perceived sound quality of the system is its frequency response, measured not only on-axis, but in a defined listening window. It is known that the smoother the spatially-averaged frequency response, the higher is the fidelity rating.

A flat on-axis response is all very good, but of little use to a recording engineer, moving position along the mixing desk. Although interesting in itself, the effect of phase response in fidelity rating is found to be magnitudes smaller than that of the amplitude response. This was verified in very carefully organised and statistically valid listening tests organised by Dr Floyd Toole[1], formerly of the National Research Council, Ottawa, Canada.

With the aim of improving the absolute sound quality of the loudspeaker system, it is essential to have a working understanding of the problem sources within the reproduction environment. This refers to sources of aberration, which in practical terms implies the loudspeaker's and control room's respective intrinsic performance as well as the interface between the two.

To the critical loudspeaker designer an understanding of the loudspeaker-room interface can be very useful in helping to locate problem sources and reduce their effect through appropriate loudspeaker design.

One of the important phenomena affecting sound radiation of a loudspeaker system is called diffraction. Simply, this means that any acoustical discontinuities—for example the loudspeaker cabinet edges—act as secondary sound sources. As the cabinet front panel usually has four edges, the system thus has five radiators operating at the frequency of interest (the actual driver plus four secondary sources). The resulting frequency response, at the listening position, is the sum of all these sources. **Fig.1** is a graphical interpretation of this phenomenon. Clearly the summed frequency response has some aberrations due to the summing of components with different arrival times. As the listener moves off-axis (**Fig.1b**), the relative distances of the secondary sources will change. The summed response heard at the listening difference is now different due to the noncoincidence of the driver and secondary sources. This can be seen in **Fig.2**, measured using a 5-inch, wide-bandwidth woofer mounted centrally in a square cabinet baffle. The existence of secondary sound sources ultimately degrades the response of a loudspeaker. On-axis the ripple has large amplitude due to the summing of coincident edge diffractions. Off-axis the summing is no longer coincident and the path lengths are different, which results in lower amplitude ripple but spread over a wider bandwidth. These effects are usually seen at and above midrange frequencies due to the physical dimensions of the discontinuities compared to the wavelength of the initially radiated sound.

Diffraction in loudspeaker cabinets has been carefully documented in acoustical textbooks. For example Dunn, Muller *et al.*[2] and later followed by Harry F. Olson[3] did much research using a tiny (and thus omnidirectional) loudspeaker placed in various boxes and measured the effect of the box. The box shape and the driver location are obviously the key factors. The best measured shape being a sphere and the worst two being a cube and the circular end of a cylinder, both with a centrally (symmetrical to all edges) located driver unit.

The practical meaning of diffraction is obvious, as is the need to minimise its effects. To reduce the frequency response ripple the distances from the driver to cabinet edges should be different so that the ripple pattern is distributed over a wider frequency span. This has, in effect, the same result as that shown in **Figs 1b** and **2**. However, during off-axis listening the frequency responses of left and right speakers will differ which blurs the stereo image. This is the reason why many speakers of this type are built in mirror pairs. A lot of speakers, however, are not and they inevitably suffer, in one way or another, from diffraction problems. These may not be visible in the published frequency response graphs, as the manufacturer wants to present his product in the most favourable way, but they may still be clearly audible.

A useful and simple test of stereo performance is to use a mono signal. The centre image should be sharp at all frequencies. If the system is ▶

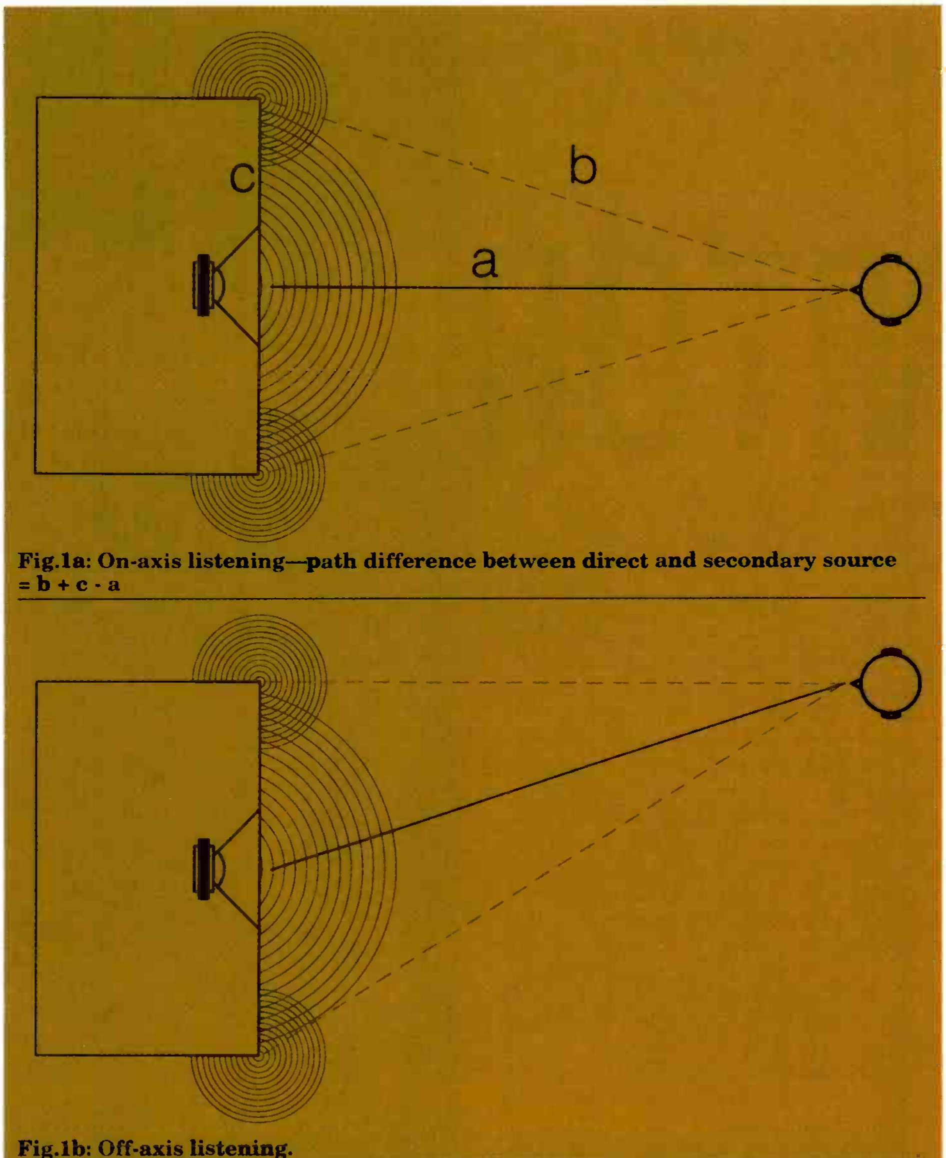


Fig.1a: On-axis listening—path difference between direct and secondary source = b + c - a

Fig.1b: Off-axis listening.

Fig.1: Sources of secondary radiation

good, the mono source seems to be inside the listener's head (like headphone listening). Reproduction of a mono signal is a simple way to check the system's ability to reproduce phantom images in general. The mono signal is only a special case of stereo, a signal where both channels are identical. If the mono image is sharp and at the centre, where it should be, then the system's stereo performance is likely to be good. If the mono image is wide, not a sharp point in space, and also frequency dependent and so on, then the stereo performance is usually also bad. However, it is worth noting that diffraction is not the only reason for poor mono (and stereo) performance. There are various other reasons but these lie beyond the scope of this article.

A second important factor that affects the loudspeakers' response, at the listening position, is the interference of first, or higher, order reflections. The loudspeaker itself is obviously not the only object in the control room that can cause frequency response problems. In fact, there are numerous machines and surfaces that reflect certain frequencies towards the listening site. From the human ear's point of view it does not matter what the cause of a delayed signal is; imaging is lost as soon as the delayed signal arrives in a suitable time window and from an acceptable direction related to the direct sound (ref. Haas and precedence effects).

Ideally the control room environment, in particular the listening path, should be clear of any equipment that might cause interfering reflections. This is, nonetheless, rarely the case. There is inevitably the mixing console and other rack equipment to cause aberration. Naturally the control room designer will (probably) have done his best to avoid early reflections from the walls and ceiling. However, an important fact is that the majority of the spaces called recording studios have had very limited advice from an acoustician.

For the loudspeaker designer this poses a problematic range of environments in which his loudspeakers must function optimally. Traditional direct radiating two and three-way designs may work satisfactorily in geometrically well-designed control rooms which are sufficiently damped. In poorer environments their responses may be much degraded due to their wide radiation angle introducing room reflections. This is far from the expectations of the loudspeaker designer, nonetheless, a widespread phenomenon.

Both the diffraction and early reflections are phenomena mainly affecting the stereo performance. The latter, especially, is well known and considered by studio designers. This has been mainly due to the widespread use of measurement equipment allowing for the visualisation of time domain impulse responses, where the room reflections are clearly visible.

There is, as yet, one important parameter which has not been covered, when considering the perceived balance of the reproduced sound. This is the loudspeaker's power response. To expand; the power response is the loudspeaker's total radiated output, not just on-axis, but in all directions around it. It can be measured in a number of ways[4], for example by measuring the loudspeaker's sound pressure level in an anechoic chamber at various points around a sphere (or hemisphere), and integrating the results. A more direct method is to place the speaker in a reverberant room and measure the power response directly, as the room integrates the signal by its very nature.

Inevitably, when listening in a control room, ▶

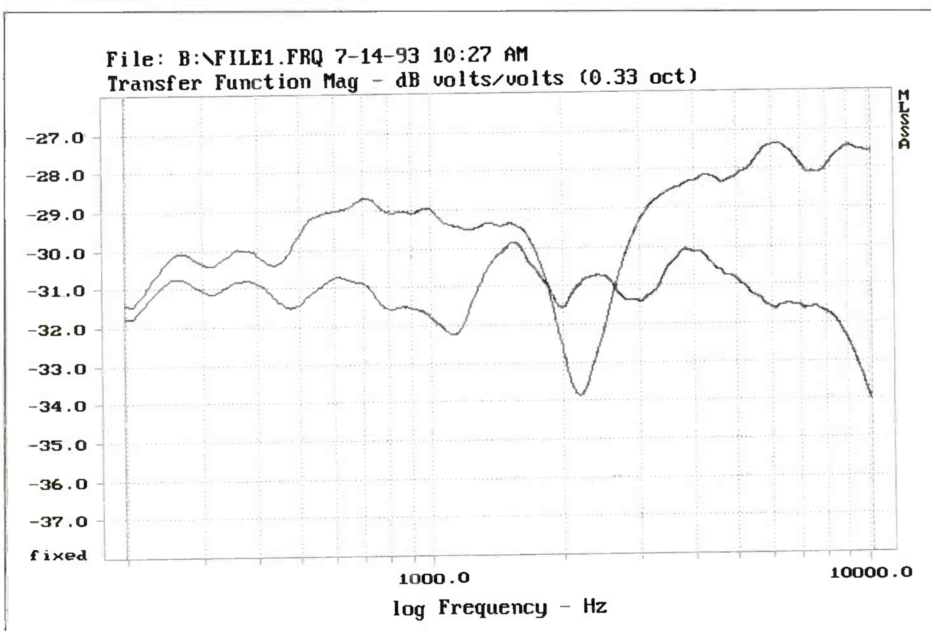
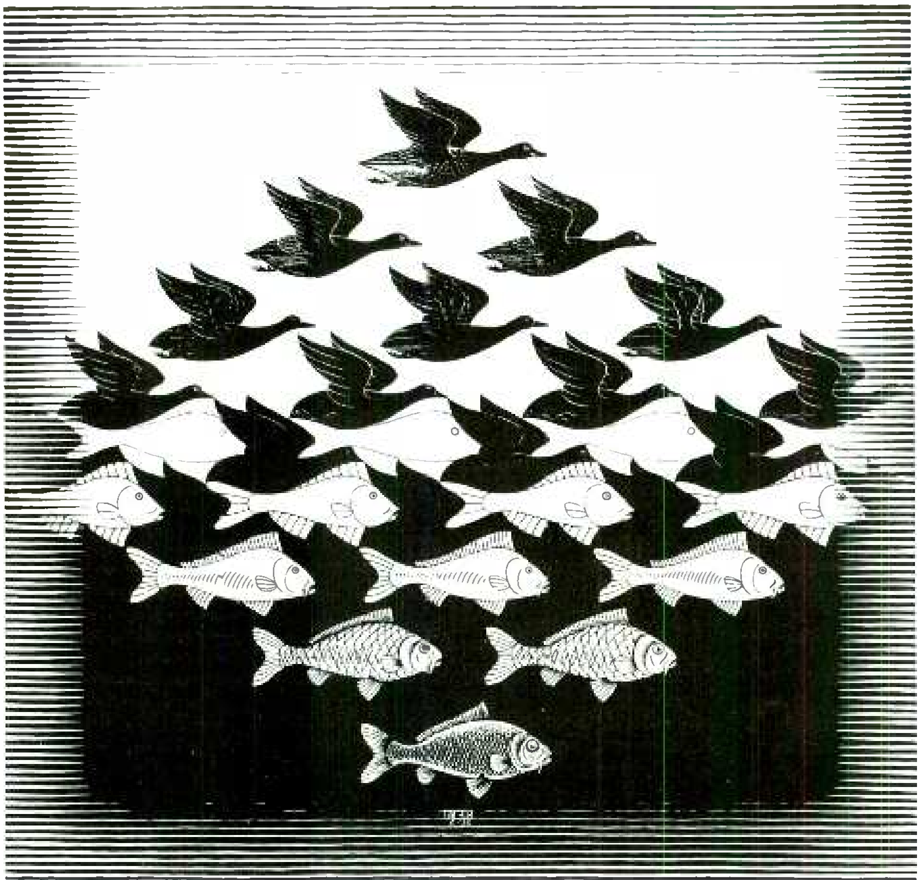


Fig.2: Measured edge diffraction effect at 0° and 30° off-axis using a 5-inch, wide-bandwidth woofer in a square cabinet baffle. First on-axis dip occurs when the path difference, b + c - a = l/2 (see Fig.1a)

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



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.

This is exactly why our engineers strive to produce studio monitors that deliver sound with unfailing accuracy. And, why they create components designed to work in perfect harmony with each other. In the laboratory they work with quantifiable parameters that do have a definite impact on what you may or may not hear. *Distortion*, which effects clarity, articulation, imaging and, most importantly, listener fatigue. *Frequency Response*, which measures a loudspeaker's ability to uniformly reproduce sound. *Power Handling*, the ability of a loudspeaker system to handle the wide dynamic range typical of the digital domain. And, finally, *Dispersion*, which determines how the system's energy balance changes as your listening position moves off axis.

Models pictured (L-R)
3-Way 10" 4410A, 2-Way 8" 4408A and 3-Way 12" 4412A

The original 4400 Series monitors have played a major role in recording and broadcast studios for years. Today, 4400 Series "A" models rely on low frequency transducers with Symmetrical Field Geometry (SFG™) magnet structures and large diameter edgewound ribbon voice coils. They incorporate new titanium dome tweeters, oriented to create "Left" and "Right" mirror-imaged pairs. Refined crossover networks use conjugate circuit topology and tight tolerance components to give 4400A Series monitors absolutely smooth transition between transducers for perfect imaging and unparalleled power response.

If you're looking for a new pair of studio monitors, look into the 4400A Series. We think you'll find them to be a sight for sore ears.

JBL

JBL PROFESSIONAL, 8500 BALBOA BOULEVARD, NORTHRIDGE, CA 91329, USA PHONE (818)893-8411

Unit 2, Borehamwood, Industrial Park, Rowley Lane, Borehamwood, Herts WD6 5P2. Tel: 081-207 5050 Fax: 081-207 4572. H A Harman International Company

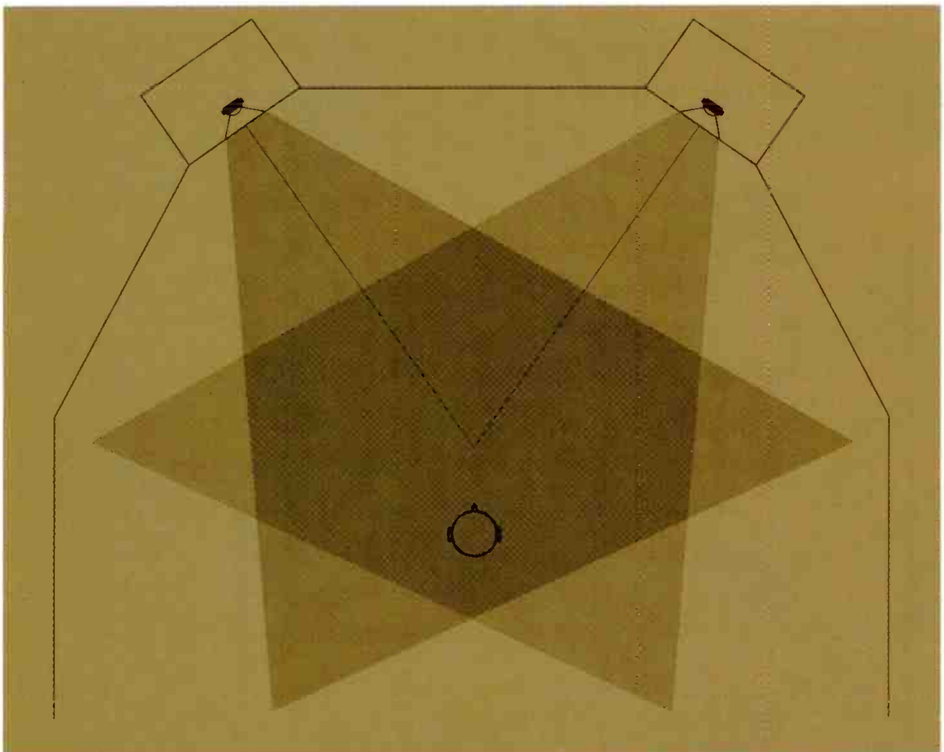


Fig.3: Desirable radiation angle of a loudspeaker

the engineer hears both the direct sound and the reverberant field. The perceived frequency balance at the listening position is thus directly linked to the power response of the system. The more omnidirectional the loudspeaker, the more the listener will hear of the rooms reverberant field. Considering its importance, it is quite surprising how rarely the power response is measured, specified or even understood.

Let us now imagine a true omnidirectional loudspeaker. Its on-axis frequency response is flat and as it is omnidirectional, it radiates identically in all directions, thus it also has a flat power response. In reality loudspeakers are only omnidirectional when the drivers' physical dimensions are small compared to the wavelength of the radiated signal. The result is that as frequency increases (wavelength shortens), the radiator becomes large with respect to the wavelength, and thus increasingly directive. Consider a system consisting of a 250mm woofer and a 25mm direct-radiating tweeter, assuming that their on-axis frequency responses are flat. The woofer becomes increasingly directive with frequency and so its radiation angle is narrower at the crossover frequency. At this point there will be a radical change due to the fact that the tweeter is acting virtually omnidirectionally and thus both its on-axis and off-axis response are flat. It is thus radiating much more power into the room than the woofer. The power response will therefore have a peak just above the crossover point, compared to the upper working range of the woofer. As frequency increases further, so the tweeter also becomes more directive and so the power response begins to roll off. If no effort is made to correct this and the on-axis response remains flat, the net result will be an uneven power response, usually having dips below and peaks above the crossover frequency. The practical meaning of this is that when a peak occurs, more room effects will be heard and less of the direct sound. The net effect is thus dependent upon the listening environment, resulting in the perceived balance varying from room to room.

Having gained a certain amount of understanding into some of the major sources of anomaly within the loudspeaker-room interface, methods of avoiding these errors can be sought in the loudspeaker design.

To ensure that the loudspeaker performs as optimally as possible, both in good and poorly designed control rooms, both the diffraction and early reflection sources should be minimised or, preferably, eliminated.

If we take a step back in the design chain to the driver design and testing stage, we can learn something fundamental about avoiding diffraction. At this point the drive units are tested for their absolute performance, before there is any knowledge of what type of cabinet with which the driver will be used. To avoid all problems associated with diffraction, the driver is flush-mounted into a so-called infinite baffle.

We can use this idea more practically by carefully flush-mounting the loudspeaker into the control room wall. In this way, basic cabinet edge diffraction can be avoided and several other sources of discontinuity also eliminated. This result can also be achieved by making the cabinet edges acoustically smooth at the frequencies of interest.

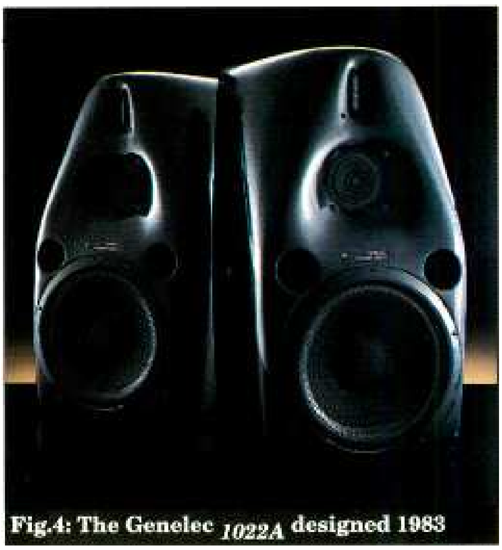


Fig.4: The Genelec *1022A* designed 1983

Flush, or soffit, mounting also has the added advantage of improving the low-frequency response through avoiding a back wall reflection. An alternative solution to soffit-mounting could be considered in the following way: if there is no radiation towards the discontinuity, nothing can be reradiated from it.

The second condition requires a way to limit the radiation angle to a desired direction only. This is only possible at mid and high frequencies and is also found to be very beneficial in other respects. **Fig.3** is an example of a loudspeaker with a preferred radiation angle, in a geometrically ideal control room.

The benefits of a limited and controlled radiation pattern become more and more obvious as the acoustical conditions worsen. The controlled and constant directivity gives great improvement in the ratio of direct sound to early room reflections. The listener is thus more in the direct field and is therefore able to listen to more of the programme material and fewer of the room effects. Subjectively, this is perceived as superior stereo imaging and detailed definition. If the radiation pattern is kept constant, a uniform and flat power response will result. In addition to improving the stereo imaging, the system will be increasingly immune to differing room conditions.

From these concepts an interesting design criterion can be developed for the loudspeaker designer: to develop loudspeaker systems that, regardless of varying enclosures and drivers, share practically the same sound radiation characteristics, both on-axis and off-axis. Although easily said, this criterion is a formidable engineering challenge.

Directivity control waveguide systems

In 1983 Genelec designed a loudspeaker system that was radically different from anything seen in monitoring, to date. The *1022A* combined both minimised diffraction and controlled directivity, (**Fig.4**). The work of Olson and others was taken a step further by matching the directivities of multiple real sources, while minimising diffraction through contoured cabinet edges. It also matched the directivities of the midrange and high-frequency drivers to allow for a smooth crossover both on-axis and off-axis (**Fig.5**).

This was a major progression from the popular two-way designs of the time, being a truly constant directivity loudspeaker. The Genelec *1022A* was the first development of the Directivity Control Waveguide (DCW).

The DCW is a novel acoustical device, which shapes the emitted wavefront in a controllable way, thus allowing control of dispersion. It is a specially curved, rigid surface fitted in front of the driver unit. It can be dimensioned for constant directivity and this feature may extend down to low midfrequencies depending on the DCW frontal area. Basically, the low frequency control limit for constant directivity is determined by the size of the DCW. The current seven DCW designs no longer require cabinet contouring to minimised the diffraction effects. ▶

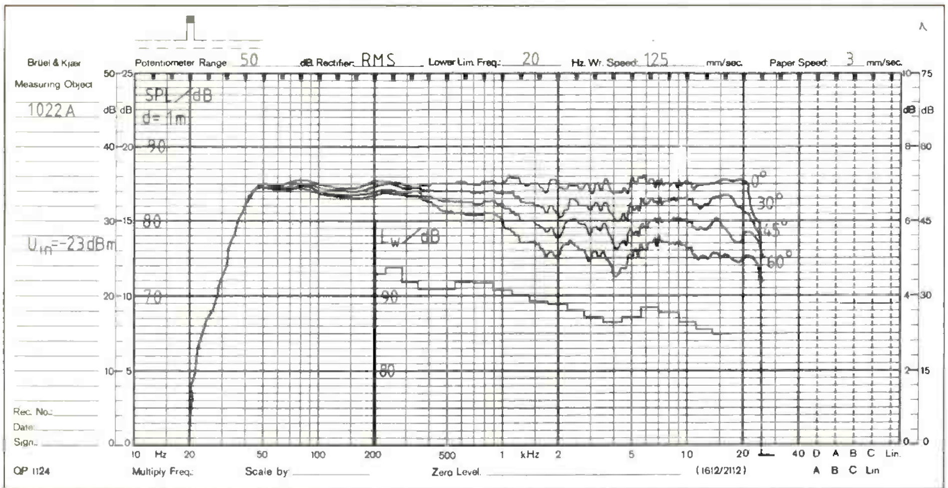


Fig.5: Genelec *1022A* frequency response and directivity curves. Note: 0°, 15°, 30° and 45° directivity curves shown. (Scale: 1dB/division.)

Wide stereo listening area

The controlled directivity has some interesting effects which, until now, have not been available in professional monitoring. Although the listening geometry follows the normal rules of equal distance between the speakers and the listener, the main monitors have traditionally been aimed to a focal point behind the engineer's position. It is believed that this practice improves stereo summing along the console (it widens the stereo listening area).

The human ear and brain use both amplitude and time cues in the localisation of sound sources. The stereo imaging thus depends on both time and amplitude differences between the left and right signals. Moving to either side of the centreline of the speakers will usually cause an image shift to the nearest loudspeaker, because the listening level of that loudspeaker is higher. With reference to **Fig.3**, imagine a system where the dispersion pattern is constant and controlled at all frequencies. Its off-axis response is flat, but the level is lower than on-axis. Next we aim this pair not behind the listening position, not even at, but in front of the listening position, that is we listen slightly off-axis to both speakers (there are no ill effects because the response is flat both on-axis and off-axis). Now assume that the listener moves to the right of the centreline; as the loudspeaker axes cross in front of him, and as he moves further off-axis of the loudspeaker, the signal level decreases. However, at the same time the distance to the right loudspeaker shortens and thus the level slightly increases. Simultaneously the listener moves increasingly on-axis of the left-hand loudspeaker. The level of the left loudspeaker will thus increase but it will be compensated by the increasing listening distance. The net result is that it is possible to aim the speakers in such a way that the imaging remains at the centre of the speakers even if the listener should move off the centreline.

Fig.6: Genelec *1038A* DCW with proprietary midrange driver

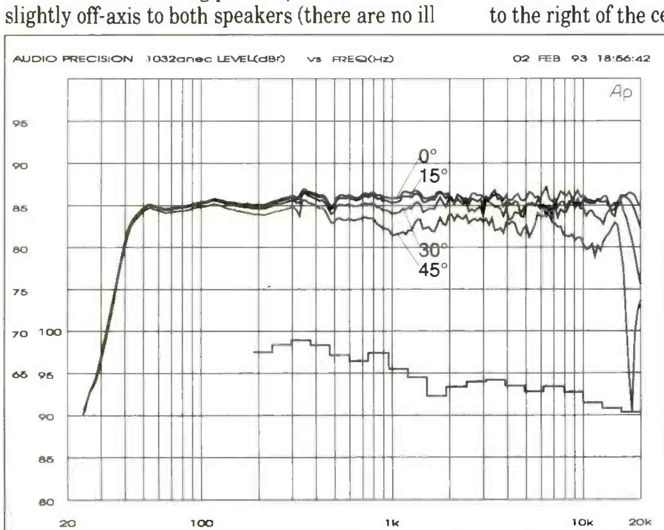


Fig.7: Typical free standing DCW monitor frequency response and directivity curves. The lower curve is the power response of the system, measured in an IEC approved reverberant chamber

The DCW systems can create surprisingly wide and good stereo listening areas; in fact, no other technology can come even close in this respect. Though the ideal listening positions will lie along the mixing console, the producer position will also have a good response.

The DCW increases directivity and thus the diffraction phenomena are minimised not only at the cabinet edges but also from any other object located sufficiently off-axis. The power response is also uniform and without peaks or dips at the crossover, see **Fig.7**. In the case of freestanding loudspeaker designs the DCW can be made a part of the enclosure construction, which enables us to reduce sources of diffraction.

f/Hz	d_2/%	d_3/%
500	0,2	< 0,1
1 k	0,25	< 0,1
2 k	0,3	< 0,1
4 k	0,4	< 0,1

Fig.8: Harmonic distortion figures of the Genelec midrange driver at 110dB continuous SPL measured at 1 m

High sensitivity and low distortion[5]

Limiting the radiation space will boost the output because the radiation resistance will increase. This basic acoustical fact helps in the DCW, because the increased directivity means that the energy is radiated to a smaller solid angle and thus the total efficiency is higher. The sensitivity of a good DCW design is about 10dB better than the sensitivity of a typical direct radiating driver.

Furthermore, the distortion of a DCW loaded driver can be very low. Genelec have achieved harmonic distortion of less than 0.5% between 500Hz and 4kHz at 110dB SPL, see **Fig.8**. This is one tenth of the distortion of compression drivers at the same SPL. At higher SPL's the difference in distortion is still higher.

The DCW technology thus allows us to control the directivity (dispersion) pattern in a predictable way. The frequency response is uniform in the listening window as is the power response. Diffraction and distortion are minimised while sensitivity increased. All of these aspects greatly enhance stereo imaging and the overall fidelity of the system with a wide variety of programme material, allowing for reliable listening in vastly differing acoustic environments. ■

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RED MEAT

Paint the town red with the Focusrite *Red 1*

Focusrite users are seeing red—at least, those who are using the latest outboard units. Patrick Stapley gets into the guts

Designer outboard has arrived! Focusrite's Red range with its lustrous anodised aluminium finish and individual styling has been turning many a head at recent exhibitions. Similarly, the first studio units are getting quite a reaction, being admired by the most unlikely of people, from cleaning ladies ('Oh that's nice dear, what a pretty colour'), to A&R men ('Wouldn't mind one of those in the motor')—I even caught one individual (who shall remain nameless) fondly stroking the review units with a faraway look in his eyes. If Focusrite's intention was to create a design to stand-out in the outboard crowd, then they must be heartily congratulated. However, for the lesser aesthetes among us, the good news is that the units also sound excellent, are easy to use and competitively priced.

There are currently two Red range units: *Red 1* is a 4-channel mic preamp, and *Red 2* is a 2-channel equaliser. Both 2U-high units have practically identical circuitry to the ISA Blue range and Focusrite Studio console, including transformers on all inputs and outputs, plus integral power supplies. The main difference between the two ranges tends to be in the manufacturing process: red units are assembled using very modern machining and production technology, which includes extensive PC mounting, while the Blue units are more traditionally put together and hand wired. This results in the Red range being cheaper and quicker to produce.

Red 1 is divided into four identical front panel sections. Each contains an input gain control (-6dBb–60dB in 6dB steps), illuminating phase reversal and phantom power switches, a circular back-lit VU meter, and a circular scribble disc for channel ID. The remaining control is a large illuminating power switch, flush mounted at the left-hand side of the unit. The only facility that some users may miss, is a gain trim for fine level adjustment.

Performance-wise, *Red 1* is excellent, having a noticeably broad frequency response with well defined low end and an open high end; mid frequencies sound both natural and well focused. The unit also deals admirably with wide dynamics, due to its low noise floor and generous headroom. In A–B tests, *Red 1* showed a marked improvement against the mic input stage of a popular console.

Red 2 has two identical front panel strips providing 4-band EQ plus high-pass and low-pass filters. EQ is arranged into two midfrequency parametrics both with x3 switching (providing a range of 40Hz–1.2kHz, and 600Hz–18kHz with a Q of 0.3 to 1.0); and two switched frequency LF and HF shelf filters (providing 33Hz, 56Hz, 95Hz, 160Hz, 270Hz, 460Hz, and 3.3kHz, 4.7kHz, 6.8kHz, 10kHz, 15kHz, and 18kHz respectively). Cut and boost is ±15dB in each case via centre detented controls. High-pass and low-pass filters are 12dB/octave ►

At the ready—the *Red 2* EQ

(Blue range uses 18dB/octave designs) each offering six frequencies between 36Hz–330Hz, and 5.6kHz–22kHz respectively.

The remaining controls are bypass switches for each channel, and input level controls (±12dB continuous) which, rather curiously, have been placed to the right of the unit where one would normally expect to find output gain controls—in fact, like the Blue units, signal flows sequentially from right to left controls. The same type of power switch is used as for *Red 1*.

Control inscriptions have been kept to a minimum, which has certain advantages and disadvantages. On the plus side is that it improves the overall appearance of the unit not to be cluttered with graphics, and this forces engineers to use their ears rather than setting up EQ by numbers. On the minus side is that it's difficult to accurately return to previous settings without first covering the panel with chinagraph marks. However, Focusrite do include a front panel layout drawing that can be photocopied and used as a setup reference sheet in the best tradition of early analogue synthesisers.

The Red range currently includes one further product, which to Focusrite's surprise has been selling like hot cakes. *Red 0* is a blanking panel (or in Focusrite jargon, a 'Rack Enhancer')

Like *Red 1*, the unit is quiet with good signal-to-noise characteristics, and silent bypass switching. Signal passed through the unit set flat and A-B'd showed no noticeable signs of degradation or coloration. The sound of the EQ has all the well documented characteristics associated with Focusrite products, and offers exemplary quality.

One of the signs of a well designed equalise is its immediacy—the 'fiddle factor' if you like. *Red 2* gave excellent, fast results on a wide range of programme material, with minimum fuss. Whether fattening-up bass, hardening or softening middle, or adding airy top, it was quick and intuitive to use.

The Red range currently includes one further product, which to Focusrite's surprise has been selling like hot cakes. *Red 0* is a blanking panel (or in Focusrite jargon, a 'Rack Enhancer') from the same thick, extruded anodised aluminium. Not only does it perfectly complement the units but it also plays an essential role in studio safety, as described in the six-page manual that accompanies each panel:

'*Red 0* is a blank of immense strength which acts as a safety barrier to prevent customers or users accidentally falling into unused rack spaces: we are very concerned that no harm should befall Focusrite customers.'

And quite rightly so. Having spent a lot of time and effort designing these stunning new units, the last thing one wants is the client bleeding all over them—although, on reflection it probably wouldn't show anyway.

The Red range will shortly be extended to include *Red 3*, a dual compressor-limiter, and *Red-4*, a 7-input stereo preamplifier designed to interface both professional and consumer devices. A further six Red units are in the pipeline and are scheduled for launch at major audio shows over the next 12 months. Also to be released soon are three new Blue range units. ■

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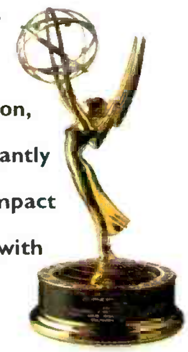

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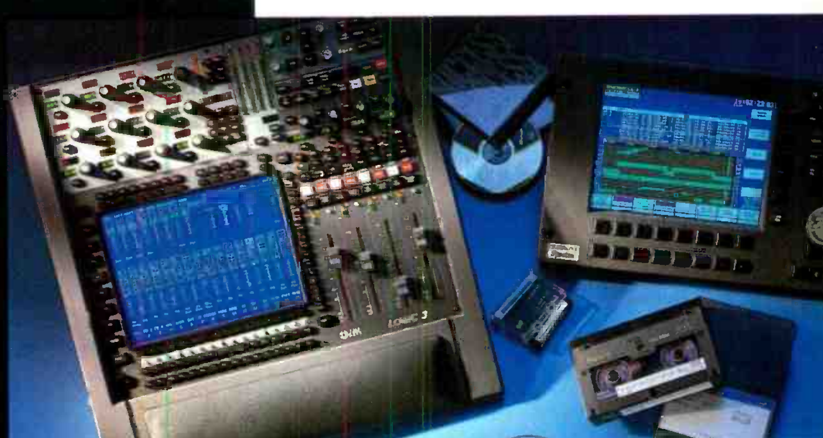

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BEAT THE CLOCK

Bob Adams of Analog Devices examines the evolution of the *AD1890* asynchronous sample-rate conversion chip—and takes in some conversion theory along the way

It is hard to argue with the benefits of the modern all-digital studio. The age of digital mixing, electronic editing, and all-digital processing is upon us, and with prices dropping we may soon see even the smallest studios adopting a mainly digital signal path. There are, however, some areas in which the all-digital approach falls short. A prime example of this is interconnectivity: the ability to readily plug different pieces of digital equipment together.

A decade ago, the average studio might include only one or two pieces of equipment that were digital in nature, a common example being digital reverberation processors. Interconnection between different pieces of equipment was mostly done in the analogue domain. An engineer could take an XLR cable and plug it into any equipment in his or her studio, and while there might be problems with hum and noise pickup, at least the signal would get through.

In the modern all-digital studio, the XLR connector usually carries an AES-EBU digital signal, and it is not at all uncommon to experience a total loss of signal when attempting to plug the output of one box into the input of the next. The thorny issue of digital sample-rate synchronisation becomes a major issue that can turn the simple task of plugging in a connector into an all-day debugging session; probably not what your client had in mind.

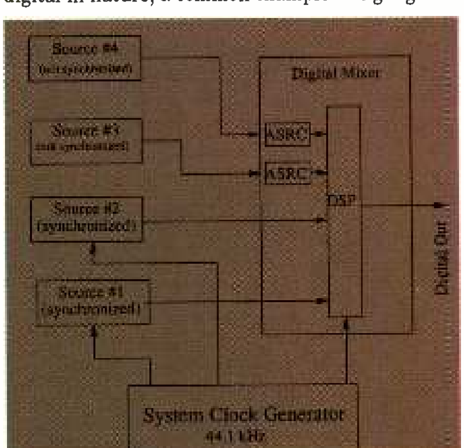

Fig.1: Typical application of asynchronous sample-rate convertor. Unsynchronised sources could include: CD players, satellite feeds, LAN connections, etc.

The digital mixing problem

Consider the example shown in **Fig.1**. Here an all-digital mixer is receiving inputs from a variety or sources. These sources might include external A–D boxes connected to microphone preamps, the digital outputs of outboard signal processing devices such as digital reverbs or dynamics processing units, the digital output of an RDAT machine or CD player, and possibly an external source coming from a remote sight over a satellite feed or high-speed network connection. In an ideal world, all of these sources would generate their digital outputs at precisely the same sampling rates, and on every tick of the clock the input samples are added together with the correct weighting applied to generate the desired mixed output signal. But what happens if one source is producing samples at a slightly faster rate than the internal clock of the digital mixer? In this case, the mixer will occasionally miss a sample on that input. In the opposite case where the input is slightly too slow, the mixer will occasionally repeat a sample. How often this happens depends on how large the sample rate difference is—a 0.1% sample rate error would cause a data error once every 22ms or so, while a 10% sample rate error would cause a data error to occur every 220μs. To ensure that this type of degradation does not occur, most digital studios are forced to purchase outboard equipment with 'sync capability', usually in the form of a back panel connector that can accept one of several types of possible synchronisation signals. By using a master sync generator (often in the form of a blank AES-EBU stream), all the equipment in the studio can be locked to the same system clock.

This ideal world is often hard to achieve in practice. In many cases, equipment with external sync capability sells for a hefty premium compared to more consumer orientated products (CD players and RDAT machines are good examples). This fact makes it difficult for the smaller studio to be upgraded to digital, as the studio owner must also upgrade much of the existing outboard signal processing equipment.

Problems can occur even in properly equipped all-digital studios with reference sync generators. Source material that was recorded at different rates (44.1kHz or 32kHz) may need to be imported into the mix. Sources that are coming in 'live' from outside the studio also represent a difficult problem, as they cannot be synchronised to the local reference clock. In many cases these external signals are nominally at the same sampling rate as the studio reference, but differ by the accuracy with which reference clocks can be generated. In this case, the audio may get through, but with periodic occurrences of skipped (or repeated) samples, often causing clicks in the program material. This situation occurs often in the video studio, where external feeds are common.

SRC to the rescue

All of these problems can be solved by a piece of equipment known as a sample-rate convertor (SRC). There are two types of sample-rate convertors. The first type is known as a synchronous sample-rate convertor, shown in **Fig.2a**. In this type of convertor, the input sample rate is converted to a new rate at the output, and the ratio between input and output sample rates must be specified by the user (normally it is also restricted to simple integer ratios like 3:2). Unfortunately, this type of convertor is of little practical use when it comes to solving real-world sample-rate conversion problems. Consider, for example, a case where we have a 32kHz input to a digital mixer running at 48kHz. A synchronous sample-rate convertor could convert by a ratio of 3:2 to give an output rate of 48kHz. So far, so good. But what happens if the 32kHz is not quite 32kHz, but 32kHz plus 1Hz? This will always happen in practice, as even the best crystal oscillators are off by 10ppm or so in their resonant frequency, and usually drift over time and temperature as well. In this case the output will not be 48kHz, but 48kHz plus 1.5Hz. Since the mixing console is running at 48kHz, it has no choice but to drop a sample every (1/1.5) = 0.66 of a second. This is a potentially audible error, especially for signals with high slew rates. Unfortunately, the mixing console cannot lock its internal sample rate to the incoming rate, because this would cause errors on the other inputs that are presumably locked to the master sync generator. ▶

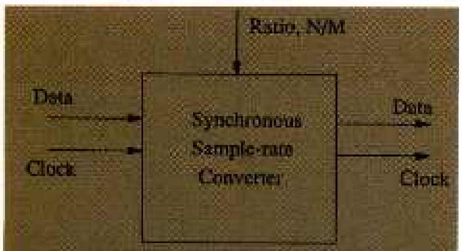

Fig.2a: Synchronous sample-rate conversion. Problem: output clock rate is still the 'master'

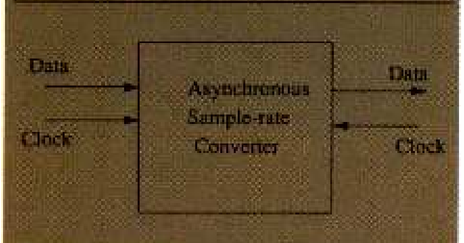

Fig.2b: Asynchronous sample-rate conversion. Ratio is sensed from user-supplied clocks and may be irrational, asynchronous and even time-varying!

Fig.2: The difference between synchronous and asynchronous sample-rate conversion

This problem is solved by the second (and more useful) type of sample-rate convertor known as an asynchronous sample-rate convertor (ASRC). **Fig.2b** shows this type of convertor, and we note that there is no longer a need for the user to tell the convertor what the sample-rate ratio is. An ASRC senses the required ratio from the input and output clocks. Note that in this case the output clock of the ASRC is an input to the sample-rate convertor, and can therefore be connected to the internal system clock of the digital mixer. This type of convertor thus operates as a clock slave for both input and output clocks.

In a true ASRC there is no requirement that the input and output clocks be related by some simple integer ratio. Ideally, even clock frequencies that are changing with time (as might be encountered in a varispeed application) should cause no audible errors. In this sense it is truly the equivalent of an analogue sample-rate convertor that uses A–D and D–A convertors, only hopefully without the signal degradation of the analogue system.

ASRCs have so far been available only as outboard signal processing boxes, often costing anywhere from $4000 to over $10,000. The reason for this high cost is a combination of hardware cost (at least two full DSP chips are required for any decent implementation) as well as the complexity of the algorithm itself, which requires a serious programming effort by a very knowledgeable DSP expert.

When evaluating an ASRC, you need to know what specs to look for. The hardest signal for any ASRC to handle is a full-level signal at 20kHz, which has the highest slew rate of any signal that can be passed through a digital audio system. The 20kHz unweighed THD+N spec is often conspicuously absent from the manufacturer's specifications, and indeed the poor performance of many units under these conditions may be one reason that many engineers claim to hear audible artifacts when using sample-rate convertors. Another area of potential trouble concerns how the unit reacts to sample rates that are nominally the same, but may drift slowly with respect to one another. Many existing units have two distinct modes of operation; one for 'up' conversion and one for 'down' conversion. Almost-synchronous clocks may cause frequent mode changes in these units, resulting in audible clocks.

To make digital interconnection as simple as analogue, the sample-rate conversion problem must become a 'jelly-bean' function; small, low-power, inexpensive, and with specs that remove any lingering doubts about sound quality. Towards this end, Analog Devices have developed a single all-digital IC (the *AD1890*) that contains a complete high-quality ASRC, with specs that meet the demanding requirements of pro audio.

While this IC will undoubtedly appear in the heart of many traditional box-level solutions, the longer term implications are more interesting. It is now possible, for example, to build a mixing console where every input contains an ASRC. Such a console would then look exactly like an analogue console; any AES-EBU signal could be plugged into any input, and the internal sample-rate of the console would be effectively decoupled from all of the external sample rates.

Numerous other applications come to mind as well; so far we have considered only the case where the input sample-rate is unknown and the output sample rate is fixed. The opposite problem exists in the design of varispeed tape recorders, where the digital output rate must remain fixed while the input rate varies with the transport speed. Again, the power of a chip-level ASRC solution lies in the fact that the functionality may now be built into the unit itself at reduced cost and power.

Theory made easy

The *AD1890* is a very complex dedicated DSP chip, with an architecture that has been specifically tailored to the task of sample-rate conversion. A comprehensive treatment of sample-rate conversion theory is beyond the scope of this article, but a brief overview will help give the reader an intuitive feel for what goes on inside the chip.

The simplest sample-rate convertor is one that uses D–A and A–D convertors, as shown in **Fig.3**. Here the digital signal is converted back to analogue, filtered with a 20kHz brick-wall analogue filter (hopefully linear phase), and then reconverted back to digital using an A–D convertor. While this system would work, it is obviously undesirable to pass the signal through so many conversion stages.

The role of the anti-aliasing filter in this diagram is complicated by the fact that the input and output sample rates may be different. If the output sample rate exceeds the input sample rate, then the brick-wall filter only serves to remove the images produced by the D–A convertor output, and hence the cut-off frequency of this filter can be fixed at half of the input rate. However, if the output sample rate is smaller than the input sample rate, then the cut-off frequency of the brick-wall filter must be lowered to prevent frequencies higher than half of the output sampling rate from appearing at the A–D input. In other words, the function of the filter changes from being an anti-image filter for the D–A convertor (in the up-sampling case) to an anti-alias filter for the A–D convertor (in the down-sampling case). For example, when converting from 48kHz down to 3kHz, it is necessary to prevent frequencies higher than 16kHz from appearing at the A–D input, and therefore the brick-wall filter must remove components above 16kHz. One of the major challenges of the *AD1890* design was to mimic this effect digitally, with a smooth analogue-like adjustment of the filter cut-off frequency during dynamic sample rate changes.

We can consider the brick-wall analogue filter of the **Fig.3** to be an infinite interpolator; that is, it produces an output voltage at every instant of time, no matter how closely one zooms in on the waveform. This leads to an interesting question; in the digital world, what interpolation ratio do we need before the difference between the analogue filter output of **Fig.3** and the output of a digital interpolation filter become negligible?

This question may be answered by analysing the worst-case signal—one that has the highest slew rate possible. In digital audio with typical sampling rates, this worst-case signal is a full-level 20kHz sine-wave. A simple mathematical analysis says that an interpolation ratio of 2^16 (65,536) is required in order to reduce the difference between analogue interpolation and digital interpolation to less than 1 LSB at the 16-bit level. This is the interpolation ratio used in the *AD1890*, and as a result of this choice the worst-case THD+N performance is 96dB with a full-scale 20kHz signal. The performance for lower-frequency signals is much better than this, and is mostly limited by the stopband attenuation of the digital interpolation filter (about -110dB).

Conceptually, all we need to do is to digitally interpolate by 65,536, and then resample this interpolated waveform by picking off the nearest computed point when an output sample clocks occurs. This may sound easy until you compute the frequencies involved; if you multiply the input sample rate times the interpolation ratio (say, 4kHz times 65,536), you get an interpolated sampling rate of about 3GHz. Fortunately, there are shortcuts. First, if we are smart we will realise that we are only using one out of 65,536 or so interpolated samples; the rest are simply thrown away. If you know the exact arrival time of the output sample clock, you can use this ▶

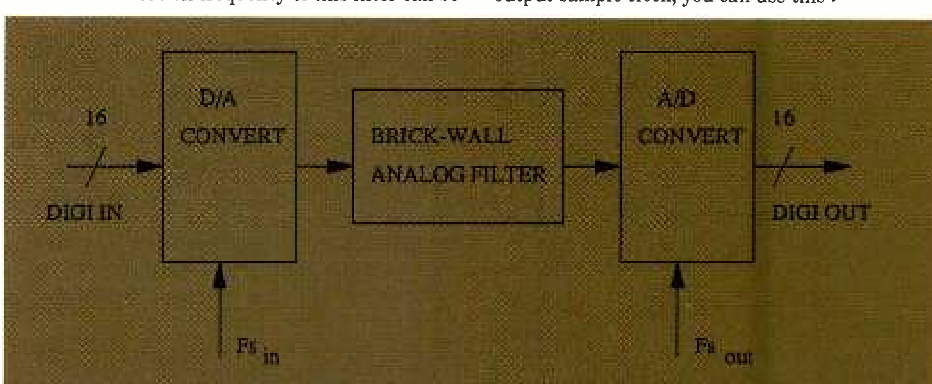

Fig.3: Analogue method of sample-rate conversion

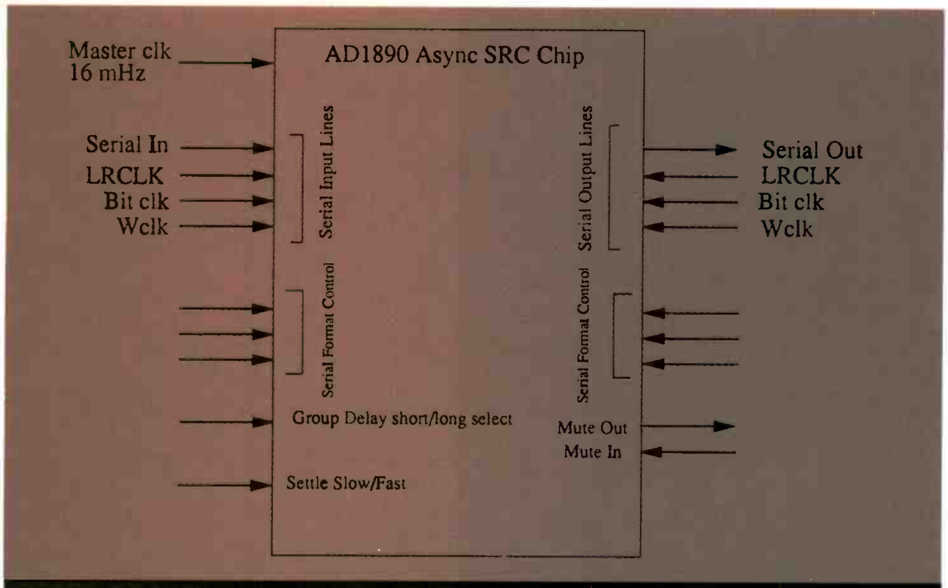

Fig.4: Control signals for *AD1890* VLSI SRC chip

information to compute only the requested output. Secondly, although the interpolation filter is extremely long (four million taps) due to the high oversampling ratio, the vast majority of the data in the FIR filter are zeros, due to the zero-stuffing operation that is an integral part of digital interpolation. Together, these two simplifications allow the interpolate-resample process to be realisable in a chip with relatively sane clock rates (16MHz for the *AD1890*). The final difficult problem to solve is setting the cut-off frequency of the digital interpolation filter depending on whether you are up-converting or down-converting. As mentioned previously, as the output sample rate falls below the input sample rate, the cut-off frequency of the interpolation filter must be reduced to prevent aliasing when the interpolated signal is resampled at a lower rate. This problem has been solved in the *AD1890* by dynamically stretching the time-domain response of the interpolation filter as the output rate falls below the input rate, which causes the frequency response of the filter to contract in the frequency domain. All this happens automatically with no programming required by the user. For example, if the input and output rates are both 44.1kHz, the frequency response of the chip is flat to 20kHz. If the output rate is then reduced to 32kHz, the edge of the filter passband is scaled downward by the ratio of 32kHz/44.1kHz, resulting in a passband of 14.5kHz.

Jitter rejection

Now we move on to perhaps the trickiest part of the algorithm. The discussion above assumed that we could determine the exact arrival times of the input and output clocks. Unfortunately, to measure this clock with enough accuracy to insure that we pick the correct interpolated output sample, we are back to needing a 3GHz clock. This problem can be solved by using the concept of internal versus external resampling clocks. The *AD1890* employs a servo-control loop that is in many respects just like a phase-locked loop(PLL). This servo loop responds very slowly to changes in input or output sample rates, and therefore tends to average the instantaneous arrival times of input and output clocks over many thousands of cycles. It produces the equivalent of an internal clock that has sufficient accuracy to reliably pick the correct interpolated sample, even though each individual clock edge coming from outside the chip is measured with relatively low accuracy (again, using a 16MHz clock).

This servo loop has several interesting implications. The first implication is that jitter on the input or output clocks will be attenuated by the transfer function of the servo loop, which has a natural frequency of about 3Hz. Jitter frequencies above 3Hz are attenuated by 6dB per octave. This is enough to remove even large amounts of clock jitter.

Another implication of this servo control loop is that step changes in the input or output sampling rate cause the internal clocks to slowly change to the new rate. During the time they are changing, the internal and external sample rates may be different. For this reason, the *AD1890* uses buffer memory to temporarily absorb this rate difference without causing data error. This buffer is large enough that real-time variations in sample rates (varispeed operation, for example) may be tracked with no audible errors. Under certain extreme conditions, a large enough step change in sample rate may cause the buffer to underflow or overflow, in which case the output is muted.

The jitter rejection capability of this chip is of extreme interest to many companies building digital equipment. One obvious area of application is in outboard D–A boxes, where jitter on the clock fed to the D–A convertor may cause many undesirable effects on the reconstructed analogue signal. This jitter is normally produced by long AES-EBU cables and the resulting high-frequency rolloff of the signal going down the cable. This rolloff causes jitter in the recovered clock. Most PLLs, especially those built in to AES-EBU receiver chips, do little to filter out this jitter, and may even add jitter of their own to the clock signal. By using an ASRC chip between the recovered AES-EBU signal and the D–A convertor chip, it becomes possible to provide a stable clock produced by a local crystal oscillator to both the D–A convertor and to the output side of the ASRC chip, and use the jittered recovered AES-EBU clock to drive the input side of the ASRC chip. Clock jitter on the input is therefore filtered by a 3Hz filter, effectively removing the effect of the jitter on the output data fed to the DAC. Other products may also benefit from this jitter-rejection capability. For example, digital power amps and digital speakers are really just glorified D–A convertors, and therefore must provide a low-jitter reference clock for high quality signal reproduction. ▶

Using the *AD1890*

For all its complexity, the *AD1890* is a breeze to hook up. **Fig.4** shows the important signals. There are two serial ports, one each for the input and output. Each serial interface is quite simple and consists of an L-R clock, bit clock, word clock (optional), and data. Because this is an asynchronous SRC, the serial control signals are inputs to the chip, not outputs.

The input and output L-R clocks are generally square waves at the input and output sample frequencies, and it is these clocks that are measured by the internal circuitry to determine the sample rate ratio. The bit clock and word clock are only used as timing signals to get the data in to and out from the serial port.

Various serial options are available to meet the most common interface standards, including the popular 'I-squared S' standard. Most common AES-EBU receivers and transmitters can interface to the 1890 with no 'glue logic' at all. In fact, a complete sample-rate convertor with AES-EBU input and output can be made with only three chips; an AES-EBU receiver, the *AD1890* ASRC, and an AES-EBU transmitter. A small amount of extra hardware may be required for AES-EBU electrical interfacing (transformers and RS422 drivers, for example). We have built a small board to test this design, and it consumes so little space and power that it fits on a 3-inch by 3-inch board and runs off an external DC-output plug-in supply that you can buy in any electronics store.

A single master clock with a frequency somewhere between 16MHz–20MHz is required for operation. There is no need for this clock frequency to have any relationship to the input or output clocks rates.

Measurement results

An Audio Precision *System One* was used to make measurements of the *AD1890*'s performance entirely in the digital domain. All measurements assume an input sample rate of 48kHz and an output sample rate of 44.1kHz. Results at other frequencies are very similar. Because this is an asynchronous sample-rate convertor, there are no magic sets of sample frequencies with markedly worse (or better) performance.

Fig.5 shows the distortion characteristics of the chip versus level for both a 1kHz input as well as a 20kHz input (which represents the worst-case input signal). Note that as signal levels are reduced, the distortion relative to full-scale is also reduced until the quantisation noise of the 20-bit input word length becomes the limiting factor. **Fig.6** shows an FFT of a full-scale 1kHz input, with an input sample rate of 48kHz and an output sample rate of 44.1kHz. Note that all distortion components are below -115dB.

In summary, a chip is now available to solve what used to be a formidable problem: connectivity in the all-digital studio. Equipment with this IC built into every input will look and feel more like an analogue system than a digital one, as any digital cable can be plugged in to any input with no regard for its sample rate. In addition, the jitter rejection properties of this chip solves the problem of recovering a clean clock from a jittery AES-EBU interface, and hence may significantly improve the sound quality of many D–A convertors. ■

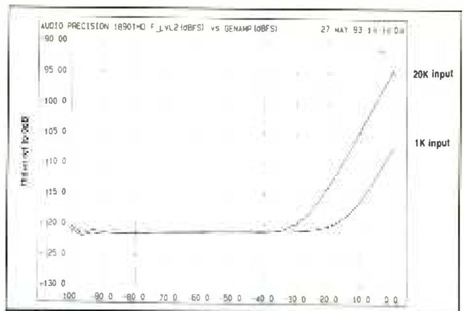

Fig.5: THD+N vs level for 1kHz and 20kHz inputs

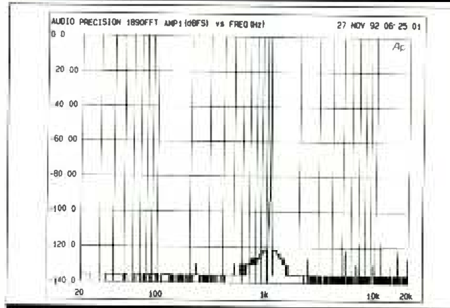

Fig.6: 48kHz to 44.1kHz input signal

AD1890 SPECIFICATION

Conversion range	2:1 up or down, with maximum output rate = 60kHz
Dynamic Range	120dB
THD+N, 1kHz	full-scale input 107dB
THD+N, 20kHz	full-scale input 96dB
Frequency response	0–20kHz (output Fs = 44.1kHz) ±0.004dB
Settling time	(to a 2:1 step change in input or output sample frequency) 200ms, fast mode; 800ms, slow mode.
Group delay	700µs (short group-delay mode) 3ms (long group-delay mode)

Note: Long group delay and fast settling time modes included for tracking very fast dynamic sample-rate changes without data errors.

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Series III Rev 7
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1987

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CAPS music sequencer
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1985

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16 Voices
Shared Memory Architecture

1982/3

CMI Series IIX
HiF sampling
MIDI implemented
Page R sequencer

1980

CMI Series II
First Real Time Graphics based Music Sequencer

1978

CMI Series I
First Sampling Digital Workstation
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1976

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8 channel Fourier synthesis
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Dual CPU architecture

AIR MAN

The decision by AIR Studios to move to bigger premises set them on a big roller coaster that is still rolling. Big budget, big rooms and big publicity. Even the problems in the building of Lyndhurst Hall (already partly documented in *Studio Sound*) have been on a large scale—exploding cement pipes; flooded basements; rotting woodwork. But on all the best roller coasters there is the *big* finish, and in AIR's case that is business through the door—and here we are talking the Big Time.

In six months names like Dire Straits; George Solti; The London Symphony Orchestra; Gloria Estefan; Neville Mariner; The London Philharmonic; Neil Diamond; The Britten String Quartet and Henry Mancini, have recorded here. Waiting in the wings are names like Steven Spielberg; The Medici string quartet with John Williams; Elton John; Colin Davis; cellist Julian Lloyd-Webber and his brother Sir Andrew Lloyd-Webber. Like moths to a flame, the cream of the world's music and movie talent are falling over themselves to see what all the fuss is about.

Now accepting its first clients, AIR Lyndhurst has crossed the line from fantasy studio to commercial studio. George Martin tells Julian Mitchell how it is being received

Keeping the flame alive is George Martin; 'When we moved from Oxford Street there were two ways of going, either we went down or we went up. I saw there was a definite niche in the market for big rooms, there aren't enough really good orchestral rooms. I saw this place and I wanted to do it—I knew we could build the best studio in the country.

'I persuaded the investors to do this. Pioneer of Japan and Chrysalis of England have put in a lot of money and a lot of faith into it, and they've been very supportive. In the end it won't be a quick return, but both these investors obviously look at it from the point of view of longevity.

'The initial capital cost will be amortised more quickly particularly if we get rampant inflation again which is perfectly possible.'

The hall of light

Usually if you wanted to record an orchestra in a room with daylight you would have to hire a hall or church. AIR's main hall has kept its full size and its tall windows and through triple glazing remains completely free from outside noise (usually a downside of hiring halls or churches).

George Martin: 'The acoustics of the hall were so good, so natural and kind. There were no twitters, no rumbles, there was no compression when you have very heavy sound. I didn't want to tweak too much, it just needed reduction of the reverb time which untreated was about four seconds.

'The first thing which we had to do was cope with the Henry Mancini session in January and I knew I had to bring the reverb time down to under two seconds. So I got the rigger of my own boat to come up from Southampton and erect a cat's cradle of rigging on which we could place deading material. We found out that the best solution was to have a combination of absorbers and reflectors, because the orchestra needs to hear each other and with 70 feet of air above you it's too far away.

'We constructed convex reflectors which we slung at an angle around the edge of the hexagon and then above on the cat's cradle we actually hung a few of those convex reflectors interspersed with panels of a material called Millitec, which is very high sound absorbent material.

'Its OK but its not the final solution. From the point of few of acoustics now its ideal for most things, everyone seems very happy with it. But I want it to look better than it does and I ►

The Hall of Light—Lyndhurst's front hall before latest acoustic treatment want it to be adjustable, so what we're going to do is to have a hexagonal reflector in the middle with six panels, interspersed with lights, suspended from the roof and these will go up and down.

'The absorbent panels will come out of the base of the organ like a kind of concertina pleat, which will then go out above the auditorium, and it will be motorised so you can retract it. That will give you in the middle range exactly what we've got now and if you take more out it'll give you a deader sound and if you take less out it'll give you a liver sound. So it will be a nice adjustable solution, and I think, quite elegant.

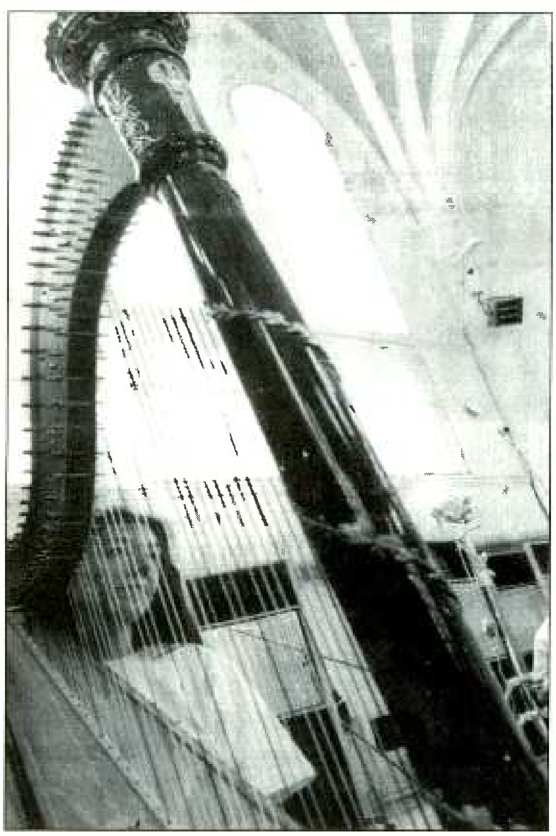
'At the moment we have Millitec lying on the galleries to soak up some of the sound, and that area will be replaced by a combination of banners hanging between the windows covered with material, and flaps in the galleries which will reveal Millitec.

A veteran's view

A man who has spent 43 years in the music business, produced the most influential band in the world and who has now opened one of the biggest studios in the world, must be worth asking what he thinks about the role of audio and how its recorded in the present day.

'These days recording engineers in the pop world haven't really got any experience of acoustics, and I've always held that its the real basis of good recording technology—that good recording starts with a good room. Because pop music has become so indoctrinated and dominated by the computer buff at home with a board, injecting stuff and mixing it which has nothing to do with acoustics at all, engineers are losing that skill.

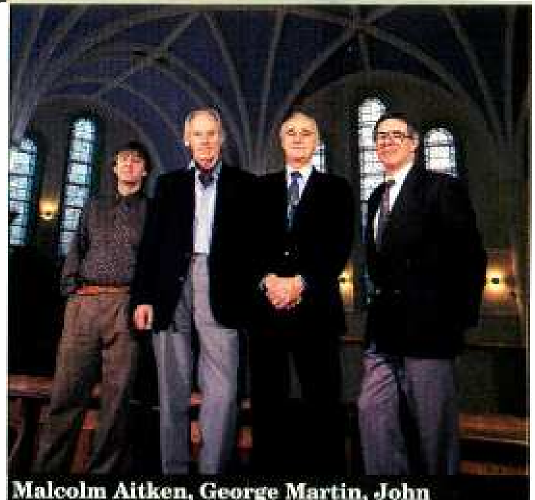
'For young musicians starting out computers are invaluable, you can make good sounds quite quickly, but where I think computers have actually made our music boring is in the kind of devastatingly accurate response they give which is not human. You have a drum box which gives you a clinically accurate beat, doesn't vary; and you have synthesisers which you can correct your own errors and get perfect reproduction, which has generated in the past a series of records which have been enormously successful of, what I call, computerised backings. It's sterile, music should flow, a heartbeat isn't programmed like a quartz clock.



'I've still got classical recordings I made in 1951 which have recently been issued on CD because they're such good recordings. I think some of those recordings, even the mono ones, were great.

'I regret the passing of the valve console, I think it had a sound which was pretty good and which you can't get with today's transistorised console and I regret, a little, the passing of vinyl.

'There's little doubt that audio and video are getting closer together and there's little doubt too that people's perception of audio has changed, there are very few real audiophiles left. In the pop world and the classical, particularly in the pop world, most people when they listen to music they see things, young people if they listen to a Michael Jackson song they are seeing him perform. They like to watch video and they like to watch TV and they buy the records because of that.



Malcolm Aitken, George Martin, John Burgess and David Harries

'In the olden days television wasn't the dominating factor; the linking of audio and video is becoming closer and closer which is why we have to cope with it.

'Anyone in the audio business who just does audio and nothing else is making a mistake.

The video union

AIR has two new TV dubbing suites each with AMS-Neve digital *Logic* desks and use the Pioneer professional Laserdisc video system which offer near-instant access and lock-up to audio and video. George Martin acknowledges the importance of having the opportunity to use this technology.

'I want to institute the same system for our film recording, but it's a case of educating the film people. When I did the Mancini picture (*Son of The Pink Panther*) we had a video wall against what is the organ wall and we also had television monitors because there is so much daylight in the room that a projection wouldn't be clear enough. If I could educate the manufacturers to put the stuff onto Laserdisc or let us put it on, then when we're recording the lock-up is great, the picture is better and you can move rapidly from one thing to another.

'I intend to start producing concerts here for television and other people have expressed an idea to do so also. I think its important to do concerts and its important to do television because it has a nice atmosphere and a nice area too.

'I'm going to propose, if we can arrange it, that on certain recording sessions with special artists who have their own control that we should actually set up videos, so they can be filmed while they're recording if they want to. It'll be very useful in the future, I wish I had videos of The Beatles stuff. Can you think how valuable that would be now? Or stuff I was doing with Peters Sellers, for example, it would be marvellous.'

Studio completion

During July and August, the main hall will be completed and the motorised sound treatment will be in place. The rear hall at the moment needs a lot of work but will be finished by next February, in the meantime George Martin has no doubt about the worth of the project.

'I've worked in many studios in America, one of the reasons I built Monserrat was that I always felt we could have a better studio than the ones I worked on in the States. It's obviously very subjective and you're used to your own space, but with my hand on my heart I can say that this studio is really the best studio I've ever worked in, that front room I think is great, I love working there.

'In ten years time people will say what a good idea this place was. It won't deteriorate, it will only get better, like a good wine.' ■

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HOME ALONE

The case for careful monitoring—studios like London's Dijiland, where MIDI is king

Philip Newell looks at the problems involved in bringing lower-spec equipment into a professional environment—and offers a little useful advice

When I engineered Mike Oldfield's *Ommadawn* album in his home studio in 1975, the term 'home studio' did not mean what it does now. Oldfield had two mixing consoles (Neve and Helios), the latest Ampex 2-inch 24-track, 24 tracks of noise reduction, 30ips Ampex stereo machines, the finest Neumann and AKG microphones, huge Westlake TMI monitors multiamped with Crown *DC300As*. The best of everything. The Helios console was the main recording console, the Neve effectively taking the 24-track outputs so that the monitor mix was gradually building itself into readiness for the final mix. Today, Oldfield's idea of a home studio is a Harrison *Series 10* desk, 48-track Sony digital machine. Again, the best equipment currently available.

A large proportion of all today's recordings are made in home or private facilities. Together with the changing face of music, the rapid advance of recording technology has made possible single-person operations where previously an entire studio staff would once have been required. But there are a limited number of people who can afford to observe Oldfield's standards. Consequently, the great majority of domestic or private project studios rely on Fostex, Tascam and Alesis products intended for private use. The performance of such equipment is quite remarkable for its price and size, but in professional terms, it remains marginal. In skilled hands, results may be indistinguishable from fully professional recordings, but the optimum performance windows are very tight.

Potential

When building large mobile recording vehicles, one of my prime specification criteria has been headroom throughout the whole system. Sound checks virtually never give a true indication of the levels to be found on the concert itself, so a wide window of acceptable performance from the recording equipment is essential. If the operational window between unacceptable noise and unacceptable distortion is too narrow, then making allowance for unexpected peaks gives a noisy recording, while aiming for an optimum noise performance ensures that unexpected peaks drive the system into distortion.

In a permanent studio installation using similar equipment, the performance ▶

windows remain the same, but circumstances usually permit more rehearsal and the possibility of retakes if things do go wrong. Another limitation of domestically-orientated equipment is the sonic performance of each component part.

Many home facilities only exist as a result of falling equipment prices. If Tascam's cheapest 24-track recorder possessed entirely pro capabilities, there would be no point making upmarket models; if they did, they probably would not sell any.

In short, the differences between the top and bottom of the range machines are the performance tolerances, sonic neutrality and the working life of the machine.

In specific instances, the sonic performance of a budget and a pro machine may be so close as to be inseparable in listening tests. But, take two of each machine and transfer four times between the two top models and the two cheaper models, and the sound on the fourth generation from the cheaper machine will almost certainly be significantly poorer than that of the pro recorder. This holds true for a great deal of semiprofessional equipment. One piece of such equipment in a professional recording chain can make one wonder why one ever paid ten times the price for its professional equivalent, but a chain of semipro equipment eats away at the performance of the overall system.

Interfacing and limitations

The input and output circuits of professional equipment are usually carefully designed to possess not only excellent headroom reserves, but also good rejection of extraneous noises likely to be picked up in the cabling. Furthermore, inputs and outputs are usually designed to operate at professionally standardised levels and impedances, which help further reduce these problems. Unfortunately, such circuits are not cheap, so semipro equipment often employs circuitry operating at lower levels and with poorer noise rejection performance. Again, one unit in an otherwise professional chain may well perform acceptably, but a complete semipro system can require very careful interconnection. Pro equipment may be less critical, but care should always be taken over the interconnection of equipment. Live recording situations, where the recording staff may not be in control of the entire chain, require particular attention to such matters. Here the tolerance of fully pro equipment can be a life-saver—sometimes literally when better earthing and isolation can (and does) save human lives.

Limited use of semiprofessional equipment can work well, but if a system is built up from as many channels, tracks, effects, processors as required in a professional studio, it is doubtful that noise, distortion and HF compression performance, and general sonic neutrality will match that of a similar chain of professional equipment. This does not seem to have registered in the minds of many private studio users, and I have even seen one commercial operation using equipment with a -37dB noise floor at the stereo outputs of the mixing console. Believe it or not, people were paying to use it, and not complaining unduly about the results.

Levels of experience

In certain areas of the world, even in parts of Europe, where top-line recording studios are scarce, a whole recording culture tends to build up based around semipro equipment. When there is no professional reference against which standards may be judged, such facilities get away with productions which are really not up to scratch. If this becomes widespread enough, it may insidiously lower general standards, with old standards becoming available to a sort of privileged class only. Such erosions are neither desirable nor healthy.

In inexperienced hands, however, semiprofessional equipment can produce remarkable results. When the limitations of the equipment are fully recognised and respected, working within these constraints can be very satisfying. Unfortunately, the ever increasingly widespread use of 'low-budget studios' virtually ensures that operations cannot develop the necessary experience quickly enough to keep pace with the continued appearance of further low-budget equipment.

With high-quality equipment, error margins are relatively wide and most people with working knowledge of the system, can achieve good results. On the other hand, a highly experienced engineer with an insight into the details of the system is required to achieve quality results when faced the narrow optimum performance targets of semipro equipment. For the average operator of such equipment, however, the best hope of reliably achieving 'expert' results is through the use of high-quality monitoring conditions.

> If Tascam's cheapest 24-track recorder possessed entirely pro capabilities, there would be no point making upmarket models;

In far too many instances, the monitor systems used in private project or semipro commercial studios are not seen as being in the direct signal chain. Consequently, as little money as possible is spent on them. Without monitors of a revealing nature, however, there is little hope of achieving reliably consistent and desirable results. With less expensive equipment, aiming for the general target area is not good enough, only hitting the bull's eye every time ensures consistent high quality results can be achieved. Great care must be taken over each stage of the recording process, and good monitoring is the only way to ensure that quality is being maintained. Somewhat ironically, when one is less certain about the sonic integrity of the recording chain, the more critical one must be in the assessment of it. The development of high quality monitoring has not produced the same reduction in cost-performance ratios as other technological developments in the recording chain. There are numerous high quality monitor systems available (by system, I am referring to the loudspeakers, amplifiers, and crossovers), but I know of none of them costing less than about two or three thousand pounds.

Getting it right

Studio monitors are *not* hi-fi loudspeakers; they are specialist devices designed to reveal details of the sound which need to be judged and dealt with before they ever reach a pair of hi-fi loudspeakers. Even top-of-the-range hi-fi loudspeakers are not generally sufficiently damage tolerant to withstand a bass drum, accidentally soloed with the master solo level set 20dB too high. With 'medium quality' hi-fi speakers and amplifiers, there is no way of addressing the subtleties which are exposed by more expensive systems. After all, it is quite reasonable for the owners of expensive hi-fi equipment to expect reputable record companies to issue high quality recordings. To allow hi-fi enthusiasts to be the first to notice that a recording is substandard is an insult to the whole recording industry.

There is, of course, an argument that a finished recording is an artistic entity in itself, and it is what it is, faults and all; but this is an insult to the record buying public (who indirectly provide our food). What is more, many substandard recordings are not sold any more cheaply than the best, and passing off substandard product at full price is an insult to the musicians and recordists who are trying to maximise recording quality.

With good monitoring conditions, quality degradations will become immediately obvious as they occur, and the recordists' attention will be immediately called to the fact. Almost certainly, if something is going to sound undesirable to the purchasers of the finished product, then it is also unlikely to be considered acceptable to the people making the recording. They may then track down the source of the degradation and make whatever adjustments are necessary.

In some ways, with clear monitoring there is more work involved, as one becomes aware of more details which need attending to. But the doubts surrounding any aspect of a recording are removed from the process. There is a satisfaction in hearing things going down as intended, and being confident that what is being heard is honest. A recording made in this way is not in danger of being shown later to be wanting when played on a top-class system. What is more, when one hears the whole process unfolding on good monitors, any adjustments to the recording chain are heard clearly, and much knowledge is soon gained about the strength and weaknesses of each piece of equipment. Noises are heard, distortions are heard, equalisation is easier to judge, reverbs are easier to judge, and the correct operating windows of each piece of equipment are clearly obvious. To anybody of reasonable intelligence and with an aptitude for music and recording, good monitors can be as good a teacher as any human being.

Room acoustics

Most studios based on the type of equipment which we are referring to here are in small rooms. Only rarely are these rooms treated acoustically in any serious manner. To a greater or lesser degree, any untreated (and some treated) rooms cause confusion. Taking any monitor system into a tiled bathroom or empty, panelled room, with rapidly reduce its ability to resolve fine detail; the sound becomes messy and confused.

It is relatively easy to deaden down the mid to high frequencies. Those that are not absorbed blur the stereo imaging, colour the sound and mask much detail by filling in the spaces in the sound with reverberated sound. When low frequencies resonate within the room, the ear will sum the ▶

The world of 'virtual acoustics'

power of all the direct and reflected sounds giving a perception of there being more bass—as the bass is all that is left in the reflected-reverberant character of the room. As well as creating a more muddy, boomy bass sound in the room itself, mixes performed in such a room may be short on bass when played over another system, if the engineer performing the mix has tried to get a 'correct' sound within the context of that environment.

Recently, some progress has been made in the low frequency control of small rooms. These techniques effectively make the rooms acoustically larger than they actually are. Some of the developments have come out of research being done in conjunction with Tom Hidley, who has very recently been building control rooms (notably the BOP TV studios in Bophutatswana) which are controlled so as to have very low reverberation times, even below 10Hz. The control rooms in question were huge—over 16 metres front to back—but the methods used 'scale' well, so control of frequencies down to 50Hz seems feasible in rooms of only about three metres in size from front to back.

Given that many rooms of the semiprofessional and private project type employ monitors which rarely respond much below 40Hz or 50Hz, even with small, high quality monitors, rooms which are highly controlled over the full frequency range of the monitors become a viable proposition. In conjunction with Reflexion Arts (Portugal) I have recently built several rooms testing these techniques, and the results have been quite spectacular. Rooms are now being built in Spain and the UK using similar techniques.

In the whole of Portugal, the number of mixing desks of Neve or SSL standard can be counted on the fingers of one hand. It is one of the poorest countries in Europe and is just finding its feet after years under dictatorial control. Life is still very controlled despite its entry into the European Community. Licences are required to ride bicycles in the cities, and all the nationals have to carry their fingerprints with them everywhere. It is certainly the closest thing to a police state that I have encountered in Western Europe. Money and power is still largely in the hands of the older people, and it has been very difficult for many younger persons to enjoy the freedom of expression which we are accustomed to in Northern Europe.

Many of the front-line studios are based around Fostex *E16*s and similar equipment. In many studios, acoustic control is rudimentary to say the least. In autumn 1991 we began building a studio for Discossete, a big-selling Portuguese record company; these studios were to be well equipped, with a Raindirk *Symphony 3324A* digital 24-track, and a moderately large control room with the absolute latest in full-range monitoring and acoustics. Many people from the recording industry came to see and to listen, and were impressed by the linearity and transparency of the sound. What is more, the entire design and construction of the studios, with a $20m^2$ granite live room, a general recording room of more dead nature, and $30m^2$ control room with the newest Reflexion Arts *234* monitors, cost less than £35,000. (The studio was featured in 'Come in Portugal', *Studio Sound,* October 1992).

Although most of the other studios in the region are more modest affairs, people began to ask us what could be done on a smaller scale. By trimming down the requirements, we managed to significantly improve existing control rooms for as little as £3000, giving a marked improvement in monitoring clarity. While some of this may not seem cheap in domestic studio terms, it soon became apparent that by spending, say £6000, on a first-rate monitor system and good acoustic control, problems which had dogged the systems for years were revealed in great detail. Once identified, these problems could be rectified. A great deal of the guess work was taken out of the recording process, so work became quicker and more efficient. What is more, the debugged systems suddenly began producing the sort of results which the owners previously thought could only be achieved through expensive upgrading and replacing of equipment. Suddenly £6000 did not seem so much when compared to the £20,000 the owners might have believed needed spending on new equipment. (The heavier theory behind all of this was described at some length in 'Monitor Systems', *Studio Sound,* March 1991 and 'The Nonenvironment Control Room', *Studio Sound* November 1991 . Under the headings of 'nonenvironment', 'monitor-dead', and 'absorbent' control rooms)

In essence, smaller rooms are fitted with free-hanging resonator panels, positioned to break up the room modes. Air gaps behind the panels are lined with absorbent, felt or Dacron, to provide good damping of the panels. The panels themselves were also treated with Dacron to kill any higher frequency reflections, and where possible, a series of angled, dead-sheet curtains were arranged in front of the panels to help to destroy low-frequency wavefronts. Ceilings are fitted with a series of sloping, full-width panels, either of deadsheets or damped resonator panels, or both, and where possible a complete membrane of Noisetec *PKB2* was fitted above. In all cases, the floors were hard and the equipment was arranged so that hard surfaces gave life to the voices of people working in the room, but did not unduly cause reflections which might disturb the monitor response. If walls are made hard to function as effective baffle extensions, and freestanding monitors used, the front walls are trapped acoustically to prevent the omnidirectional low frequency directivity of the loudspeakers causing sound to bounce off the wall and return to the listener in undesirable phase relationships with the direct signal. All of the treatment is hidden behind open frames covered in stretch fabric, giving the rooms a clean and spacious appearance.

Psychoacoustics and artistry

It is unreasonable to take an untreated room, pile in a load of equipment, all sited so that the knobs are in easy reach, put a pair of inexpensive loudspeakers in some physically convenient location, and expect to achieve adequate monitoring. Far too many people view the recording process merely in terms of equipment —this concept seems to be being reinforced by the ever increasing use of electronically-generated sounds. Where no natural sound reference exists, discrepancies in any recording chain are less obvious, a similar parallel exists in the 'virtual realities' of computer games; these stimulate our senses in a believable way, but can be so removed from reality that after extensive use, readjusting to the real world can be somewhat strange. Likewise, people spending many hours engaged in intense work in a studio, can begin to believe the what they hear is 'the truth'. In fact, as one puts more and more energy, concentration and emotion into a project, the project itself can become a reality of great intensity. The ear and brain become conditioned to the room and monitor responses; and other representation, even on a more accurate system, may sound 'wrong'.

The emotional and psychological aspects of an artistic process such as music creation and recording means that we must look at our assessment of the processes in a highly objective way. Quality control is not an artistic function, it is one of the more objective features of the exercise. Any sort of distortion is acceptable as long as that is what that the artist intends. It is then down to the consumer to choose whether or not to buy it. What is not artistically acceptable however, is when noises, distortions, poor frequency balances, harsh sounds or other afflictions affect the recording because the people involved in the recording could not hear them on the monitoring system. Because the recording process is increasingly in the hands of people who have never worked on state-of-the-art equipment or properly constructed studios, the knowledge of what is possible in terms of detailed resolution is often lacking. Experience is gained by trial and error on simple systems, rather than being passed down from highly experienced personnel in high quality studios. If more people appreciated the importance of good acoustics and monitoring, ▶

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they would realise that any extra cost is an excellent investment, allowing them to squeeze every last ounce of performance from their equipment, and giving pleasure and confidence in use of the system. The big advertising money however, is in the hands of the equipment manufacturers and suppliers, not the acousticians and consultants; the publicity battle is presently being won by people who want to sell more equipment.

Consultants and acousticians are often thought by the uninformed to be an expensive luxury, but just because they may have built multimillion-pound studios does not mean that they are all inaccessible or unavailable for small projects. When the front line of the battle shifts,many of them must get involved in order to gain a feel for the future needs and likely directions. As the tide of a battle changes, the front line is not always to be found with the generals, the elite troops, and the best equipment. Major studios are still absolutely necessary for orchestral recordings for films or large-scale productions for rock bands. The bulk of electronic-based music has shifted permanently into project studios where time and external pressures are less demanding. Like any other major change in direction though, a learning curve has been defined which deals with the development of techniques to achieve the best from what is available and to find new ways of making improvements.

> The ear and brain become conditioned to the room and monitor responses; and other representation, even on a more accurate system, may sound 'wrong'.

Trends and goals

When new possibilities arise, there can be an excitement which is reward in itself. Musicians can find much satisfaction in the creative process, but this, excitement can only be sustained if it is shared. Many musicians can be satisfied by their music in the early stages, yet the acceptance of the music by the public is usually the main source of satisfaction. It is the same for me, whether I am producing a record or designing a studio; I may be very happy about the outcome itself, but the true satisfaction comes when you see other people excited about listening to the music or working comfortably in the studios. It is the impact of the lasting impressions which is usually the true reward. It is how the end result is perceived by the consumers and users that really counts. This has to be the case unless the musician is privately wealthy—otherwise nonacceptance of the end result means no money for food, premises, or for new equipment.

There is no average concept of a domestic music system which has any meaning in terms of a reference standard for domestic listening. It is down to the music-buying public to decide for themselves how important music is to them, and according to their resources, to spend what they think fit to give them the level of enjoyment that they consider appropriate. It is therefore incumbent upon the persons making the recordings to produce products to a standard which will not leave anybody short-changed.
I believe that the only way to do this is to take the recording control room out of the equation to as great a degree as possible. This is even more appropriate now that so many people listen to music on headphones or in cars, where no room exists in the listening environment. To achieve the required sensory and emotional impact on such a wide range of possible reproduction systems, the psychoacoustic aspects of the recording process become of paramount importance. Ultimately, the 'psycho' half of the equation is the process which leaves the lasting impression in the mind, but to allow that part to be allowed to develop its full potential, it is essential that it should not be inhibited by restrictions placed on it by deficiencies in the acoustic part.

Many equipment manufacturers will not want people to read this for several reasons. Firstly, they hold the big advertising budgets which urge users to constantly update to 'latest versions', which now seem to come out at six-monthly intervals. With quality monitoring and the acquired skills which comes from its use, many people would be surprised how they could actually achieve an equally good sound from their old systems. Secondly, when people hear the degradations caused by much inexpensive—and some expensive equipment—they may make greater demands on the details of sonic performance. We have recently been pointing this out to many people when we have completed new rooms with clear monitoring. When playing a good CD through a good player, straight into the monitors, then via the console stereo returns, many studio owners and users have been shocked by the degradation through the simple stereo returns of the console. If the stereo returns can degrade to such an extent, then what is happening in the rest of the console? When typical project studio loudspeakers are used, the degradation is almost undetectable. We have challenged console manufacturers directly on this point; the response usually is 'Nobody usually complains', but with better listening conditions in the control rooms, more people certainly would. In fairness to the manufacturers, if we keep demanding more facilities for less money, something has to give. At some point, reason must prevail.

The techniques described above are relatively new in their application to small spaces, as is the really significant shift towards producing such a great proportion of finished product in what were previously considered to be only writing and preproduction environments. Even in these applications, the response is usually that working becomes both easier and more enjoyable. The subsequent transfer of such work to the final production process also becomes more rewarding. It is worth serious consideration. ■

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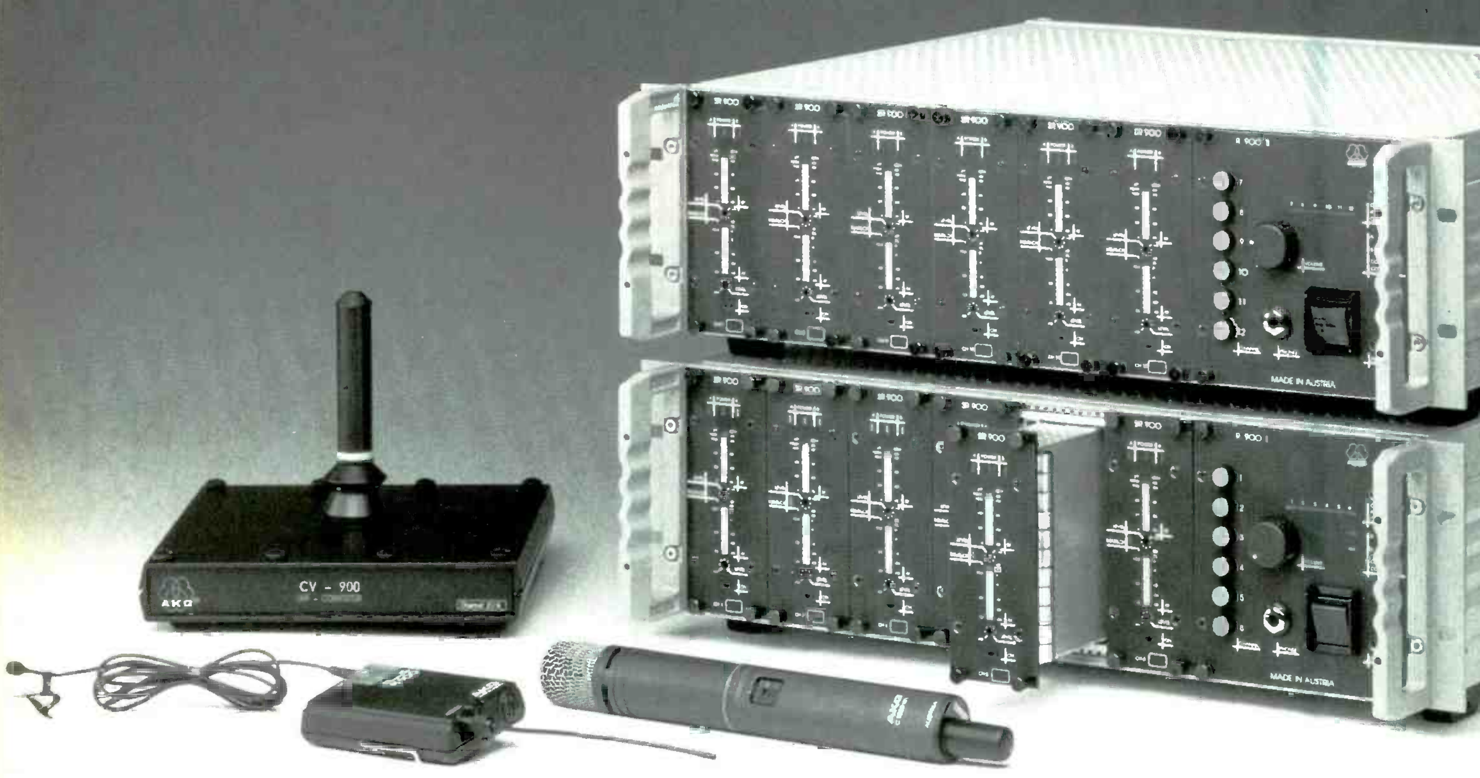

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THE PLEASURES OF MEASUREMENT

Assessing the future of audio measurement techniques. Francis Rumsey can hardly believe his ears

Ever since most of us can remember, debate has been rife in the audio industry between those who listen to and those who measure the performance of the equipment we use. Those with their heads screwed on realise that the truth is to be found in a proper synthesis of the two approaches, and that the results of both listening tests and measurements can tell us useful things about the device under test.

At one extreme there are those who will say that if a distortion, noise or error can be heard in a listening test, then it must be measurable—the worst excesses of this approach arise when measurement shows up no reason for believing that an artifact exists, engendering statements to the effect that the artifact does not exist and that anyone who thinks they can hear it needs their ears testing! At the other extreme there are those, sometimes branded the 'golden ears brigade', who pride themselves on their ability to hear minute differences between pieces of equipment (or even pieces of wire), often producing results in a form which gives grades or percentages based on subjective performance. If subjective tests indicate a difference in sound quality between pieces of equipment, but measurement shows up no good reason for this, the 'golden ears brigade' have every right to suggest that the right measurement has not been found to highlight a particular problem.

Background

Published measurements on commercially available audio devices have tended to be fairly simple in the main, although research of various types has been conducted in an endeavour to make measurements more relevant to subjective results. In this article I shall be considering measurements of electronic and recording equipment rather than transducers or acoustic measurements, since the latter field is something of a separate issue. Although in operation audio equipment is used to process complex signals, tests often involve fairly simple signals since they are easier to analyse. As Bob Stuart of Meridian has pointed out, the devices we test are intended for use as conveyors of meaningful programme material for human listeners, and this involves a cognitive element, based partly on memory and the higher levels of human information processing. We may be able to devise tests to determine the detectability of an artifact in isolation, but it is much more difficult to determine how significant that will be to a listener who has a certain set of memories, knowledge and expectations.

I spent some time at a consultancy in the early 1980s testing analogue recording equipment and performing listening tests for reviews. The measurements that we made involved such conventional things as signal-to-noise ratio, maximum output level, HF saturation, stability, wow and flutter, and so forth. We also attempted to develop more sophisticated tests to determine the effects of modulation noise which had a significant effect on sound quality, by looking at the spectrum of AM and FM sidebands which resulted, together with the increase in average wideband RMS noise floor in the presence of a signal, and discovered that we could come up with figures from these tests which seemed to correlate well with the subjective listening tests on the same machines and tapes. The differences between analogue tapes and recording devices were really quite significant compared to the smaller differences which often exist between recent digital systems, and thus the challenge to today's measurers is to find tests which are capable of distinguishing between systems which arguably sound much more similar to each other than their former analogue counterparts.

It has been argued that digital audio does not have a sound quality, and that the quality of sound depends only on the quality of conversion, both analogue-to-digital and digital-to-analogue, but this is increasingly not true. Digital signal processing (DSP) in audio equipment is capable of introducing all sorts of audible artifacts, both intentional and unintentional, the unintentional artifacts being due to factors such as requantisation and choice of filter algorithm. Sound quality in digital audio is only unaffected by a system if it has been passed through or stored entirely unmodified, such as might be the case in a simple recording system or audio interface. Even recording systems may offer options for gain control or digital redithering, which involve processing.

Acknowledging that digital audio measurements would in some cases be different to those needed for analogue equipment, the AES published guidelines on the measurement of digital audio equipment in the early 1990s, in the form of the AES17 standard. This standard documents tests which can be used to characterise the performance of digital equipment, omitting those tests which would previously have been used for analogue equipment which might give irrelevant or misleading results, and introducing new ones which would be more appropriate to the types of imperfections which arise in digital equipment (such as tests of linearity, idle channel noise, aliasing components and noise in the presence of a signal). At the time it was acknowledged that devices such as low-bit-rate coders, based as they are on psychoacoustic masking, would require additional techniques to be developed which would assess audible performance. It is true, though, that there would be advantages in relating measurements on all types of equipment to the actual audibility of imperfections, and this is the thrust of recent research work.

The goal, then, is to design systems such that the errors introduced stand the least chance of being perceived, and this requires that the measurement process takes into account the characteristics of the human hearing process.

Measurement and psychoacoustics

The availability of fast, reasonably-priced DSP has made it possible to model the human auditory system with varying degrees of accuracy, such that signals may be analysed by simulating the effect of auditory filters, taking into account such factors as masking and the perception of loudness. The processing involved in experimental systems has similarities with the auditory models used in psychoacoustic low-bit-rate coders, but in the case of at least one approach takes the form of non-real-time postprocessing of input data derived from other measurement or data analysis systems, such as the Audio Precision *Dual Domain*, or from spreadsheets of spectral data. We are not dealing here with commercially available products, but research systems which are beginning to be involved in the design process of digital audio devices such ►

> It has been argued that digital audio does not have a sound quality... but this is increasingly untrue

as convertors and processors. It is probable, though, that the fruits of this research will appear in due time both in commercial test equipment and in improved sound quality from audio systems.

Differences in approach to the problem of deriving some sort of objective assessment of audio quality using a psychoacoustic model lie principally in whether the system attempts to come up with a specific value or grade for the performance of the device under test, or whether it aims to indicate the probability that a certain distortion or noise will be audible. The latter approach is useful to the designer since it allows him to determine design trade-offs with some knowledge of the significance of the distortions allowed. Clearly there is no point in over-engineering digital audio equipment in order to make inaudible distortion even more inaudible.

Experimental results from auditory models are often quite surprising, since distortions, nonlinearities and noise spectra plotted on simple linear scales often do not correlate at all well with the same data having been analysed using an auditory model. This is particularly true with different types of dither noise and when investigating the effects of timing jitter in convertors. Most people are aware that the ear is a highly nonlinear device, and that the ear's sensitivity varies with both signal level and frequency, and in conventional measurements of noise we have used relatively simple fixed weighting filters such as the CCIR 468 family or 'A' weighting, which were designed some time ago to make noise measurements more related to their annoyance value. DSP is making it possible to use what amounts to a much more sophisticated weighting characteristic, and the results should thus be much more closely related to any subjective results.

Auditory models

A significant amount of work has been carried out in the last year or two on noise shaping in digital audio, with particular concentration on devising the 'correct' spectrum for dither noise. Clearly such research models as outlined above come into their own here, since they can help in the design of audio processors whose errors are squeezed into those regions of the spectrum which may be considered inaudible. This is also the principle on which the various noise-shaping systems work that are beginning to be used in CD mastering, such as Sony's Super Bit Mapping, SBM, which set out to give the audible impression of a dynamic range greater than unprocessed 16-bit audio.

More than one person has noted the apparent conflict inherent in testing devices which incorporate an auditory model using test equipment which is also based on an auditory model. The question would be: 'What is testing what?' Results will only be useful if the model in the test equipment is better than that in the device under test, and presumably manufacturers on both sides of the fence will be adopting the best model they can. In the end one might assume that it would come down to cost, since the price of a single piece of test equipment could probably justify more serious DSP than mass-produced audio equipment, allowing it to adopt a more comprehensive model. This is not a new problem, though: test equipment has always had to have better performance than the equipment it is testing, otherwise the results mean very little. Clearly developments in design and in testing will go hand in hand.

Given that the aims of some research projects are to design test systems which will provide a measure of perceptual quality on a graded scale, it would not be unreasonable to suggest that we should expect to see devices appearing which claim to perform 'objective listening tests' without the need for human listeners. Clearly there is some way to go here, especially if any account is to be taken of the cognitive problems, such as how the brain-memory interprets what it hears, but, since there is also significant work under way in modelling that process as well, it may not be as far-fetched as it seems. There is, of course, no substitute for human ears as the final arbiter of quality, but real human hearing can be inconsistent in judging quality, and can be fooled fairly easily. The perceptions of two people are never entirely the same, a statement which both acknowledges the blessing of diversity in humans (thank God we are not all clones of a 'standard model' as current politicians would have us be) and at the same time indicates the difficulty of determining on whose perception auditory models should be based. Notwithstanding this, tools such as those described will provide engineers with a means of approaching much more closely the limits of human perception in the design of audio equipment. ■

Further reading

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3. Beerends, J. and Stemerdink, J. *A perceptual audio quality measure.* Presented at the 92nd AES Convention, Vienna, preprint 3311, 1992.
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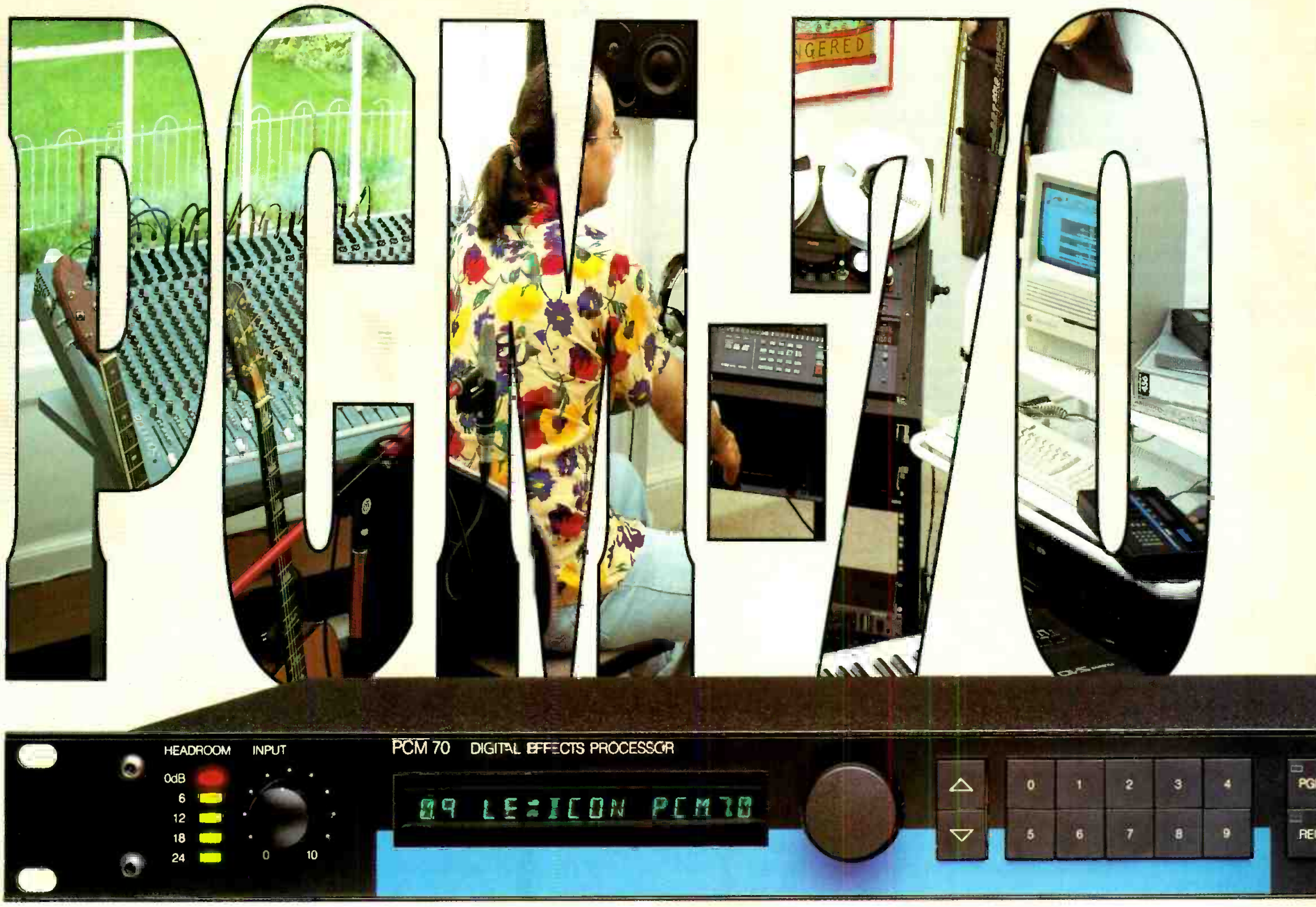
Dr FRANCIS RUMSEY is Chairman of the British Section of AES, and a lecturer on Surrey University's Tonmeister degree course in Music and Sound Recording. He is the author of numerous conference and convention papers for AES, the Royal Institute of Acoustics and the Royal TV Society, and six books on audio technology including *Digital Audio Operations* and *MIDI Systems and Control*



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4D defence

Dear sir, Mr Fox's comments in last month's *Studio Sound* are merely the latest in a series of highly personalised attacks he has made against Deutsche Grammophon's engineers and their work in developing the 4D Audio Recording system. Mr Fox's apparent wish to disparage our engineering achievements is, however, somewhat undermined as he exposes some fairly basic errors in his understanding of his subject.

His *Studio Sound* outburst reiterates several misconceptions regarding digital audio technology and adds a few more. It is bad enough that this kind of misconstrued criticism should appear in hi-fi publications but bizarre that it should appear in the pages of an international professional audio magazine; if you did not spot the errors a good number of your readers will have done so.

To begin with Mr Fox conveniently ignores any difference between recognising exemplary recording techniques and actually implementing them. Perhaps Mr Fox might provide your readers with a list of the recording companies who are in fact able to place their A–D convertors on the recording stage in venues where the mixing console may be sited up to two hundred meters away; in the robing room or crypt as he puts it. I'm afraid for the foreseeable future that list will be a list of one. Those few companies who are able to record exclusively with digital consoles remain constrained by long analogue cable runs or are very limited in their choice of venues. Only the 4D Audio Recording system is able to operate in this manner, both for multitrack and direct to stereo recordings.

The suggestion that 4D Audio Recording is based on work by any other recording company, pioneering, patentable, or otherwise, is scurrilous. Mr. Fox's comparisons of the work of Hannover Recording Centre and Decca is insulting to the engineer in both companies. The entirely separate development programmes of both engineering centres is a long established mark of each companies independence under the Polygram banner. The details of 4D Audio Recording were first made known at an internal meeting during the 1990 AES at Montreux, under the chairmanship of a Decca Recording Centre engineer. None of the sentiments exposed by Mr Fox have ever been expressed by anyone from our sister company, and they can be most readily dispelled.

Mr Fox's bases his accusation solely on his confusing the early use of dither to alleviate quantisation distortion and the modern development (by DG, Sony, Harmonia Mundi, Decca and several other companies) of noise shaping techniques to transfer properties of high-bit digital audio signal to 16-bit CD. Very complicated noise shaping algorithms cannot be compared with simple noise generators. Mr Fox might also be interested to know that Deutsche Grammophon engineers have being using time delay techniques since 1982 for mixing down when they designed and built one of the very first fully-digital delay lines (with digital inputs and outputs) to allow them to do so. Digital delay during on-location recordings has been used by DG Teams since 1986. 4D Audio Recording encapsulates years of pioneering development work by Deutsche Grammophon Recording Centre.

On the question of lunch, Mr Fox has, indeed, been invited to Hannover Recording Centre—to inspect any aspect of 4D Audio Recording he so wished and put whatever question he like to our engineers. On every occasion he has refused, preferring to quote spokesmen from other recording companies then enter into any direct dialogue with the engineers whose work he presumes to criticise. I enclose copies of two communications we sent to Mr Fox, one providing him with my direct fax number and another from Klaus Hiemann, Director of the Recording Centre, inviting him to Hannover. You have my express permission to print both these communications in order that no one should doubt our goodwill in this matter.

Mr Fox has pressed us for statements concerning the work of Sony and Decca in this field. Even, if we were in possession of this information, it is hardly our place to issue statements to the press, concerning the work of other recording companies. Providing Mr Fox with any further information on our own noise shaping techniques would involve revealing the algorithms we use and the particular way in which we implement them. This we are most definitely not prepared to do because this is the proprietary development of Deutsche Grammophon.

Mr Fox further reveals how we might improve our recording methods. Perhaps he might care to name any major classical recording companies who record large scale orchestral works with just two microphones. Decca certainly do not, the famous 'Decca tree' consisting of five microphones augmented by as many spot microphones as required.

Concerning Mr Fox's 'final thought'; nobody adds dither during copying of digital tape with the same word length, only during requantisation when mastering at 16-bit from a high-bit master. This is a practice universal to all recording companies and not unique to any one. How could such a basic error make it onto the pages of *Studio Sound*? Deutsche Grammophon—like everyone else—has done this from day one.

Mr Fox might also acknowledge, in any future correspondence on the subject of our recording practices, that far from being 'a fancy name for a balance engineer', Tonmeister is, in fact, a recognised degree qualification introduced at the end of the 1940s, both in Germany and in Poland (and later by John Borwick in the UK). Qualified in acoustic engineering, electroacoustics, mathematics, physics, score reading and, additionally, he is also required to be musically proficient to a high degree on at least one instrument. Tonmeisters presenting the missing link between technicians and artists, are qualified in engineering, producing, and editing within the whole recording area.

A more appropriate final thought concerns the comments of those who have so far reviewed 4D Audio Recordings: 'the Deutsche Grammophon sound, incidentally is superb', 'the 4D sound is utterly splendid', 'adds to the immediacy and drama of the work'. These people have, of course, only listened to our recordings.

We do not expect that our claim for our development need necessarily go unchallenged, but we do expect that, in the case of editorial copy, those who challenge us are equipped to do so, otherwise, it is difficult to justify the time that we must spend writing letters such as this in order to defend ourselves.

Deutsche Grammophon and its engineers are due an apology. Unless, of course, the Barry Fox column is some type of spoof and we have simply missed the joke.

Stefan Shibata, Head of Audio Engineering Deutsche Grammophon Recording Centre, Hannover. ■

Martin Polon

Movie theatre sound—the good, the bad and the ugly?

Over more than ten years, I have periodically lamented the state of film soundtracks in obvious contrast to the continuous improvements in film sound. This summer, centred around what could be the year's biggest movie hits, the motion picture industry has begun a campaign to bring a new and more diverse customer base to the movies with 'CD-quality digital sound'.

Foremost in the effort, Dolby Labs continue to expand their installed base of theatres equipped for Spectral Recording-Digital (*SR-D*). The Sylvester Stallone blockbuster *Cliffhanger* uses the Dolby digital format, as do 15 other films made since 1992. Sony Studios (once Columbia), intend a limited release of Arnold Schwarteznegger's *Last Action Hero* in Sony Dynamic Digital Sound (SDDS) format. Universal Studios are premiering Spielberg's *Jurassic Park* with a new CD-ROM-based digital sound system, the Digital Theatre System (DTS).

The system most likely to catch the ear of recording studio practitioners (because of its use of CD-ROM technology) is DTS. Manufactured by the Westlake Village, California firm, and having the programming as well as financial support of Universal-MCA (once Matsushita), the system uses two CD-ROM players and operates with 16-bit, 44.1kHz precision. A separate pickup is installed above the 35mm xenon projector, below the supply rollers leading from the film platter (modern theatres use platters holding the entire film rather than individual reels).

The film's soundtrack and associated time code data are stored on the CD-ROMs shipped with the film. Measured by obsolete film industry definitions, the scheme is considered 'double system', since the sound is provided by media and mechanisms separated from the film. The precision of the CD-ROM and its internal time code plus the time code on the film and the microprocessor technology in the system box guarantee absolute synchronisation. Even if there is a loss of sync or if the reels or the CDs are out of order, the system finds synchronism. The architecture of the system will even default to analogue stereo if a non-DTS 'reel' is used. Fifteen per cent of the CD-ROM capacity is used for error-reduction, yielding an average rate of one error in one trillion bits. *aptx-100* data compression is used. A single CD-ROM can store up to 3h 20m of 2-track sound plus sync information. Two CD-ROM discs are required to provide up to 3h 20m of 6-track digital sound.

The DTS-encoded film itself contains a conventional optical dual variable area stereo soundtrack in addition to the DTS time code track which conforms to the appropriate Dolby standard. If the DTS system on start-up or during projection senses the lack or loss of the CD-ROM component it switches automatically to the optical track; a similar scheme is used in competing systems.

DTS comes in two different modes: a 2-track DTS-S unit provides the conventional stereo matrix theatrical configurations plus subwoofer. The 6-track discrete DTS-6 provides L-C-R channels, split surrounds and subwoofer. The multitrack unit automatically configures itself to provide 2-track matrix or 6-track discrete digital sound depending upon the mode of the CD-ROM that is supplied with the film.

Printing the completed digitally-encoded film has been a major issue in the past. Since the DTS system does not use digital sound on film; film duplication via conventional and even high speed film printers is possible. 'Single Inventory' is one of the driving phrases of the motion picture industry and the adoption of digital sound as a motion picture industry exhibition standard has to fit the concept. Neither the major Hollywood studios nor the exhibition chains want to have to keep track of separate prints for digital and analogue theatres. So all three systems are compatible with any form of exhibition, though emphatically not compatible with each other.

Incompatibility between the systems is absolute. To begin with, all three utilise different formats and location on the film for digital sound track or time code. DTS-Universal format places the time code within the sound track printing aperture, between the conventional optical left-channel soundtrack and the picture area. The Sony system places digital audio at the inside edges of the perforations on both sides of the film. The Dolby system places digital data in the area between the perforations. Dolby use their proprietary *AC3* compression and stress the unique sophistication of their system. Sony utilises an 8-track, 5:1 compression system conceived for the Mini Disc and stress the presence of front left-of-centre and right-of-centre channels in addition to the six offered by its competitors. Data on the DTS CD-ROM is compressed via *aptx-100*. The conventional Dolby optical analogue soundtrack present in all three systems and the fact that all three systems must use digital compression schemes to cope with data density and transport speed, are the only technological conventions.

The most rapid adoption rate appears to be that of DTS. Theatres are being offered all the necessary hardware for $5000 or less—an introductory price. It is reported that more than 1000 theatres agreed to install the system for the mid-June opening of *Jurassic Park*. The system is supposed to retail at approximately $6000 after introduction.

> some analysts predict digital film audio wars could descend into a financial blood bath

The Dolby *SR-D* processor and film head-penthouse, first used for *Batman Returns*, carries an approximate $7000 price tag, and is estimated to be in over 150 theatres by midsummer. The Sony system, to be used in a limited number of theatres in Los Angeles and New York this summer for *Last Action Hero*, will have a general release at the beginning of 1994. The Sony technology has been subject to film industry speculation as to its being priced at a competitive level with other systems at $14,000, but this ignores the impact of the lower-priced CD-based system offered by Universal. If Sony take the 'we will not be undersold' approach that some industry analysts predict, digital film audio wars could descend into a financial blood bath for one or more of the three companies as well.

The fact that two of the three companies pushing specific incompatible digital sound release formats this summer are motion picture studios, with considerable direct and-or indirect theatre holdings, will have some impact on the ultimate outcome of the battle. That the third competitor has brought motion picture sound out of the dark ages and has set the standard for each successive improvement in film sound and has a name recognised since the late 1970s for optical surround in over 15,000 theatre installations will also weigh heavily on the minds of those who must make the ultimate decision to buy.

It is important to remember, however, that the improvement of recording and playback methodologies is just that; the generally dreadful state of speaker and amplifier systems in movie theatres has to be addressed as well. In fact, no attempt to install digital sound in theatres could succeed without providing the means to capitalise upon upgrade. One could make a generous rough estimate of the 23,000 motion picture theatres in the US and say that less than 5% are equipped with either the Allen *HP5-4000* sound system or the Lucas Film *THX* sound system, both considered by most analysts to be the acceptable norm for digital reproduction. In fact, if you accept the estimate for *HPS-4000* and the *THX* installed base in addition to any other number of other 'quality' system options, you are hard pressed to reach 5%.

The bottom line is more important than a battle over digital formats. Firstly, the prestige of major multinational electronic entertainment providers is on the line. Secondly, home theatre is perceived to be of major importance by the merging studio, telecommunications, TV and consumer industries—if people are to order downloads of movies for their living rooms, they must have sound quality equal to the best in the theatre. Thirdly, a 'win' in the theatre could propel a surround format into contention for HDTV—in prestige if not technology alone. Lastly, the issue is not whether we use eight channels, or digital 5.1 channels (the subwoofer is said to need only 0.1 of a channel), or six channels or two discrete matrixes into six... Ultimately, we need a standard to improve the 'quality of the breed'. ■

FLYING THE FLAGS

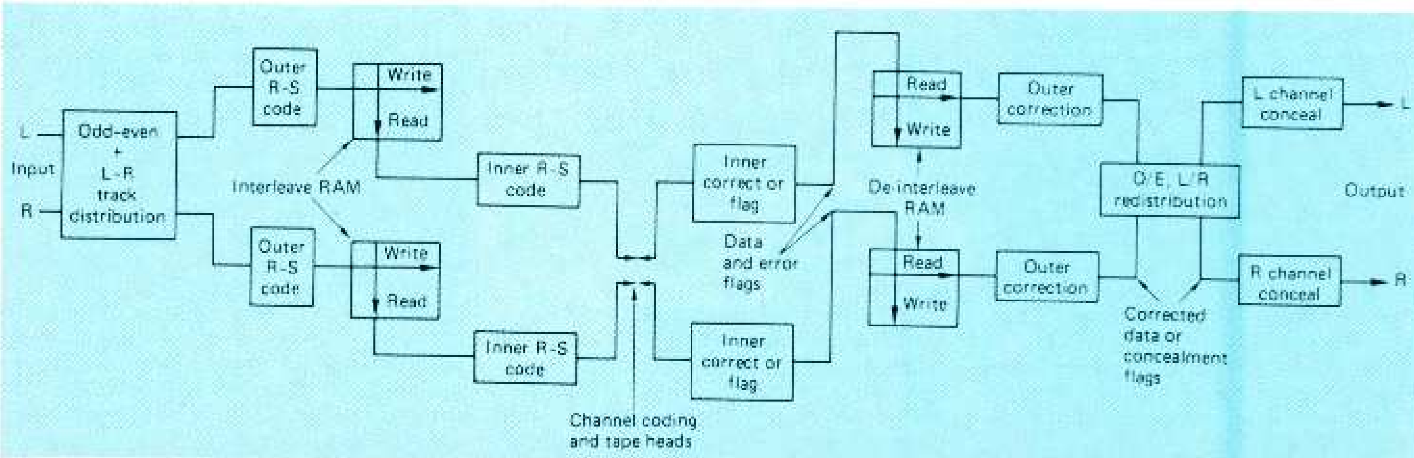

Fig.1: The error protection strategy of RDAT. To allow concealmeant on replay, an odd-even, left-right track distribution is used. Outer codes are generated on RAM rows, inner codes on columns. On replay, inner codes correct random errors, flag burst errors. Flags pass through de-interleaved RAM to outer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

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

Studio Sound's DAT Test has raised questions regarding the DAT format and testing. Here John Watkinson casts an eye over the results and draws some conclusions about their significance for the DAT user

Fig.1 shows a block diagram of the RDAT error handling system which uses product codes formed by producing Reed-Solomon codewords at right angles across memory arrays. The audio samples corresponding to one head revolution are routed to a pair of 4kB RAMs, one for each track, which have 128 columns of 32 bytes each. RDAT works with 8-bit symbols, and so each sample is divided into high byte and low byte and occupies two locations in RAM.

Fig.2 shows only one of the two RAMs. Incoming samples are written across the memory in rows, leaving an area in the centre, 24 bytes wide. Each row of data in the RAM is used as the input to the Reed-Solomon encoder for the outer code. The encoder starts at the left-hand column, and then takes a byte from every fourth column, finishing at column 124 with a total of 26 bytes. Six bytes of redundancy are calculated to make a 32-byte outer codeword, also called a C2 code. The redundant bytes are placed at the top of columns 52,56, 60, and so on. The encoder then makes a second pass through the memory, starting in the second column and taking a byte from every fourth column finishing at column 125. A further six bytes of redundancy are calculated and put into the top of columns 53, 57, 61, and so on. This process is performed four times for each row in the memory.

The inner, or C1, codewords are encoded when the memory is read in columns. **Fig.3** shows that, starting at top left, bytes from the 16 even-numbered rows of the first column, and from the first 12 even-numbered rows of the second column, are assembled and fed to the inner encoder. This produces four bytes of redundancy which when added to the 28 bytes of data makes a 32-byte inner codeword. the second inner code is assembled by making a second pass through the first two columns of the memory to read the samples on odd-numbered rows. Four bytes of redundancy are added to these data also. The redundancy in each case is placed at the bottom of the second column. Each column of memory can be accommodated in one sync block on tape. The effect is that adjacent symbols in a sync block are not in the same codeword. The process then repeats down the next two columns in the memory and so on until 128 sync blocks have been written to the tape. This uses up the PCM audio sector of one track. the contents of the second RAM are written to the other track of the frame.

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

As there are four bytes of redundancy in each inner codeword, a theoretical maximum of two bytes can be corrected. The probability of miscorrection in the inner code is minute for a single-byte error, because all four syndromes will agree on the nature of the error, but the probability of miscorrection on a double-byte error is much higher. The inner code logic is exposed to random noise during dropout conditions, and the probability of noise producing what appears to be only a 2-symbol error is too great. If more than one byte is in error in an inner code all bytes are flagged bad as they enter the de-interleave memory and the outer correction must be used. The interleave of the inner codes over two sync blocks is necessary because of the use of a group code. In the 8/10 code, a single mispositioned transition will change one 10 (channel) bit group into another, potentially corrupting up to eight data bits. A small disturbance at the boundary between two groups could corrupt up to 16 bits which would be beyond the correcting power of the inner code. By interleaving two inner codes into two sync blocks, the worst case of a disturbance at the boundary of two groups is to produce a single-symbol error in two different inner codes. The inner code interleave halves the error propagation of the group code, which increases the chances of random errors being corrected by the inner codes instead of impairing the burst-error correction of the outer codes.

After de-interleave, any uncorrectable inner codewords will show up as single-byte errors in many different outer codewords accompanied by error flags. To guard against miscorrections in the inner codes, the outer code will calculate syndromes even if no error flags are received from the inner code. If two bytes or less in error are detected, the outer code will correct them even though they were due to inner code miscorrections. This can be done with high reliability because the outer code has 3-byte detecting and correcting power which is never used to the full. If more than two bytes are in error in the outer codeword, the correcting process uses the error flags from the inner code to correct up to six bytes in error.

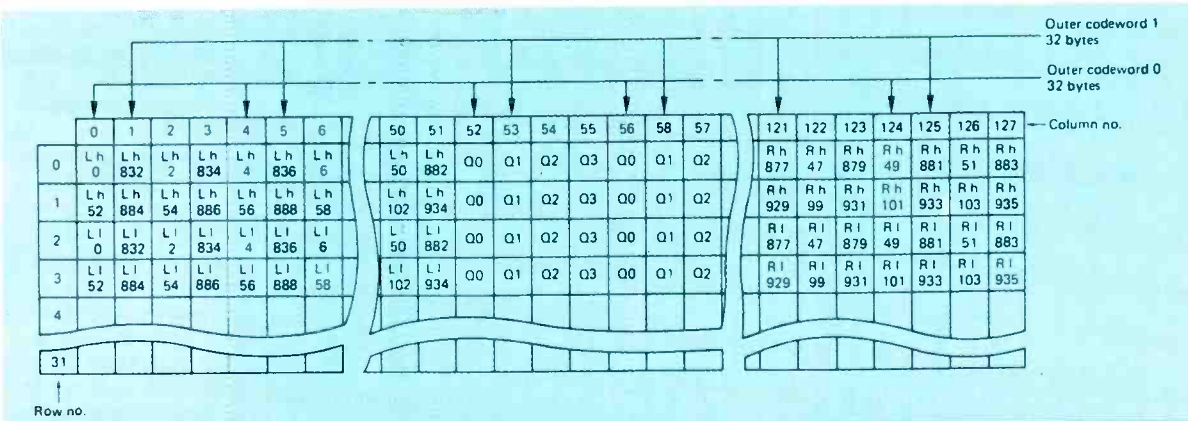

Fig.2: Left even-odd interleave memory. Incoming samples are split into high bit (h) and low bit (l), and written across the memory rows using first the even columns from L0–830 and R833–1439. For 44.1kHz working, the number of samples is reduced from 1440 to 1323, and fewer locations are filled

Fig.4 shows the correcting power of RDAT graphically. Owing to the four-way interleave of the outer code, four entire sync blocks can be destroyed, but only one byte will be corrupted in a given outer codeword. As an outer codeword can correct up to six bytes in error using flags from the inner code, it follows that a burst error of up to 24 sync blocks could be corrected. This corresponds to a length of track of 2.64mm, containing 6336 bits, and is more than enough to cover the tenting effect due to a particle of debris lifting the tape away from the head. The track angle of 6.35° means that a scratch about 0.3mm wide along the tape will cause damage of this length in every track. This limit is also reached by a transverse area of damage 2.6mm long.

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

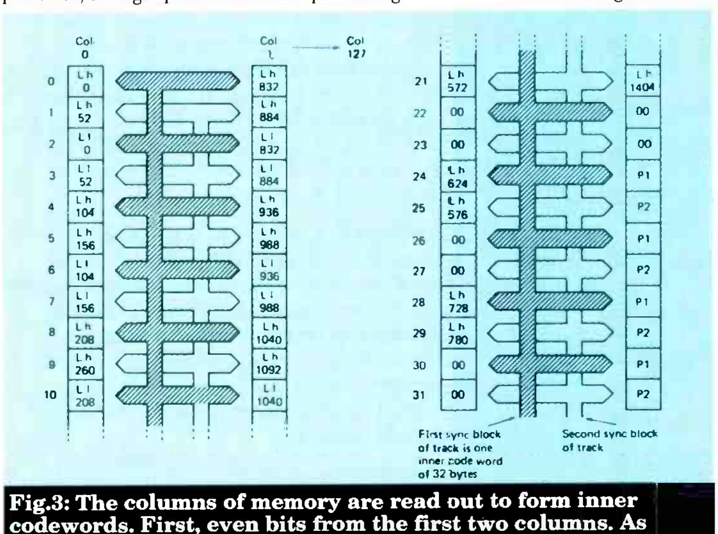

Fig.3: The columns of memory are read out to form inner codewords. First, even bits from the first two columns. As there are 128 columns, there will be 128 sync blocks in one audio segment

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

The tests

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

The Tascam *DA30* uses a Hitachi HD49211FS error correction chip which has an error flag on output pin 18. The data sheet on this chip reveals that the significance of the flag depends on when it occurs. **Fig.5** shows that, in addition to the flag bit, there are two gating signals—FSYNC and C2. FSYNC is a rotary-head-speed square wave which is high when one head is selected, low when the other is selected. This signal can be used to compare the error rate of the two heads. Following a transition in FSYNC (in either direction) and prior to C2 going high, the flag output pulses are due to inner (C1) code failures. In other words, a pulse is produced every time an inner 32-byte symbol is found *not* to be a codeword. Thus the generation of the flag simply indicates the existence of a disparity between the original and replayed data. It does not tell us how big the disparity was: it could have been as small as one bit or considerably larger. However, if the inner code is capable of correcting the error, it will do so, and only a C1 flag will result. The inner code can correct up to eight bits in error. If the error is too large for the inner code to correct—more than eight bits—the inner checker will declare the entire codeword incorrect and attach error flags to each of the 28 data bytes prior to writing them in the ▶

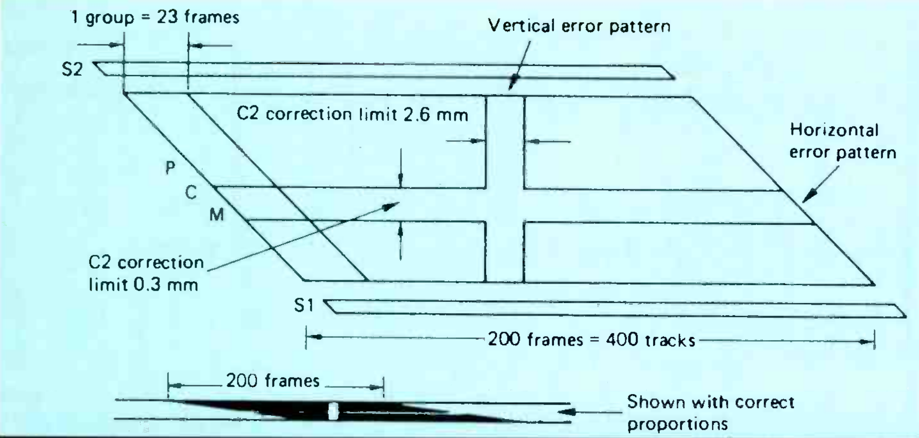

Fig.4: The correcting power of the C1, C2 codes of RDAT. The maximum correctable burst error length is 792 symbols (6336 bits). Since the linear recording density of DAT is 61,000 bits /inch, this is equal to 2.64mm along a helical track. The horizontal correctable error width is 2.64 x 6.35°= 2.6mm. The physical defects are shown conceptually in the upper diagram, and to scale in the lower diagram

de-interleave memory. On reading the memory, the outer (C2) code finds the flags and uses them as pointers to aid the correction process.

During the period when C2 is high, pulses from the flag pin indicate that a C2, or outer code, correction was needed. Between the termination of C2 and the next transition of FSYNC, the output represents flags only from the subcode. If the inner code is capable of performing correction, no C2 flags will result. However, if pointers from the inner code are passed to the outer code, or if the outer code detects inner code miscorrections (these are quite rare and can be discounted for analysis purposes), the C2 flags will result. As can be seen from **Fig.2**, there will be a maximum of 128 flag pulses during the time when C2 is high. Clearly a C2 flag indicates that there has been a larger error (up to 28 bytes) than does a C1 flag.

Unfortunately, the test procedure treated all flags as being of equal status and did not distinguish between the two types of flag, and this is, a shortcoming in the test method. The C1 flags are an indicator of the noise level of the head-tape-preamp, whereas the C2 flag results from a greater magnitude of error and is an indicator of tape surface quality rather than the noise level. I would have liked to see a separate comparison of the C1 and C2 counts.

As a result of treating the flags as being of equal status and considering the spread of the control tape, the ranking in Tables 1 and 2 cannot really be justified. In any statistical exercise, ranking should not be used if the difference between ranks is similar to the experimental accuracy. Some of the results are very close, especially in the final error rate in Table 2 for the Apogee, 3M and TDK tapes. It would have been preferable to have placed these together in the list. Similarly, the HHB and Maxell consumer tapes could have been placed equal first in Table 2.

Having said that, the control tape test shows a spread which is much less than the overall spread from best result to worst, so I must conclude that the overall results of the test are statistically significant in that the different brands of tape really do vary in the rate at which error flags are generated. The test, however, only specifically reveals a different integrated error flag rate and the results cannot be used to determine if the lower scoring tapes carry more noise or more surface defects.

The results

The tests ran for a minute of tape, and in that time 4000 tape tracks were played containing half a million sync blocks. In the worst result shown in Table 2, around 15,000 error flags were measured. If the errors were uniformly distributed, this figure would correspond to almost four flags per tape track, or about one sync block in 35 in error. This is well within the error correcting power of the RDAT code, and so there is no cause for jumping out of high buildings. On the strength of these results alone I could not refuse to use any of the tapes tested for professional recording purposes. I do, however, await with interest the results of the accelerated life tests, as I feel that these are more significant. It would have been useful to see if any of the tapes contained bursts of C2 error flags at some point within their length as this would indicate a potentially serious dropout. Performing such a test would, however take an extremely long time.

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

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

What concerns me about these periodic error rate variations is that these were observed on new media. The friction liner construction of the DAT cassette can only cause this to get worse. With repeated winding, the liner sheet wears, and with ageing the lubricants leach out of the tape and liner. The cassette itself may warp with time. The result is that the tape itself is fine, but it cannot be played because tension variations cause tracking errors which cause excessive errors. I have seen plenty of old Compact Cassettes which suffered appalling wow and flutter which were rendered quite playable by fitting the tape in a new shell. This may become necessary in the future when DAT cassettes start to get old.

Another factor which is of significance is the disparity between the rate at which contamination builds up with different tapes. Regular cleaning of RDAT transports is probably a more important factor than the variation in error rates in the brands tested. ■

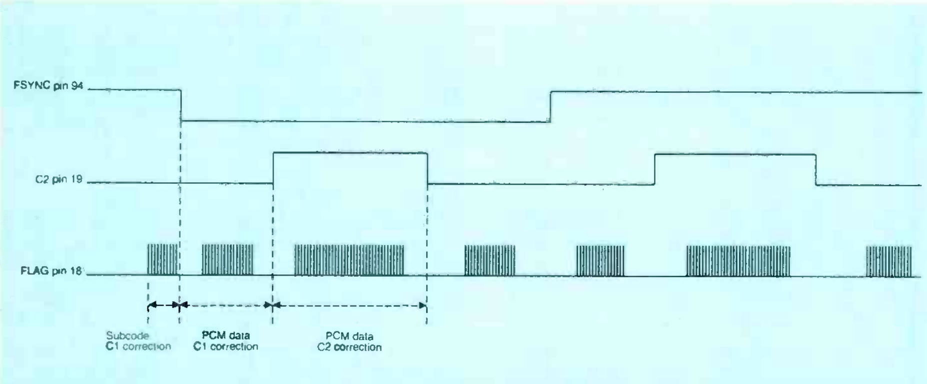

Fig.5: The *DA30* error-test points allow errors in selected areas of the tape to be gated. A maximum of 128 flags per audio blocvk and 16 flags per subcode block can be present

An alternative version of this article appeared in the July 1993 issue of *Studio Sound*. Since then more detailed information on the Hitachi chip has become available allowing the author to update certain aspects of the content and also to correct a production error that compromised its detailed description of the error correction system's operation.

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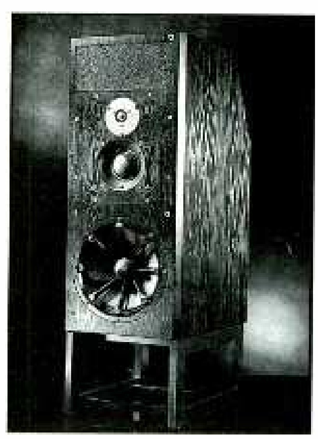

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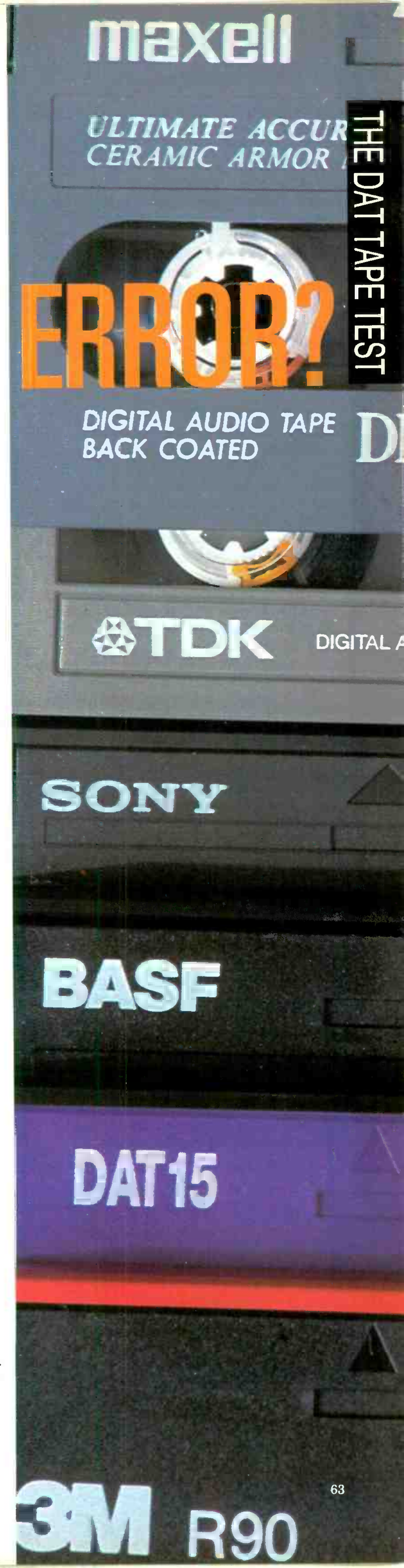
WHAT'S DAT ERROR?



Sam Wise updates the initial DAT test with new information, new tapes and new results

My poor Audio Precision *System One* has never worked so hard in its life. Round and round go the tapes, out come the graphs. Day and night it carries on its duty to professional audio investigating variable after variable to get to the roots of RDAT.

Being a protagonist of honest reviewing, and taking seriously every comment and criticism has this time taxed this poor reviewer's budget to breaking point. Yet we diligently carry on, ensuring as far as possible the unimpeachable verdict. Tape may be tape, but we are continually and repeatedly verifying that not all DAT tape is created equal. Along the way though, a few lessons have been learnt, some new activity has started in the industry, and I have received more attention from tape manufacturers than I ever considered possible. Many have been concerned, at least initially, but to a man (and occasionally a woman) they have been helpful.

Tape and only tape

As we have attempted to make clear in the initial review, the aim has been to get as close as possible to the tape itself and avoid getting entangled in questions of whether or not the errors we were seeing would become immediately audible. If it had been possible to dispense with the DAT machine and error correction system altogether, we would have done so. For what we were trying to examine is the tape, and only the tape. Unfortunately, the DAT mechanism and electronics are involved in the whole process, so what we are actually seeing is the tape and the tape machine, though the differences in tapes have been demonstrated to be almost entirely tape variables rather than machine variables.

John Watkinson's article on error correction explains the two layers of correction applied to DAT tape, and makes it clear that the likelihood of C1 errors making it through to the ear are low. However, detecting these errors, along with C2 errors does give us a closer measure of tape error rate, which is what we are interested in this case. To aid this exercise the test rig has been modified to separate C1 (inner code) and C2 (outer code) errors. The Hitachi chip inside the *DA30* produces an error flag whenever the particular level of correction has failed. A look at the tapes which we have measured using this new system reveals that all of the errors are of the C1 or inner code type. The C1 error flag is toggled whenever the inner code fails to fully correct an error, and the problem is passed further up the error correction ladder to the outer code correction system. On the tapes measured, there were no C2 errors flagged up, meaning that the outer code managed to fully correct all of the inner code errors. This means that there would be no audible errors in any of the tapes tested over the number of passes which were used in our tests. However, the possibility of audible errors is higher with the tape showing more extensive C1 errors.

I agree with John also that my original ranking cannot be taken too exactly—this was not what was intended. However, I would not have ranked Apogee with TDK, perhaps due to factors not too evident at first glance. TDK not only had the highest error counts, but also showed evidence of the worst tape deposits. But the differences between HHB and Maxell are not statistically valid without a lot more measurements—both were shown to be good and clearly better than other tapes (which is the important thing), while in our tests Maxell did consistently perform that bit better. I would not hesitate to purchase either based on initial test results.

Hardware hassles

The selection of the *DA30* as a test bed has proved contentious; some have questioned the mechanism, others the electronics. We purchased the *DA30* originally because of ease of parallel remote control required for acoustic analysis processes, because it offered other features we required, and because a review showed that it worked well. So far we have not regretted it. Maybe it is better than some would think, and maybe it is not. But, what do studios use? Probably every variety of machine under the sun.

If we were seeing inconsistencies in our measurements, we would have changed our machine. But no, we can repeat the same results time after time. If our electronics are noisy and corrupting the data, so will many users' machines.

If our mechanism is less than the best, this will also be the case with users' machines.

The little testing I have been able to do using other machines indicates that the best tapes continue to perform better than the others when the machine is altered. That also gives me ►

confidence. The point is that good tapes, with good cassettes, will most likely work better in all machines. The best machine may overcome some tape deficiencies, but we would rather see these tape differences in our measurements.

At the time of publication of the DAT review, Apogee were just about to launch their first tape. Priding themselves on the superb quality of their products, they were a little upset that their tape did not perform well. A flurry of faxes, phone calls day and night, fast air parcels, and so on ensued —Bruce Jackson of Apogee assisted in getting us information which UK sources seemed unable or unwilling to provide. Both of us would now say that we know more about DAT tape than we ever wanted to. That is probably a good thing for the audio industry.

Bruce Jackson initially tested tapes on other machines, and got wildly varying error results. We responded by borrowing a Sony *7050* (from HHB hire stock). This has a front panel readout of 'block error rate' as defined by Sony—which is not the same as the error flag we used for our initial testing on the Tascam *DA30*. Using five tapes representing low, medium and higher error levels from earlier tests, we once again got low, medium and high error rates on the Sony. Tape ranking remained approximately the same (with one readout every six seconds, it was not possible to differentiate between two 'good' tapes, but 'good' tapes still had the lowest readings).

We reported this to Bruce Jackson who by then had obtained a Tascam *DA30* and started to duplicate our test setup, as did his tape supplier in Japan. Their results placed ours in question since they were getting readings 50 times lower than ours, but consistent with each other. This concerned us because we had quality assured the consistency of our own measurements between tapes, but had not compared different DAT machines.

Our next step was to get our portable Technics *SV260A* DAT recorder back and to try tapes from our *DA30* on that. An error count output cannot be obtained easily from the *SV260A*, but audible tests indicated that our 50 times higher error rate was not producing what should have been audible errors. The *SV260A* does not mute when errors are unconcealable, so it is useful for listening to the audible effects of serious error levels.

Tascam in the UK were preparing for the APRS show and could not loan us another *DA30* for comparison, but Raper & Wayman did. Now we also started getting 50 times lower error counts —would this affect the ranking of tapes? We decided not to alter the adjustment of the original *DA30* or send it for repair if necessary until all tests were finished; to do so would introduce another variable we did not want.

We immediately got down to retesting all of the tapes to find out if tape error rates had changed between tapes, altering their ranking. If so, we wanted to issue a correction to earlier information as early as possible. Our first tests were of two minutes duration and once again gave a tape ranking similar to that obtained on our first *DA30*.

The lower error count made it necessary to look at measuring a longer portion of tape, and after experiment a period of 15 minutes was selected. This meant that testing time for a set of three manufacturer's tapes extended to 11 hours—thus the day and night testing mentioned above. Can I say that we were relieved when the first three tapes tested, selected to be of low, medium and high error count once again turned out to be low, medium and high. Something was wrong with the DAT machine, but we really had managed to look at the tape.

Meanwhile, Apogee and their supplier, KAO, have been diligently working to get their tape to

Something was wrong with the DAT machine, but we really had managed to look at the tape

the standard that others may have lead you to expect. The latest sample arrived here three days ago, with another expected in a few weeks.

My further opinion of the Tascam *DA30* will be formed from testing and evaluation rather than marketing hype or brand recognition. When this series of reviews is over, both of the *DA30*s will be taken in for an evaluation of the mechanical and electronic alignment, to first determine the cause of our 50 times error count discrepancy, and secondly to experiment with error count sensitivity to various factors of transport adjustment.

Record alignment

Those of you who have grown up with pro analogue machines will know the grind of ensuring that the machine is aligned properly for the tape being used. With rotary digital machines, this has all gone out of the window—but should it? Is digital tape really so consistent that the recording level can remain constant as the formulation varies? Sony test tapes are used for playback quality assurance. Does this mean that all machines are setup to perform best with parameters Sony have set and therefore favour Sony tape?

The differences between the *DA30*s have proved one thing—variation can cause serious tape interchangeability problems. These two machines have an alignment difference; the detail of what is different has yet to be investigated. Playback occurs without audible artifacts when tapes are exchanged between the two machines but for some tapes recording a second time on a second machine produces audible errors (and the only instance of measured C2 error flags). One should hardly be surprised at this but in contrast with analogue machines, little matter of alignment is made by the manufacturers— and alignment is anything but convenient and easy. But it must be done.

Formula variations

Remembering some old principles of scientific investigation, and noting and publishing the batch numbers of the tape samples reviewed has revealed a few more things you might wish to know. One of these is that the tapes tested were up to 14 months old, though shipped to *Studio Sound* by manufacturers for an up-to-date comparative review. So much for any suspicion of manufacturers having supplied specially 'tweaked' tapes. With analogue tape, which is now unlikely to change its formulation often, this is not a problem, but with DAT tape, things are rather more dynamic.

Take Sony tape for example: through administrative error at their end, no tape was submitted for the initial review. We had some Sony tapes in stock here and put them through the tests with the other tapes. The results were not printable. Error rates were significantly worse than anything published, and tape deposits apparently quite serious; if Sony had sent us this tape we would have published the results. However we suspected that things might have changed since we knew that some of the cassettes we tested from other sources were also loaded with Sony tape. Some of these performed very well. When the Sony tape finally arrived, it proved to be good, and utterly different to my 18-month-old samples.

Phone calls to other manufacturers revealed similar stories, usually relating to tapes said to originate from Sony—the newer batch codes performed much better.

The next question to arise is whether the differences are due to tape ageing, or to formulation changes. We are convinced that both factors are involved and, in a way, compound each other. Older tape formulations were not only poorer, but will deteriorate more quickly. And yes, they are still on the shelf for you to buy today. Have you ever asked yourself why batch numbers are written in a secret code? Now you know: they enable the supplier to keep track of himself without giving the game away to you, the trusting customer.

Cassette shells

Examining the cassette shell visually does not reveal much to the untrained eye, but we have it on good authority that one of the best tapes is made of a carefully selected pair of components:

TABLE 1

Tape type	Write Once 1 Mean Error Count eight passes Standard Deviation	Write Once 2	Write-Read	Summary
Apogee new formulation	Mean count 252 Std. dev. 26	Mean count 164 Std. dev. 22	Mean count 287 Std. dev. 48	=1st
HHB	Mean count 230 Std. dev. 47	Mean count 225 Std. dev. 13	Mean count 300 Std. dev. 32	=1st
Ampex	Mean count 256 Std. dev. 28	Mean count 210 Std. dev. 32	Mean count 2934 Std. dev. 1676	=2nd
Fuji	Mean count 612 Std. dev. 43	Mean count 168 Std. dev. 14	Mean count 243 Std. dev. 40	=2nd
Maxell Consumer	Mean count 199 Std. dev. 18	Mean count 155 Std. dev. 21	Mean count 659 Std. dev. 53	=2nd
Sony PDP Series	Mean count 276 Std. dev. 42	Mean count 1111 Std. dev. 486	Mean count 376 Std. dev.85	=2nd
3M	Mean count 335 Std. dev. 102	Mean count 435 Std. dev. 33	Mean count 671 Std. dev. 315	=2nd
Apogee previous	Mean count 633 Std. dev. 202	Mean count 1064 Std. dev. 43	Mean count 729 Std. dev. 106	=3rd
TDK	Mean count 622 Std. dev. 39	Mean count 1715 Std. dev. 212	Mean count 921 Std. dev. 183	=3rd

tape from one manufacturer and a cassette shell from a competitor.

Careful examination of our earlier results, as mentioned by John Watkinson, can reveal visibly obvious error rate variations which are caused by tape rotation. An example is the 14-second cycle visible in the Apogee error curves.

What will happen to the cassette over time? One would expect that liners will wear and collect debris which will roughen the running of the tape, introducing contact pressure variation at the head along with timing errors and possible longitudinal vibration of the tape. All will affect errors. And, as John Watkinson says, ageing is likely to affect the shell also. But, at least it is possible to replace a shell should it be necessary to save an historic recording.

Replay noise levels

The main difference in the present Apogee tape as compared to the previous versions is output level. Raise the output level a few dB and the error rate drops considerably. On average, its error level has dropped threefold when measured on our test system. At first glance the finger is pointed at the DAT machine with accusations of poor replay noise levels corrupting the digital data. But, looking at information from tape manufacturers relating to their testing of tape life, one can see that when measured tape level drops only 2dB, errors increase tenfold.

Questions must be raised about noise issues, possibly about our *DA30*, and possibly about the DAT format as a whole.

Erasure

One further thing which came to light during the retest concerns erasure of the tapes. When a tape recorded on the original *DA30* was replayed on the new machine, there was no problem. However, when some tapes were rerecorded on the new machine the error rates shot up. These were very audible when replayed on the *SV260* machine. The tapes which introduced this apparent new problem were those which gave the very best results on the original machine. Until mechanical alignments are checked, it is not possible to be clear on what is happening, but it does appear that the poorer tapes can be rerecorded more reliably than the better tapes. This may be due to a higher remanence remaining from previous recordings on these better tapes, or it may be some other factor, but it is interesting and occurred with both Maxell and HHB tapes.

Tape quality assurance

Several things seem to have occurred in the industry, at least in part as a result of these reviews. Most are to do with quality assurance. Firstly, several tape vendors are working on tightening up the controls on their supplier manufacturers—this will benefit all parties in the end. Secondly, for some tapes the development cycle has been sped up to bring a tape up to the mark—the Apogee tape is an example, now performing with the best. Thirdly, some technical staff inside the manufacturers are beginning to demand standard ways of measuring tape quality which could be published in specifications. The Sony 'block error rate over six seconds' method seems mainly to benefit the machine owner in helping to determine head cleaning and replacement needs. It provides little information to quality assure tape. Finally, there is an increasing recognition that machines must be carefully checked following purchase, and routinely aligned—though not as often as analogue machines.

Further testing

The results of the repeated tests are given in **Table 1**, ordered (rather than ranked) according to average error count over eight runs of the tape. Note that though error count is lower, the order has remained almost the same. Note also where the newer entries lie compared to earlier ones. All of the results below are from new tapes obtained from manufacturers following the last review. Earlier tapes were checked to ensure that ranking had not been compromised in the previous review, but are not now applicable, and are omitted.

In summary, HHB tape and Apogee's new formulation (which we are told is the one which is on sale) show the most consistency over the three samples. Fuji, Sony PDP, Maxell consumer and 3M tapes are up there with the best, though not as consistent. One tape out of the three had lower performance. Two of the Ampex tapes were also good, with the third very much worse. TDK remain clearly following behind.

We did not receive any newer BASF tape.

Most of these tapes have now completed the ageing process in our environmental chamber. Next month will see a report on tape ageing, which should be of even more interest than initial tape performance. ■

INTERNATIONAL STUDIO DIRECTORY

LEGEND TO SYMBOLS:

DIGITAL CONSOLE:

DIGITAL EDITING:

DIGITAL - TRACKS: D

ANALOGUE - TRACKS: A

AUDIO FOR VIDEO:

FULL-TIME MAINTENANCE:

MOBILE RECORDING:

RESIDENTIAL STUDIO:

CANADA

LITTLE MOUNTAIN SOUND STUDIOS

201 West 7th Ave. Vancouver BC, V5Y-7L9, Canada. (604) 873 4711; Fax: (604) 873 4718. Studio/Bookings Manager: Tilde Cameron. **No. of studios & dimensions:** Studio "A" - 33ft. x 50ft. 22ft.H (Plus loading bay). Studio "B" - 31ft. x 33ft. x 22ft.H. **Mixing consoles:** SSL G-Series 64?? E-Series 48???(studio). **Recorders:** 3 x Studer A-800III/Sony 3402 Dash Digital. **Monitors:** Urei 813C/NS-10M/Auratones. **Specified outboard:** AMS RMX/DMS/Yamaha Rev 1/Neve EQ, 1084/ SPX-90II/TC2290/PCM 90/Neumann Tuve 47/Sony C-37.
Special Services: Mixing for three channel film L-C-R in studios "A" & "B"/Full album tracking & Mixing facility.

D A X48

STUDIO MORIN HEIGHTS

201 Perry, Morin Heights, Quebec, J0R 1HO, Canada. +1 514 226 2419; Fax: +1 514 226 5409. Owner: L'Equipe Spectra. **Studio/Bookings Manager:** Judy Smith/Peter Holmes. **No. of studios & dimensions:** Studio 44ft. x 30ft., Control Room 18ft. x 15ft. **Mixing consoles:** SSL 4056 G Series: + Total Recall, Events Controller plus 8 extra stereo patchable VCA's assignable to stereo bus. 1 Neve 12-4 with 1073EQ. **Recorders:** 1 Studer A800/II, 1 Otari MTR90/II, 1 Studer A80 1/2 inch. 2 Studer B67, 1/4 inch. 2 Studer A710 cassette. 2 Panasonic SV3500/3700 DAT. 1 Revox B225 CD player. 2 Timeline Lynx Synchronizer. **Monitors:** 1 pr Acoustic Research AR18S, 1pr Auratone 5PSC main monitors, 1 pr Quested 412/II, 1 pr Yamaha NS10M, 1 MacIntosh Mc2300 (near fields), 1 Quested DX3000E (main monitors), 2 Quested A900E (main monitors), 3 Studer A68 (headphones), 2 BSS FDS360 (Modified) Crossover (main monitors). Instruments: 1 Hammond B3, 2 Leslie 122, 1 Yamaha 9'Concert Grand Piano.
Specified outboard: 8 Focusrite ISA110, 12 Assorted Equalisers, 1 AMS RMX16, 3 EMT 140 Plates, 1 Eventide H3500BV, 1 Lexicon 224XL, 1 Lexicon 480L, 7 Assorted Digital Reverb/FX, 7 Assorted DDL, 1 Gates M3529B Tube Limiter, 2 Neve 2254E, 3 RCA BA6A tube limiter, 2 Urei LA3A, 2 Urei 1176N, 8 Assorted Compressor/Limiter.
Special Services: The studio, our six bedroom guest house and cottage are situated in the heart of the Laurentian resort area, with panoramic views of our private lake and forest. However, we are only 10 minutes from St-Sauveur, the major centre of the region, and less than an hour from Montreal. We have a cappuccino bar, satellite dish, games room, band office, boats on the lake and can arrange participation in a multitude of seasonal sports including skiing, golf and horseback riding. A wide range of in-house catering can be arranged of alternatively there are over eighty restaurants within a few minutes drive. Our 'tech shop' is well stocked and equipped, including an Audio Precision System One. Apart from our spacious studio area, we can offer as an option the use of our large 'live room' for recording or as a real reverb chamber.
Association Member: SPARS.

A X48

VANCOUVER STUDIOS

3955 Gravely St., Burnaby, B.C. Canada. +1 (604) 291 0978; Fax: +1 (604) 291 6909. Studio/Bookings Manager: Tilde Cameron. **No. of studios & dimensions:** Studio 1: 34ft. x 27ft. studio - 23ft. x 26ft. control room. Studio 2: 24ft. x 17ft. studio - 19ft. 10inches x 17ft. 6inches control room. Studio 3: 23ft. 8inches x 11ft. 10inches - 20ft. 8inches x 16ft. 7inches control room. Studio 4: (attached to studio2) 39ft. 6inches x 38ft. 4inches. **Mixing consoles:** Studio 1 - Neve 8058 MK24 input with 4 inboard compressors. Studio 2- SSL 4056 G-Series with total recall & plasma meters. Studio 3 - MCI JH-500 32 Input in-line console. **Recorders:** Otari MTR-100 Multitracks (3). Otari MX-80 Multitracks (1). Otari MTR-12 1/2inches & 1/4inches CTTC 2-TRK. Panasonic SV-3700/3500 DAT recorders. **Digital audio workstations:** SSL Screensound Digital Edit Workstations (2). **Monitors:** Urei 813C/NS-10M/Tannoy FSM 215/Auratone Soundcubes.
Specified outboard: Lexicon 480L/Eventide H-3000/Drawmer DS-210 TC Electronics 2290/Lexicon PCM-70/Lexicon PCM-42 Yamaha SPX -90/AMS RMX-16/KORG DRV-3000, Aphex Expander - Gate/DBX-160X Compressors/ Valley Dynamite Expander/Gate/TC Electronics 2240 Preamp & EQ (Tube Tech Pe-IB EQ) John Hardy M-I Mic Pre-Amps(4). **A/V equipment:** SSL Screensound digital workstations, JBL Projector/100" Screen, Timeline Lynx syncronizers with KCU controller, Sony BVU-900 3/4inches Video recorders(2), Sound ideas sound effects library, many other CD sound effects libraries in edit suites.
Special Services: Full Album Tracking/Mixing facility and feature film mixing and editing.

A X48

DENMARK

AIR PLAY RECORDING STUDIO

Montanagade 29D-E, 8000 Aarhus C, Denmark. +45 861 91212/+45 861 91272. Owner: Lars Alsing. **Studio/Bookings Manager:** Lars Alsing. **No. of studios & dimensions:** 1 studio, 300m² with daylight, 50m² control-room, 100m² recording room, both with air condition, Mahogany floors. **Mixing consoles:** AMEK Angela 28/56 inline with automation. **Recorders:** Otari MTR 90 mrk. 2 -24 track. **Digital audio workstations:** Fairlight series III. **MIDI set-up:** Yamaha TG77, Ensoniq KMX 16, Roland A880, Atari Mega ST4 & Cubase, Fairlight series III. **Monitors:** JBL 4412, Dynaudio Acoustics M1, Yamaha NS 10M, Auratone.
Specified outboard: Tube Tech, Urei, Drawmer, DBX, Aphex, TC2290, Lexicon 480L, PCM 70, Klark DN780, Sony MUR201, Yamaha SPX90, Alesis Quadraverb, Eventide H3000S, SPL Vitalizer, Microphones: Neumann, AKG, Bruel & Kjaer.
Special Services: Inhouse engineers & producer, catering, leisure facilities, satellite TV, video, kitchen, acoustical design by soundtechical institute or Copenhagen.

A X24

HOLLAND

STUDIO 150

Lauriergracht 150, 1016 RV Amsterdam. +31 0 20 625 95 85; Fax: +31 0 20 620 49 82. Studio/Bookings Manager: Peter Richeck. **No. of studios and dimensions:** 1, recording area 80m² control room 40m². **Mixing consoles:** SSL 400G. Multitrack: Otari MTR-100 with Dolby A & SR. **Recorders:** 2 track analog recorders; Studer A810, Studer B67. 2 Track digital; Panasonic 3700

DAT, Fostex D20 DAT with timecode. **Digital audio workstations:** DAR Sigma with DSP, 16 channels playback and 16 trackhours recording capacity. **MIDI set-up:** Atari Mega ST with Cubase and 44MB Syquest. SMP24 synchronizer, various synths. Instruments: Yamaha C3D Grand Piano, Vox AC30. **Monitors:** Genelec 1034A, Nearfields; Genelec 1031A, Yamaha NS10, Tannoy Eclipse.
Specified outboard: Lexicon 224XL, Lexicon 480L, Lexicon PCM70, Yamaha SPX900, TC Electronics TC2290, Lexicon Prime Time II, Roland SDE3000, A/DA STD1, AMS 2-20, AMS DMX 15-80 S, Urei 1178, Aphex Compellor, DBX 165, DBX 900 series compresser, gates and De-essers, Audio & Design Compex F760, Kepex II gates, Valley People Dynamite, Focusrite RED 1 pre-amp, Focusrite RED 2 Eq, Focusrite ISA 115 HD, Tube Tech MP1A tube pre-amp, Tube tech PE1C tube-Eq, Aphex Aural Exciter, BBE 882 maximizer, Behringer Denoisers, Klark Teknik DN 360 1/3 octave graphic Eq. Microphones tube: 4 Neumann U67, 1 Neumann U47, 1 Neumann SM23, stereo, 8 Neumann KM64, 2AKG C28, 1Telefunken CMV3. Microphones other: Neumann; 2 x TLM170, 2 x TLM50, 2 x U87Ai, 1 x U89, 4 x KM140, Neumann KMS. AKG; 1 x D12, 2 x 460 Sennheiser; 4 x 441, 2 x 421 Shure; 6 x SM57, 1 x Beta58, 1 x 55SH, Electro Voice; 1 x P120. **A/V equipment:** Sony 9600 U-Matic, Sony Jumbo Monitor, Adams-Smith Zeta-3 Synchronizers with remote, DAR Sigma Audiocomputer, House sync.
Special Services: Studio 150 is in the centre of Amsterdam, on a canal.
Association Member: AES.

INDONESIA

STUDIO 15 (SAGITARIUS RECORDING STUDIO)

Jalan Petojo Selatan VII No. 19A Jakarta 11001-9-Indonesia. +62 (21) 3865775-3857911-3450142; Fax: +62 (21) 3450141. Owner: Leonard Handhi Kristianto and Edward Indra Kristianto. **Studio/Bookings Manager:** Moh. Stev. Heru Purnomo & Tono Libel. **No. of studios & dimensions:** studio 1 control room, 16 m² (day light) recording room 20 m², Studio 2 control room 16m², recording room 16m². **Mixing consoles:** Soundcraft 6000-44/24 & Soundcraft 2400 LED 32/24. **Recorders:** Studer 827 & Studer A80 Mk.IV. **MIDI set-up:** Akai S-1000, Roland S-770, McIntosh computer. **Monitors:** Auratone QC-66, Yamaha NS-10M Studio Monitor, Urei 813.
Specified outboard: Lexicon 240 XL, Lexicon PCM 70, Klark Teknik Equaliser, Urei, Compressor/Limiter, Yamaha SPX 1000, Yamaha SPX 900-II, Eventide Harmoniser DHM 98, EMT 240.
Special Services: Inhouse engineers and producers, Analogue-Digital transfer. Studio design by: Jeka Records, Jakarta-Indonesia.
Association Member: ASIRI (Association Recording Industries of Indonesia).

ITALY

CAPRI DIGITAL STUDIOS

Via Tudro 11, Capri (NA), Italy. +39 837 5157/5158; Fax: +39 837 5141. Owner & Studio/Bookings Manager: Carloquinto Talamona.
No. of studios & dimensions: Control room 80sq. metres. Recording areas 150 sq. metres. **Mixing consoles:** SSL 4072 G Series with Ultimation. **Recorders:** Sony PCM 3348, 2 x Mitsubishi x 880, 2 x Studer A820. **Digital audio workstation:** MacIntosh and Sound Tools. **MIDI set-up** Opcode Studio Vision, Notator, Cubase, Function Junction Plus, 16 x16 MIDI, Akai 1100, Proteus 1 x R, Yamaha TC 77, Roland R8M, Akai ME 35T, Real World M48. **Monitors:** Kinoshita with TAD components, Yamaha NS to M, JBL Tannoy.
Specified outboard: Lexicon 480, AMS/Eventide, Roland, Yamaha TC Electronic, Drawmer, Focusrite, Neve, Valley, Orban, DBX, Urei, Aphex, Summit, Teletronix, Tubetech, Art. **A/V equipment:** Sony Projector, with 100 inch screen SVHS Multi Standard, Full Video Sync.
Special Services: Residential studios, Dash PD Dash Digital domain transfer available.
Association Member: World Studio Group.

FONOPRINT RECORDING STUDIOS

V. Bocca di Lupo 6, Bologna, Italy. +39 51 5852 54; Fax: +39 51 3340 22. Studio/Bookings Manager: Nicolini Luciano. **No. of studios & dimensions:** Studio 1 Control Room 60 m² Recording room 85 m². Studio A Control Room 26m² Recording room 56m². Studio B Control Room 27m², Recording room 30m². **Mixing consoles:** SSL 4064 G Series with total recall, MC1 500 28CH with Automation, DR 8000. **Recorders:** 2 x Sony PCM 3324, 1 x Otari DTR 900 II, 2 x Otari MTR100. **Digital audio workstations:** Sound Tools with Macintosh II Ci. **Monitors:** Quested Q412B, Urei 813B, Yamaha NS10M.
Specified outboard: Lexicon 480, 224, PCM70, AMS RMX16, 1580S Yamaha Rev5 Rev7 Eventide H949, H3000S, K.T. DN780, Prisma Neve.
Special Services: Studio Design By: ADG Acoustic Design Group. Kitchen, Catering, Satellite TV, Billiard Table.

MULINETTI RECORDING STUDIO

Via Bordigotto, 5 - 16036 Recco (GE), Italy. +39 (185) 75017; Fax: +39 (185) 722525. Owner: Alberto Parodi. **Studio/Bookings Manager:** Alberto Parodi.
No. of studios & dimensions: 1 x control room, 1 x live room and all the villa. **Mixing consoles:** Neve VR 60 with flying faders and recall. **Recorders**: Mitsubishi X850, Studer A827 with SR. **Digital audio workstations:** Digidesign PROMASTER 20 up two hours. **Monitors:** Quested 212b, Dynaudio M1, Genelec S30, Yamaha NS10.
Specified outboard: Lexicon, AMS, Eventide, Roland, Quantec, Klark Tecnik, TC Electronic, Bel, Yamaha, DBX, Urei, Summit Audio, Tube Tech, Aphex, Drawmer and more...
Special Services: the studio is a residential facility located in a beautiful old villa by the sea with private beach, 5 bedrooms with telephone, TV, air conditioned and sea view, satellite, TV, gymnasium, good Italian food and hospitality.

JAPAN

SOUND DESIGN

2-32-2 Sendogaya, Shibuya-Ku, Tokyo, Japan 151. +81 (3) 3423 0481; Fax: +81 (3) 3423 0480. Owner: Sound Design, Inc. **Studio/Bookings Manager:** Keizo Suzuki. **No. of studios & dimensions:** (1) W. 8m x H.3m x D7m. **Mixing consoles:** Focusrite Studio Console + GML MFA. **Recorders:** Sony PCM-3348 x 2, PCM - 7030 x 2, Studer A800 MKII-24, Sony PCM -3402 x 2, Panasonic SV-3700 x2, Studer A 80 1/2 inch, 1/4 inch. **MIDI set-up:** SBX-80. **Monitors:** Ray Audio Kinoshita Original TAD. **Specified outboard:** Neve 1081, Tube Tec CL-1A, ME-1A, GML 8300, ELA-M251, V-47,67, C-12, 12A, John Hardy M-1, PulTec E2P-1A3. **A/V equipment:** Sony 8/4 Umatic, BVU-950, Victor VHS BR-7700.
Association Member: Audio Rents, Co. Ltd. (Tel: 813 3402 2291). Sound Design Inc. (Tel: 813 3423 0481).

SOUND INN STUDIOS

Yonban-cho Annex, 5-6 Yonban-cho, Chiyoda-ku, Tokyo 102, Japan. +81 3 3234 4311; Fax: +81 3 3234 4385. Owner: Tsuneo Hayakawa. **Studio/bookings Manager:** Yoichi Aikawa.
No. of studios & dimensions: Four studios - Studio A: Cont. 92m²/St. 246m², Studio B: Cont. 90m²/St. 101m², Studio C: Cont. 90m²/St. 35m², Studio F: Cont. 85m²/St. 25m². **Mixing consoles:** SSL 4056G, SSL 4072G, Over quality OQM8180 + 8S(GML), OQM8196(GML).

Recorders: Sony PCM 3348, PCM 3324, Studer A800. **Monitors:** Quested Q412, Yamaha NS-10M, Autotone 5C, Genelec S30NF.
Specified outboard: Focusrite 115HD, AMS rmx16, dmx-15-80S, Lexicon 224XL, 480L, Quantec QKS Neve 33609, Urei 1176LN, LA-4, LA-2, Eventide H-3500, GML 8900, TC 1210, 2290, Roland, Yamaha. **A/V equipment:** sony BVU-800DB; LINX/SAL.
Association Member: JAPRS, WSG.

D A

MALAYSIA

ADDAUDIO POST PRODUCTIONS SDM BHD

11, Jalan 17/56, 46400 Pethling Jaya, Malaysia. +60 (3) 7560600; Fax: +60 (3) 75 60470. Owner: The Music Machine Holdings. **Studio/Bookings Manager:** Daniel Tang. **No. of studios & dimensions:** Studio 1, Studio 2, Studio 3. **Mixing consoles:** DDA DMR12, DDA D Series. **Recorders:** Studer. **Digital audio workstations:** 2 DAR Soundstation II. **MIDI set-up:** Atari, Emulator III, Akai S1100, Microwave, Proteus. **Monitors:** Quested H210. **Specified outboard:** SSL, Klark Teknik Equalisers, Summit compressors, EMT & Urei Compressors, Lexicon, Yamaha.
Association Member: SPARS

SWITZERLAND

CONCEPT STUDIO

Leman 7, 1020 Renens, Switzerland. +41 21 634 5000; Fax: (same).
Owner: Patrice Collet. **Studio/Bookings Manager:** Patrice Collet. **No. of studios & dimensions:** 1 studio - 60m². **Mixing consoles:** Soundcraft 6000 - 36/24. **Recorders:** MCI JH-24 Analog 2 inches/Sony DT55, Denon DTR 80-DAT. **Digital audio workstations:** Akai DD-1000. **MIDI set-up:** Notator Logic on MacIntosh/C-LAB creator on Atari/Akai S-1000/EMU-systems EMAX, Vintage key, proformance, Roland D-550/Yamaha DX-7...**Monitors:** Westlake BBSM-6/Yamaha NS-10/AKG LSM-50. **Specified outboard:** Lexicon 200/PCM 70/PCM 60/PCM 42/ LXP-AS/ Valley Gatex/ Dynamite /DBX 902/ 903/Aphex Type-C/Dominator II/Neumann U-87...**Special Services:** Recording/ Mixing /Pre-mastering/

A

GAMMA RECORDING

Bahnhofstrasse 50, 4663 Aarburg, Switzerland. + 41 62 414 470; Fax: +41 62 414 471. Owner: Lachmann/Schwitter & Co. **Studio/Bookings Manager:** Martin Lachmann. **No. of studios & dimensions:** 1 x control room: 25m² (daylight) with separate machine room, 1 x recording room: 250m², 1 x booth: 16m² (daylight). **Mixing consoles:** Harrison MR-3 (aux-update, 10 discrete sends), Optifile 3D computer, separate effects-returnmixer. **Recorders:** Studer A820/24 with Dolby SR. **Digital audio workstations:** Sonic Solutions (available from autumn '93). **Midi set-up:** Proteus "World", TG77, D50, Matrix 1000 etc. Atari Mega 4 computer and SRC AT synchronizer. **Monitors:** Andy Munro M3, Yamaha NS-10M, Auratones. **Specified outboard:** Lex 300, Lex 224 (2), Lex PCM70 (2), EMT 240, Eventide H3000SE, TC 2290, TC 1210, Dynacord DRP20X, Yamaha SPX 1000 plus many more.... Focusrite ISA 115, Tube Tech PE1B, CL1A, Summit TPA200A, UREI LA4 (6), 1178, Publison Relief Enlarger plus many more... Fine collection of Mic's including Calrec Soundfield, B&K's, Sennheiser MKH's etc. **Special Services:** Very large recording room complete with Bosendorfer grand piano, experienced house engineers, producers-office, fully equipped kitchen/lounge and a hotel just next door to the studio. **Association Member:** AES, Re-Pro Associate.

A X24

UK

ABBEY ROAD STUDIOS

3 Abbey Road, St. Johns Wood, London NW8 9AY, UK. +44 (71) 286 1161; Fax: (071) 289 7527. Owner: Chairman- Ken Townsend MBE. **Studio/Bookings Manager:** Colette Barber. **No. of studios & dimensions:** Studio 1: 94ft. x 55ft. x 42ft.H, Studio 2: 58ft. x 37ft. x 28ft.H, Studio 3: 28ft. x 23ft. x 24ft.H. Penthouse: 20ft. x 20ft. x 9ft.H. **Mixing consoles:** Studio 1 - Neve VRP Legend 64ch. Penthouse - Neve Capricorn Dig. Desk, Studio 2 - SSL 4000E G Series computer. Studio 3 - SSL G Series W' Ultimation. **Recorders:** Studer A820, Sony 3324A, Mitsubishi 32T, Sony 3348 1630 and Mitsubishi 20 bit. **Digital audio workstations:** Sonic Solutions. **MIDI set-up:** Atari Mega 4 with cubase & Notator software. Akai S1100 & S1000. Keyboards on request. **Monitors:** Studio B&W 801 series 3, studio 2 &3 - Quested, Penthouse - JBL's.**Special Services:** Sonic Solutions no-noise and hard disc editing systems. CD prep/PQ Encoding, Digital mastering, Lacquer cutting, DMM, Digital remastering, cassette duplication, digital editing, digital copying, Cedar, 3X location recording units, accommodation, bar and restaurant.**Association Member:** APRS.

D X48 A X48

HELICON MOUNTAIN STUDIOS

The Station, London, SE3 7LP, UK. +44 81 858 0984; Fax: +44 81 293 4555. Owner: Jools Holland. **Studio/Bookings Manager:** Richard Holland. **No. of studios & dimensions:** 1 studio. **Mixing consoles:** Soundtracs "Quartz" 48 Channel In Line Desk. **Recorders:** Saturn-824-24 track (Dolby SR) by arrangement. **Digital audio workstations:** Revox PR99, Casio DA1 DAT/DA2 DAT. **MIDI set-up:** Atari 1040 STE - Seinberg: 24/C Lab 3.00 XR300 SMPTE Synchroniser. **Monitors:** JBL 4430 Studio Monitors, Yamaha NS 10M's. **Specified outboard:** Lexicon 480L, PCM 70, Rev 7, RV 1000, SPX90 etc. **A/V equipment:** Yamaha C5 Grand Piano - Lesley 145.

A

MILO MUSIC

43/44 Hoxton Square, London N1 6PB, UK. 071 729 4100; Fax: 071 729 7400. Owner: Henry Ckallan. **Studio/Bookings Manager:** Nick Young. **No. of studios & dimensions:** Studio 1: Control room 31m², recording room 33m²; Studio 2: Control room 20m², Recording Room 48m², Studio 3: Control room 30m², recording room 35m². **Mixing consoles:** Studio1: Amek G2520 + Automation; Studio 2: Soundcraft 400; Studio 3: Amek G2500. **Recorders:** Analogue - Studer A80 MKIV, Fostex E16. Digital: 2 x 12 track Akai Adam. **MIDI set-up:** Akai S1100, Akai S1000, Atari 11040, Atari Mega 4, Korg SG10, Korg MIR, Kurzweil 100PX, Roland D1101 SH101, Korg M3 R, Roland MKS 50, Yamaha TX802, Roland Jx8P/PG800, Roland D50, Casio C25000, Casio C2101. **Monitors:** Sean Davies LS841, Eastlakes, Yamaha NS10S, AR1815, Auratones, Tannoy Golds. **Specified outboard:** AMS RMX15, Lexicon PCM 70, Yamaha SPX90, Roland DEP5, Alesis Quadraverb plus, Urei 1176, DBX 165, Drawmer DL221/LX20, Eventide 910 Harmonizer/H949, AMS 15805, Roland SDE 2000, Aphex exciter B, Roland dimension D, Delta Lab DL-2CR. **A/V equipment:** Adams-Smith Zeta three, Sony U-Matic, XR1300, Synchroniser, Unitor Synchroniser.
Special Services: Yamaha and Steck BabyGrand Pianos, Rhodes 88 Piano, Hammond 101. Full range valve mics.**Association Member:** APRS.

A

SAWMILLS STUDIO

Golant, Fowey, Cornwall, PL23 1LP, UK. +44 (0)726 833752; Fax: +44 (0) 726 832015. Owner: Dennis Smith & Simon Fraser. **Studio/Bookings Manager:** Ruth Taylor. **No. of studios & dimensions:** One Studio 250sq ft. Control room 700 sq ft. main room. **Mixing consoles:** Trident 80B 54-24-24 & Automation. **Recorders:** Otari 24th &

Ampex ATR 102 1/2inch. **Digital audio workstations:** Sound Tools. **MIDI set-up:** Extensive. **Monitors:** Quested. **Specified outboard:** AMS (15.80) & Lexicon 480L + Neve, Urei + BSS + EMT.**Special Services:** Fully residential including good food.**Association Member:** APRS (Full member).

A D

SHAMBLES STUDIO

The Shambles Westhorpe Road, Marlow, Buckinghamshire, SL7 1LD, UK. +44 628 891003; Fax: +44 628 485363. Owner: Chris Rae. **Studio/Bookings Manager:** Adam Vanryne. **No. of studios & dimensions:** 1 studio, 18 ft. x 12, Phil Newell Disign Granite Room. **Mixing consoles:** SoundCraft 3200. **Recorders:** SoundCraft. **MIDI set-up:** Atari & Cubase. **Monitors:** Reflextion Arts. **Specified outboard:** Lexicon, Yamaha, Roland, Drawmer Gates & Compressers. **A/V equipment:** UMatic/Betacam (1inch Sony).
Special Services: Music composition. Production.

D A X24/48

STRONGROOM

120, Curtain Road, London, EC2A 3PJ, UK. +44 71 729 6165; Fax: +44 71 729 6218. Owner: Richard Boote. **Studio/Booking Manager:** Siobhan Paine. **No. of studios & dimensions:** 2 x 24 track analogue studios, 8 x Pre Production/programming rooms. **Mixing consoles:** Studio 1: 1 x Neve V3 with Flying Faders, Studio 2: 1 x SSL G+ with utimation. **Recorders:** 3 x Otari MTR 90 MKII. **MIDI set-up:** Studio 1: Akai S3200 16 Meg. JD800. MIDI routing matrix. Studio 2: Akai S3200 16 meg. extensive range of keyboards, vintage keys. MIDI routing matrix. **Monitors:** Neil Grant Boxer Five System in Studio 1 and 2. Yamaha NS10's. **Specified outboard:** Studio 1: 480L Eventide H300SE, Massenberg, EQ, Neve compression. Studio 2: 224 x L. Eventide DSP4000. Massenberg EQ, Reverbs etc. **A/V equipment:** Studio 2.
Special Services: Studio 1: Large live room. Naturally lit Grand piano.
Association Member: APRS and Accord.

A

USA

ARDENT STUDIOS

2000 Madison Avenue, Memphis, TN 38104, USA. +1 901 725 0855; Fax: +1 901 725 7011. Owner: John E. Fry. **Studio/Bookings Manager:** Susan Allred. **No. of studios & dimensions:** Studio A - control - 20 x 24, studio - 25 x 40. Studio B - control - 25 x 20, Studio - 24 x 17. Studio C - control - 18 x 25, studio - 25 x 35. **Mixing consoles:** Neve VR 48 x 48 with Flying Fader Automation, Neve V Series 40 x 48 with Necam, Solid State Logic 6056E with G Computer 56 x 32. **Recorders:** Mitsubishi X-850 32 track (3), Mitsubishi X-80 2 track (3), Mitsubishi X-800 32 track (1). Studer A827 24 track (1), MC1 JH24 24 track (1). **Digital audio workstations:** Studer Dyaxis. **MIDI set-up:** Fairlight Series III. **Monitors:** Audio Consultants/JBL, JBL 4435, Yamaha NS-10 KEF. **Specified outboard:** Lexicon, Quantec, SSL, Publison, Summit, Fairchild, UA, Pultec, Drawmer, Yamaha, Eventide, Aphex, dBx, Urei, Marshall, Valley People, DOD, Deltalab. **A/V equipment & Special Services:** *Consoles:* Studio A - Neve VR with Flying Fader Automation 48 x 48. Studio B - Solid State Logic 6056E with G Computer 56 x 32. Studio C - Neve V Series with Necam 96 Automation 40 x 48. *Tape Machines:* (2) Mitsubishi X-850 digital 32-track, (1) Mitsubishi X-800 digital 32-track, (3) Mitsubishi X-80 digital 2-track, (1) Studer 827 analog, MCI JH24 (16 track available for both machines), Dyaxis digital disc editing system. *Monitor Amplifiers:* Studios A & B - Main Speakers: Audio Consultants/JBL Custom, Studio C - JBL 4435, Minis - Yamaha NS-10, KEF. *Outboard Gear:* Lexicon: (1) 480L, (1) 224XL, (1) LXP 15, (1) Prime Time 93. Quantec: (2) Room Simulator. Solid State Logic: (1) Infernal Machine 90. AMS: (1) RMX 16. Summit: (1) DCL 200. Fairchild: (1) 670 Stereo Tube Limiting Amp, (2) 660 Mono Tube Limiting Amp. UA: (2) 176 Mono Tube Limiting Amp. Pultec: (1) EQP1A Equalizer. Drawmer: (1) DL 251 Spectral Compressor. Yamaha: (6) Rev 5, (1) Rev 7, (2) SPX 90 I, (2) SPX 90 II, (2) SPX 1000. Eventide: (2) H949, (1) H3000ES/95 sec. mono sampling Mod Factory, (1) H3500/23.7 sec. mono sampling. Aphex: (2) 612 Gate. dBx: (7) 165 Compressor/Limiter, (2) 160 Compressor/Limiter, (4) 263 De-esser. Urei: (1) 1176. Marshall: (2) Time Modulator 5002. Valley People: (2) 440 Compressor. DOD: (2) EXR SP II Projector R860. Delta Lab: (1) Effectron II. *Microphones:* AKG: (4) C451L, (2) C460B, (1) C422, (3) D112, (3) C452L, (4) C414, (1) D12. Sennheiser: (1) MD441, (15) MD421. Shure: (1) SM53, (1) SM58, (1) SM56, (13) SM57. Electro Voice: (3) RE20, (1) PL20. Neumann: (1) U47 FET, (1) KMI, (4) U87, (3) M249 tube, (2) KM84, (3) U64 tube, (2) KM86i, (1) U67 tube. Beyer Dynamics: (2) M201. Crown: (6) PZM 3OR13. Altec: (1) 150A.

D A

CARIBBEAN SOUND BASIN

135A Long Circular Road, Maraval, Trinidad, W.I. +1 809 628 6176/77/78 (Tel. + Fax). Owner: Amar Holdings Limited. **Studio/Bookings Manager:** Sarah Mohammed. **No. of studios & dimensions:** Studio A 60 ft. x 70 ft. x 18 ft. Control room: 32 ft. x 30 ft. Studio B: 35 ft. x 28 ft. x 10 ft. Control room: 20 ft. x 8 ft. x 10 ft. **Mixing consoles:** SSL 4064G, Neve VR48. **Recorders:** Mitsubishi X880 Digital, Otari MTR 9011 Analogue. **MIDI set-up:** MacIntosh & Atari computers with performer, Cubase & Notator software. Complete selection of keyboards. **Monitors:** Westlake TSM 3 and BBSM 12. **Specified outboard:** Lexicon 480L, Quantec QRSL, TC 2290 Eventide M3500, Pyltec EQ, Massenberg EQ, AMS 15805, AMS RMX 16. **Specified services:** Elegant bedroom suites with a swimming pool, gym and sauna for use by guests.
Association Member: World Studio Group

D A

CRYSTAL CLEAR SOUND

4902 Don Drive, Dallas TX 75247, USA. 214 630 2957; Fax: 214 630 5936. Owner: Sam Pavlos. **Studio/Bookings Manager:** Keith Rurt. **No. of studios & dimensions:** Multitrack room: 1400 square feet. Digital editing room: 10 ft. x 12 ft. **Mixing consoles:** DDA 224V with uptown moving fader and comprehensive switch automation. **Recorders:** Studer 827 24-track with Dolby SR, MC1, Studer & Tascam 1/4 inch, 1/2inch & DAT machine. **Digital audio workstations:** 4-track, 1.3 gig, Pro Tools running on a Mac Quadra 700 with 20 megram. **MIDI set-up:** Akai S-1000 with meg ram, Ensoniq VFX-SD, Roland R-8m, Akai MG-35, Adam-Smith & Mark of the Unicorn, MIDI/SMPTE Syncronisers. **Specified outboard:** T.C. Electronic M5000, Ensoniq DP/4, AKG 68k, Eventide H3000, Yamaha SPX-900 and SPX-90 multi-effect processors. Lexicon model 200 digital reverb processor. Korg SDD-3000 and Lexicon PCM-41 digital delay units. Tube Tech Pe1B, API 5502 and Orban 622b parametric EQs. Summit Audio LA100 tube, JBL 7110, DBX 160 & 165 compressors, Aphex 612 gate/expanders and Compellor. Valley DSP de-esser. Baldwin SD-10 concert grand piano, Hammond B-3 & Leslie cabinet. **Monitors:** KRK, Tannoy, Urei, Yamaha, Auratone.
Special Services: Digital editing/marketing. CD-R recorder. KABA Realtime cassette duplication with imprinting and packaging. CD manufacturing. Regional music distribution.
Association Member: Texas Music Association, NAIRD.

HOWARD SCHWARTZ RECORDING INC.

420 Lexington Ave., Suite 1934, New York NY 10170, USA. (212) 687 4180; Fax: (212) 697 0536. Owner: Howard Schwartz. **Studio/Bookings Manager:** Beth Levy-Davs. **No. of studios & dimensions:** Seven Audio form video Post Production suites and recording studios. **Mixing consoles:** SSL and Sony. **Recorders:** Studer, Sony, Otari. **Digital audio workstations:** SSL Screen Sound (5) and Slynelevier Comp. post production SD. **Monitors:** Urei 813's and 811's; Yamaha NS10. **A/V equipment:** Time Code DAT; 3/4inch, 1inch, Beta SP and D-2 Video.
Special Services: Satellite up and down link; Edne & T-1 fiber and switch 56 digital communica-

tors capability.
Association Member: AES, SPARS, ITS.

A X24 D X24/48

NEW RIVER STUDIOS, INC.

408 South Andrews Avenue, Fort Lauderdale, FL 33301, USA.
+ 1 (305) 524 4000; Fax: +1 (305) 524 3999. Owner: New River Productions. **Studio/Bookings Manager:** Virginia Cayia. **No. of studios & dimensions:** 2 studios: "A" 35 x 25 ft. Live Tracking area with tall ceilings in center. "B" MIDI Studio: 15 x 15 ft. with ISO booth (6 x 9 ft.). **Mixing consoles:** "A" Neve 8108 56 in 48 out with Flying Faders. "B" Trident Series 65 32 in 16 out. **Recorders:** 1) Mitsubishi X850 with Apogees, 2) Studer A800 Mark III 24 track Analog. **MIDI set-up:** Studio B: MacIntosh plus computer with performer software, Vision software, Rold D50, D550, Yamaha TX 802, Akai S950, Sampler with 16 bit Upgrade, 360 systems MIDI Bass & Patcher. **Monitors:** "A" Westlake BBSM-10's with Meyer 833 Subwoofers, Westlake BBSM 6, Yamaha NS10M. "B" Westlake BBSM-10's. **Specified outboard:** AMS RMX 16, DMX 15-80S, GML E.Q., EMT 140S Stereo Tube Plate, Lexicon 480L, PCM 70, SPX 9011, API 550 & 550A, Pultec, Eventide H3000, H949, DBX 165, 162 & 902 Desser, Urei 1176 LN. **A/V equipment:** Sony BVU 850 3/4 inches SP Umatic and Zeta 3 Synchronizers.
Special Services: In house maintenance. Assist with housing, rental cars, etc. Location along the banks of the New River in a private historic Mediterranean Village. Close to the airport, beaches and fine restaurants.

A D

ONE WORLD RECORDING CORPORATION

72 East Dedham Street, Boston, MA 02118, USA. +1 (617) 426 8078;
Fax: +1 (617) 426 3709. Owner: Steve Van Natta. **Studio/Bookings Manager:** Alexander Milne. **No. of studios & dimensions:** B room MIDI pre-production 15ft. x 15ft. control. 15ft. x 15ft. live room 6 ft. x 9ft. ISO. Main control room 30ft. x 20ft., live room 60ft. x 30ft., ISO 15ft. x 20ft. **Mixing consoles:** Neve 8038 with 32 1081 EQ and preamps. **Recorders:** Studer 827 24 track. **Digital audio workstations:** Mac FX with 2 gigabytes running Studio Vision & Soundtools. **MIDI set-up:** Studio Vision, IBM: Voyetra Gold Mark III. **Monitors:** JBL 4435 Yamaha NS10m, ARBXI modified, Tannoy 6.5.
Specified outboard: Tape Machines: Studer A827 (2inches) 24-track, Studer A807 (1/4inches), Otar 5050B (1/4inches), Panasonic SV-3700 DAT, Panasonic SV-3500 DAT, Nakamichi MR1 Cassette Deck, 5 Nakamichi MR2 Cassette Decks, Tascam 122 Cassette Deck. Monitors: Steven Durr/JBL 4435 (mains), Audio Pro AR 18 Bxi (near field), Yamaha NS10m Studio (near field), Tannoy PBM 6.8 (near field), Urei 809 (studio), 10 AKG 141 Monitor Headphones, 2 AKG 240 Monitor Headphones. Monitor Amps: Belles OCM 500 (mains), Belles OCM 200 (near field), 2 QSC 1400 (cue system), Ashley Mos-Fet 200 (studio). Computers: MAC II FX (running Sound Tools and Studio Vision), IBM PC (running Voyetra MK 4 Gold), 360 Systems MIDI Patcher, JL Cooper PPS-100, Opcode Studio 3. Acoustic Instruments: Steinway 6.5' Grand Piano (1901), Fender Rhodes ('73), Yamaha Recording Series 9 piece drum kit, Tama Swingstar 7 piece drum kit, Vintage guitars & basses from Fender, Gibson, and Guild. Synthesizers: 2 Yamaha DX7's, Yamaha TX816, Yamaha DX7 II FD, Emulator II+HD (with full library), Emu SP-12 (with sampling), Alesis HR-16, Kurzweil Midiboard, Roland S550, Korg Mono/Poly, Moag - Mini Moag, Oberheim OB-8, Roland JX8P, Roland TR 909, Yamaha CS 5. Mics: 2 AKG 414 B-ULS, 2 AKG 451 CK1 (matched), 1 AKG 451 EB, 1 AKG 224, 2Beyer Dynamic M69, 2 Electrovoice RE-20, 2 Electrovoice PL-20, 1 Electrovoice 55, 1 Neumann U87, 1 Neumann U47, 2 Neumann TLM 170, 2 PZM, 2 Sennheiser 441, 8 Sennheiser 421, 2 Shure SM 58, 6 Shure SM 57, 1 Shure SM 7, 1Sony C37, 1 Sony C33, 2 Vintage Sony ECM 56F (matched).
Special Services: One World is a full service music production facility with a client list that includes commercial applications as well as album orientated projects. We produce everything from traditional jazz to urban comtemporary. Call for Studio Gduie and booking information.

A

THE PRODUCTION BLOCK STUDIOS

906 East Fifth Street, Austin, Texas 78702 USA. +1 (512) 472 8975; Fax: +1 (512) 476 5635. Owner: Joel C. Block **Studio/Bookings Manager:** Delaine Frasier. **No. of studios & dimensions:** 2 studios and one free listening room. **Mixing consoles:** Otari 54 and Yamaha MR-1642. **Recorders:** Sony/MCI JH 24-track and Tascam MS 16-track. **Monitors:** Sony 29inches, Samsung VM 3105, Sony KV20TS30. **Specified outboard:** Adams Smith Zeta 3, Alesis PA-100, Crown D-150, Crown D-75, DBX 160x compressors/limiter, DB 165A, Eventide HD3000 Harmonizer, Alesis 3630 Compressor, Yamaha SPX-900, Yamaha SPX-9011 effects processors Lexicon LXP-15 reverb, sound performance "The vitalizer" equalizer. **A/V equipment:** Adam Smith Zeta 3.
Special Services: Advertising voice-overs, industrial and slideshows narrations, radio shows music libraries, SFX libraries, Phone patch jingle packages/post scoring. Time compression, DAT mastering, remotes, automatic dialogue replacement, R-T-R- and cassette duplication, night music packages.

A

RECORDING ARTS

Box 121702 Nashville, TN 37212, USA. +1 615 321 5479; Fax: +1 615 321 0756. Owner: Carl Tatz. **Studio/Bookings Manager:** Lou Johnson. **Mixing consoles:** SoundCraft 3200. **Recorders:** Sony 3348, Mitsubishi X-850 Apogee Filters. **MIDI set-up:** Name it. **Monitors:** Yamaha NS-1000s, NS10 Studios, Name It. **Specified outboard:** Lexicon 480L, (2) Lexicon PCM70, Lexicon Prime Time, Eventide H3000SE V, Klark Teknik DN780, Roland SRV-2000, Roland DEP-5, Roland Dimension D, Yamaha REV7, Yamaha SPX90 II, Teletronix LA-2A, (3) UREI 1176LN, (2) LA-4 , (3) dbx 160X, Drawmer DL241, Valley People 440, (2) Focusrite ISA 110 mic pre/EQ, Adams-Smith Zeta-3, Alesis D4 drum sampler, J.L. Cooper PPS sync, Neumann U87 modified, (2) Akg 414EB, EV RE20, (4) Sennheiser 421, (2) Sennheiser 441, (3) Shure SM57, (2) Bryston 6B amp, (3) Bryston 4B amp, Recording Arts comp/ease vocal compilator, (8) Judenus headphone boxes. Sony CD player, (2) LofTech TS-1 RMX test set, Alphatron PC-100 Phase Checker.
Association member: NARAS, CMA, NEA.

REFLECTION SOUND STUDIOS

1018 Central Ave. - Charlotte, N.C. 28204, USA. +1 704 377 4596; Fax: +1 704 375 9723. Owner: Wayne Jernigan. **Studio/Bookings Manager:** Kelly Bright. **No. of studios & dimensions:** Studio A: 44 x 32 with 18 ft. ceiling, control room 19 x 24. Studio C: 8 x 24 with 12 ft. ceiling, control room 18 x 14. Studio B: 10 x 12, control room 20 x 12 (MIDI suite). **Mixing consoles:** Sony MXP 3036 36-ch. with hard-disk automation, Sony MXP 3036 32-ch, Allen & Heath Saber Plus 36-ch JL Cooper MAGI automation. **Recorders:** (2) Sony PCM-3324A 24-track digital, Sony APR 24-track analog, Sony/MCI 24-track analog, Sony PCM 3402 2-track. **Digital audio workstations:** Digidesign Sound Tools with 1.8 gigbyte hard drive. (Studio B). **MIDI set-up:** Mac IIcx and SE running Digital Performer interfaced with (2) Motu MIDI Time Pieces. Roland D-550, Korg M1Rex, Akai S1000 with 20 meg RAM, E-mu Proteus 1, Proteus 2, Procussion, Yamaha TX7 and TX81Z, Oberheim Matrix 1000, Alesis HR-16 and HR-16B, KX-88 and Octapad controlers. **Monitors:** Studios A and C-TAD double 15 inches systems, powered by Hafler and AB systems amp with White EQ. Tannoy Series 8 with subwoofer (Studio B). **Specified outboard:** T.C. 2290 (2), Eventide H300 ultra-harmonizer (3) Lexicon PCM 70, Lexicon 224XL with LARC controller, EMT 140 stereo reverb. (3) dBx 900 frames. Too much more to list. **A/V equipment:** Sony V09800 3/4 inches VTR, Sony monitors, Motu Video Time Peice, Adams-Smith Zeta Three Synchronizer.
Special Services: On staff composer, 24 and 48 track digital or analog remote recording.

D A

SHEFFIELD AUDIO/VIDEO PRODUCTIONS

13816 Sunnybrook Road, Poenix, MD 21131, USA. +1 410 628 7260; Fax: +1 410 628 1977. Owner: John Ariosa. **Studio/Bookings Manager:** Richard Van Horn.
No. of studios & dimensions: (1) 50x40 with 18ft. ceilings. **Mixing consoles:** SSL E/G. **Recorders:** Sony 3324 Digital Otari MTR 90-Analog. **Digital audio workstation:** Pro Tools - Mac Quadra. **Monitors:** Urei 813B, Yamaha NS-10. **A/V equipment:** Lynx Timeline, 1 x Beta Machines.
Special Services: 48ft. Remote Truck w/ SSL + Neve Consoles - 30ft. Video Remote Truck.
Association member: AES

SOUND TECHNIQUES INC.

1260 Boylston Street, Boston, MA 02215, USA. +1 (617) 536 1166; Fax: +1 (617) 536 4446. Studio/Bookings Manager: Susie Pottery, Gina Romani.
No. of studios & dimensions: CRA 510sqft. Studio A 775 sqft. ISO1 168 sqft. ISO2 95 sqft. ISO3 42sqft. CRB 510 sqft. Studio B: 400 sqft. ISO1 48 sqft. ISO2 30sqft. CRC 220sqft. ISO 88sqft. Scenaria Suite 500sqft. ISO 120sqft. **Mixing consoles:** SSL 4056 G3.2, SSL Scenaria, Neve Flying Faders, Allen & Heath Saber. **Recorders:** Sony 3324 (2), sony APR 24, Ampex ATR 102(1/2"), Sony 7030 DAT with TC, Sony 5003 with TC, Otari MTR10 (3), Pansonic SV3700 (5). **Digital audio workstations:** SSL Scenaria, SSL Screen Sound (2), all networkedvia, SSL Sound NET, DigiDesign Sound Tools. **MIDI set-up:** MAC IIs in all studios, pro, performer, S1000, K2000, K250, Proteus (2), 01w(2), R8M (2), M1, D550,MKS80, MKS70, Young Chang 7ft Grand. **Monitors:** Westlake BBSM-12 (3 Pair), meyer hd-1 (3 pair), Westlake BBSM-4, Yamaha NS10, Tannoy PBM 6.5. **A/V equipment:** Ampex VPRS, Sony BVH 3100, Sony BVW75, Sony BVU800. **Specified outboard:** Reverb Effects: Lexicon 480L, 224, PCM70(2), Quantec QRS XL, TC M5000, TC 2290(3), Eventide H3000(3), Dynacord DRP20(2), KT DN780(2), Roland SDE 3000(5), Yamaha REV5(2), SPX9011(2). Compressors: Urie LA2A(2), Tubetech CL1A, UA175(2), DBX165(3), DBX160X,DBX900. Equalizers: Pultech EQP1S(2), Summit EQF100, DBX900's, Dolby 740, SPL Vitalizer,BBE 804, Orban(2). Gates: Drawmer DS201(4), Ashley SC33, DBX 900's. Synchronizers: 6 Timeline Lynx with SSU+KCU, SSL Screensounds, JSK1128 Timecode Machine. Microphones: U47(Klause Heine), U47Fet, U67(Klause Heyne)(2), U87A(5), KM84(2), AKG C-24, 414(6), D112, D12E 460(6), Beyer MC740N(2), 160N(2), Senheisser 421 (6), 441(4), B&K 4003, PML DC96,EV RE20(3), Shure SM57 (10), SM81(2).
Special Services: In house engineers, original compose work, satellite, talent casting, duplication, located in centre of Boston, near clubs hotels and restaurants.

TRUSTY TUNESHOP RECORDING STUDIOS

8771 Rose Creek Road, Nebo, Ky. 42441 USA. +1 502 249 3194(Tel.+Fax). Owner: Elsie Trusty Childers.
No. of studios and dimensions: 2 Studios: 32 x 36 11ft. ceiling; 12ft. x 18ft., control room- 20 x 22ft. **Mixing Console:** Tascam M3700 (24 tracks), Studiomaster 16 into 8, **Recorders:** Tascam DA-88 (3). **Monitors:** JBL 4412A (2). **Specified outboard:** PMC-70 Digital Effects Processor, Roland R-8 MK II, Sv-3700, RTU-11, Magnavox - Custom made (2), Yamaha NS-10(2), Yamaha PS-35 synthesizer, PAIA -Strings & Things Synethesizer, Jesse Green Grand Piano, Otari (8 track), Baldwin Acousonic Piano.

ULTRASONIC STUDIOS

7201 Washington Ave; New Orleans, LA 70125 USA. +1 (504) 486-4873; Fax: +1 (504) 488 1057. Owner: Jay Gallagher. **Studio/Bookings Manager:** Steve Reynolds.
No. of studios & dimensions: Two studios: "A" 45 x 50 ft., "B" 15 x 15 ft. **Mixing consoles:** MCI/Sony JH 652. Soundcraft Delta 200. **Recorders:** Studer A827 with SR. MCI JH24. **Digital audio workstations:** Pro-Tools 8 tracks. **MIDI set-up:**SoundTools II, D-4, Roland S-50. MAC II CI/ Opcode Studio Vision; Kurzweil K2000R; Wavestation AD. **Monitors:** JBL 4331A; Tannoy SGM 10 with M.L. x overs.
Specified outboard: Lexicon 300; Summit EQP 200A; Drawmer DL241 & 1960. **A/V equipment:** Sony 3/4 inch recorder.
Special Services: Remote recording; ADR; Sound Design.
Association member: NARIS.

WALLY CLEAVER'S RECORDING

2200 Airport Avenue,Fredricksburg, CA 22401-7220, USA. +1 (703) 373 6511 - (Tel. + Fax). Owner: Peter Bonta. **Studio/Bookings Manager:** Buffalo Bob.
No. of studios & dimensions: Room 1: Studio 20 ft. x 40 ft., control room 18 ft. x 22 ft. Room 2: Studio 8 ft. x 8 ft., Room 3: Studio 8 ft. x 9 ft. **Mixing consoles:** Custom ESP 44 Input with JL Cooper Magi II Automation. **Recorders:** MC1 JH-114 24 track with AL III, Tascam 85-16B 16-Track 1inch, Otari 5050 2 -track, scully 280B 2 -track, Revox A77 2 -track, Panasonic SV-3500, and SV-3700 DAT Recorders, 2 Tascam C-3RX Cassette Recorders. **MIDI set-up:** Alesis MMT-8 Sequencer, Alesis Data Disk, Akai ME-35T Audio/MIDI Trigger, J.L. Cooper PPS-1 SMPTE/MIDI Generator, Atari 1040 STE Computer with MIDI sequencing software, 2000 + voice DX-7 Library, 300 + Voice Akai S-900 Sample Library. **Monitors:** Urei 809 Time aligned monitors, Tannoy PBM 6.5, Near -Field Monitors. **Specified outboard:** Urei 1176 Limiters, Urei LA-4 Limiters, DBX 160 Limiters, Aphex aural exciter, Neumann U-87, U-47 FET, KM-84; Sennheiser MD-441, MD-409, MD-421; RCH 44A, 77DX, BK5; Yamaha Rev-7, Sony R-7, Alesis Quadraverbs, Alesis MIDI verb III, Lexicon PCM-41, SDR-1000 Digital Reverb.
Special Services: In House engineers and producers.
Association member: AES, AF of M.

WENDELL RECORDING STUDIO

Lockes Hill Road, Wendell, MA 01379, USA. +1 (508) 544 8288 - Tel+Fax. Owner: Jeffrey Bauman. **Studio/Bookings Manager:** Judith Bauman.
No. of studios & dimensions: 1 studio - playing room 25 ft. x 28ft. x 19, control room 22 ft. x 16ft. **Mixing consoles:** Trident 24 Recording console, with 32 channel mega -mix automation. **Recorders:** Otari MTR 90 II 24 Tracj; 3M M79 1/2 inch 2 track; Otari 56/50 1/4inch, Panasonic SV 3700 DAT, Sony PCM 501 Digital 2 track processor, Esoteric V9000 cassette, Aiwa AD-S 40 Cassette. **MIDI set-up:** Opcode 2 + 2. **Monitors:** Gauss 7258; Yamaha NS-10M; Aurotone T6; Aurotone Sound Cubes; Electro-Voice Sentry 100A; ADS L700; Altec 604. **Specified outboard:** Lexicon PCM70, 6042, Teltronix LA2A, (2) Pultec MEQ-5, Drawmer 201 Noise Gates, Valley People 610, Quadraverb plus, Yamaha SPX900, DBX166, Ashley SC50, 2 Ashley SG-33 Noise faders, DBX 150L101, Ashley SE66 Parametric EQ; Aphex Aural Exciter, Delta Lab Super Prime Time Digital Delay, Urei 1176 compressor; Valley People Dynamite, Klark Teknik, DN360 Equalizer, Orban Co-Operator, Master Room XL-305.
Special Services: Residential facility.

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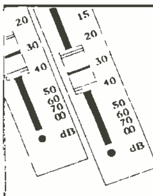

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DDA AMR24 36 fitted 28, Mastermix II	£26,500	TAC Matchless, 36 channels, 1988	£7,975
Harrison MR3 36 channels, auto, 1985,	£18,000	Studer A800 mk III, SSL interface,	£14,500
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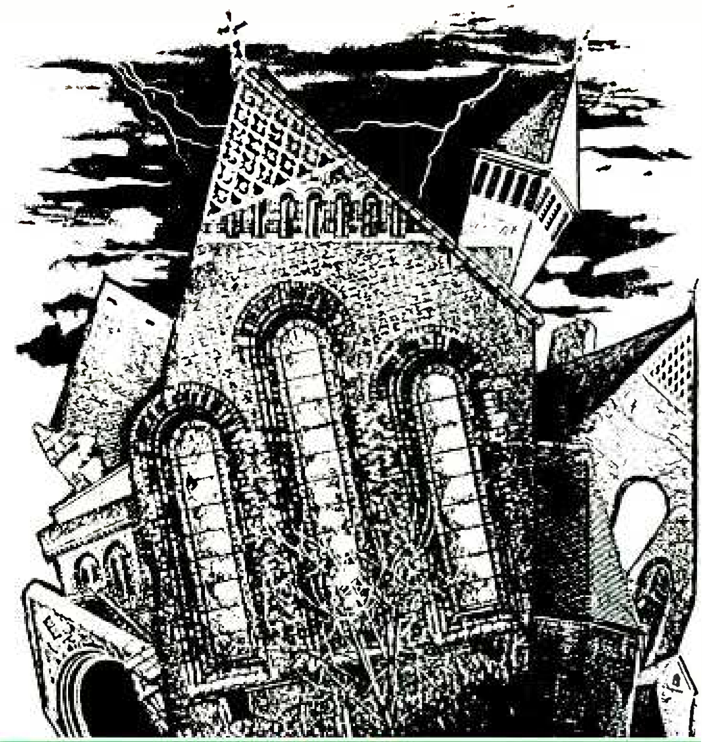

Barry Fox

Sounding out Super Bit Mapping and airing incompatibilities at Lyndhurst Hall

It was predictable, but a pity, that no-one from Deutsche Grammophon accepted Sony's invitation to attend the open day on Super Bit Mapping held at Air Lyndhurst at the end of May. It was predictable because SBM is a rival for DG's 4D-Authentic Bit Imaging-High Bit system. Significantly, DG's parent, Polygram, was there at Air, and cannot have failed to notice the big difference between how DG and Sony have pitched their stalls.

It is a pity DG did not show up because the Germans would have seen an object lesson in how to present the record industry with a clear proposition, backed by comparative demonstrations in a fine acoustic and with engineers on hand to answer questions with authority.

Indeed, the Air demos may well have won Sony the high ground on SBM, and nailed the coffin on DG's plans, irrespective of system merits.

To recap briefly. All digital recording involves chopping the smooth analogue waveform into discrete steps, for measurement. The stepping injects white noise (equal level at all frequencies) into the final sound. The more bits in the digital word, the more steps there can be. The smaller the steps, the less noise injected. The CD standard uses 16-bit words and although the noise is too quiet to be recognisable, it subtly coarsens very low level sounds—like the natural ambience and echo of a concert hall. The new 18 and 20-bit recorders define smaller steps and thus add less coarsening. But the advantage is lost when the higher bit recording is, of necessity, down-converted to 16-bit words for commercial release on CD.

The noise created when the extra bits are thrown away forms a floor under the full range of recorded sound. SBM shapes this noise; it redistributes its energy, unevenly across the frequency range, so that there is less energy in the range 3kHz–5kHz, where the human ear is most sensitive to quiet sounds, and more in the frequencies where the ear is less sensitive.

Sony have so far been cagey about how this is done, but have now explained that the SBM processor uses a feedback loop which compares the original higher resolution signal with a simple down-conversion of it to 16-bit code. The difference gives a mathematical telltale of the noise content. This telltale is used to set a digital filter which changes the shape of the signal as it is being down-converted, so that the down-converted signal has a shaped noise floor. Unlike some systems on offer, SBM is fully compatible with all existing CD players.

I sat in (uninvited, but allowed to stay) at the Air session given to APRS members. Sony compared the sound of music, and subtle effects like water dripping on metal, in three ways; from an original 20-bit recording, after crude truncation to 16 bits, and with noise shaping. Without exception, every engineer judged the noise-shaped sound as very close to the 20-bit original, and far clearer than the truncated sound.

Said Avi Landenberg, of Chop 'Em Out, the first mastering house to install SBM equipment, 'SBM will do for CD what the Dolby logo did for cassettes. The cost of the hardware is small and no extra work is needed, so we do not intend making any increase in mastering charges.'

Said Alan Philips, Vice President Sony Software Corporation of Sony Music: 'We want to encourage the widespread adoption of SBM, both for new releases and back-catalogue reissues. We are giving the technology away. There is no licence fee. The record industry has nothing to lose and a lot to gain. SBM makes CD the premium sound carrier for the next ten or 20 years.'

Sony had flown in George Massenburg, respected sound engineer and producer, and audio system designer, from Los Angeles. He played a tape he had made of a James Taylor live concert which lost almost nothing in sound detail when down-converted from its 20-bit original to 16-bit code through the SBM processor.

Engineers expressed only one concern. How will Sony police the licence? Although SBM can improve the sound of a transfer from high quality analogue original to CD, it can do nothing for a 16-bit digital master, or ropey old analogue tape. What is there to prevent record companies slapping the SBM logo on any old recording?

Sony have filed patents, and trademark applications on the technology and SBM logo. Record companies can now apply to Sony for a free licence to use the trademark on CDs made using Sony's down convertor.

'We cannot act as policemen,' admit Sony, 'but the licence does give us right of sanction.'

So what's in it for Sony? In the short term Sony gain only on the sale of SBM convertors to recording studios, at £10,000 each. In the long term Sony gain by turning the industry onto the idea of 20-bit recording, and selling it 20-bit, and later 24-bit, studio hardware.

Sony Music in the US have already reissued Miles Davis' classic *Kind of Blue* album, after going back to the original master tapes and using SBM. The company are currently reprinting a rave revue of the reissue from the *Wall Street Journal*.

Moving on to Air Lyndhurst, to boggle at the wonderful job George Martin had done on the old church building—although everyone is very tactful, it is clear that English Heritage have given them all a terrible time, even insisting that the old organ (hacked apart by a previous show-biz user of the building and then further damaged by rain and previous builders) be replaced with a lookalike mock-up.

Speaking as someone who knew the church when it was still 'working' (my kids went to Cub Scout classes in the room upstairs which is now a video editing suite), I can tell you it looks a whole lot better now than it ever did before. And just over the road a similar church is quite literally falling apart because no-one can afford to meet the requirements for restoration.

Air's video editing suite is equipped with Pioneer's magneto-optical video disc recorder, ganged to an AMS *AudioFile*. So editors can now search through synchronised audio and video, with none of the time usually lost on waiting for the video tape to spool.

Pioneer first showed the VDR a year ago, in American NTSC format. Air Lyndhurst has one of the first European PAL models. But a lot of Air's clients come in from the USA. The studio's first big recording session was Henry Mancini on *Son of Pink Panther*. As sure as night follows day, American producers will bring in NTSC VDR discs of movie sequences, and they won't play on Air's PAL VDR.

Pioneer considered the idea of building a dual standard VDR, which could both play and record NTSC or PAL, but rejected the idea as too expensive. Air's back projection video monitor is dual standard PAL-NTSC and studios like Air only need to play back from NTSC discs, not record in NTSC. Dual standard playback technology is cheap, and a routine facility on Laser Disc players. So why not build NTSC playback into the PAL VDR?

'Good idea,' say Pioneer, 'what a pity we didn't think of that.'

A pity for Air, too, when someone flies in with an NTSC disc to edit. ■

ILLUSTRATION: CARL FLINT