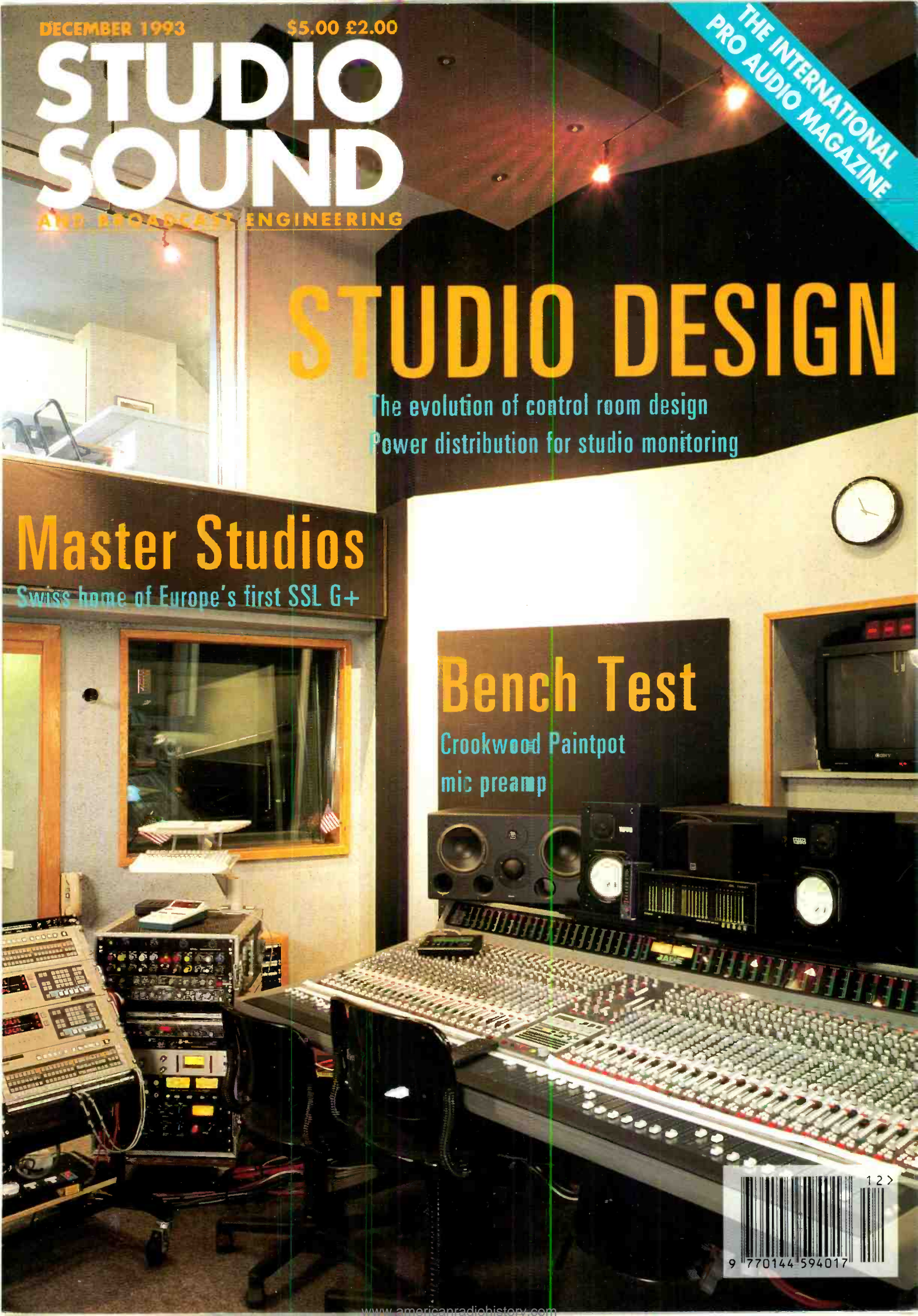# STUDIO
SOUND

STUDIO DESIGN

The evolution of control room design
Power distribution for studio monitoring

Master Studios

Swiss home of Europe's first SSL G+

Bench Test

Crookwood Paintpot
mic preamp

STUDIO SOUND

AND BROADCAST ENGINEERING

Take a cable car
to the Säntis
mountains
when you visit
Master Studios.
See page 31

New foundations. See page 14

3

The media and the message

As the December issue of *Studio Sound* goes to press, there is chaos in the Editorial offices. Not only have we been busy with the last issue of 1993, but also with a special Mastering, Duplication and Replication supplement (*MDR*). We have also come up with something we believe is a bit of a first—*Studio Sound Interactive*.

MDR has been produced in response to an ongoing expansion of interest in the art and mechanics of mastering. The differences between European and American attitudes to mastering music are both long standing and well recognised. Until now, the response of the British—given their historical importance in dictating the direction of popular music—has been disappointing to put it mildly. Finally, however, it seems that a renewed effort is being made to bridge the gap between the two national attitudes. The new mastering facility at London's Metropolis is covered in this issue but it is accompanied by similar activities at the former London Hit Factory, now Sony studios.

These activities are also reflected in directions in technological development. Requantising processes such as Sony's Super Bit Mapping, Prism's DRE (on their *AD-1*) and Apogee's UV22 (and also Deutsche Grammophon's Authentic Bit Imaging system, although this is not presently available in a commercial unit) are changing the nature of mastering for digital media. In addition to these processes, there has been a steady trend on behalf of the manufacturers of hard-disk editors (such as those from Digidesign and Studio Audio & Video) to extend the capabilities of their systems into the realm of premastering. It is hardly surprising, therefore, that recording studio personnel are expanding their interests and activities accordingly. This supplement is another facet of this development.

It should be pointed out that the circulation for *MDR* has been derived from the mailing databases held by Spotlight Publications and is not simply that of *Studio Sound*—if you are a *Studio Sound* reader not in receipt of *MDR* and you feel that you would benefit from receiving a copy, please contact the *Studio Sound* circulation department at the usual editorial address.

The diskette many of you will have found included in this issue of *Studio Sound* is what we have chosen to call *Studio Sound Interactive*. On it you will find a *Macintosh*-based listing of significant product releases that have occurred during the last year, along with useful contact details for the relevant manufacturers and other useful information concerning purchasers and references to editorial coverage that has appeared in the pages of *Studio Sound*. Apart from being of general interest, this disk is intended to help you with purchasing projects and to provide you with a convenient database on many of the companies active in pro-audio. The program is intended to be simple to navigate, and will help you find your way around.

Although not every issue of *Studio Sound* contains *Studio Sound Interactive*, at least one copy should have gone to each studio, manufacturer and so on. Please copy the program to your hard drive for ongoing use, and circulate the disk among your colleagues and friends.

For those of you not using *Macintosh* machines, a PC version of the program is available if you contact the magazine. We believe *Studio Sound Interactive* to be a publishing first and welcome feedback on any aspect of its presentation and content. The time and effort involved in coordinating and producing these additional facets of *Studio Sound* has been considerable. I hope they are are valuable to you. ■

Tim Goodyer

Cover: Soundtracs *Jade* console in Steve Lipson's studio.
Studio design by Architecture Design

International News

NAB seek help to defeat increased artist royalties

Edward Fritts, president and CEO of the National Association of Broadcasters, has asked broadcasters to stand up for their rights and fight the newly proposed charges by the recording industry.

Fritts was addressing the Federation of Australian Commercial Television Stations and proposed that the NAB would become a 'clearing house' for all broadcast groups fighting both new fees and the expanded application of performance rights in the dozens of countries where such rights already exist.'

He continued, 'American broadcasters already pay $300 million each year for music rights, and performance royalties could cost them hundreds of millions of dollars more. Worldwide, the economic threat is many times that.'

NAB and other broadcasting groups claim that the recording industry already receives huge marketing and promotional value from the public exposure radio and television give to recorded music, and that additional performance royalties are neither warranted nor fair.

Fritts concluded: 'It is imperative that we unite with the commercial broadcasting around the world to oppose this scheme by the powerful record companies to squeeze windfall royalties from broadcasters.'
NAB. Tel: +1 202 429 5350.

Swiss AES Digital Audio Seminar

The Swiss Section of the AES ran a three-day seminar on all aspects of digital audio recently and the success was confirmed by the attendance of nearly 200 delegates against the 50–60 anticipated. President for the AES organising committee, Peter Joss, was extremely encouraged by the results which he felt reflected on the quality of the organisation and committee members.

The seminar was held at the Swiss PTT Centre in Bern and included a first-day workshop covering all aspects of digital audio in preparation for the intensive sessions to follow.
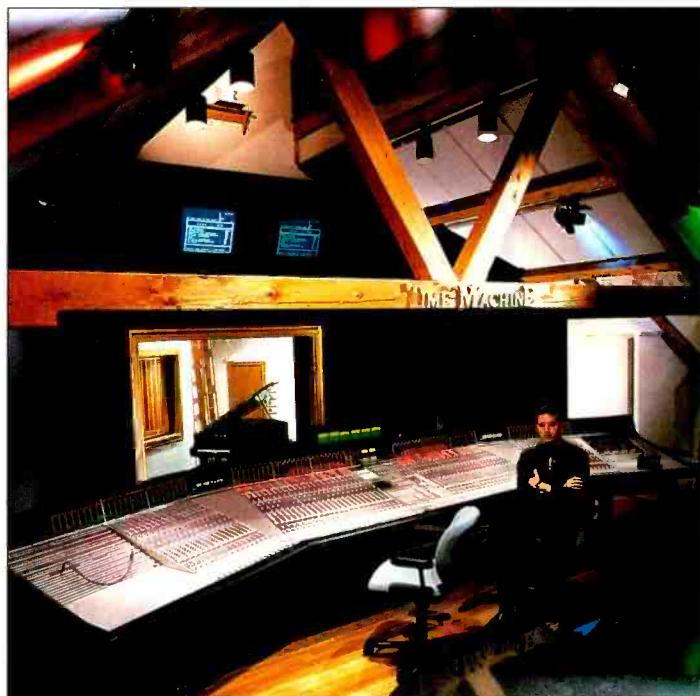
The Time Machine Studio based on a mountain in Vermont, US, has recently installed this 80-input winged Solid State Logic *SL 4000 G Plus* console in its Studio A. The 11- building, 150-acre complex houses two independent but interconnected recording environments. The Time Machine and The Time Capsule which includes Sony *PCM-3348* multitrack, multiple Studer and Otari recorders and a Euphonix *CS2000* console

Papers were supervised by two of Switzerland's leading digital engineers, Markus Erne and Claude Cellier, and lecturers represented both Switzerland and abroad, including the UK, Germany, France and Holland.

Recognising the need for 'hands-on' tutorials, a Digital Audio Workstation workshop was organised with systems supplied by Fairlight, Studer-Dyaxis and Sonic Solutions and demonstrating the various applications for workstations in different fields.

The evening was concluded by a lecture from Alan Parsons who compared the advantages of digital and analogue systems and encouraged response from the floor. A lighter note was struck when Parsons asked from what walks of the industry the audience were from and found that the percentage from the working audio fields was quite low.

Delegates came from all parts of Europe, which showed that the seminar was more international than regional and demonstrated the need for this kind of event. The fact that the attendance was so high also made the seminar financially viable and thus paved the way for future events.

One result of the success of the seminar has been the founding of a new organisation, The Society for the Furtherance of the Swiss Audio Scene (a loose translation from the German), which came into existence at the beginning of December, 1993.

There has also been demand for another seminar, probably to be held in the French part of Switzerland, and feedback has shown that a diminished lecture programme, but with further workshops, is desirable.
Terry Nelson
Swiss AES: Call Peter Joss,
c/o J&C Intersonic.
Tel: +41 56321850

Mics and Men

Further to 'Of Mics and Men' (*Studio Sound*, September 1993), more information on the European Telecommunications Standard Institute (ETSI, not ETFI as credited in the article) has become available.

In 1990, the ETSI established the RES-08 group to write a standard for wireless mics. At its inaugural meeting in Italy, sufficiently diverse interests were identified to prompt the formation of a special group called Working Party 3. This group —now consisting of Chairman Brian Copsey (ASP FM, UK), John Wykes (Micron, UK), Dr E Werner

(Sennheiser, Germany), Bernard Bond (DTI, UK), Clive Lorton (Sony, UK), Stephen Frese (AKG, Austria) and John Sykes (BBC, UK)—have invested considerable effort in pursuit of two documents: the *Wireless Microphone Standard* (radio regulatory) and the *EMC Standard for Wireless Microphones* and similar RF audio link equipment. It is anticipated that this standard will become an ETS by 1996.

The RES-08 team had no set policy regarding the involvement of individual companies, and membership would have been open to any interested parties at the time of its establishment.

It is largely due to the effort of this group that Europe enjoys its present number of available frequencies and freedom of use.

Thanks to Sony's Clive Lorton, Micron's John Wykes and Sennheiser's Dr Werner for further information on this issue.

Virtual orchestra plays the classics

The orchestra of the future from Oscar Technology

The 1993 Classic Music Show provided the stage for Oscar Music's latest demonstration of the sophistication of MIDI technology. A full-size walk-through orchestra was constructed, with 13 Quad electrostatic speakers representing the different instruments.

Eight Apple *Mac* computers running *Notator Logic* were used to play Oscar's MIDI transcripts of Mozart and Beethoven, and scroll the score as the music played. Visitors were able to conduct the orchestra using a special dataglove. The orchestra sounds were provided by a Yamaha *SY77*, a Roland *S750* sampler and a *Proteus II* module. Also featured in Oscar Music's technology area was the world's first VR musical instrument, created by VR company Virtual S. Oscar Music are also bringing together independent CD-i developers to focus on the creative and technical aspects of working with CD-i. Alternative ways of publishing and distributing product will also be explored.

Oscar Music Productions.
Tel: 071 377 6294.

Events of 1994

January 18th–19th	The Live Show, Royal Horticultural Halls, London, UK
January 21st–24th	Winter NAMM, Anaheim, US
January 30th–February 3rd	MIDEM, Cannes, France
February 8th–9th	Sound '94, Sandown Exhibition Centre, Esher, Surrey, UK
February 8th–10th	The ISDN User Show Exhibition, London
February 13th–16th	SIEL, Paris, France
February 26th–March 1st	96th AES Convention, Amsterdam, The Netherlands
March 9th–13th	ITA, Tuscon, USA
March 16th–20th	Musikmesse, Frankfurt, Germany
March 20th–24th	NAB, Las Vegas, USA
April 12th–14th	Replitech, Europe, Munich, Germany
May 7th–11th	Pro Audio Light & Music, Beijing, China
May 16th–20th	AV and Broadcast China, Guangzhou, China
June 1st–4th	Broadcast Asia, World Trade Centre, Singapore
June 7th–9th	Multimedia '94, Earls Court 2, London, UK
June 14th–16th	Replitech International and ITA, Santa Clara, California
June 22nd–24th	APRS, Olympia, London, UK
July 6th–8th	Pro Audio Asia, Singapore
July 24th–26th	BMF, Olympia 2, London, UK
July 30th–31st	NAMM, Nashville, USA
September 11th–14th	PLASA, Earl's Court, London, UK
September 18th–20th	IBC '94, Amsterdam, The Netherlands
September 19th–23rd	Image World Video Expo, New York, USA
September 22nd–27th	Photokina, Cologne
October 12th–15th	NAB, Los Angeles, USA
November 10th–11th	SBES, Birmingham, UK
November 10th–13th	97th AES Convention, San Francisco, USA
November 16th–18th	Tonmeister, Karlsruhe, Germany
November 16th–18th	Interbee, Makuhari, Japan

Information correct at the time of going to press

Contracts

● **Solon upgrade with dance**
The Solon Corporation have upgraded their facility in Brixton to include a *Pro Tools* hard disk multitrack system, Timeline *MicroLynx* and DynaudioAcoustics monitoring. A new office and dance label is also planned.

● **John Henry buys EAW Series**
London-based John Henry Audio have expanded their sound reinforcement rental division with the purchase of £144,000 28-enclosure Eastern Acoustic Works *Forsythe Series* Virtual Array touring and concert system.

● *Vienna II* **for B&H Sound**
B&H Sound Services have taken delivery of a 40-channel Soundcraft *Vienna II* console, supplied by LMC Audio Systems. The desk will see use in London's Royal Albert Hall, to which B&H are contracted to supply sound reinforcement.

● **Meyer for Stavanger Konsert Hall**
Norway's prestigious Stavanger Konsert Hall has recently upgraded and expanded its Meyer Sound system, taking delivery of four more *UPA-IC UltraSeries* loudspeakers from local Meyer agents. The overall system stands at a total of 12 Meyer Sound *UPA-IC* loudspeakers and four *USW-I UltraSeries* subwoofers, all with the appropriate control electronics units.

● **Satellite orbits with ARX**
Indonesian TV company SCTV have bought ARX *EQ60* equalisers and *Afterburner* comp-limiters for uplink EQ and gain control for the Palapa Indonesian Satellite.

● **BRMB order Xtra Discarts**
The Midlands Radio Group have recently ordered Sonifex *Discart* machines for the studios of BRMB and Xtra AM. Each On-Air studio will be equipped with two *Discart DX30* triple stack playback machines, production areas will have *Discart DX300* player-recorder machines.

▼ **AIR ravin' for Craven in TV show**
The Great Hall at AIR Studios in London played host to singer-songwriter Beverley Craven for a one hour TV show due for a December transmission. Air's Neve *VRP* desk and a Sony 48-track digital were used for the sound.

Beverley Craven in the spotlight with a Neve *VRP* console in AIR's Great Hall studio, London

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Solitaire joins Sequel II

The *Solitaire* console from Soundtracs is a 24-bus in-line design available with the proprietary ADP dynamics package providing gates, compression, limiting, expansion, modulation and autopanning on each of 24, 32 or 40 channels. Automation options of either VCA or moving fader will also be available soon. With dual inputs on each channel the *Solitaire* allows 88 inputs on mixdown, combined with automation and dynamics.

The *Sequel II* sound reinforcement console offers 4-band FdB EQ, group muting, VCA muting, VCA grouping as standard and the ADP assignable dynamics package providing gating, compression, limiting, expansion, modulation and autopanning.
**Soundtracs, 91 Ewell Road, Surbiton, Surrey. KT6 6AH, UK.
Tel: 081 399 3392.
Fax: 081 399 6821.**

Augan aim to make hard disk systems obsolete

Augan Instruments have introduced a new type of magneto-optical disk drive for their *408 OMX* optical multitrack recorder-editor. The new disk drive, which uses standard 1.3Gb disks, promise to change the audio industry's attitude towards the traditional hard-disk-based digital audio workstations.

The new optical disk drive take the specifications of the *408 OMX* one step further. Firstly, the use of a new optical disk with an increased capacity of 1.3Gb means that the standard dual-drive *408 OMX* system now has an on-line audio capacity of 4 hours. This can be extended by connecting external disk drives, bringing the total capacity up to 18 hours. The new disk has a much higher data throughput which means that more channels of audio can be played and recorded simultaneously. Augan now claim it is possible to create a complete multitrack project, including edits and crossfades, on only one optical disk.
**Augan Instruments BV, Wilhelminastraat 31, 6881 LH Velp, The Netherlands.
Tel: +31 85 648966.
Fax: +31 85 644785.**
UK: Studer Revox UK, Foster House, Maxwell Road, Elstree Way, Borehamwood. Herts WD6 1JH.
Tel: 081 953 3533.
Fax: 081 207 5103.

AKG focus on home recording

AKG have introduced the *C3000*, a large diaphragm condenser vocal and instrument microphone aimed at the home and project studio market. The *C3000* features include a gold-plated capsule that allows the microphone to emulate the sonic characteristics of the classic *C414*. Features also include two polar patterns, hypercardioid and cardioid; built-in elastic spider-suspension; bass roll-off and -10dB pre-attenuation switching for close miking of loud instruments.
UK: AKG Acoustics, Vienna Court, Lammas Road, Godalming, Surrey, GU7 1JG, UK.
Tel: 0483 425702 Fax: 0483 428967
USA: AKG Acoustics, 1525 Alvarado Street, San Leandro, CA 94577.
Tel: +1 510 351 3500.
Fax: +1 510 351 0500. ▶

In-brief

● *FastTrax* from Synclavier Co.
The ultra-high-speed *FastTrax* 2.4Gb removable disk drive system assures users of virtually unlimited recording time. Whenever one set of drivers is filled, the user simply dismounts them, inserts new drives, and continues recording without interruption—just like tape recording. In addition an Apple *Mac*-based application called *BaxTrax* allows the user to backup disks off-line to a DAT drive using any stand-alone *Macintosh* computer.

● **DAR Feet steps**
DAR have released software upgrades for their *Sigma*, *Delta* and *Sabre* workstations. It's a feet-and-frames facility which provides film editors with familiar position indicators, even when working with time code on a video tape. DAR workstations can be locked to film transporters via additional external hardware, making it possible to combine traditional user practices with modern techniques. Other software updates include Laser Disk Interface; automatic and flexible time code standard switching; and expanded storage options.
**DAR. Tel: 0372 742 848.
Fax: 0372 743532**

● **Celestion's latest Pro Monitors**
Celestion Industries have unveiled their latest Pro Monitor Series loudspeaker. The already established *Studio 3* and *Studio 5* loudspeakers have been upgraded to Series Two models, while Celestion has also introduced a *Studio 1* monitor to complete the line. Each speaker features Celestion's proprietary 1-inch titanium-dome tweeter as well as felted fibre-cone drivers that are now ported.

● **Verity help S-VHS users**
Verity Systems have introduced the *V870* lightweight degausser especially for users of S-VHS tapes. Verity say that reusing the S-VHS cassettes without completely 'deep erasing' previously recorded tracks, often leads to degradation of the cassette recording. **Verity Systems Ltd. Tel: 0252 317000.**

● **Fishman's new *Purple Microdot***
Fishman Transducers have made improvements to both design and construction of its *Purple Microdot* head-mounting acoustic drum-trigger. The new trigger will interface with all electronic drum-trigger modules.

Solitaire mixing console

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Sonifex delay audio

Sonifex digital products have added a digital programme delay to their range. The *DPD 2000* is designed for applications that include audio time shifting, delaying or buffering audio. The maximum active delay periods can be programmed in excess of 14 hours with a minimum delay of 20 seconds.

The *DPD 2000* has a desk-top control panel that can be used manually or set to give automated timed starts and fixed delay periods on command. Search and Retrieve is available on any part of the recorded audio during a live recording and also Playback selected segments at any time without affecting the recorded material.

Sonifex, 61 Station Road, Irthlingborough, Northants. NN9 5QE, UK. Tel: 0933 650 700. Fax: 0933 650 726.

Tahiti Drives for Fairlight

Fairlight are now able to playback typically 12 to 16 simultaneous tracks of audio from the new technology *Tahiti-III* optical drives.
This represents a 50% increase in storage time, to 4 hours per disk (2 hours per side). The new playback capacity means that quite large postproduction projects can be run directly from removable media.

Also in the area of transportable media, Fairlight have announced support of lossless data compression for tape backup. This process has no effect on audio quality as data going to and from tape is bit-perfect. Backup and restore now takes place at 8x mono play speed.

Fairlight Europe, Whiteposts, High Street, Barley, Herts. SG8 8HY, UK. Tel: 0763 849090. Fax: 0763 849090.

Samson in Europe

Samson Technologies Corporation are stepping up their operations in Europe by opening an office in London and launching a new range of audio products.

The audio range consists of three mixing consoles, two power amplifiers, a headphone amplifier and two powered mixers. The mixing consoles are all rackmount with three-band or four-band equalisers and up to 30 inputs. Each of the consoles uses low noise discrete preamplifiers, hybrid circuits for low crosstalk, and have balanced inputs and outputs.

The *Servo 150* and the *Servo 240* rackmount power amplifiers provide 75W/channel and 120W/channel at under 0.1% distortion.

The powered mixers are the *S6* and *S8*. Each is mono in operation and built with the same preamp as the consoles. The *S8* has eight channels, each with 4-band EQ. The *S6* six channels each with 3-band EQ.

Samson Tech Corp, 262 Duffy Avenue, Hicksville, NY 11801. Tel: +1 516 932 3810. Fax: +1 516 932 3815. Europe: Samson Technologies, 154 Clapham Park Road, London, SW4 7BZ. Tel: 071 498 4861. Fax: 071 498 4861.

3M go short for wider choice

The professional audio-video Group of 3M have announced further expansion of their range of professional DAT cassettes. The new cassettes are available in ▶

Samson herald their European arrival with a new audio range. Shown above is the *S8* powered mixer

www.americanradiohistory.com

PRODUCTS

15-minute and 30-minute lengths and can also be supplied in 3M's unique hanger box. These provide a secure container for two cassettes, including full APRS-standard labelling.

The new short-length 3M DAT cassettes use a high-performance ultra-fine metal particulate tape and feature an Anti-Stat treated back-coating. A record lock-out switch is provided and the binder formulation is designed to handle the long term effects of the format's 200rpm head scanner, together with high speed search of 2000x normal speed.
UK: The Professional Audio-Video Group, 3M United Kingdom plc, 3M House, PO Box 1, Bracknell, Berks. RG12 1JU.
Tel: 0344 858614.
Fax: 0344 858614.
USA: 3M Co. Pro Audio, 3M Center, Building 223-5N-01, St Paul, MN55144-1000. Tel: +1 612 733 3477. Fax: +1 612 737 5583.

Hanger box design for 3M's new shorter length DAT tapes

maximise the available surface area, which in turn benefits the acoustic absorption properties of the panels. The panels can be adhered directly to walls, freely suspended or, as in the case of AIR studios, laid horizontally over stretched retaining wires.

Melatech is also the base acoustic absorption material used in the *SoundCheck* stretched-fabric covered-wall lining system which is also marketed by the Noise Control Centre through their Architectural and Building Services Division in Berkshire, UK.
The Noise Control Centre, Charles House, Toutley Road, Wokingham, Berks. RG11 5QN. Tel: 0734 774212. Fax: 0734 774212. ■

Melatech calms the AIR

Melatech is a low density, semirigid acoustic foam formed from the same base materials as conventional kitchen *Melamine*.

The foam is heavily featured in AIR Lyndhurst's main hall (See 'AIR Man', *Studio Sound*, August 1993.) to control potentially troublesome reverberation.

The panels come in many surface designs with an emphasis on bold profiles to

***Melatech* foam—Class 0 fire resistant with excellent sound absorption properties**

Fostex' next generation: the _foundation 2000_

FOUNDATION 2000

If its intended significance is to be understood, it is important to give Fostex' new baby its proper title—the _Foundation 2000 Digital Recorder Editor Mixer_. The words have been chosen carefully, as have those describing the _Foundation_'s hardware controller, the _Edit Controller_, removable hard drive, the _Removable Project Environment_, and the proposed _Dancing Fader Mixing System_ hardware controller.

At the heart of the _Foundation 2000_ are a design team new to Fostex—having previously been part of New England Digital—and a dedicated computer-based recording, editing and mixing unit described by the team as an Expert Audio Computer. Design of the platform has incorporated a very high real-time performance requirement—the hardware architecture is entirely new and believed to be good for the next five years without modification. Regular 2-year updates to the object orientated software is expected to prolong the _Foundation_'s life for a further five years. The machine is intended to find application in audio-postproduction, audio for film and TV and, of course, in straight music recording, and the sheer scope of its conception reflects these ambitions, as we shall see.

Getting down to some specifics, then, the _Foundation 2000_ employs 16/24-bit processing and storage (16-bit implemented on first release). All three popular sampling rates are supported (32kHz, 44.1kHz, 48kHz) with ±12.5% varispeed on 44.1/48kHz; A–D conversion is 18-bit/64x oversampling; D–A conversion is 18-bit/8x oversampling; frequency response at 1kHz is 20Hz–20kHz ±0.5dB and dynamic range at 1kHz is 103dB. The machine will handle up to 16 channels of simultaneous audio providing hardware supported 8-channel working including simultaneous crossfading and DSP functions on all tracks. The intention here is that the system will support up to six units in cascade giving 48 direct output tracks and 96 audio channels. The system providing this interface takes the form of another proprietary development, the Shared Memory Interface, which the design team are describing as a 'Vulcan Mind Meld' between machines. The aim is one of as complete integration as possible.

The 'pooled' DSP capability of the _Foundation_ is one of the keys to its power. The system is, once again, modular, using what Fostex call ACE (Algorithmic Computing Engine) cards. These cards are actually Motorola-56000-based but are also intended to accommodate new devices (such as RISC-based chips) as they become available. ACEs take various forms are and fitted depending on the application of the _Foundation_. A Mix ACE comes as part of the basic package, and this provides 16 simultaneous channels' worth of 3-band parametric EQ as well as the power required to handle real-time mixing duties. Up to five further ACE cards can be fitted and further DSP tasks such as reverberation and comp-limiting are facilitated by downloading algorithms from the unit's onboard data card slot.

Due to its basis in software, the _Foundation_'s internal mixer structure is flexible but initial release of the unit offers an 8:2:2 structure with two auxiliary sends and four returns. All mixer functions may be automated, alternatively control may be from the _Edit Controller_ or by external controller.

The _Edit Controller_ represents a considerable investment in ergonomic study and forms the front panel of the _Foundation 2000_ and may be used attached to the unit or detached from it (as with the company's _R8_ multitrack). Attached, it sits below the display and card slots; detached, it reveals the ACE cards.

I-O configurations begin with 2-channel analogue input and 4-channel analogue output (on XLR), and extend up to 18 analogue input and 20 analogue output configurations. The basic 2-channel option also provides AES-EBU and SPDIF digital ports on XLR, RCA or EIAJ optical (TosLink). A multichannel analogue I-O option offers eight ins and outs. A future digital 8-channel option will give eight ins and outs on AES-EBU.

In keeping with its multirole aspirations, the _Foundation_ supports extensive sync and control options. Sony 9-pin protocol is supported and MIDI ports offer external mixer control and interfacing to MIDI Machine Control and MIDI Time Code; the system clock can be referenced to a variety of sources and the high-speed LTC-VITC time code reader will chase in both forward and reverse directions—audio output is maintained in reverse. Time code formats supported are 24, 25, 29.97, 30 nondrop and 30 drop frame.

In spite if its name, Fostex maintain that the _Removable Project Environment_ is not the product of ingenious marketing strategy, but is fundamental to the conception of the _Foundation_ itself. The _RPE_ uses a unique file format which is claimed to circumvent such problems as file fragmentation to allow not only comprehensive interchange of work projects between _Foundation_ stations, but also permits projects to be moved between systems of differing capability to enable work to continue with as great a flexibility as possible. Shipping as standard with initial units will be a Seagate _ST 3610N_ 540Mb drive (giving 90 minutes of mono storage) but it will be possible to substitute other drives—a 1.3Gb alternative giving almost twice the 540Mb capacity is already available. External hard and M-O drives are supported via a SCSI2 port as is background working for archiving to Exabyte or Wang DAT drives. ∎

Tim Goodyer

Francinstien

Coming from an outfit not famed for giving its products dull yet sensible names, the *Francinstien* (sic), complete with its Gothic script, almost challenges you to take it seriously. The name is not a misspelling, but a rather tortuous acronym for FRequency Adaptive Nonlinear Crosstalk Injection Network for STereo Image ENhancement—as I am sure you had already deduced.

The phrase 'Stereo Enhancement System' can be taken two ways, and in a sense both are true. It is stereo, and it does have familiar controls for tonal enhancement, but clearly its main objective is enhancement of the stereo image itself. The manual goes into some of the theory of hearing and stereo placement, pointing out that pan-potted stereo (and indeed coincident pair recordings) lack the arrival time differences which are the principal cues at low frequencies as to where a sound is coming from. This is regarded as a fundamental flaw in most stereo recordings which is crying out for the means to put it right, and the unit sets out to add the required timing information to the low frequencies in order to produce a more realistic spatial impression.

The principle used is acknowledged to be the EMI 'Shuffle' circuit developed, after his death during World War II, by Alan Blumlein's team. It is claimed that the technology of the 1950s was not up to the job of rectifying the perceived stereo problems without unacceptable distortion and tonal colouring, but that *Francinstien* has cracked it. The manual proceeds to go into detail about filters in the difference circuits which sadly fall down because none of the variables in the equations are explained, leaving the section looking very much like an attempt to blind us with science. This impression is not helped by some rather suspicious waffling jargon elsewhere in the manual, which veers so violently between whimsy and hard technical exposition as to leave one feeling a little seasick.

None of which is as important, of course, as the results the unit produces. I have never been a great fan of gadgets which claim to artificially widen or enhance a stereo image, some of which I find almost painful to listen to, so I confess to having approached *Francinstien* with trepidation; to my surprise, I liked what I heard. Whether it is true to say that the image 'broadens and clarifies' as the Space control is advanced is debatable, but at modest settings the feeling of space and separation was noticeably increased without any disturbing or inappropriate side effects. Taking it further did, as promised, create a super-wide image with sounds apparently beyond the speakers and added distance to the central sounds, producing what the manual describes as concave distortion of the stereo image; personally the out-of-phase character of this exaggerated version of the effect made me feel uncomfortable, but some may find it exciting. The claimed mono compatibility is easily proved—with the monitors in mono the Space control has no effect whatsoever, however extreme its setting.

Added to this primary function are three tonal sections, although it is not entirely clear whether they are simple equalisers or something a little more elaborate. A Depth control appears to be straight EQ, with a WARM-DRY switch to select shelf or bell characteristics, and has an extremely musical response, eminently useable even without the stereo enhancement. A Harmonics control may or may not be a simple HF equaliser; it offers more HF boost than one is likely to want, but in smaller amounts is subtle and useful. Subtler still is the Mid-Hi Tune and Lift section, supposedly fitted to compensate for the 'concave' effect of the stereo widening by bringing required elements further forward in the mix. Compared with the other controls, this section in fact appeared to do very little.

Francinstien is available in two versions, one solid state and one with valves (tubes), although it is not clear which particular job the valves take over. The review sample was in fact fitted with the optional Groove Tubes matched valves. Not having heard the other versions, I cannot say how much of an improvement this brings, but certainly the sound was clean, musical and (with the controls flat) accurate. Since the unit by definition only works in stereo, never as two separate channels, precise matching of the circuitry is particularly important, in terms of not only the active components but the ganging of the front panel controls. I detected no problems in this respect, either by ear or with the help of suitable metering.

The front panel is simply and logically presented, and refreshingly free of the kind of pretension which occasionally infects boxes of this nature. The temptation to sprinkle knobs, switches and indicators liberally over the unit has been resisted, to the extent that the LEDs are bicolour varieties, lighting red or green to show the selected status, halving the number required. The rear carries audio in and outs on electronically balanced XLRs or unbalanced jacks. It goes without saying, of course, that this is not a send-return type of treatment but one through which an entire signal must be passed, although that signal could be a complete stereo mix or a stereo component within it.

The success of *Francinstien* is going to depend purely and simply on whether or not people like what it does, which apparently many already do. It sets out to do a job which not everybody would regard as needing to be done, and it might be safer to describe it as a creative effect rather than as a corrective treatment. Like many enhancement devices, it is easy to go over the top with it and produce unpleasant results, but it is also easy to be more discrete with it and do some quite subtle, pleasing things. I think it is fair to say that this is the first device I have encountered with the express purpose of tinkering with the stereo image that I would be happy to use. ■

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Francinstien—set for a monster success?

DMC 1000

Yamaha stress that the *DMC 1000 Stereo* is not a 'Mk II'

Since its introduction just over two years ago, Yamaha's *DMC1000* digital console has enjoyed considerable success, earning a reputation as a quality, highly featured yet affordable product suited to a broad cross-section of applications. The versatile nature of the console has been proven by its adoption into a wide range of sectors from postproduction, where it is being used in conjunction with all the major hard disk editing systems, to classical music recording—notably with Deutsche Grammophon who use the *DMC1000* as a key component in their 4D recording system and currently own in excess of 50.

This versatility has now been further extended with the introduction of the *DMC1000 Stereo*; a new version of the console which has come about in direct response to user feedback, in particular that generated at a seminar hosted by HHB Communications. Martin O'Donnell, HHB's Technical Sales Engineer, explains.

'We held the seminar about six months after the initial launch of the *DMC1000* and invited people from a total cross-section of areas—video, film, broadcast, and music recording. One of the demonstrations showed the console operating in video suite mode via the RSAM 2 edit suite protocol, whereby both picture and sound were edited by a hard disk system. After this we had a questions and answers session, and a significant point that emerged was the way in which the desk worked in stereo. Although the *DMC1000* has a Stereo Construction mode whereby two channels can operate as a stereo pair with linked functions, it was felt that for VT and postproduction applications, where so much of the work is now in stereo, that single fader stereo channels were required.'

HHB put together a proposal, along with interested parties including the BBC, outlining these stereo requirements. It was decided that no hardware changes should be made as this would result in a major rework of the desk; instead a new layout was submitted that utilised existing controls. Yamaha have now accommodated these requirements by releasing a specific and extensive software revision that restructures the console, offering four main features: individual channel configuration to either a stereo input path or two independent mono paths; simultaneous access to the routing matrix from all 22 inputs; MS decoding for any of the eight main stereo input channels; and full update of automation to cover all new features including MS decoding.

The *DMC1000 Stereo* is available either as a physical console, or alternatively existing *DMC1000*s may be converted by installing the new software and relabelling controls —a job that takes around an hour and costs around £2,000 (UK price).

Yamaha stress that the *DMC1000 Stereo* is not a 'Mk II' successor to the original console; rather it should be viewed as is an alternative version suited to applications where dedicated stereo channels are seen as a beneficial feature.

The only physical change to the control surface is its labelling, and one of the first things that is noticeable is the absence of any reference to Input and Monitor sections, these are now referred to as Left and Right inputs. This along with the relegation of the BUS TO MONITOR switch from the channel strip to an LCD menu gives a fair clue to the intended operation of the console.

Each channel can be configured either in stereo (the default state), or dual mono by selecting the MONO button (formerly the BUS-TAPE button just mentioned). In Stereo mode both channels are controlled by the fader while the rotary encoder, previously used for Monitor Level, acts as a Balance control. In Mono mode the two inputs are split so that the odd-numbered channel (Left) is on the fader while the even-numbered channel (Right) appears on the rotary control—this arrangement may be reversed using the Flip facility.

This control duality also applies to the Pan function which either operates in the conventional manner for selected mono inputs, or as a Width control in Stereo mode. The horizontal bargraph display shows the width of the signal, and if left and right channels are crossed-over, the LEDs become hard-lit to indicate stereo reversal.

Most channel functions are linked in Stereo mode; thus functions such as EQ, pre-post Aux switching, channel delays and so on will affect left and right channels equally. However ROUTING and PHASE functions, although normally stereo linked, can operate independently. For example, if independent left-right routing is required, the channel may be temporarily split into mono to allow individual assignments to be made which will then be retained once the channel is switched back.

The original INPUT and MONITOR path assignment buttons on the Routing Matrix, Aux sends, EQ, and Pan sections have been relabelled to distinguish left-right selection during split mono operation. Left and right channels can independently and simultaneously access the eight program buses, the stereo mix bus and the three auxiliary buses. Auxiliary 3 remains a stereo send, as with the previous console, and in stereo mode independent control is provided over width; however the send does not follow the channel Balance control which seems a small oversight.

Two other small changes have been made (mainly as a result of requests from the BBC) to Master Solo selection and to Control Room Monitoring selection. There is now the option to choose between two default solo configurations: the AFL switch in the Master Solo section will either switch globally between AFL and Solo-In-Place (as with the original software), or it will switch AFL-PFL. This does not affect the availability of the console's other solo modes such as Pre Switch Listen.

The control room monitoring change involves the switching speed between the main stereo bus and an external stereo input. The feeling here is that the switching was too slow during A-B monitoring, in particular when comparing off-line and on-line transmission signals during live broadcast applications, and consequently the interpolation time has been reduced from 46.8ms to just under 3ms.

Console metering has also changed so that instead of splitting Inputs and Monitors into separate banks of eight meters, the eight stereo inputs are now arranges across the 16 bargraphs in pairs. All previous metering facilities, including bus metering, remain unchanged.

The MS decoding facility is available for any of the eight main stereo input channels. The MS microphone mid and side channels are connected to the console's odd and even channels respectively. Left and right signals are then created using the following equation: $L = M+w*S$, $R = M-w*S$ where the constant, w, represents a variable percentage (-100% to +100%).

The percentage of S content in the equation is controlled via the console's data wheel and displayed in the LCD. The process can be automated to capture dynamic stereo effects, or to synchronise static settings to scene changes.

The *DMC1000 Stereo* was officially released last month (November) and according to Yamaha considerable interest has already been shown, predominantly in the postproduction and broadcast areas where a number of users have elected to convert their existing consoles. ∎

Patrick Stapley

Switchblade

Building on the success of the *MC8* switching matrix, Sound Sculpture have released the *Switchblade* matrix switching system. The unit doubles the number of inputs and outputs which can be connected to the unit (to 16 each) and offers eight programmable relays through four stereo jacks which can mimic the action of push-on/push-off or momentary push-on/release-off footswitches for such things as guitar amp reverb switching.

While the unit will immediately appeal to guitarists as a sharp way of reconfiguring a rig, its uses extend well into the realms of studio automated routing and expanding desk aux sends, for example, and any other application that requires fast and efficient rerouting and mixing of numerous inputs and outputs.

Cosmetic and operational tweaks to the *Switchblade* over the *MC8* include a larger better-lit LCD, 75 programmable presets and a wider programmable gain range plus MIDI continuous controller influence on gain. Apart from more than twice the number of unbalanced standard jack sockets on the back of the 1U-high unit, not a lot else has changed externally. A MODE button on the front panel steps through menus, Select moves a cursor (unfortunately still only in one direction) around the screens, and UP-DOWN increment buttons alter values.

The MIDI In port permits the unit to be driven by a MIDI pedal board while a MIDI Out allows daisy chaining to effects units, set to receive on different channels, for patch change information which can be programmed into a *Switchblade* preset along with appropriate routing and gain data.

The 75 presets can be arranged into 20 banks of 10 for performance purposes and their selection can also be performed from a momentary mechanical footswitch although this seems like a crude way of controlling such a sophisticated unit.

Setup

This area of the unit concerns itself with the creation of names for all devices connected to the in and outs —something that was noticeably missing on the original *MC8*. It is here that you also set MIDI channels where applicable and the receiving channel of the *Switchblade*.

This really must be done prior to programming presets as this determines what the unit will present in its menus, for example, and once it is done connections will become clearer as the names of the units involved will be displayed.

While this is a time consuming process, especially if you can fill up 16 ins and outs, hindered by an erratic cursor it pays to be accurate and descriptive as the more information you put in the easier life will be afterwards in programming. It also means that once you have effectively customised the *Switchblade* to your rig and connections you can forget all about the physical side of the unit and get onto the creative side of patching and mixing. It is here that you also enter the crossfade time between presets, variable from 0–1000ms, and set MIDI channels.

For later reference the Set up menu is also the location for a nice simple preset Copy page and the area in which the entire memory contents can be dumped or loaded externally.

Programming

From an operational point of view it is worth grasping the principle that in *Switchblade* everything is internally routed to everything else but what actually activates a route is the programmable gain level. The basic state is no gain, or off, and the connection is made by turning the gain on. Any programming is performed on the current preset in memory, there is no buffer in which you can experiment and then decide to Save somewhere else.

The unit passes audio while being programmed allowing complex strings and parallel combinations of effects to be heard as they are created. The display presents you with a named source input and a named destination output with a variable input gain setting for the transactions thus it goes further than simple routing as signals can be mixed together by input gain. You can rest assured that any noise build up that results from this is derived from the connected devices and not the *Switchblade* which is about as silent as they come. Intelligent use of gain at different stages of a configuration can help to improve matters a lot.

Dealing in the single connections that the LCD shows you can become a little mind-boggling if you are determined to employ all 32 jacks worth of connections in one mega patch. You really have to be alert and on top of what you are doing or risk sending the wrong things to the wrong destinations without realising it immediately. It is something of a knack if you are into daisy chaining your entire rig as is going back and unpicking connections.

Patch change commands for connected processors are also programmed while editing.

In use

One trick possible on the *Switchblade* is to employ MIDI continuous controller data to adjust gain settings dynamically. In its most basic form this can functions as a type of swell pedal operating between user-defined start and end gain values. A more advanced application is to use the same controller to influence two gains in two different connections with one of them inverted relative to the other. Continuous controller data will then effectively balance between the two signal paths permitting, for example, a clean guitar tone to an amp to be suitably melded with a chorus tone on the pedal.

This may seem to take *Switchblade* into multieffects territory but its approach has the benefit of being significantly simpler to program in this respect than your average effects unit is.

Other tricks possible include the ability to set the change between the two aforementioned signal paths to be swept automatically on an internal LFO, variable from 250ms/cycle to 10s/cycle, or a multipan which uses four LFOs working relative to each other but at different speeds for more complicated arrangements. While these permit intricate autopans to be realised they have got to be of relatively limited use in comparison to the continuous controller mode. However, all are zipperless.

On the downside the rackmount holes are too small to take M6 screws but I was looking at an early model. The front panel LEDs which indicate the unit's mode while bright are recessed and very directional so can be viewed clearly only from directly in front of the device in bright light on a sunny day facing south.

Conclusion

To all intents and purposes, once a *Switchblade* is plugged up with its connections named and MIDI channels selected the fiddly stuff is done. Operationally it is worth bearing in mind the alternative —repatching everything physically and resetting levels? I don't think so.

It is a very good way of experimenting with a rig and adding value to it by trying different combinations of effects in a manner that would verge on the tedious and unrewarding if you had to rewire just to try something out for fun. In this light it is positively fast.

It is a shame some of the connectors are not balanced but bearing in mind the variety of sources that can be plugged in over the -42 to +6dB of gain available—but what the hell!

Switchblade has dozens of applications away from its core target of guitar rigs for which it is ideally suited. It is not cheap but combined with just about any basic MIDI pedal board on the market you are still talking about substantially less than half of what you would gladly pay for a name custom job. Add a really clever pedal board you are really cooking with gas. ■

Music News is compiled by Zenon Schoepe

Switchblade—'cooking with gas'

www.americanradiohistory.com

AES NEW YORK

Zenon Scheope and Terry Nelson report from the 95th AES show which displayed a genuinely positive attitude to pro audio and a healthy crop of products.

October's AES Convention again set new records as a pro audio event but also achieved the often quoted goal of getting the entire exhibition, barring a couple of demo rooms, in one hall and on one floor. It was a popular show for visitors and exhibitors alike in a superb venue but the preponderance of computer-based recording systems could not be ignored.

The launch of the *Foundation 2000* recorder-editor-mixer by Fostex typified the product direction with many of the developments in software enhancement and hardware add-ons. There was also a mood of alliance in the air.

Hardware

Studer Editech and **Avid Technology** demonstrated the first exchange of Open Media Framework (OMF) format files between digital audio workstations and were swarmed because of it. Audio media data and sequences were transferred between *AudioVision* and *Dyaxis II* on optical disc in what was termed an audio milestone. Both manufacturers are now shipping product which support OMF Interchange.

Studer introduced the sharp-looking MultiDesk assignable control surface for the *Dyaxis II* incorporating moving faders, a scrub wheel, rotary knobs for EQ and pan, transport buttons and dedicated function keys. There was also the *D827 MCH* DASH machine heralded

as third-generation (there was an 8-track digital prior to the *D820*)which comes with MADI as standard and is field upgradable from 24 to 48 tracks with a head block change and added cards. Its lighter, smaller and faster than its predecessor and weighs in at around £83,000 for a 48-track with a simple remote and without A–D convertors ready for connection to a suitably equipped and complementary digital desk. The upgradable 24-track version will ask about £55,000.

Gone are the flashes of Swiss chalets and real wood end-cheeks replaced by a more forceful implementation of the pale grey and burgundy colour scheme introduced at the Berlin AES. Studer were presenting themselves as a modern and progressive company.

Avid Technology added Release Version 2.5 to *AudioVision* which now offers time shift, pitch shift, waveform display enhancements, on-the-fly and automated punch-in recording, better machine control, multichannel off-speed playback during scrub and shuttle, clip cut within editing, blip audio scrubbing for pin-pointing audio cues, seamless loop playback from in-to-out marks, improved Avid video support and, of course, OMF Interchange.

AudioStation was introduced as an entry-level system with a price tag of around $30,000 and designed as a prep station and for dialogue editing in particular. It is significant as an Avid product in not including digital picture facilities, but is upgradable to *AudioVision* status. Features and options include master-slave machine control, EDL import and autoconform, time compression-expansion, pitch shift,

an ADR-Loop record mode, and a range of storage options.

Digidesign were spilling out into the aisles but this was after all the first public showing of its *Trans-system Digital Matrix* (TDM) bus and all this after the introduction of *PostView* random-access video at the APRS exhibition. The enhancements through the TDM Bus to the company's *Pro Tools* workstation include automated digital routing and patching, new digital mixing options, an open system for DSP hardware and software options from Digidesign and their development partners and the ability to route and automate external equipment within the TDM environment.

The *Miles* hardware control surface for *Pro Tools*, shown glowing gently in a glass case and expected early next year, is linked intimately to these goings on. The work surface uses motorised faders, a touchscreen, 'soft' pots and transport control, autolocate and dedicated control buttons. Its touchscreen can display, change or recall signal routing and default and custom configurations can be stored and recalled. Its compatibility with the *TDM Bus* makes it a potential controller for other equipment as well.

And there was more, *MasterList CD* software allows the creation of full Red Book glass-master ready CDs with PQ encoding on a selection of Digidesign systems and certain CD-R machines.

Upgrades to **DAR**'s *Sigma*, *Delta* and *Sabre* workstations included foot-frame display, film transport follow, instant offset, a laser disc interface for the likes of the Pioneer unit, automatic Time Code standard switching, reel split and merge, and instant remote machine ▶

Sabine *ADF2400* workstation

autolocate and play. Hard disk-based *Sigmas* and *Deltas* are now offered with 16 or 32 track-hours, all systems are available with M-O drive and *Sabres* can be fitted with two 1.2Gb M-O drives.

Version 5 of **SSL**'s *ScreenSound* added a faster processor with enhanced editing options, better graphics and more machine control. Most notably reconforming is now supported along with *Scenaria*'s VisionTrack random access video option for 525 or 625-line operation. With its own 2-channel audio recorder, VisionTrack can be used to load sound, in addition to the picture, off-line for importing into *ScreenSound*.

Sonic Solutions added a DiscVideo option to its CD premastering system for the inclusion of video which will adhere to the Karaoke CD, CD-I/FMV and CD-DV formats.

Version 2.0 for **Roland**'s *DM80* added more than 40 new or improved features including an audio profile display on the *DM80R* remote, group move, enhanced backup, recovery of unused memory, fader grouping, mixer snapshots and threshold editing. The ability to control four *DM80s* for 32-track recording will be included in Version 2.0 for the *DM80S* Multitrack Manager System software. Rewritable magneto-optical drives are available for the *DM80* each providing 4-track recording-editing in 650Mb and 1.3Gb capacities.

Creation Technologies' long awaited *RADAR* hard-disk-based 24-track stand-alone recorder had another outing looking like a good absolute replacement for multitrack tape at $18,500 for a 20-minute version with syncability and an AES-EBU. Backup is to Exabyte or DAT. *RADAR* should ship in March 1994 with a small remote, the larger *RE8* remote being required for comprehensive editing.

With the *StellaDAT* back in action under the

auspices of its new governors **Sonosax**, DAT specialist **HHB Communications** pulled the *Portadat* out of the hat after a two-year R&D programme. Aimed at location recording using a 'ruggedised' 4-head, 4-motor transport derived from DDS technology, two units (the *PDR1000* and the *PDR1000TC*) are available with balanced inputs, phantom power and digital I-Os. The *PDR1000TC* differentiates itself from the cheaper unit only in its time code abilities which equip it to jam sync, convert A time to time code and record, generate, and reference to all standards. It is neat, incredibly well laid out and presented and even resorts to the use of Nickel Metal Hydride rechargeable batteries to prolong its operating life away from a mains plug. Prices are £2,895 and £4,395.

Demeter announced the *VTCL-2* stereo or dual-channel, all-valve compressor-limiter with variable attack, release, sensitivity and input and output gain. VU metering sources are selectable and the unit has a useful 10dB boost setting for metering -10dB levels.

Focusrite revealed the *Red 3* dual-stereo compressor-limiter with transformer coupled I-Os and the same VCA used in the *ISA130* and *131*. Attack and release are continuously variable along with gain make up and threshold. Compression ratio is switched as is the limiter threshold.

Sabine, manufacturers of the *FBX* feedback exterminator, released the *ADF1200/2400* digital equalisers starting for around $2,700. Available in single-channel or dual-stereo versions each channel features 12 independent adaptive digital filters for 12-band parametric EQ, automatic feedback elimination, shelving filters, programmable delay, a programmable noise gate and real-time analysis.

Yamaha made additions to their product range with the *YDG2030* digital 2-channel, 30-band

graphic EQ with four notch filters, high-pass and low-pass filters and the *YDP2006* digital parametric EQ offering 6-bands, four notches and high-pass and low-pass filters in 2-channel mode and twice that in mono. Convertors are 20-bit with 40 memory locations and the 1H-high devices can be integrated into computer controlled systems with devices like the *DEQ5*.

Urei revealed a modular line of signal processors called the *Platform Series*. Central to the operation of the 3U-high rackmount, which can hold 11 modules, is a computer module which allows 100 memory settings of primary functions to be recalled. Five module types in addition to the computer interface are initially available: gate, compressor-expander, parametric EQ, and input and outputs.

An enormous amount of interest and disbelief was generated by the appearance of the **Tactile Technology** *M4000* digitally-controlled analogue console which will be offering a snapshot automated 24-channel, 24-monitor, 8-bus desk for around $30,000 at the beginning of next year. Using the now recognised approach of digital control surface linked to an analogue rack the board assigns channels to a master control strip for getting to the 3-band EQ (swept HF, parametric mid and swept LF), six auxes and the routing and unusually also has one stereo digital input channel. Five analogue racks can be controlled from one surface. There is a 32:24 configuration, the *M6000*, on its way with 3-band parametric EQ, 12 auxes, four stereo digital inputs and optional dynamics modules for around $100,000.

Taking the themes established by its *MPC* and *Series 10* digitally-controlled analogue consoles and putting a curve on them, **Harrison** came up with the *Series 12* using a mute control surface that is

IF SILENCE IS GOLDEN, THIS CO
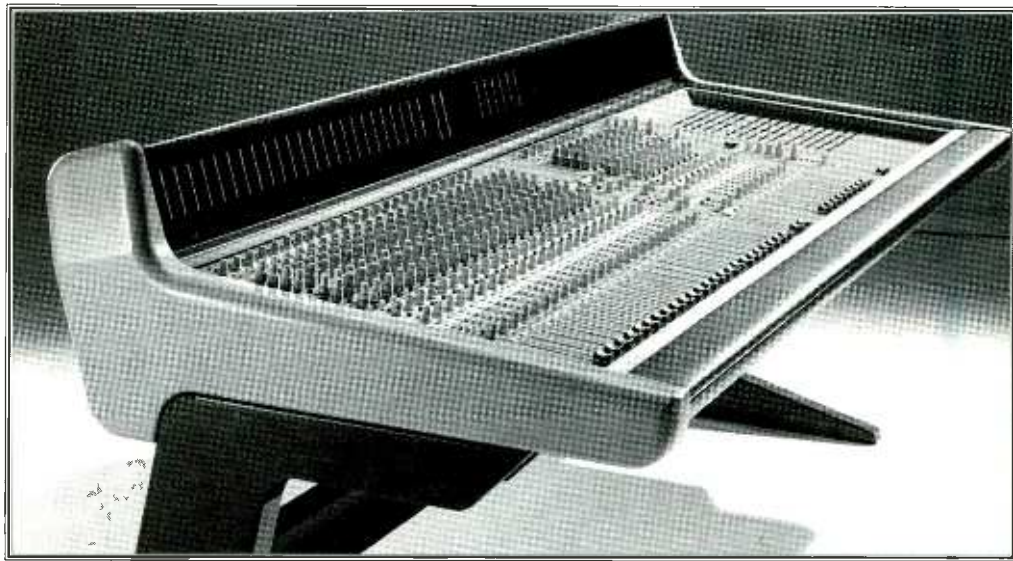
The D&R Orion. From its Hi-Def™ EQs to its fully modular design, from its custom-welded RFI-killing steel frame to its incredibly flexible floating subgroups, the handcrafted Orion is every bit a D&R.

considerably more approachable than either of its predecessors and an analogue audio rack that can be replaced or supplemented by an all-digital rack at a later date. The trademark of total dynamic control is preserved in a dual channel architecture that features mono or stereo operation, 48-track routing, four stereo buses, 16 auxes, two 4-band parametric EQs, two high-pass and low-pass filters, two gates and compressors and motorised long and short faders.

CM Automation showed an affordable VCA mix system running *Mac* software ($249), *MX816* VCA hardware and connector MIDI controllable rackmount ($799) and an 8-fader and mutes *FX100* automation controller ($499). The controller is ingenious in using twin-colour LEDs for nulling purposes. Of more general appeal was the *PM216* ($799) automated audio patchbay with 16 inputs and outputs bused in blocks of eight, and controlled by a 64-patch MIDI controllable memory.

On the problem-solving front, **Tektronix** showed prototypes of products that will be released next year. The *AM700* audio measurement set is a portable test and measurement instrument with analogue and digital generators and acquisition units, internal CPU and DSP, VGA display and storage. There was also the *Pathfinder AM70* hand-held digital audio analyser and analogue and digital audio generator and the *ASW100* series audio input and output multiplexers.

Digital Domain extended their product range with the *VSP* digital audio control centre which uses a crystal oscillator device to clock digital audio signals and thus virtually eliminate clock jitter. A sample-rate convertor is included and its 20-bit compatible format convertor accepts any conventional digital source to output any standard signal format.

New company **Z-Systems** showed a digital sample-rate convertor called the *Z-1SRC* which is input and output selectable between XLR, optical and coaxial and supports AES-EBU, IEC958, SPDIF, EIAJ CP-340. Features include autodetection of input sample frequency and six set output sample rates plus fully arbitrary conversion and external synchronisation.

Sony's MiniDisc raised its head in the pro environment. **Denon** introduced the *MD Cart* recorder (*DN990R*) and player (*DN980F*) as a replacement for traditional carts. **Otari**'s *Pro-MD* is aimed at broadcast storage and playback with all the speed advantages of the format plus TOC editing, memory start, stop-standby and single-repeat play function modes, a front panel headphone output, level meters and enhanced read-out of time, track number, title.

For the *ProDisk 464* came the *SoundShare* removable media system which combines systems configured with removable hard disks and magneto- opticals with Otari's off-line *BackUp Station* for fast transfer of files and projects between studios, background restore or backup and the storage of projects on removable hard drives, magneto-optical or 8mm tape.

Alliances

Arboretum Systems revealed their *Hyperprism* real-time audio signal processing program for the *Mac* and **Digidesign Systems** allowing nondestructive dynamic effects to be applied on-the-fly to *Sound Designer II*, AIFF sound files or real-time audio routed through the sound-card digital or analogue inputs.

Avid Technology and **Digidesign** announced plans to strengthen their links with **Digidesign** planning OMF support and **Avid** intending to incorporate the *TDM Bus* in its digital audio and video-editing workstations.

Pro Spatializer announced the development of a plug in option for **Digidesign**'s *Pro Tools* system to enable real-time spatial expansion and dynamic sound movement and localisation in 3-D which will work with the *TDM* digital audio bus.

Sonic Solutions, meanwhile, incorporated Super Bit Mapping into their products while **Avid Technology** agreed to incorporate **Sonic Solutions**' *NoNoise* audio restoration system into its *AudioVision* and *AudioStation* systems. While this is going on **Sonic Solutions** will port its *Mac*-based Sonic System to **Silicon Graphics**' *Indigo* line of workstations.

Sonic Solutions will be offering *Pro Spatialiser 3D* audio processing as a sonic effects option on the Sonic System.

Esoterics

Burgess Macneal premiered his latest addition to the legendary *Sontec* range in the *MPA1* 2-channel mic preamp, now in its third generation of improvement according to the man. It features an ultra-clean signal path (discrete high slew-rate, low-noise amps with no electrolytic capacitors or transformers in the signal path), a choice of battery or AC power supplies, level indicators and selectable EQ to plug-in EQ modules. Each channel has a 12-position rotary gain control, a polarity reverse switch and phantom power. Using a minimalist circuit design with specially selected components Macneal said the unit is extremely close to 'the mythical straight wire with gain'. ▶

Similar qualities were being claimed for **Mytek's** *Hi-8* line-level recording console for multiple high quality mic preamp mixing. With 16 inputs, 8 groups, 4 auxes and 4 stereo aux returns *Hi-8*s can be daisy-chained for more inputs if required.

ADgear introduced the *Over Quality OQM 8100* desk on the **George Massenburg Labs** booth with **ADgear** *DOP210A* discrete op-amps used throughout, no mic inputs and operating at line level. Remote *KZ-902U* mic head-amps complete the picture for a desk in which all internal signals are summed in balanced active mixes. $780,000 buys an 80-channel desk with **GML** automation, two remote head-amps, one multitrack remote interface, two stereo patchable faders, LCD monitor display, EIA 19-inch rack for the power supply and automation, and a full bantam patch bay with cables. (Sounds like a bargain to me.) ∎
Zenon Schoepe

Live Sound

The AES highlighted the important place in the audio industry that is held by sound reinforcement and live performance systems and the exhibition saw the launching of a wide range of new products.

Loudspeaker systems in evidence included products from **TOA**, **EAW**, **JBL**, **Community**, **Yamaha**, **Pioneer**, **Turbosound**, **Weisberg Audio** and **Bag End**.

TOA launched the *SR-FO5-MO5L (R)* 2-way processor-controlled systems for house and monitor use respectively. The enclosures employ a 10-inch (25cm) bass-mid driver 1.75-inch (45mm) titanium compression driver coupled to a 90 x 40 CD horn for the HF. Each speaker has its own dedicated controller with the necessary EQ corrections and a separate subwoofer output with limiter.

EAW announced a 'strategic partnership' with **Siemens** Austria for the contracting market in Central Europe, together with an Engineering White Paper to back up the launch of the *AS943* high-output, long-throw system and the *KF852-BH852 Stadium-Array VA* systems. The systems feature greater output together with smoother response and precise pattern control. The *852* enclosures match the *KF850* dimensions for greater compatibility.

JBL released new additions to the *Array Series* of loudspeaker enclosures as well as to the *Architectural Series* and *SR Series II*, the latter featuring new large format driver technology. Of particular interest to those interested in true modular systems is the introduction of the *2350 Series* CD horns, together with the *2337* and *2451* 1.5-inch (38mm) compression drivers, and the *2490* 3-inch (76mm) midrange compression driver and *2394-2393* biradial horns. The new large format horns are claimed to provide smooth response over a bandwidth of 300–4000Hz.

'You've seen the film, now hear the monster!' This could almost apply to the **Community** *Leviathan*, possibly the ultimate full range enclosure-horn assembly. Designed for very large installations, the *Leviathan II* is a point-source system that features a 34Hz horn flare powered by six 15-inch (38cm) drivers, a Pattern Control mid horn with *M4* driver and twin HF horn PCD Focused Array with 2.8-inch drivers.

Yamaha joined the list of manufacturers who are discovering, or rediscovering, the spherical horn with their new *WaveForce* series of loudspeakers which features models for small and large venues, a dedicated floor monitor and a subwoofer. Special features include an integral design for the HF compression driver and horn assembles, or wave guides, and the elimination of sharp edges and discontinuities at the subwoofer and waveguide boundaries.

The **Pioneer** *Pro* series derives from the larger sound reinforcement range and consists of the *S-V7000* full-range enclosure housing four 6.3-inch (16cm) woofers and six 3.06-inch (7.7cm) cone tweeters plus the *S-V5500W* subwoofer which has the same dimensions and houses two 10-inch (25cm) woofers in an opaxial (opposite axis) configuration. Applications include discotheques and 'entertainment' areas.

The result of three years intensive research and development, the **Turbosound** *Floodlight* high power sound reinforcement system features new generation **Turbosound** horn-loading technology designed to provide exceptionally smooth voice and music reproduction for touring and fixed installation applications. Developed as a wider dispersion extension of the *Flashlight* system, the *Floodlight* features a 4-way active design with a constant-coverage polar response and a seamless transition from enclosure to enclosure, together with a dedicated controller, the *LMS-660* dedicated management system.

The compact *TFL-760H* mid-hi enclosure utilises a high power 12-inch low mid driver on the new 'Axehead' device operating in the 180–1300Hz range, a 6.5-inch paper cone high mid driver mounted on a special 'Axehead' device for the 1300–8000Hz band, and a 1-inch titanium compression driver on a proprietary waveguide horn. All proprietary transducers are mounted vertically in-line and are physically time-aligned in the enclosure, eliminating the need for external time correction. The *TFL-760H* is available in both rectangular and trapezoidal enclosure formats.

Weisberg Audio design speaker systems for most sound reinforcement applications and have launched the new *Arena* series 3-way loudspeakers, designed for both single enclosure and array installations. The *Arena 1* is completely full range and features a very high SPL output. Control is via DSP processor and separate subwoofers are available should they be deemed necessary. Also from **Weisberg** is the *Drum One* low profile drum wedge which again features very high power handling and output.

Bag End Loudspeakers have made a new addition to the *ELF* range of compact processor-controlled subwoofers, the *S10E-B*, which claims a flat frequency response down to 8Hz (1) with a 10-inch driver.

Controlled units

The **Sytek** *A-SYS MPX-4D* digitally controlled 4-channel microphone preamplifier. Each input stage uses a Class A, Auto-Bias bridge configuration for lowest noise and optimum impulse response and extremely low phase distortion. Gain may be remotely adjusted via a digital control amplifier run on a PC host computer using *Windows* 3.1. Future developments will support *Macintosh* and MIDI via a free (upgradable) program on floppy disc.

Yamaha released the *YDG2030* digital graphic EQ and the *YDP2006* digital parametric equaliser.

The graphic is a 2-channel 30-band EQ with four notch filters and high-pass and low-pass filters. The parametric may be used as either a dual channel or mono unit and, depending on the mode, features 6 or 12 parametric bands, 4 or 8 notch filters and high-pass and low-pass filters. Common features for the two units include balanced inputs and outputs, 20-bit convertors, 44.1kHz sampling frequency, variable output delay up to 730ms, large LCD and three dedicated rotary controls for frequency, gain and Q. Up to 40 user-programs can be stored and recalled using either MIDI or a computer (via Y-485 connection).

Ramsa introduced the *WZ-DE40* digital multiequaliser. The unit is two channel and operates in a variety of modes, namely: automatic notch filter, 27-54 band sixth-octave graphic, 8-band parametric, limiter-compressor and spectrum analyser. Twenty-bit A–D/D–A convertors are used and the equaliser can be controlled via MIDI or D-SUB 25-pin port.

Sabine showed the prerelease model of the *FBX1800 Feedback Controller*. Essentially a 2-channel version of the *FBX900*, the unit also incorporates some new developments, such as being able to user-select the number of frequencies for notching. Watch this space.

Processing

The *EQ2* from **Night Technology International** is a 5-band (fixed frequencies) equaliser module for use on final programme sends or critical channels. One of the major characteristics of the unit is extremely low phase-shift and extraordinary definition in the high frequencies, coupled with low noise and very high transient response. These comments are based on actual listening tests and though the unit is not cheap, it merits attention.

White Instruments have introduced the *Model 4856* dual third-octave equaliser that fits into a 3U-high rack chassis. Features include 28 filters with ±15dB range, variable high-pass filter from 15–175Hz, separate sub-bass output with 12dB rolloff at 100Hz, servo-balanced inputs and outputs and six-LED headroom metering.

Computer control

The buzz word here is *MediaLink* from **Lone Wolf** and there is already an impressive list of signatories to the system. Time will tell how this works out in practice but it is already an encouraging start.

One of the latest systems using *MediaLink* and the VNOS software Interface is QS Control from **QSC** for the *EX* series of power amplifiers. The system allows elaborate control over large installations, together with status reports on systems and individual units.

The latest from **Yamaha** in computer control is the *IFU485 Interface Unit* which is a bidirectional RS-232C-RS422 to Y-485 protocol convertor. The **Yamaha** *RCX2* digital equaliser software package can be used with either *MS-Windows* or *Macintosh* to control the *DEQ5, DEQ5E, YDG2030* and *YDP2006*.

Klotz Digital have introduced the *VADIS* (Variable Audio Distribution and Interface System) which is a more economic development of the *Oak Link* system. *VADIS* has a wide variety of applications in different types of audio systems and for sound reinforcement finds uses in medium to large fixed installations and touring systems. Control is either by PC or *Macintosh* computers.

The technology available to live sound is increasing by leaps and bounds and so much the better. However, we do still need to stand back at times and ask the question: 'Does it really *sound* better?' ∎
Terry Nelson

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SWISS RÔLE

A ny commercial recording studio which has successfully been in business for over ten years could be forgiven for resting on its laurels, especially when all the equipment is paid for and the studio is so busy it doesn't even need marketing.

But when Master Studios in Switzerland found itself in this enviable position, studio owner Victor Waldburger decided things had become a little too dull for his taste. So he ripped out both of the control rooms in Studios A and B, called in top studio designer Andy Munro to redesign them, put DynaudioAcoustics monitors in both rooms and became the first studio in Europe to install one of the new breed of Solid State Logic consoles —the *G Plus*.

The new look, two-studio residential facility is based in St Gallen—a 40-minute train journey from Zurich airport. It is now operational again and is already pulling in clients from Switzerland, Germany, Austria and the UK who are eager to try out the SSL console for mixing projects. Waldburger is

View over the Säntis mountains, with their large skiing area up to 2,500m, and only half an hour away from the studio

delighted with the response he has achieved so far, but regards this as 'just the beginning', and is actively marketing the studio complex as a truly international facility suitable for top bands and top record producers.

Waldburger explains: 'I needed a challenge and so did the studio. I really had reached the point where I could have left the whole place empty for five years without suffering financially. Everything was jogging along quite smoothly, everything was paid for and it all worked without any effort. We were getting plenty of music recording, radio and TV postproduction to keep both rooms busy but I felt the potential was there to do much more, and that was the challenge I was looking for'.
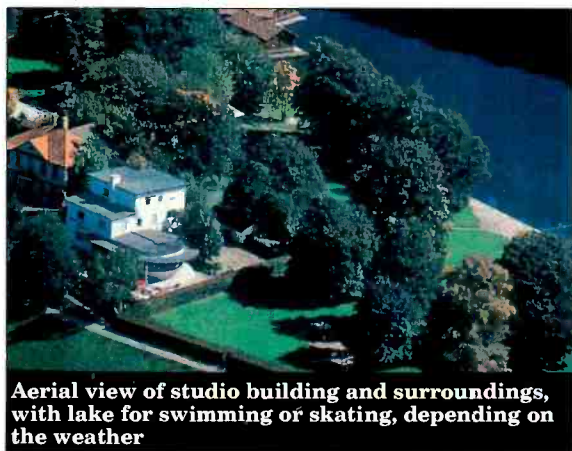
Aerial view of studio building and surroundings, with lake for swimming or skating, depending on the weather

The potential that Waldburger wanted to exploit was Master Studios location. The studio is located in a detached house on a hill just outside the centre of St Gallen, and boasts picture-postcard views of the town with Lake Constance with the Alps in the distance. The area is so pretty that it has been allocated 'recreation-only' status by the Swiss Government which strictly controls all building so that the natural beauty of the landscape is not spoilt.

'Master Studios really can boast a unique location,' Waldburger asserts, 'every artist who has worked here has loved it, which is why so many of our clients keep coming back. They feel at home and because both of the studios have natural daylight they don't feel hemmed in.

'When I first started running a studio in 1978, I had a room in a basement and I hated it because I felt like a mouse that never saw the light of day. No one can be expected to be creative in that kind of environment; it is much too depressing. I had to get out of the basement and back into the daylight, so in 1982 I started looking for new premises. I found this building and fell in love with it because the views were so inspiring. But the authorities would only let me have it on condition that I opened a restaurant on the ground floor, because they felt it would be good for the town—so in one quick move I became the owner of a studio and a restaurant!'

The restaurant is still running, although Waldburger is no longer ▶

The phrase 'if it ain't broke, don't fix it' means nothing at Master Studios in Switzerland—Sue Sillitoe found out

Studio A control room with night view on the old town of St Gallen with its world famous cathedral

involved. However, bands working at the studio need little encouragement to use the facility as the food is excellent and in the summer one can eat *al fresco* on the terrace.

With so many natural advantages at his disposal, Waldburger says that going upmarket with the studio equipment was an obvious step. He chose a 48-channel SSL *4080-G Plus* console with studio computer, *Total Recall* and both *G* and *E-Series* EQ for his main recording room because he wanted 'the world's most successful mixing desk' in its latest generation.

'I knew that in order to attract international clients from the US, the UK and the Far East everything had to fit together like a jigsaw puzzle,' he explains. 'The equipment, the monitoring the acoustic environment, the location—they were all equally important elements.

View from border of first lake—the studio building is on the left

'I chose the SSL *G Plus* because I felt it was a significant advance on the *G Series*, both in terms of enhanced sound quality and it operational requirements. Also SSL have an international reputation with artists—more than 80 percent of all chart songs worldwide have been produced on an SSL—so having this console adds to the studio's appeal.

'At the moment there is only one other *G Plus* console in Europe and certainly no other in Switzerland. If bands and producers want the latest Solid State Logic technology they can either work here or at the Strongroom in London—that is the situation.

'My decision to buy the *G Plus* was also based on the way in which I saw the studio developing. I think the easiest way to encourage new clients is to get them in to try mixing here first. Then, if they like the studio, they will come back to record their next project. That was why the *G Plus* made sense for us. Sure, for recording there are lots of choices and everyone has their own favourite desk but for mixing, most people prefer SSL. Statistically these are the most popular mixing desks on the market because they are fast, efficient and most producers and engineers know how to use one. They are a world standard.'

To make sure the acoustic environment matched the technical standards of the equipment Waldburger appointed acoustic engineer Andy Munro, head of studio design

company Munro Associates, to redesign both control rooms. Waldburger moved the 36-channel Soundcraft *TS24* that was formerly in Studio A downstairs to Studio B and has installed DynaudioAcoustics *M3* monitors in the main studio and DynaudioAcoustics *M2* monitors in Control Room B.

'We had Urei monitoring before,' he comments, 'but we wanted a change so we tried out every other type of monitor on the market and the DynaudioAcoustics ones were the best. There are two things people expect from a top studio—a great console and a great sounding room. As a result of the work Andy has done we now have a very precise monitor system that allows the client to judge his recordings and mixes adequately without fatigue. We had real evidence of how well the room works when Polydor Germany signing Silent Majority came here recently to mix their new album. It was taken back to London to be mastered and the mix was so good that apart from aligning the volume nothing had to be changed at all.'

Both studios are equipped with Studer *A820* 24-track tape machines—two in Studio A to give 48-track. Waldburger says he has no plans to buy a digital multitrack because he believes there is still a problem with digital formats: some clients demand Sony machines while others still want the ill-fated Mitsubishis. Instead Waldburger gets round the problem by hiring in whatever the client wants.

'We are fully wired up for digital and we can have machines here within an hour so it is hardly worth buying them,' he says.

Apart from a full range of outboard equipment and a lovely view of the Alps, Studio

The restaurant's bar

A also has a 60m² live room, a 15m² stone room and an isolation area of the same size. All of the rooms are separated by glass doors allowing full visibility between the studio areas and the control room.

Studio B, which is primarily used by Master Studios' radio and TV postproduction clients, has a smaller recording area, enough for a 5-piece band. There are separate drum and isolation booths and plenty of natural daylight. Both studios have equipment racks that are well stocked with names like Lexicon, Korg, Eventide, AMS-Neve, Roland and Tubetech.

The revamp of Master Studios has also taken in the rest and relaxation areas. Both studios have their own lounges and upstairs there is a large, comfortable rest area with TV, video, pinball machines, juke box and—arriving soon—a billiard table. Immediately behind the studio complex is a natural lake which is suitable for swimming or skating depending on the time of year. And of course, this is Switzerland so one hardly need mention the fantastic skiing. The nearest ski lift to the studio is ten minutes away—and bands who have never been skiing before might like to know that although he keeps quiet about it, Victor Waldburger is a qualified ski instructor. ▶
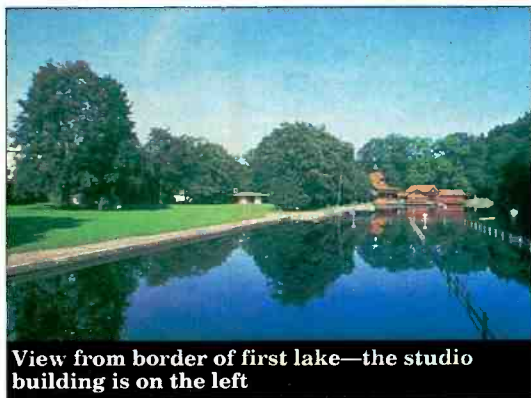
Studio B control room

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Recording area: view from the flat room

'Artists want ambience and the right atmosphere in order to be creative,' says Waldburger. 'A good studio should make you feel at home. If you have a studio where people are frightened to spill anything or drop cigarette ash on the carpet they won't relax and that's no good for anyone. We want to be efficient and friendly without being clinical and cold at the expense of the human being.'

Master Studios is a residential facility although the traditional Swiss chalet style accommodation is a 15-minute drive away. The chalet sleeps up to ten people and is fitted out with all mod cons. For bands who prefer to be closer to the action Waldburger has negotiated special rates with two local hotels in the historic part of St Gallen.

'We offer a specially tailored rate-card because there are so many options available,' Waldburger adds. 'This can include flight costs and hotel charges if that is what the client wants and because we can negotiate special rates on their behalf, recording in Switzerland doesn't have to be as expensive as people might think.'

Even before the refit, Master Studios was used by the likes of Yello, Billy Cobham, Chi Coltrane, the late Alexis Korner and Sens Unik—a Swiss band whose album Les Portes Du Temps was mixed at Master Studios and has been in the Swiss charts for four months. Now the facility is concentrating on expanding its client base as well as coping with existing demands.

As a man who also runs a successful record label, publishing company and band management company, Waldburger can honestly claim to know the recording studio business inside out because he is both a studio owner and a studio user. His label, ZAR, is not given priority treatment on studio time so he often has to put his bands into other facilities because Master Studios is fully booked.

'It certainly helps me to understand what makes one studio better than another. It all comes down to service and making people feel special. Equipment is only half the story.' ∎

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This chalet is available as accommodation

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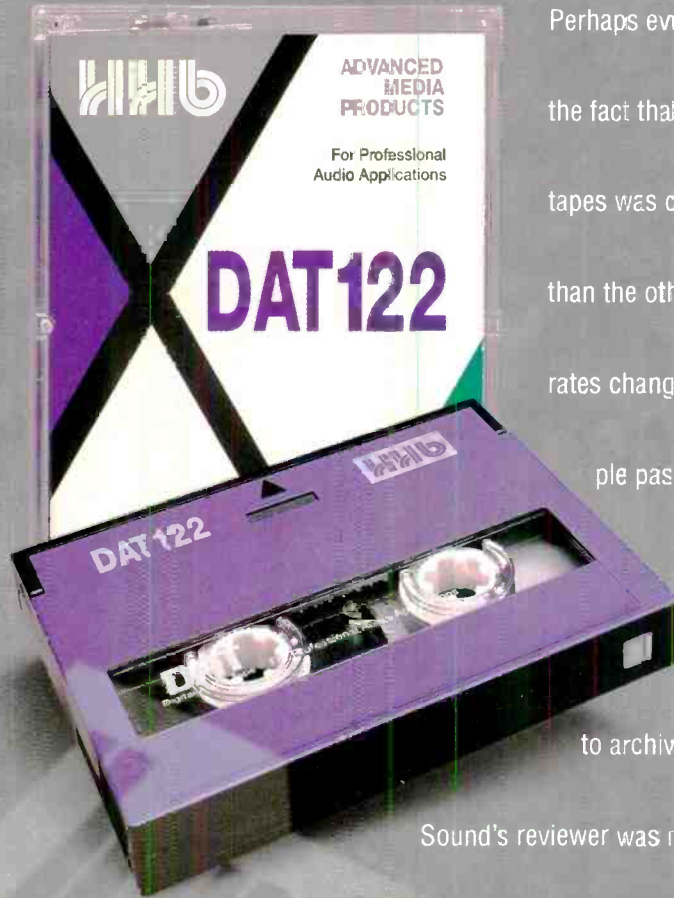
Perhaps even more significant was the fact that one of these leading tapes was clearly more consistent than the others, with its low error rates changing very little over multiple passes. That tape was HHB.

And when it came to archiving stability, Studio Sound's reviewer was moved to write:

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OUT OF CONTROL?

I once read a specification for a new television studio in a European capital city. It quoted design reverberation times in octave bands, which were to be shown mathematically when the plans and tenders were submitted.

The specification stated that the reverberation times were to be calculated according to the Sabine formula, yet in the same sentence went on to say that the Sabine formula did not necessarily have direct relevance to achieved results. This contradiction grew more apparent to me when I was told by many in that same country's television industry that sound was not of the greatest importance and that the emphasis was on visual effects. Yet, I personally know of TV companies which have gone to great lengths to provide the finest sound quality. Clearly, inconsistencies in design theory accompany those of industry commitment to the rooms themselves.

Philip Newell's design for the control room at Front Row Studios in London

You need specs

There are two obvious reasons for this state of affairs—and a third less obvious reason which casts its shadow over the concept of acoustic specifications in general. The first reason is that the overwhelming majority of people who watch television listen on equipment with poor sound reproduction. And while it is true that a growing number of people now own 'hi-fi' video systems, the percentage is still small. The second reason is that the human visual sense is so dominant that when hearing and seeing at the same time, sensory emphasis is on visual material.

The third and last reason has held back television sound for some time. Most people consider high quality TV sound worthwhile, however, few enough consumers buy hi-fi TV systems to make the financiers of the TV manufacturing industry consider improving the system as a matter of urgency. It is this which leads to the third point: who of these is in control?

There are TV visionaries (no pun intended) who look to a future of 'more hi-fi television', and commit themselves now to the installation of sound studios of the first order. Their companies will be the first to be able to take advantage of the new boom. Acousticians, technicians and musicians may also consider them visionaries come what

may, but if the boom does not materialise, then their accountants, chairpersons, bank managers and lawyers may not be so enthusiastic. This third important point is that the power in the television world often lies quite apart from pro-audio people.

TV studios cost a truly enormous amount of money to set up, and when such projects are undertaken, no one person usually has the cash in their pockets. Almost without exception there is an involvement with banks, institutions or shareholders who are considering the venture in purely financial terms. Such large amounts of money usually come with many legal restraints and require business plans to be submitted in such a way that everyone involved has the 'protection' of a specialist set of figures and qualifications. The intention is there should be nothing arbitrary in the assessment of the finished project. Under such circumstances, the provision of room acoustics which cannot be truly specified tend not to figure in a project unless someone within the management structure has the experience to take care of sound and to ask the financiers to back an area lacking provable specs.

Large sums of money may be involved in the sound recording or music industry in general, but its history has been a story of the development of previous

successes rather than of written specifications. This is partly a function of evolution because many successful recordings have been made in low-budget studios, and partly because the hierarchy of the industry appreciates the artistry of the music. In radio and television, football matches, drama, documentaries and political events—the programming—are the majority of the work and for these current television sound is usually deemed 'good enough'. I wonder if Neil Armstrong's moon landing would have been quite so dramatic *without* the noises, bleeps and the strangled sound from the mic within his space helmet?

Sound personnel in the broadcast companies strive to achieve the best results, but it is in the music business that companies gamble readily with designs of an innovative nature. The divide between broadcast and record companies as discussed here is really one of priorities; music people choose to gamble because they face competition in terms of original, ▶

Philip Newell examines the contradictions of studio control room design

creative music output, whereas broadcasters largely deal with music which has already been created and is merely (or not so merely) being reinterpreted or presented complete.

Correlation

A strange situation exists, whereby (in many instances) the greater importance the sound assumes, the less possible it becomes to specify acoustic parameters in meaningful objective terms. The further one progresses towards subjective acoustics, the further one moves from provable facts. In the above design proposal, the consultants had opted for the Sabine Formula which requires a diffuse, reverberant soundfield such as can be approximated in some large halls and reverberation chambers. When absorbers are introduced, however, the diffuse nature of the soundfield is lost and the Sabine Formula becomes at best approximate, and at worst, very misleading. Indeed, a totally diffuse soundfield could not exist as it implies a totally random energy flow with a net energy flow of zero, whereas in reality there is always an energy flow away from the source of the sound. Nevertheless, in a highly reverberant space, a good approximation can be achieved via the above formula.

In small acoustic spaces, Sabine formulations

may tend to fail because of higher levels of clearly definable reflections or echoes. Plus the fact that anything introduced into the room (such as a carpet or a person), has a greater influence on the acoustics of a smaller space than a larger one.

Furthermore, five live rooms built to identical reverberation time characteristics, (especially in terms of RT60 alone) if built, one of wood, one of smooth stone, one of concrete, one of rough stone, and one of plaster, will all have radically different timbral characteristics, yet may all have similar written specifications. In such instances, a room may meet a written spec, yet be deemed to be sonically inferior to a room of identical conventional written specifications but built of different materials.

Many designers use controlled specular reflections, or multiple echoes to achieve, or at least to bolster, the reverberation. The principle is similar to the way that old tape echo machines used to synthesise a sort of reverberation using multiple repeats from a number of replay heads and appropriate feedback loops. When this synthetic reverberation is mixed with the true reverberation, it is not easy to find any readily accessible or easily understandable process which specifies the perceived sonic performance in such a way that (from the written specifications alone) an identical sounding room could be built. ▶
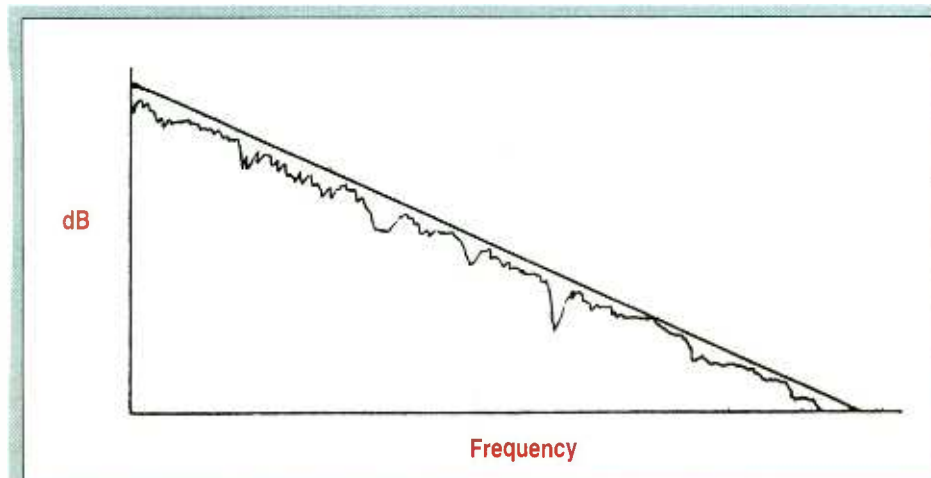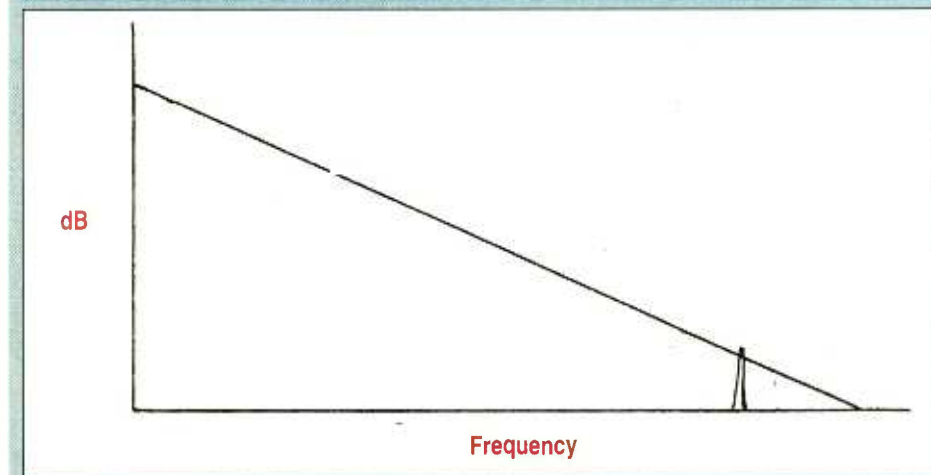
Fig.1: The above plots represent sections of a noise-criteria boundary by which the noise represented in Fig.1a would be numerically acceptable but the noise in Fig.1b would not. Clearly, the noise in Fig.1a would contain vastly greater energy than in Fig.1b, and in many circumstances Fig.1b would be more easily masked in virtually all real situations. Fig.1b would be greatly preferable to Fig.1a. Too strict an adherence to statistics will create absurdity!

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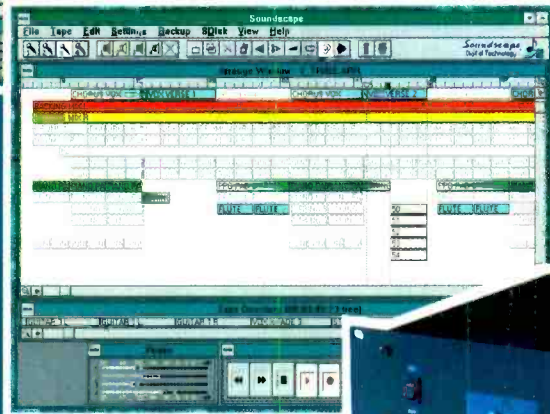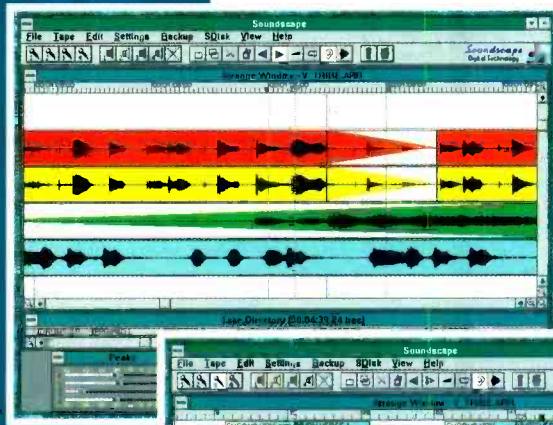
Operation is from an IBM PC™ or compatible and runs under Windows 3.1™. Software allows up to 64 virtual tracks to be recorded in stereo, edited (non-destructive) and digitally mixed down to four outputs. As the system is modular, several Soundscape units can be synchronised with full sample rate accuracy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

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Data format: 16 bit linear

Signal processing: 24-bit internal N.L.

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

2nd internal IDE drive can be fitted.

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

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

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

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

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Telephone

RT60 measurements only show the time for the reverberation to decay to a point 60dB below the initial sound pressure levels. While it is true that in most genuinely reverberant spaces this decay is relatively uniform, in more complicated or less homogeneous rooms, the RT20, RT40 projections may be very different. The energy–time curves of the different rooms can be very dissimilar indeed.

On the subject of sound isolation, similar caveats exist. More specification conscious broadcast industries usually work to Noise Criteria (NC) figures. These give an envelope of maximum levels of noise against frequency which must not be exceeded if the specifications are to be met. Yet from *Fig.1* it can be seen that if a single spot frequency in a relatively unimportant part of the spectrum exceeds the NC, the result is deemed unacceptable. If, on the other hand, a broad-band signal remains 0.5dB below the NC at all frequencies, the specification is met. In most circumstances the much greater energy of the sound in the second case would be more objectionable than the relatively innocuous failings of the spot frequency. But to rectify this, the cost of the building could be greatly increased if the NC figures were rigidly adhered to—even though in practice no problem existed. Conversely, where NC figures are used for justifying digital data compression techniques, a broadband noise slightly exceeding the NC can be less obtrusive than a spot frequency in a sensitive area which does not exceed the NC, but conditions vary with masking effects.

A problem often arises when acoustically inexperienced architects specify to 'known' criteria without referring to acousticians. Under these circumstances, errors of judgment can easily be made in terms of subjective results, despite objective acoustic specs being satisfied.

Subjective acoustics is much like an iceberg, with 90% beneath the surface and not perceived. A change of 50% to what is perceived, may seem significant, but this may only represent a change of 5% to the whole structure. Any such change to an iceberg would alter its balance and hence the angle of flotation, revealing unseen sections, and submerging parts which were formerly visible. Without a precise knowledge of what lies beneath the surface, no change can be entirely predictable. Such is the capricious nature of acoustics: the deeper one gets, the deeper one's awareness needs to be.

In an address to the 72nd AES Conference in Anaheim in 1982, Ted Uzzle summed things up by saying: 'No sound system, no sound product, no acoustic environment can be designed by a calculator, nor a computer, nor a cardboard slide-rule, nor an Ouija board. There are no step-by-step instructions a designer can follow; that is like Isaac Newton going to the library and asking for a book on gravity. Design work can only be done by designers, each with his own hierarchy or priorities and criteria. His three most important tools are knowledge, experience, and good judgment.'

Quoting Lettinger from his book *Studio Acoustics* (Chemical Publishing Co, New York 1981): 'Nothing is gained by specifying the noise level limits in a room using an NC curve. According to this method, a noise is not acceptable when any part of the spectrum exceeds the limiting curve, no matter how narrow the frequency band which surpasses it is. But then, a noise is also unacceptable with a spectrum equal to that of the pertinent NC curves but which slightly transcends these curves. Yet the two noises carrying the same rating number, may differ widely in their

A-weighted sound levels.'

There is nothing new about any of this, yet it is surprising how many people in the recording industry are still unfamiliar with these aspects of design. Back in 1963, Schultz in his *JAES* paper (Vol. II, pp 301-317) *Problems in the Measurement of Reverberation Time* stated: 'In a large room, if one has a sound source whose power output is known, one can determine the amount of absorption in the room by measuring the average pressure throughout the room. This total absorption can then be used to calculate the reverberation time from the Sabine formula. This method fails badly in a small room however, where a large part of the spectrum of interest lies in a frequency range where the resonant modes do not overlap but may be isolated. In this case, the microphone, instead of responding to a random soundfield (as required for the validity of the theory on which these methods depend), will delineate a transfer function of the room. It does not provide a valid measurement of the RT in the room.'

Evolution

When it comes to designing a control room, the problems described above almost entirely preclude designs based on rigid formulations. Control rooms began in the first days of electric recording as 'control booths'; very small rooms with the monitoring consisting of a very basic loudspeaker. Even 50 years on, control rooms were still largely what would today be considered rudimentary designs. Any acoustic treatment consisted of Helmholtz resonators and absorbent panels and, if one was lucky, some geometrical 'control'. One was very lucky if any two locations in the room sounded

similar. Although almost every control room for record production was then stereo, there were still mono disc-cutting rooms and mono broadcast control rooms. Few if any control rooms had been designed with stereo as a prime consideration. Most were updates of old mono rooms and many still had four-speaker monitoring from the early days of four-track multitracking—not the quadraphonic layout of the mid-1970s, but four speakers in a row across the front of the room, one per track. Eight and 16-track machines were routed through these via switching matrices.

These rooms were often inadequate for good imaging or frequency balance when mixing to stereo. 'Good' rooms became renowned—though many of their designers could not pinpoint exactly why one room 'worked' yet another failed to please. Few genuinely good rooms existed even when 24-track machines began to appear in 1973. Around this time, Tom Hidley (then at Westlake Audio in Los Angeles) began to make international waves with a new concept in control room design. This was intended to produce repeatable results around the world, but the claims were too exaggerated and people began to complain that work done in one room of Westlake design did not necessarily sound reasonably similar in another. It was, however, a bold step, and though he did not achieve his goals during the 1970s, Hidley did change the face of recording studio design throughout the world.

Hidley later sold his shares in Westlake, moved to Switzerland and started Eastlake Audio. He later passed Eastlake to David Hawkins who had been his representative in the UK, and moved to Hawaii. Since his return to studio design in the mid 1980s, Hidley has worked alone, nowresiding in the Caymen Islands.

What Hidley did not know—and could not have known at the time—were the implications of the then yet to be defined 'chaos principle'. There were just too many small changes from room to room which would conspire to produce radically different sounds. What Hidley and most other studio designers were using as their major reference, was third-octave spectrum analysis with graphic equalisers on the monitors. Most studio designers now accept that third-octave analysis and monitor equalisation were disasters, but Hidley had conceived a system of control room design and monitoring which, possibly for the first time on a large scale, was enabling wide-band flat amplitude responses without excessive use of equalisation. It was a beginning, and while by no means perfect, some of his rooms of almost 20 years ago are still in use by major record companies: a testament to his achievement.

The problems which fooled people at the time, stemmed from the concept that between reverberation time and the pressure amplitude response around the position of the engineer, the performance of a room could be specified. As we now know, in a well-designed room, it is the culmination of the minor bumps of the response which dictate the overall character, and that a third-octave equaliser is far too crude a tool to do anything but move the bumps around.

Reverberation time affects the steady-state response of the room, but the direct signal would only be affected by the monitor system and its method of mounting. Each affects the other in terms of perceived responses, but act differently on steady-state and transient signals. In 1974 when I first approached Tom Hidley about building a studio for Virgin Records (for whom I was Technical Director at the time) we had been using

JBL and Tannoy monitors. Hidley told me that the Westlake systems which I had heard were using JBL drivers. In those days, JBL monitors sounded hard—or so I thought. Hidley suggested that we used Gauss drivers for the low and middle frequencies. Once completed, the room measured identically in third-octave, pink noise terms as the JBL-equipped room in Los Angeles, yet the two sounded very different, and could not be equalised to sound similar.

What was going through our minds in those days was short of concepts of phase response. Though many papers had been published indicating its importance, classical acoustics was still largely adhering to Helmholtz' philosophy of phase being imperceptible by humans, which we now know not to be so. But even then, the fact that I had asked for Gauss instead of JBL drive units in an equalised room, should have lit a beacon in our minds—if the Gauss and JBL units could be equalised to sound alike, then no preference should exist. This was missing from our concepts of achieving similarity via 'repeatable' room designs and third-octave equalisers. I remember Hidley finally saying to me in the late 1970s, 'It's phase distortion; that's what produces fatigue and inconsistency!'

Looking back he was right, though what could have been done about it at the time is uncertain. Since then, however, as the general requirements for control rooms have multiplied, so have the approaches of different designers, most of whom attempt to produce a relatively neutral acoustic. By definition, neutral implies that it neither adds nor detracts anything, and hence, moving from one neutral control room to another—even of different design—one should enjoy a consistent sound character. This is still not the general case; in ▶
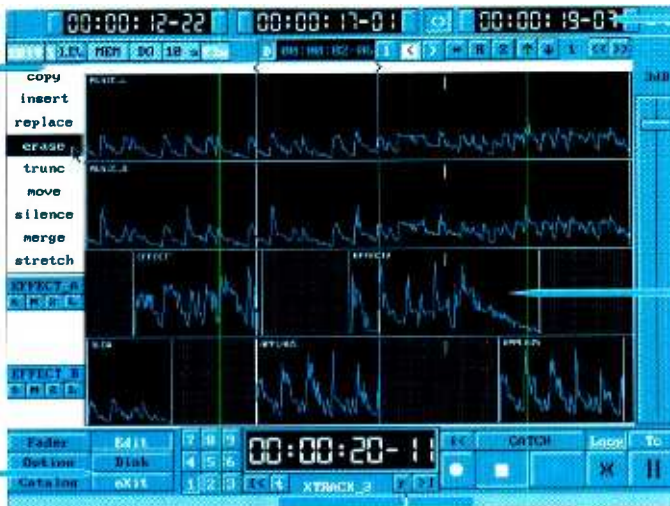
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reality, many rooms sound so different, and even identical rooms with different equipment in different locations can sound very different. This is one of the major reasons for the widespread adoption of nearfield monitors (or more properly 'closefield' as 'nearfield' is an acoustical term for the zone immediately in front of a sound source where the air acts as a 'lumped element' and the sound does not decay by the usual 6dB with doubling of distance).

With a small monitor so close to the listener, the room plays less of a role in the perception of this sound, but necessarily, desk reflections and reduced frequency range mean that there is a price to pay for their use.

Acoustic control

There have been many approaches to the problem of linearising the response of a control room by acoustical means. The first attempts were based on 'tuning' the room by means of resonators and absorbers to produce a uniform reverberation time over frequency. The problem with this approach is that even if different rooms of different designs achieved the same measured response with frequency (for example 0.4s from 50Hz to 10kHz) then, as discussed earlier in the context of live rooms built from different materials, their construction will substantially change their perceived sonic performance. What is more, if they had either different monitor systems with different directivity characteristics, or areas of different tuning or absorption (or both), they would be perceived to be very different.

This was discussed above with regard to the broadcast industry, working to predetermined specifications. It necessitates each company's own set of 'golden ears' having to make their individual assessment of fine tuning.

Another approach towards linearising the performance of a room is to use a technique of geometrical control, providing many nonparallel surfaces and selected areas of absorption and reflection. In a rectangular room there are three types of resonant modes; axial, between parallel walls and parallel to four other walls; tangential travelling round four walls and parallel to the other two; and oblique, travelling round all walls and parallel to none. The axial modes are the strongest in terms of energy, whereas the oblique and tangential modes lose more energy both by striking more surfaces per given distance travelled, and because surfaces tend to absorb better when the incident sound strikes them obliquely rather than at right angles. In such rooms, the spread of the modes can be changed, generally levelling the response to some degree, but at low frequencies angles cannot be made large enough, so low frequency control is again achieved by absorption; often by 'bass traps' placed strategically at the suitable points of problem modes. Such was the early Westlake approach which produced some very pleasant rooms to work in—but they were not identical as intended. They never could have been because, again, specifications could not take into account the effect of differing shapes, sizes, decorations and equipment. However, they could, and were, built to be capable of providing uniform third-octave pressure amplitude responses, but the phase, and hence transient responses were different. These rooms could never be accurately assessed until built.

The ear has an ability to detail as separate events the early and late arrival of sounds. If a sufficient time delay exists between the direct sound from the loudspeakers and the first

reflections from the room prior to any general 'pseudoreverberation', then ears, especially 'trained' ears, can usually begin to 'lock' on to the incident sound quite soon, and allow the brain to consider the room separately . This relies on the Haas effect, named after its first proponent. Taking into account this phenomena were the 'Live End, Dead End' rooms, as proposed by Wrightson & Berger, having the front half of the room made very acoustically dead, and the rear half of the room made reverberant to the desired degree, sometimes by means of Schroeder diffusers. The concept prevented early reflections from reaching the mixing position. Thus the 'first pass' of the incident wave from the loudspeaker from the effects of the room. Once again, consistency was uncontaminated by reflections for a sufficiently long period of time for the ear to clearly differentiate the direct sound from the loudspeaker and depended upon both the monitor system (for the direct sound) and the design of the rear of the room (for reverberant sound). As with other approaches, there are inconsistencies between measured and perceived performance, though it must be said that this need not be a problem for engineers familiar with certain rooms and *can* offer consistent results.

The seemingly obvious choices, if the rooms are to be entirely consistent to one another, are: to standardise on room shape; size; equipment location and design. Sadly, this is impractical given the differing requirements of different people, and the suitability of available property.

The second option is the anechoic chamber, but such rooms are not pleasant to be in. Indeed, some people find them disturbing as all normal auditory cues on wall positions are missing—you do not hear the room which your eyes see. Such rooms also run against the long-held concepts of rooms bearing some general similarity to the 'average' domestic room, though by now, it is generally agreed that any average is itself so unrepresentative of the majority of rooms, especially from country to country, that little relevance exists between the concept and the reality. What has further strengthened the argument against the representation of domestic rooms is that many listeners now use cars or personal stereo systems, where no reproduction room exists at all.

Dead end

When Tom Hidley returned to studio design around 1984, he returned determined to achieve the room-to-room consistency which had eluded him in his first decade and a half of trying. It does not require many reflections to make a person feel comfortable in an anechoic chamber—two or three restore a sense of ease. He thus concluded that by making the front wall and the floor the only reflective parts of the room, other than the equipment such as the mixing console, the remainder of the room could be 'trapped' to as low a frequency as possible without producing any unpleasant characteristics. If the front wall is the only vertical hard, reflective surface, it provides an effective monitor baffle extension but cannot produce any reflections of the music. Further, reflections from the floor have to arrive at the listener in the same vertical plane as the direct sound, so no disturbance of the stereo image can occur. In practice, a soft back to the mixing console effectively stops all but the low frequency floor reflections from reaching the engineer's ears. When one adds to this concept the use of only one type of monitor system, it is easy to see how a great degree of consistency can be achieved from room to room.

Significant differences arise with the sound of speech within the room, which varies with the distances to reflective surfaces, and the responses below 40Hz or so, dependent upon the amount of trapping space available for low frequency absorption. This approach has probably come the closest yet to achieving room-to-room consistency.

After becoming familiar with Hidley's concepts, he and I cosponsored further research from Brazilian acoustician Luis Soares, at the Institute of Sound and Vibration Research at Southampton University, England. This was an attempt to find mechanisms by which to achieve similar absorption in smaller rooms, or more effective absorption in larger rooms. Hidley himself refuses to drop below certain room sizes as he is rightly very conscious of previous criticisms of his claims of uniformity of listening conditions for the pre-1980 rooms. These concepts are so powerful, however, that I have used them to great effect even in quite small rooms, although these rooms are very demanding of their monitor systems, requiring specifically designed units. They soak up almost all of the incident wave, so little reinforcement of loudness is delivered; furthermore, off-axis irregularities of the monitor system are heard off-axis with ruthless accuracy. Effectively, the rooms need high power, smooth, wide-band, coincident source monitor systems. Such systems are not easy to locate commercially, so most come specifically with the rooms.

It is ironic that these rooms which have begun to achieve consistency on a subjective basis, have only done so by having a dual specification. They are 'monitor dead'—in other words, the monitors drive into a largely anechoic termination, hence the rooms have no specification in terms of reverberation time as they are approximating to a free field. There is, however, a second specification which could relate to the direction and distance of echoes or specular reflections from a sound source inside the room, such as from the listening position, but this relates only to the ambience of the room from the point of view of persons within it. It has no bearing on the monitoring, except in some extreme psychological concept of 'comfort factors'. Not surprisingly Tom Hidley refers to these rooms as 'Nonenvironment Environments'. You feel as though you are inside a room, yet you listen to the music as though you are in a field. The logic behind this is that even if small changes in terms of specifications can have what would appear to be disproportionately large subjective repercussions, then the only way to make the practical differences approach zero, is to make the specification approach zero; ±20% of error on two dead rooms is still two dead rooms!

Subjectively, there are no rights and wrongs in control room acoustics. Some people have made great recordings in 'difficult' acoustics, but this is no justification for haphazard approaches to design. Many 'difficult' rooms have either been built by people who thought that they knew more than they did, or designed by designers working under constraints from their clients.

Too much has sometimes been expected from technical specifications alone and, without a demonstrable relationship between subjective and measured results, many studio owners have understandably shied away from what they have seen as 'the black art of acoustics'. Hopefully a clearer understanding of the degree of complexity involved in acoustic interactions can be achieved in the future—and a better relationship between studio designer and owner built upon it. In the meantime, Ted Uzzle's comment should be taken as the golden rule: there is absolutely no substitute for experience in such a capricious 'science'. ∎

GET IN LINE!

No professional would dream of making valuable recordings on an analogue tape recorder without verifying the alignment of the transport and circuitry. To do so would be to invite interchange problems and deterioration in sound quality. Yet people do exactly this with RDAT machines. The machine comes out of the box and goes to work. Digital audio was heavily oversold in its early days—error correction may be extremely clever, but it does not help if the head is dirty and nowhere near the tape tracks. Subsequently, many of the difficulties RDAT users experience are due to lack of maintenance or poor alignment.

Digital audio is wonderful because it expresses delicate detailed analogue waveforms as robust data which can only be zero or one. If the numbers output by the A–D convertor reach the D–A convertor unchanged, then the sound quality is determined only by the convertor quality. Convertors can vary in quality, of course, and some RDAT machines are notorious for their poor convertor quality. This article is not concerned with convertor quality, perhaps that will be covered at another time. What I want to deal with here is the situation where the numbers from the A–D do not make it intact to the D–A because there is an alignment problem in an RDAT transport or signal system.

The error correction system of RDAT is designed to correct random errors due to noise or slight mistracking, and burst errors due to dropouts. It is more than powerful enough for the job. It is not intended to correct the enormous error rate which results from serious mistracking, nor is this practicable in any digital recorder, audio or otherwise. If mistracking is bad

enough, the error correction cannot cope and the audio mutes. Mistracking occurs when the tape and the machine playing it have different ideas about where the tracks should be. Interchange alignment eliminates mistracking and leaves the error correction to do its intended job.

From the tape-head point of view RDAT has more in common with a video recorder than a traditional audio recorder. The tape passes around a cylindrical drum shown in **Fig.1**. The upper part of the drum revolves and carries the heads. The lower part of the drum is stationary and carries a helical ramp which guides the lower edge of the tape. A pair of fixed guides, known as the entry and exit guides, lead the tape to and from the ramp. The tape may wrap around the scanner by various amounts, typically a little more than 90°. The total angle of scanner rotation for which the heads touch the tape is called the *wrap angle*.

In RDAT the scanner rotates with the direction of linear tape motion. It will be evident that as the tape is caused to descend the scanner axis by the ramp, it actually takes on the path of a helix. The head rotates in a circular path, and so will record diagonal tracks across the width of the tape. If the tape is stationary, the head will constantly record the same track, and the angle of the track with respect to the edge of the tape will be at the *helix angle*. When the tape moves, the angle between the tracks and the edge of the tape will be different from the helix angle. **Fig.2** shows how this occurs. When the head makes contact, the tape will be at a given location, but when the head loses contact, the tape will have moved a certain linear distance. To obtain the *track angle* it is necessary to take into account the tape motion.

If the linear tape speed and ▶

What needs to be aligned in a typical RDAT machine? John Watkinson precisely sets out the facts and challenges users to get professional
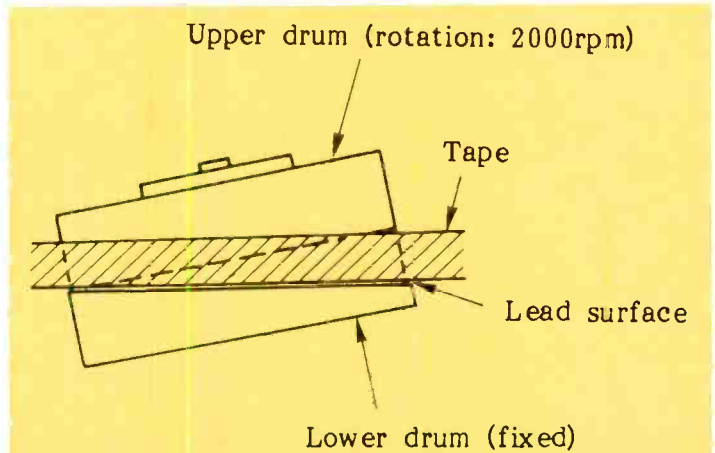
Fig.1: The tape is guided around the drum by a helical step. The rotating upper drum produces an air film such that only the edge of the tape touches the drum

$$\frac{w}{L} = \tan \hat{H}$$

$$\frac{w}{L-d} = \tan \hat{T}$$
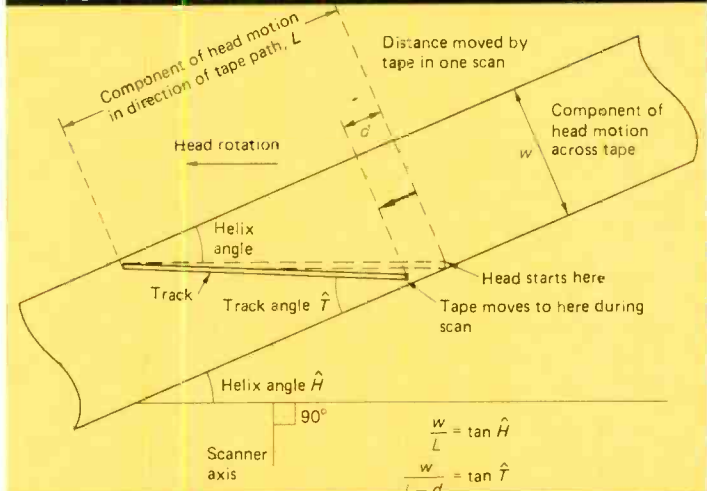
Fig.2: When the tape moves, the track angle becomes different from the helix angle. It is the track angle which is standardised in the format
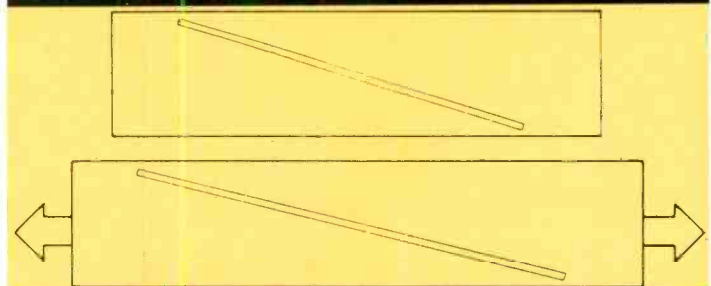
Fig.3: Tension changes cause the track angle to change and this results in tracking errors which become worse at the ends of the track

Fig.4: The drum step only locates the lower edge of the tape so the entry and exit guides are set to contact the top edge. In this way the path of the tape is constrained

Fig.5: Conceptual view of RDAT transport showing the guides which are slotted for screwdriver adjustment. The entry and exit guides are the most critical for interchange

scanner speed remain the same, the theoretical track angle will remain the same. In practice this will only be the case if the tape tension is constant. Tape is flexible and if the back tension change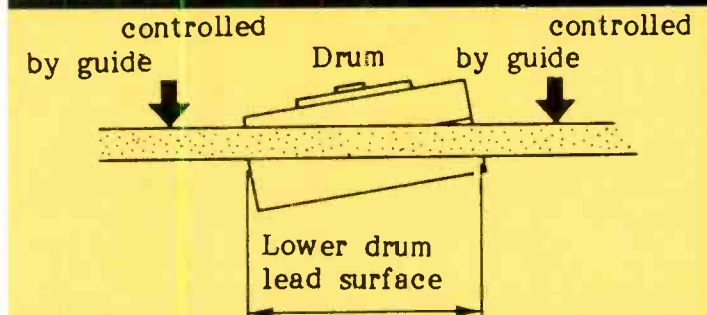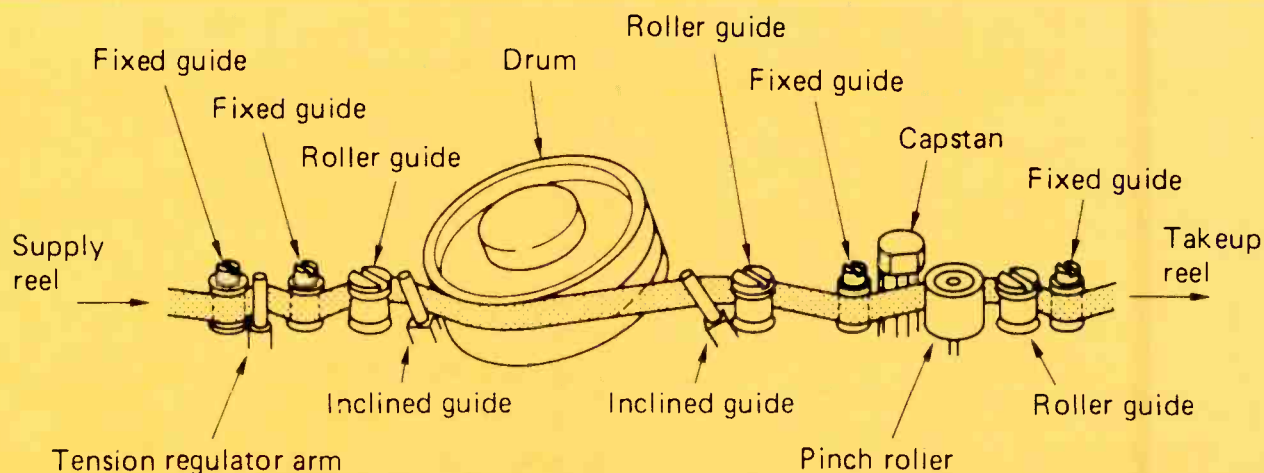s, the effective length of the tape will also change, and with it the track angle, a phenomenon known in video-recorder land as *skew* (**Fig.3**). All digital rotary-head recorders need some form of tape tension servo to maintain the tape tension around the scanner constant irrespective of the size of the pack on the supply hub. As there will be friction between the tape and the scanner, the tape tension will gradually increase between the entrance and exit guides. When the tape direction reverses, the sense of the friction in the scanner will also reverse, so the back tension has to increase to keep the average tension the same as when going forward.

The spinning drum builds up a flow of air which lubricates the tape. The tape does not touch the upper or lower drum surface at all, but is guided around the drum by its lower edge only. The tape is prevented from rising by the entry and exit guides. The entry and exit guides are adjusted to contact the top edge of the tape so that it must run along the drum step (**Fig.4**). Also shown in **Fig.4** is that if the tape sheds particles of binder, these may adhere in the angle of the drum step and cause the tape to lift over the debris, resulting in mistracking. Unfortunately a cleaning tape will do the same and will not necessarily remove the debris. The best way of ensuring the drum step is clean is to run along it with the end of a wooden cocktail stick soaked in alcohol. The wood is strong enough to dislodge the binder, but not sufficiently hard to scratch the drum surface. Clearly there is no point in attempting alignment on a machine which has not been scrupulously cleaned. As Sam Wise's recent tests have shown, some tapes are capable of considerable shedding.

When the cassette is lowered into the transport it seats on pillars which hold it level. The tape within the cassette is guided by liner sheets which determine the height of the tape pack above the transport baseplate. The first step in aligning the transport is to ensure that all of the guides the tape runs past on its way to and from the scanner are at the same height as the tape. **Fig.5** shows that the guides are threaded so that they can be screwed up and down. In the correct position, the tape will stay in the cassette plane and distortion will be avoided.

Once the tape can be passed through the machine without damage, the basic transport functions can be checked. Since tape tension affects the track angle, obtaining the correct tension is essential before attempting any adjustments at the drum. Normally tape tension is measured with a gauge such as a Tentelometer, but the small size of the RDAT transport rules this out. Instead a special cassette is used in which a tension gauge is built in to each hub. The test cassette is loaded into the machine as normal, and the tension can be read off a scale on the body of the cassette. Tape tension needs to be adjusted in several modes of operation, typically normal play, reverse and shuttle, and adjustments are provided for each mode. In some transports the tension-sensing arm is not statically balanced, and the tape tension becomes a function of the orientation of the machine. In this case the adjustment must be made with the machine in the attitude in which it is to be used.

The track spacing on record is determined by the capstan speed, which must be checked. As the capstan speed will be feedback-controlled by a frequency-generating (FG) wheel on the capstan shaft, it is generally only necessary to check that the capstan FG frequency is correct in record mode. A scratch tape will be used for this check.

Helical interchange can now be considered. Tape passing around the scanner is guided in three ways. On the approach, the tape is steered by the entrance guide, which continues to affect the first part of the scanner wrap. The centre part of the scanner wrap is guided by the machined step on the scanner base. Finally the last part of the wrap is steered by the exit guide. Helical interchange is obtained by adjusting the entrance and exit guide heights so that the tape passes smoothly between the three regions. In video recorders, which use wider tape, it will often also be found necessary to adjust the angle of the guides. In RDAT the tape is so narrow that it will flex to accommodate angular errors, and the guides only need a height adjustment.

As tape is flexible, it will distort as it passes round the scanner if the entrance and exit guides are not correctly set. The state of alignment can be assessed by working out the effect of misalignments on the ability of the replay head to follow tape tracks. **Fig.6a** shows an example of the entrance guide being too low. The tape is forced to climb up to reach the scanner step, and then it has to bend down again to run along the step. If straight tracks were originally recorded on the tape, they will no longer be straight when the tape is distorted in this way. **Fig.6b** shows what happens to the track. Since in an azimuth recording machine the head is wider than the track, small distortions of this kind will be undetectable. It is necessary to offset the tracking deliberately so that the effect of the misalignment can be seen. If the head is offset upwards, then the effect will be that the RF signal grows in level briefly at the beginning of the track, giving the envelope an onion-like appearance on an oscilloscope. If the head is offset downwards, the distortion will take the track away from the head path and the RF envelope will be waisted. If the misalignment is in the exit guide, then the envelope disturbances will appear at the right-hand end of

the RF envelope. The height of both the entrance and exit guides is adjusted until no disturbance of the RF envelope is apparent whatever tracking error is applied. One simple check is to disable the track-following system so that the capstan runs at approximately playback speed with no feedback. The tracking will slowly drift in and out of registration. Under these conditions the RF envelope should remain rectangular, so that the amplitude rises and falls equally over the entire head sweep. A rough alignment can be performed with a tape previously recorded on a trustworthy machine, but final alignment requires the use of a reference tape. **Fig.7** shows that the reference tape has only alternate tracks recorded. This prevents crosstalk from the tracks of wrong azimuth confusing the testing process and allows the RF signal to fall as the head mistracks.

Once the mechanical geometry of the transport is set up, straight tracks on tape will appear straight to the scanner, and it is then possible to set up the automatic track-following system so that optimum tracking occurs when the tracking error is zero. This is done by observing the RF level on playback and offsetting the tracking adjustment to each side until two points are found where the level begins to fall. The adjustment is then placed halfway between these points. If the machine has a front panel tracking adjustment, it should be set to zero while the internal adjustment is made.

The final interchange adjustment is to ensure that the scanner timing is correct. Even with correct geometry, the tracks can be laid down at the correct angle and spacing, but at the wrong height on the tape, as **Fig.8** shows. The point where recording commences is determined by the sensor which generates a pulse once per revolution of the scanner. The correct timing can be obtained either by physically moving the sensor around the scanner axis, or ▶
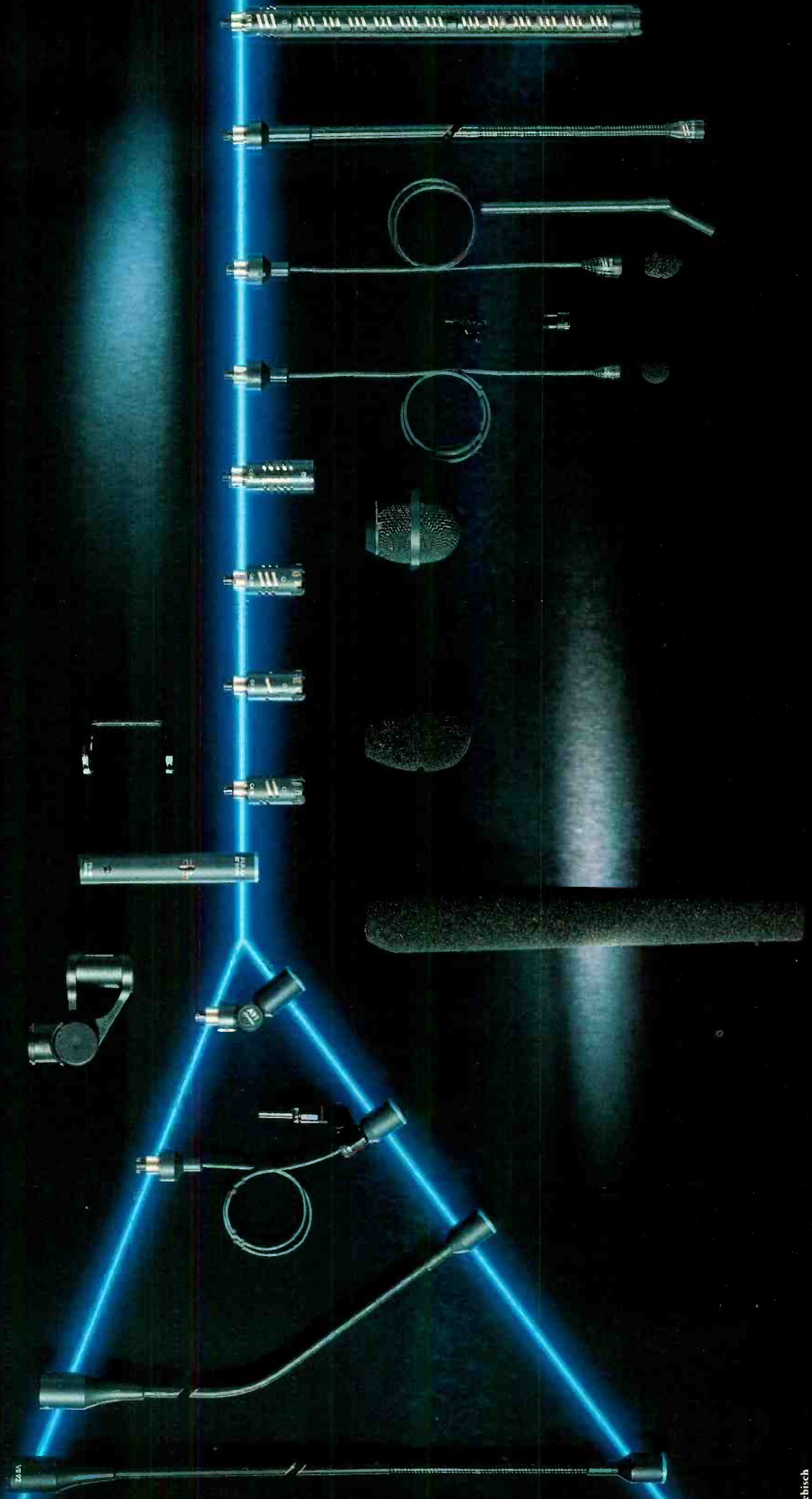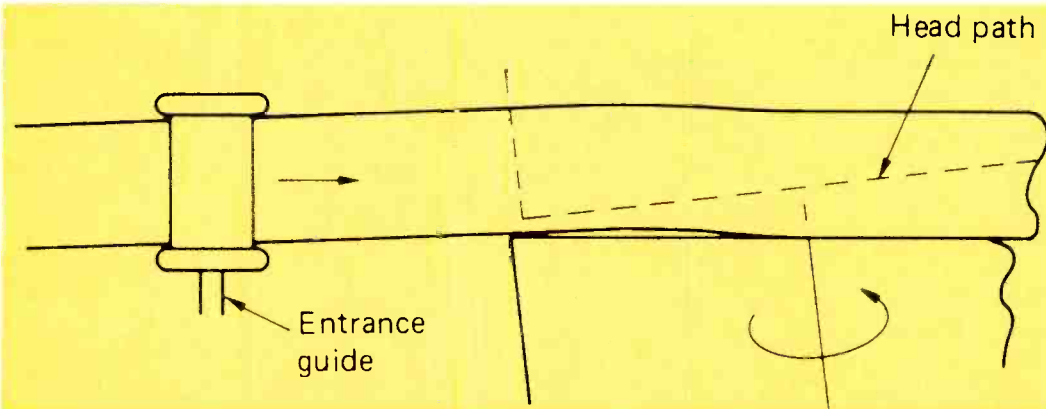
Fig.6a: Here the tape is not approaching the drum cleanly and the tape is forced to flex
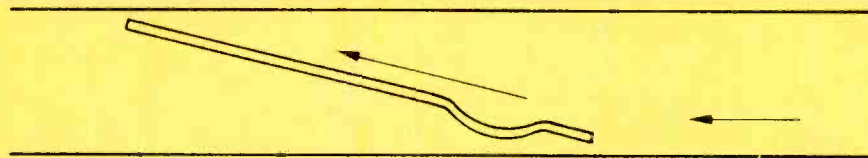

Fig.6b: Here following recording the tape relaxes and the track becomes distorted. This will cause a tracking error when played on a correctly aligned transport
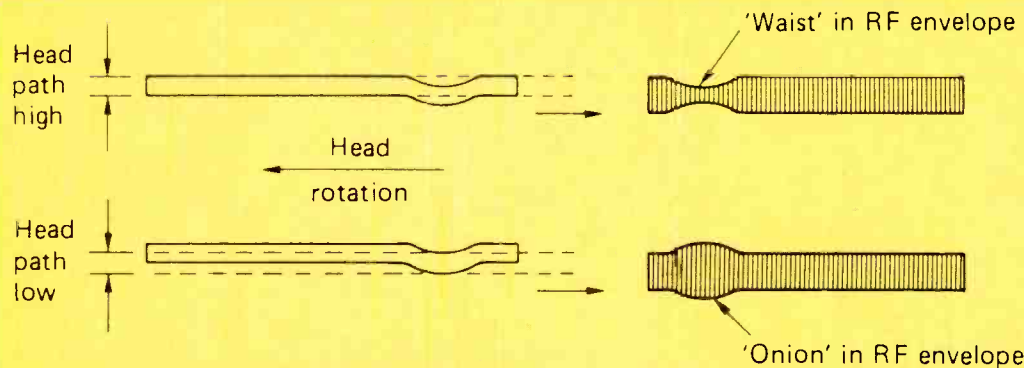

Fig.6c: Here a reference tape is being played on a misaligned transport. The RF signal shows disturbances which vary with the tracking setting. On a correctly aligned transport, the envelope should remain rectangular as it collapses
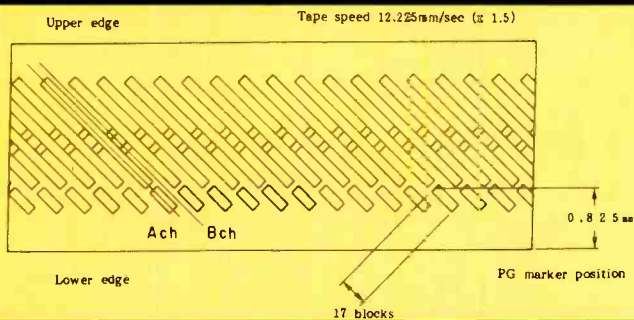

Fig.7: The reference tape has only alternate tracks recorded. This allows the RF envelope to be analysed more easily during alignment. Note also the gap in the recording which is used to adjust the phase of the drum pulse generator
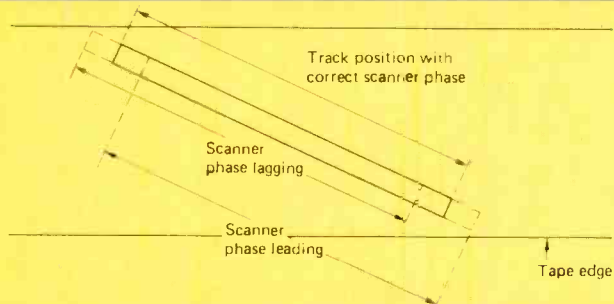

Fig.8: If the drum pulse generator is wrongly adjusted, the tracks are recorded at the wrong position across the tape

random errors. However, the record current should not be increased so much that the tape saturates, because this results in recorded pulses spreading and the increased jitter negates the noise advantage. There is thus an optimum record current which is just below the onset of pulse spreading. As the optimum current depends upon the characteristics of the tape, there is a danger that setting it for a high coercivity tape will result in overdriving a lower output tape. In theory all tapes should be the same but I wouldn't like to say what the practice is like. Thus there may be a case for erring on the low side by a few percent when setting record current.

On replay, the high density of recording results in peak shift distortion which results in jitter and raises the error rate. While space prevents details here. I have droned on about it elsewhere [1]. Suffice it to say that the antidote is an equaliser in the replay channel. The equaliser should be adjusted to compensate an individual head. This can be done by adjusting both ways in order to minimise the inner error rate. As the heads wear, the shunting effect of the gap reduces and the EQ needs to be reduced by periodic readjustment.

The major alignments have been covered in principle, but for a specific machine the maintenance manual will have to be consulted. Not all machines have all of the adjustments mentioned. RDAT machines like all rotary head recorders need care in adjustment. If in doubt about your ability to make the adjustments, they should, in the words of the famous car manuals, be entrusted to your dealer. Once all of your machines have been aligned, interchange between them should cease to be an issue. Poor quality cassettes can still cause problems, primarily binding between the pack and the cassette liners. If severe the tension servo cannot compensate and tracking suffers. ■

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John Watkinson, RDAT, Chapter 4, Focal Press, ISBN 0 240 51306 1.

by adjusting a variable or tapped delay in series with an artificially early fixed sensor. A timing reference tape is necessary that has an observable event in the RF waveform. The tape is played, and the sensor or delay is adjusted to give the specified relative timing between the event on the reference tape and the sensor pulse. When this is correct, the machine will record tracks in the right place along the helical sweep. If this adjustment is not correct, a machine will still be able to play its own tapes, but if tapes are transferred to a different machine, difficulty may be experienced with editing as the timing will change between different parts of the tape.

As with analogue recorders, the mechanical adjustment must be paralleled with correct signal system adjustment. In digital recording, the data are recorded by changing the time between flux reversals, and the record current has constant amplitude and only the direction is reversed. Clearly the larger the record current, the larger the replay signal, and this improves the signal-to-noise ratio and reduces

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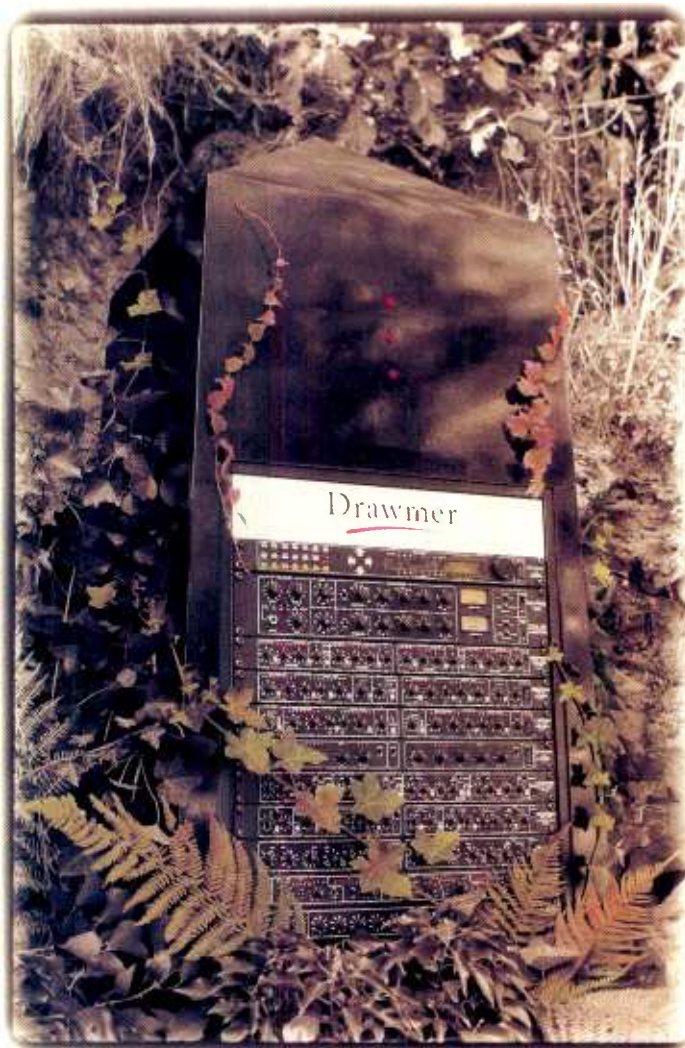
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A SAPPHYRE BEYOND PRICE

The hardware—Soundcraft's *Sapphyre* console

It is fitting that two products that share a fundamental similarity, high quality features at a low price, should be put together: Soundcraft's *Sapphyre* console and AD Systeme's *Optifile LC* can now be obtained as a package.

'There were two main reasons why we chose *Optifile* for the *Sapphyre*,' says Soundcraft Product Sales Manager Andy Brown. 'Firstly it's an excellent, easy to operate system which we feel suits the console very well, and secondly it allows us to add significant value to the product which can be passed onto the client without a substantial hike in price. Another significant point is that this is not the first time that Soundcraft and AD Systeme have collaborated—the *FAME* automation system that appeared in the *IS12* console was

actually an early AD Systeme design.'

Since its release just under two years ago, *Optifile LCs* have increasingly been retrofitted into *Sapphyres*; six months ago Soundcraft produced an automation-ready version specifically for the *LC* system. At no extra cost the *Sapphyre LC* comes fully equipped with VCAs and connectors ready for simple plug-in connection to the *LC* processor rack. It is then up to the client to order the *LC* processor via an *Optifile* dealer such as The Home Service in the UK; although in extreme cases, where a dealership is not available, Soundcraft will supply a fully fitted console.

The current UK price for a 36-channel *Sapphyre LC* with patchbay is £21,295; the *Optifile* computer adds an extra £3,650 excluding the price of a colour monitor and video cable. This offers

excellent value when you consider the retrofit price would previously have been in the region of £6,500.

Since the introduction of the *Sapphyre LC*, sales of the console have steadily increased (US sales are ▶

So pleased were Soundcraft with AD Systeme's *Optifile LC* automation that they have produced an automation-ready *Sapphyre* console just for the system. Patrick Stapley reports

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reported to be up by a third for example) and now stand at just over 650 worldwide: consequently Soundcraft expect to cease production of the standard *Sapphyre* early next year. Apart from the addition of VCA circuitry and *LC* connectors, the new console remains unchanged featuring the same compact in-line design, powerful, Trevor-Stride-designed EQ, and FET noise gates on each channel, which have all contributed to its popularity. A full review of the console can be found in the August 1991 edition of *Studio Sound*.

The *LC* System

Optifile LC is actually the fourth automation system to appear under the *Optifile* banner —*LC* simply stands for *Low Cost*. Considering the current top of the range *Optifile* system (*3D*) sells at nearly double the price, it is surprising to discover just how many similarities there are between the two products.

The major differences are that the *LC* does not include local Status switches, it uses a standard keyboard rather than a dedicated one, does not include off-line mix functions, and does not offer a machine control option.

The VCA boards inside the console have been specially designed by Soundcraft in collaboration with AD Systeme, and although electronically identical to the original *Optifile* design, the PCB layout has been changed to accommodate factory production and to minimise VCA noise and distortion. All VCAs may be bypassed when the automation is not in use or has simply not been fitted.

Three additional components are required to automate the console—the LC 19-inch rackmounting 1U processor unit that contains a 3.5-inch floppy drive, a mini PC keyboard and a 14-inch colour monitor. The console is supplied in 20 to 52-channel frame sizes, while the standard *Optifile LC* will control up to 40 channels, however this can be increased by adding a 40-64 channel-extender card.

The system controls all the long throw faders on *Sapphyre*, which are designated as monitor faders, and their associated mutes. The resolution is identical to *3D* offering an accuracy of 0.19dB (4096 steps), with both faders and mutes being accurate to one frame. Existing mute switches are utilised, while status switching is controlled from the keyboard.

The system avoids the use of a mouse or trackball but instead operates with 16 dedicated function keys, and a possible total of 26 macros which are accessed from the alphabetic keys (ALT A–Z). The system comes supplied with 13 factory programmed macros, but any of these may be overwritten by the operator.

There are just two screens the Main Screen and the Mix List Screen. The Main Screen displays up to 20 channels at a time showing fader levels and mutes plus independent status boxes for each. VCA Group information is displayed with slaves and masters being highlighted in light blue and dark blue respectively. The system provides up to nine VCA groups which can be used independently of the automation system—this is in addition to the console's four mute groups and audio subgrouping facilities.

Three time-code windows are included on the main screen; the largest displays the current position and will flash to indicate a format mismatch or code read error, while the two smaller windows above it indicate in-out registers for

programmable auto functions. The system reads all standard time-code formats, but does not include an onboard generator.

Also on the Main Screen is a horizontal bargraph which acts as a RAM used gauge; a window displaying the current operating mode (Record or Rehearse); a command-prompt field; and in case of amnesia, a panel to show the name of the studio. Unlike the *Optifile 3D*, the LC does not include a Menu Bar at the base of the screen.

Operation

Before the system can first be used there are various setting up procedures that must be carried out such as setting the correct time-code standard, entering date and time, fader calibration, disk formatting and so on. The programme itself is stored on two EPROMs, and prior to mixing, the computer must be initialised by marking the mix boundaries either by entering time-code values or by using the HERE key to capture time-code points directly from tape. These points act as finite limits outside of which the system's computer will not respond; the large time-code window will be bordered in red to indicate that the current position is within the mix area.

Status selection may be independently set for faders and mutes either globally or for individual channels; the four modes Read, Write, Update and Isolate are all available for faders, while Update is not applicable to mutes.

At start-up, status is globally set to Isolate. The procedure to switch the entire console to Write is to key in 'ISOLATE' followed by 'WRITE'. If just Faders or mutes are to be switched to Write, the FADER or MUTE key is added at the start of the command line, and similarly if individual channels are to be specified, actual channel numbers or groups of numbers are included.

Once channels have been removed from isolate status, they will respond to commands such as FADER UPDATE which switches all faders to Update, or MUTE WRITE FROM 01 to 10 which sets the mutes on channels 1–10 to Write.

There are two other keys that affect the way status operate—RECORD and DROP. The RECORD key toggles the system between Record and Rehearse modes thus allowing moves to be auditioned before writing them; there is also an Auto-Record mode that provides a means of accurately programming Record-In and Record-Out points which will follow Record-Rehearse selection.

The DROP key acts as a status safe-enable switch and will toggle status between an active and deactive state. Thus, channels set to either Write or Update can be dropped into record and out from a single switch. The function is taken a stage further with the Auto-Drop facility which automatically switches channels into record at the point the tape was stopped and wound back from. Both Auto-Drop and Auto-Record time-code positions are displayed on the Main Screen.

Currently the only means of nulling faders is by visually matching levels on the screen, the bargraphs display mix levels while the fader heads represent their absolute positions. To help match levels, the fader-head icon will change colour at the null point.

Optifile LC operates with a double-buffer system a Base Buffer and a Temporary Buffer. This arrangement allows mistakes to be simply overwritten by running the tape back and dropping-in at the appropriate moment. This is not the case with some other low cost systems where a

single buffer design restricts the user's choice to either keeping or aborting the whole pass—which is not a very satisfactory arrangement during Update passes.

During the first pass data is written to the Temporary Buffer; modifications made in the second pass are also added to the Temporary Buffer while the first pass is simultaneously copied to the Base Buffer. This procedure results in a current and previous mix being stored in RAM. However, during Update the computer calculates moves relative to the Base Buffer and stores the sum of the previous and current data to the Temporary Buffer—obviously to operate correctly the Base Buffer must contain the current rather than the previous mix. To address this, the Base Buffer must be made current prior to an Update by either Keeping or Saving the mix. The Keep command simply copies the Temporary Buffer to the Base Buffer, and Saving the mix to disc has the same affect. When data in both buffers is identical the memory used display will be blue, as soon as a change is made the display turns red warning the user that unless the pass is kept, data will be overwritten. The Temporary Buffer can also be deleted with the Undo function—thus allowing the entire last pass to be erased.

Mixes are saved to disk under a Title which is entered during the initialisation process—just one Title is permitted per disc. Mixes can be optionally named (21 characters), otherwise they are entered into the Mix List by number along with a time and date entry. Mixes can be renamed, and deleted in groups (that is delete 1 8). Group data and channel status is stored along with a mix, while Initialisation Registers are stored with the Title. Typically the system can store between 40 and 50 average size mixes on one disk; if it becomes necessary to continue on a fresh disc, a Copy function will transfer over the last saved mix.

Mixes belonging to the same Title can be directly compared by loading a mix from disc into the Base Buffer, while globally setting the console to Update and Rehearse. In this mode the DROP key will instantly switch between the stored mix in the Base Buffer, and the current mix in the Temporary Buffer. The function can also be used to compare individual channels by selectively setting Update. If the procedure is implemented using Record rather than Rehearse, the function becomes a mix merge facility for which Auto-Record can be used to perform precise edits.

Conclusion

The marriage of the *Optifile LC* and *Sapphyre* will obviously be mutually beneficial to both manufacturers, and indeed this is already proving to be the case with Soundcraft reporting increased sales. However, the main beneficiary is the customer who can now buy a truly versatile, quality package at a significantly reduced price.

The two products complement each other very well, both being easy to operate while offering a wide range of features and a degree of professionalism that surpasses their price. ■

Thanks to Sonic Studios in London for providing us with the time to look at their Sapphyre LC.

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THE POWER CIRCUIT

Ben Duncan analyses how the mains power supply can affect monitoring systems

The quality of the AC mains supply for studio amplifiers has received marginal attention in pro audio. Yet with most monitoring systems using Class A-B amplifiers, it has a significant, even major effect on sonic quality, particularly when high power is needed. It is also straightforward to improve upon.

The current draw of nearly all power amplifiers is higher than elementary knowledge predicts, regardless of whether they are working in Class A-B (most), or A, G, H or D.[4] A primary cause is the need to get smooth DC from the 50–60Hz 'sine' wave. The most 'efficient' way is by capacitor smoothing; it is found in almost all of present day audio equipment supplies. In the future, we will see more power-factor-corrected supplies, but for now, reservoir capacitance in amplifiers is largely responsible for a peak current that is typically between three and five but also up to 20 times the rms level. In AC power engineering parlance, it is called a bad *power factor*. It happens because reservoir capacitors preferentially charge at the mains peak voltage, so the current draw is pulsating—charge transfer is 'bunched up' within a couple of milliseconds in every 10ms (8.3ms for 60Hz).

The same is true of most of the DC electronic equipment supplies that are presently connected to the mains. The outcome of the use of millions of electronic consumer goods is that the national mains waveform ends up flat-topped; the richest part of the supply has been robbed. So 240V rms, for example, rarely if ever peaks at the 340V (240V rms x 1.414) dictated for a pure sine wave. A typical peak reading in my lab is 310V to 320V. The shortfall
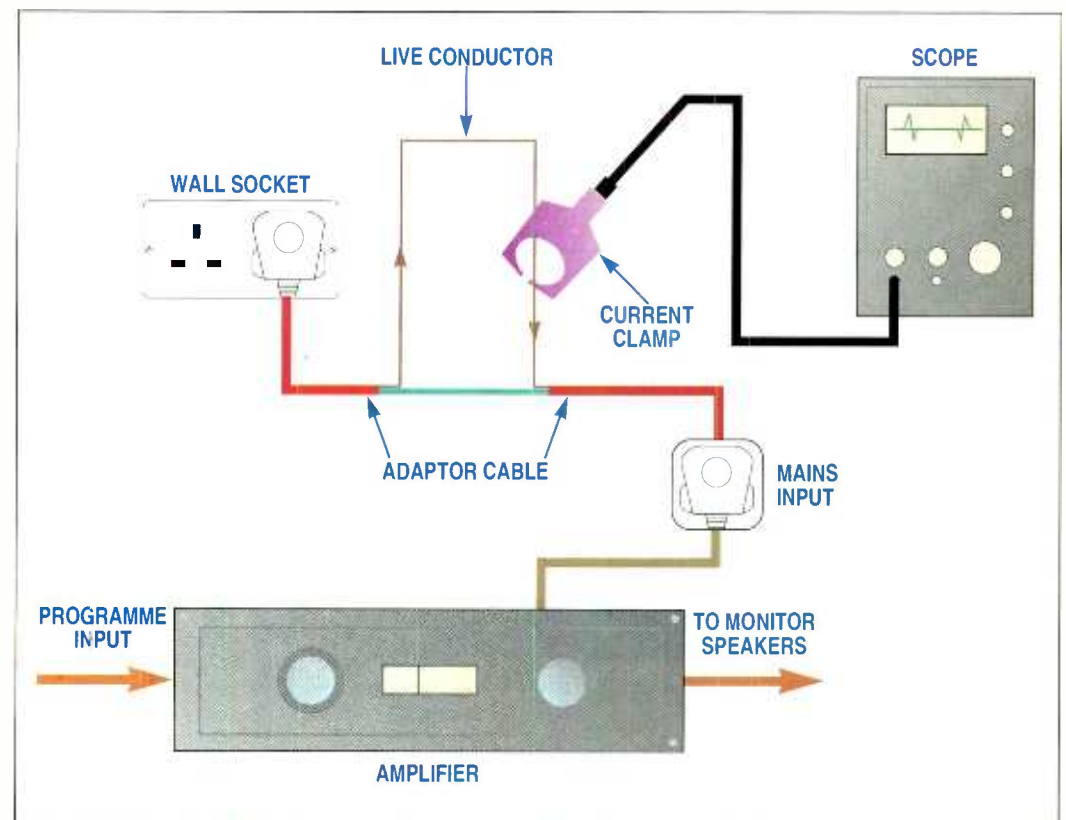
Fig.1: The peak current drawn by amplifiers can be measured and viewed with a 'scope connected to an current clamp placed over one conductor. A break-in adaptor lead with a 'split out' live wire makes this easy. With a current transformer clamp, an AC-milliamp meter must normally be connected in parallel, to show the true waveform. Scaling is typically 100mV/A. A basic scope will do, but the voltage sensitivity of the setup must be calibrated, for example with a test resistor across a 50-60Hz supply. An 11.35V rms 50Hz slate tone from a power amplifier connected across an 8Ω, 16W resistor gives a 2A reference peak current

directly limits the peak voltage to which the amplifier supply can charge, which disproportionately restricts the maximum output swing, probably when the amplifier needs it most. This is called 'brown' power.

The flat top also amounts to distortion; the mains waveform now contains harmonics. The effects of monitoring amplifier(s) on the mains purity may be minuscule on the scale of the outside world, but adds to the problem locally, within the studio. For a start, locally produced harmonics of 50–60Hz are not as attenuated as well, as more distant sources.

How much current?

Considering monitor amplifiers' supply current draw in more detail, maximum demand is a combination of the current magnitude and the Peak-to-Mean ratio (PMR). Beginning with the absolute magnitude of rms (or peak) current drawn by amplifiers, this is a product of power output and efficiency, hence rating, average speaker impedance and class of amplifier associated with the power supply. Classes G and H draw less current than Class A-B under hard drive conditions, but all three draw a varying peak current that is dependent on the music (particularly bass), ultimately modulating the mains with audio (**Fig.2**). Class A amplifiers draw the highest currents, typically between 200% to 900% of those drawn by everyday Class A-B amplifiers under varying drive conditions. This should be evident from the electricity bill. If the Class A tag is truthful, peak mains current, while pulsating, should be otherwise invariant over time. ▶

Footnote: In the UK, there are millions of TV sets, 'music' systems, and microwave ovens with largish DC supplies. Alongside these the percentage of the mains waveform distortion that is caused by all the studio monitoring and PA systems in the country is insignificant—even if all are driven at maximum power simultaneously. Surely this is sufficient argument for excepting them absolutely from impending EEC legislation on mains harmonic generation?
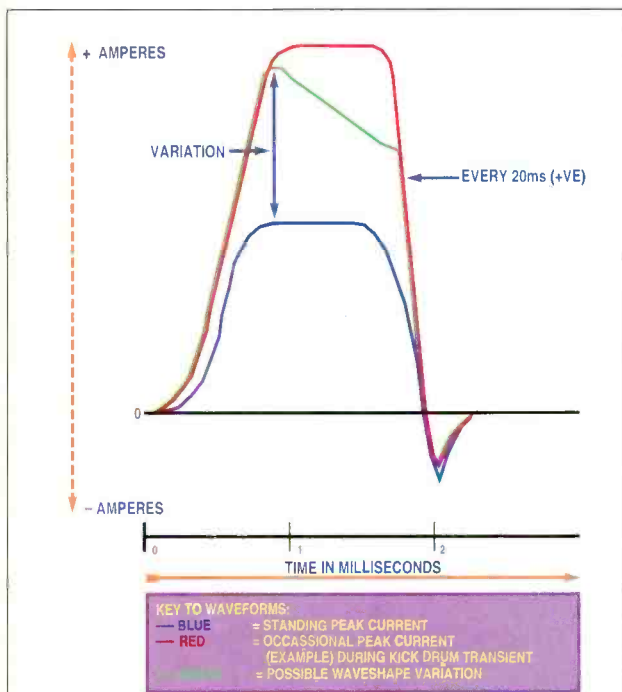
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Fig.2: The peak current waveform can vary widely; on a badly flat-topped supply it is almost rectangular, and increases little in its duration while increasing greatly in magnitude (height) with transient demands

Class D and other PWM 'digital' amplifiers, when they occur, draw much less current than any others, but then RF noise escape is an issue.

The PMR of amplifiers' current draw depends mostly on the supply reservoir's size (in μF) and in conventional, passive supplies only, on the transformer size and construction. The likelihood of a high PMR is increased by generous supplies, and by a low mains supply impedance, which depends on the gauge and distance of the studio supply wiring, rating of the incoming feeder (100A is the recommended minimum for a recording studio), and the distance to the distributor's substation transformer.

As a rule of thumb, most amplifiers rated below 100W have low absolute current demands, and with 'music power' supplies and thin wires they also have low PMR, circa 10dB/x3. Excepting Class A, the highest peak currents will be small enough to be handled by a quite light duty supply, like a 30A ring or 20A spur, without too much influence on other equipment. But higher powered amplifiers with beefier supplies and output stages naturally draw increasingly high peak currents, with Class A out ahead, followed by Classes A-B, G, H and D. But amplifiers of any rating (excepting elderly bipolar amplifiers with overzealous V-I limiting and Class A) with large, low impedance supplies and high current-capable output stages will draw occasional high peak currents when connected to current hungry loudspeakers—all the more so when driven by a dynamic, transient-rich programme. With all the existing kinds of power supply, high mains peak currents are a necessary evil for good sonics. Speaker currents exceeding 100A have been measured;[1] this figure may be

limited to the more esoteric monitoring devices, but sinking tens of transient amperes into bass drivers is not uncommon, and it has to come from somewhere.

Exactly how much peak current does a monitoring system draw? With so many variables, there is no substitute for measuring the peak current (**Fig.1** outlines how). By way of example, measurements were made with a C-Audio *TR-850* amplifier driving a 15-inch driver-cum-horn monitor, bridged with a potential capability in excess of 1500W into 8Ω, and driven with techno music, at levels peaking up to clip. Mains current regularly peaked at 4A with the (sampled) kick drum, and occasionally peaked at well over 10A per channel while the nearfield SPL reached 123dB$_{SPL}$, 'c' weighted. With intricate percussive bass guitar, the occasional maximum peak mains current was slightly less, at 8A per channel. Behind the peaks was a steady current peak of 1A. The mains current waveform only became modulated by the program when the drive level was enough to exceed this 'standing' current. With Class A-B amplifiers, this occurs typically above -20dBR. As few can afford to

use amplifiers at below ¹/₁₀₀th of their rated power, regular modulation of the studio supply by the monitoring system is on the cards. **Fig.3** shows the effect; merely by being plugged into the same ring main, an amplifier with a modest noise floor (A) is picking up highly distorted garbage from another amplifier (B). Other than forming part of a biamped or triamped monitoring system, the amplifier could be providing delicate foldback or it could be in adjacent control room, again handling delicate material. In either case, sonic quality may be marred.

While most monitoring systems' total rms (hence heating) current is likely to be safely within the capability of a 30A ring main or 15A spur, and no fuses will blow, the peak current will cause a train of voltage drops or 'sag', in keeping with the ▶

> **sinking tens of transient amperes into bass drivers is not uncommon, and it has to come from somewhere**
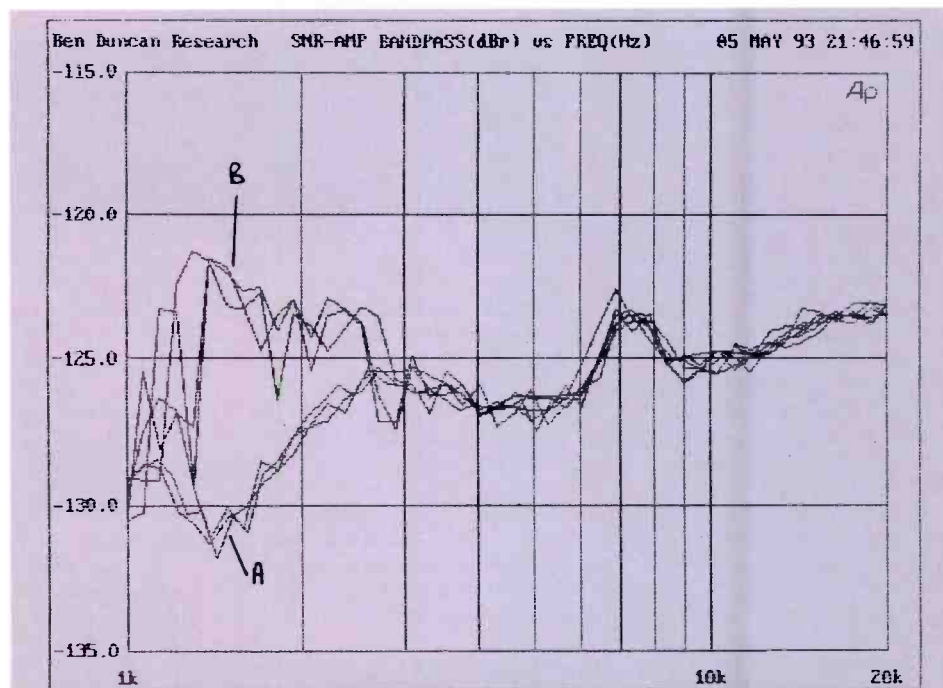
Fig.3: The noise floor of an amplifier (A) is increased by 3–8dB in the 1–3kHz region (B) where hearing sensitivity is greatest, by mains modulation caused by a second amplifier supplying high level programme into a bass driver, connected to the same ring mains supply. Each condition was swept three times

Fig. 4: The 30A ring supply (universal in the UK for domestic and light industrial premises) is unsuited to powering technical equipment, especially audio

programme. Even without music modulation, this will limit the amplifier's own performance, that of its colleagues, and may even cause other (particularly digital) equipment to mysteriously malfunction. Music modulated sags may also last longer than the stimulus—they can have a tail. The answer is heavy-duty house wiring derived directly from the incoming supply, to reduce voltage drops, hence self and mutual interference.

Wiring practice

In many installations, but only in countries adopting the custom, the mains power for domestic and light commercial buildings is wired in a ring (Fig.4). This saves copper; the idea was originated in the UK in 1939, when ships carrying copper ore were at risk from U-Boats. However, to avoid excess voltage drops and modulation effects, the AC power for studio equipment must be fed from a dedicated radial spur cable, taken directly from the incoming supply, where impedance is lowest (Fig.5). This technique of running individual cables to equipment from the lowest impedance, nodal point is analogous to 'star earthing' and virtual earth mixing. Like these, it helps each source reject external influences, and reduces the sources' own influence on all the other circuits. This means:
(i) sag (including music-modulated voltage fluctuations) caused by the amplifier(s) and experienced by equipment on other circuits will be attenuated.
(ii) the amplifiers' reception of interference from other circuits (fridges, thermostats, lifts, air conditioning and so on) will be attenuated, and these noises quite possibly done away with altogether. Typically, a spur's interference-cum-disturbance attenuation ranges from a modest (but still welcome) 8dB, up to 40dB at audio (10Hz–20kHz), and more at radio frequencies.[2] By contrast, ordinary mains filters are solely focused on RF: they provide nil protection below 200kHz, and come in strongly above, in the MHz realms.

For these reasons, radial spurs are the norm for the technical supply in most competently wired studios. If you have mains problems, it pays to

> ordinary mains filters are solely focused on RF: they provide nil protection below 200kHz, and come in strongly above, in the MHz realms

check. Installations where the monitoring or foldback never quite sounds right or changes daily are likely suffering from a congested ring main.

With a multiamped monitoring system, powering the amplifiers from individual, exclusive spur cables, one per amplifier, are worthwhile for the closest approach to a clean supply. It is best to err on the side of generosity with the cable size, as even 10mm² meter tails do not incur a great deal of added expense alongside the cost of cutting out and replastering. And even if you do not have current-hungry speakers or particularly current-capable amplifiers right now, you might have later. Cable choices may be compared by a voltage drop benchmark. For example, with a worst case 50A peak current (in large systems), cable resistance will need to below 10mΩ for below 1V mains peak deviation. At this level, a 2.5mm² cable would be limited to a run of 2m, whereas 10mm² meter tails would allow nearly 8m. When planning a new studio or a revamp, it is advisable to ▶

POWER FACTOR CORRECTION AND OTHER ESOTERICA

Bad power factors are corrected by controlling the current drawn from the supply. Repeated current peaks of varying size with nothing in between are replaced by a current waveform that changes smoothly—a sine wave.

Agile mains-connected electronics are needed to compensate for such a dynamic load as a power amplifier, and the majority of existing power amps cannot be easily modified. However, EEC and other EMC regulations may eventually make them a requirement—at least for future designs. Phased in on a global basis, power factor correction is 'green'; it will save energy and make the optimum use of existing utility power plants.

Anyone planning a serious studio rewire may like to consider the procedure adopted by a number of keen recording engineers and audiophiles who have equipped dedicated listening rooms. Unswitched outlets must be used as there are fewer contacts to cause problems. Linked conductors should be soldered and not simply clamped by a terminal screw. A torque-driver should be used to set the optimum tension on all brass terminal screws. Live and neutral conductors should be twisted to enhance RF rejection and reduce 50–60Hz radiation into the room —this means using individual tails laid in trunking.In the UK, 13A outlets should be used; the older 15A round-pin variety (surprisingly) have a higher contact resistance. All the screw terminals should be checked a few months after the installation to ensure that they are still tight. If the mains is 95–115V, the problem of mains-circuit resistance and interaction are at least twice as intense as in 220–240V territories, and it may be worth considering stepping up the entire technical supply to 240V—a practice observed by a number of US studios. ∎
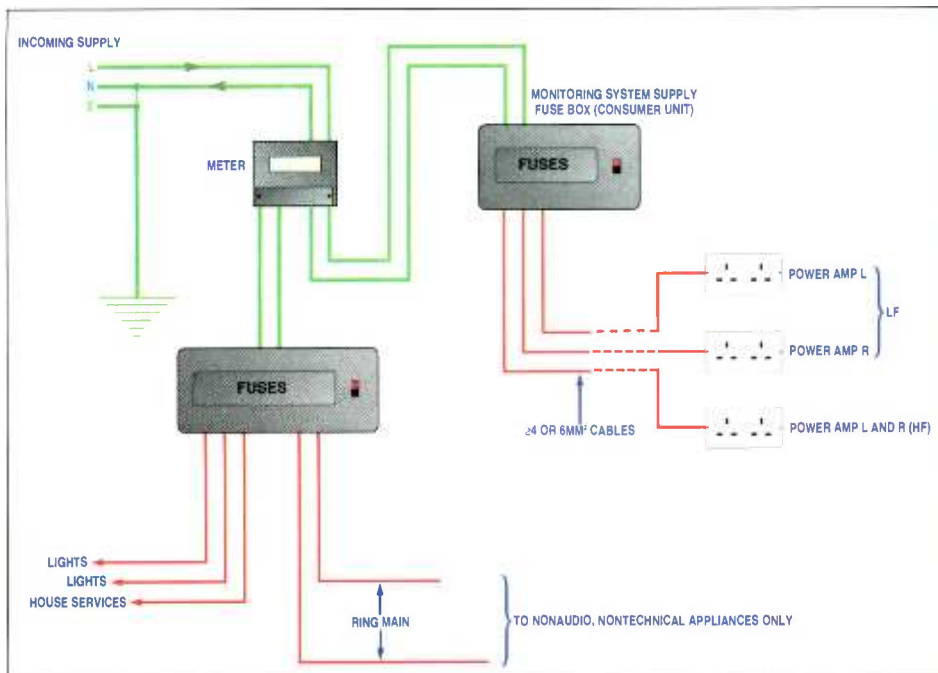
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Fig.5: In a professionally executed studio, every power amplifier and all other major equipment is supplied by an individual spur noded from the incoming supply

organise the incoming supply and the amplifier cupboard to be adjacent for the highest results-expense ratio. By increasing the potential for high peak currents, dedicated, low resistance spurs accelerate the recharge of Class A-B amplifier's reservoir capacitors after power-hungry transients. If several amplifiers must be powered from a shared spur, parallelled cables are recommended—and yes, expect strange looks from the electrician as they contemplate your request for >100A-rated cabling for a load that is under 2kW, and the quaint idea of radial spurs from the incoming supply, last used in the UK in 1949. The explanatory keyword is 'headroom'. The mains in Tokyo is generally regarded as the world's most polluted. In this instance, and wherever AC power is excessively brown, interference ridden, or has erratic voltage or frequency, a reputable studio installer will propose power conditioning plant which may range up to dedicated substation or even regenerating clean 50-60Hz on site.

Rms or peak?

To measure the rms voltage of the mains, you must use a meter with *true* rms sensing. A meter that reads average (even if it is calibrated in rms) will misread because of the mains' waveform distortion. However, the rms reading is only useful for gauging average power consumption and the PMR. For calibrating supply voltages and making close power measurements (on the majority of amplifiers, with unregulated supplies), the mains peak voltage is a better point of reference, as the DC supply charges to this level. Up to now, the supply is usually set at 240V rms when testing an amplifier. If the tests are broken off, then continued an hour or day later, then in the interim, both supply voltage and waveform distortion will have changed. So now the same rms voltage (240V) no longer provides the same peak voltage. The power rails could be now 5V different and the amplifier's spec, particularly its power rating and

clip point, will have changed. When studio amplifiers are tested in production on an unregulated supply, mains fluctuation adds to the variability of the test data.

Conclusions

All power amplifiers affect and are affected by the mains supply. Class A amplifiers are least affected, but cause the most continued disruption to other equipment sharing the supply. A limited number of modern amplifiers have regulated supplies. They are largely immune to performance variations caused by supply voltage and waveshape,[3] and while this is not automatically conducive to good sonics, it helps a great deal. But in all cases, the benefits of a dedicated, preferably low impedance, diversified, spur supply are clearly evident to the ear at higher playback levels on even the most modest monitoring setups, but are all the more apparent in high performance, audiophile grade systems. ■

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4. Ben Duncan, 'Which amplifier technology?', *Studio Sound*, December 1988, January–March 1989.

Further reading
Manuel Huber, 'Important Aspects of Power Amplifiers', *Studio Sound*, November 1985.

TERMINOLOGY

Brown: Short for 'Brownout', a mains voltage that sags for a prolonged period and has a thoroughly squashed peak voltage.
dBR: dB below an arbitrary Reference, in this case, an amplifier's full output.
EMC: Electro-Magnetic Compatibility, essentially the keeping down of RF interference and mains harmonics.
PMR: Peak-to-Mean Ratio, called 'Crest Factor' in electrical engineering, the difference between the peak and rms values in current or voltage waveforms.
Peak: Refers to the 'top of a sine wave', not to be confused with programme peaks, here called transient or loud passages.

Ben Duncan is an international audio consultant, analogue equipment designer and author. He has been involved in the design of over 70 products, and has published nearly 400 articles and papers. He is deeply committed to the refinement of sound quality for both recording and live performance

PPM5
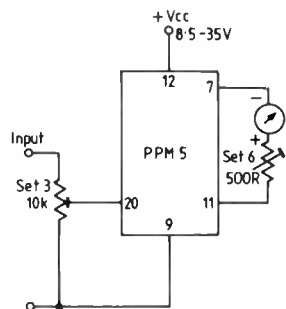
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A slow migration is going on. Established recording studios are closing, moving or opening their doors in cities previously outside the music tradition. If these moves continue throughout this decade, it is possible that Chicago and Nashville could eclipse LA and NYC for recording music and producing commercials.

In New York's Manhattan, the high prices of commercial real estate coupled with limited availability of large, unobstructed spaces promise to diminish New York's long held status as a recording centre. One studio owner complained, 'I can barely crack my financial nut each month, what with paying for new equipment and the cut throat competition in studio rates. When my lease expires, I don't know what I will do since I couldn't hack the going rate for space around here today!'

By now, the fate of one of America's great recording studios—well steeped in its own history and with it's own unique sound—has been sealed. The RCA recording studio in Manhattan has been forced from its quarters by that 1980s–1990s scourge: real estate development. At least other major studios similarly threatened have been able to move to new quarters, but the scarcity and price of real estate in New York City made this a task that could not be achieved for the RCA complex.

The RCA studio is the only US facility so dispossessed in the last 10 years but it is certainly one of the most significant—responsible for hundreds of successful projects on disc and on film. As many as half of the mainstream 'mothership' studios that have disappeared from our musical universe over the last 10 or 15 years were forced out of downtowns or other trendy spaces too valuable for music recording activities.

The history of the music recording business in America saw the 'central city' location develop as the location to choose for most radio and recording studios in the 1940s, 1950s and the 1960s. Musicians were traditionally, as Petula Clark put it, 'downtown'. Orchestras played in central city auditoriums and concert facilities, while downtown nightclubs offered big and small band entertainment. The advent of television found broadcasters located in or adjacent to their former radio studios in the central city. Everybody who wrote songs or scores or lyrics, recorded albums or broadcast music was downtown because that was where metropolitan cultural life was focused. Music satisfied the prodigious demand for entertainment by downtown dwellers who served as a captive audience.

The most important change in America after the end of the Second World War impacted recording studios as much as it did urban development. It was the flight from the central city, as the development of suburban and exurban communities around and away from downtown continued apace. New Yorkers deserted the Big Apple for bedroom communities on Long Island and in Connecticut, New Jersey and upstate New York. Los Angeles vowed to do the same, as urged on by cowboy star Roy Rogers, 'and make the San Fernando Valley my home'. The industrial fabric of manufacturing and service businesses such as printing that were centred around these hubs also

Martin Polon

Trading places: the great migration of American studios

made an exodus for the lower costs and lessened congestion of the 'burbs'.

Initially, this shift in demographics from one geography to another actually helped the recording studios—allowing existing studios to move to larger quarters in now empty real estate, and new studios to establish themselves in 'transitional' neighbourhoods. But that was before Ronald Reagan and the business boom of the go-go 1980s. The central cores of Atlanta, Boston, Denver, LA, NY and San Francisco—some of America's most important music cities—experienced a tremendous boom in high-rise and other real estate developments.

Suddenly, those recording studios that had continued to thrive through the ups and downs of big city life were threatened by the removal of the very spaces that they had occupied for years. If their building was not targeted for demolition to allow the construction of an 84-storey apartment tower, the landlord would decide that a higher rent could be obtained by bringing in an office client.

In the case of the RCA studios, a landmark Manhattan recording facility that had been optimised for orchestral and film soundtrack sessions at the end of the 1960s and which saw virtually continuous use into the 1990s, office space overcame audio recording. The studio complex which had been modernised in 1989 to the tune of three million dollars and more by the Bertlesman Music Group (BMG) as the successor to RCA—with its irreplaceable large studio spaces—was preempted by the building's management.

What is especially ironic is that recording in that famous and unique RCA facility got the heave-ho so that an important client of the building's management could find additional office space. That client is the Internal Revenue Service of the United States of America. It is as though the Inland Revenue preempted EMI's Abbey Road studios to be converted into regional London offices to conduct tax audits. The loss of the RCA-BMG complex at West 44th Street was especially vexing to the BMG studio operations people, who by all

'Between crimes, riots, gangs, earthquakes and fires, it's like Bladerunner out there.'

reports had expected to continue to use the studio into the foreseeable future.

Another take on the relationship between studios and cities and real estate is the yin and yang of LA and Nashville. No other city is experiencing such a minor exodus of its music industry like LA. There has always been a substantial country music subculture to LA's rock 'n' roll scene and a supportive recording, A&R and publishing community to go with it. Of late, a number of studios and other related businesses have picked up and moved their entire operation to Nashville. Now it seems that Nashville is experiencing a kind of mini boom as studios are moving in from elsewhere to Music Row, to settle on or near streets with names like Chet Atkins Place. To some extent, Nashvilles' success mirrors that of country music—which has posted gains in record sales from around half a billion dollars at the end of the 1980s to a projection of two billion dollars by the end of 1993. Similarly, radio stations playing a country music format in the US have grown from less than a hundred in 1960 to well over 2,000 today. Internationally country music is picking up listeners from Slough in England to Shiba Park in Tokyo.

The exodus from LA is not limited to those involved with music, since a recent survey revealed that over 60% of all LA residents wish that they could leave the city. Those in the music and recording industry making a drastic relocation cite other issues in their decision making process besides those of business. Quality of life issues dominate the thinking of many who have packed up their studio operations to move to Nashville and elsewhere. The problems of running a studio and living in LA were summed up by Chas Sanford, who moved 29,000lbs of studio equipment from LA to Nashville at the beginning of 1993. He told the New York Times that, 'between crimes, riots, gangs, earthquakes and fires, it's like Blade Runner out there. I'd say a quarter of the people I know have moved and a lot of them have come to Nashville where you don't see graffiti every time you open your eyes, and you don't hear helicopters over your house all the time.'

The most difficult issue for studio owners is finding a location that will meet with the favour of recording clients and is still central to musicians, instrument rental services, equipment vendors and the other service providers needed by a busy studio operation. Some operators have tried using the prefabricated warehouses that are clustered in industrial parks in industrial cities on the edge of major downtowns. It has worked for some operators in a place like Los Angeles, where the industrial town of Burbank houses NBC and Walt Disney studios as well as being adjacent to Universal Pictures. But for others, setting up a studio in a converted warehouse in Long Island City, NY lacks the feeling of excitement that being in Manhattan gives. The groups would rather record where there are restaurants and adjacent nightlife. It would appear that for recording studios as well as for most other kinds of businesses, there are indeed three rules of real estate selection. 'Location, location and location.' ■

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POT OF GOLD

Crookwood was established in July 1992 by Crispin Herrod-Taylor and wife Penny. Herrod-Taylor began his audio design career with Boothroyd-Stuart Meridian (a noted hi-fi company), before moving to SSL and latterly Focusrite, where he was responsible for the design of much of the current Focusrite *Studio Console*. Initially he supplemented his income during the company startup phase by undertaking technical writing—his notes to me were excellently written, so obviously he is good at that as well as electronic design.

The new company was formed to produce interesting and sonically superior equipment for the music industry, marrying purist analogue design with digitally controlled convenience. The ultimate goal is a set of stand-alone products which provide the basis for what I would describe as a 'customerisable console design'.

The purpose

The mic preamp's name comes from the similarity of shape and size to a 5-litre paint tin—including handle. In Herrod-Taylor 's words it is this shape because, 'I wanted a challenge, and because of its heat dissipation properties—*Paintpots* run hot due to the number of relays and semi Class-A output stages'. It suits Crookwood's desire to be 'interesting'.

Paintpot is intended to provide amplification for audio sources, particularly microphones, as close to the source as possible, and to provide a drive to the cables which will minimise quality loss, also enabling a tape machine to be driven directly —avoiding all of the electronics stages present in a typical mixing desk. It is optimised for microphones, but is also suitable for use as a DI box for instrument pickups, or electronic sources. The unit has two independently adjustable channels, with a switch selectable, in-built M-S decoder linking them if required.

Paintpot should provide the ultimate in performance for purist recording.

Construction

Paintpot is formed of two vertical extrusions supporting top and bottom panels, all of double anodised aluminium. The panels are finished in gold with blue lettering, easy to read

The Crookwood mic preamp takes its styling from a pot of paint

and high in contrast—should be good in low light conditions. Most controls are on the top panel, supported on a fibreglass PCB, mounted in parallel behind it.

Mounted vertically within the unit, a pair of PCBs provide the input amplifiers and attenuators on one side of the housing, while a third contains the output amplifiers which are screwed to the vertical extrusions as a heatsink. A final PCB beneath these houses the power supply, with a high-quality toroidal transformer mounted on the bottom plate.

As is evident from the accompanying photograph, the unit is circular, with side panels fixed to the vertical extrusions by allen-head screws. These panels interlock with the top and bottom plates to give

structural integrity.

With two exceptions, the construction would appear to be robust and likely to survive when carried around as a 'paintpot'. The first involves the pillars which support the PCB assembly, these could usefully be increased in diameter and made in one piece, rather than the present construction which involves joining several smaller pillars in series. The other is one bit of wiring where vibration might eventually cause failure. Crookwood are looking into these areas for the future. but claim no field problems so far.

Regarding safety aspects, all mains wiring is protected as recommended ▶

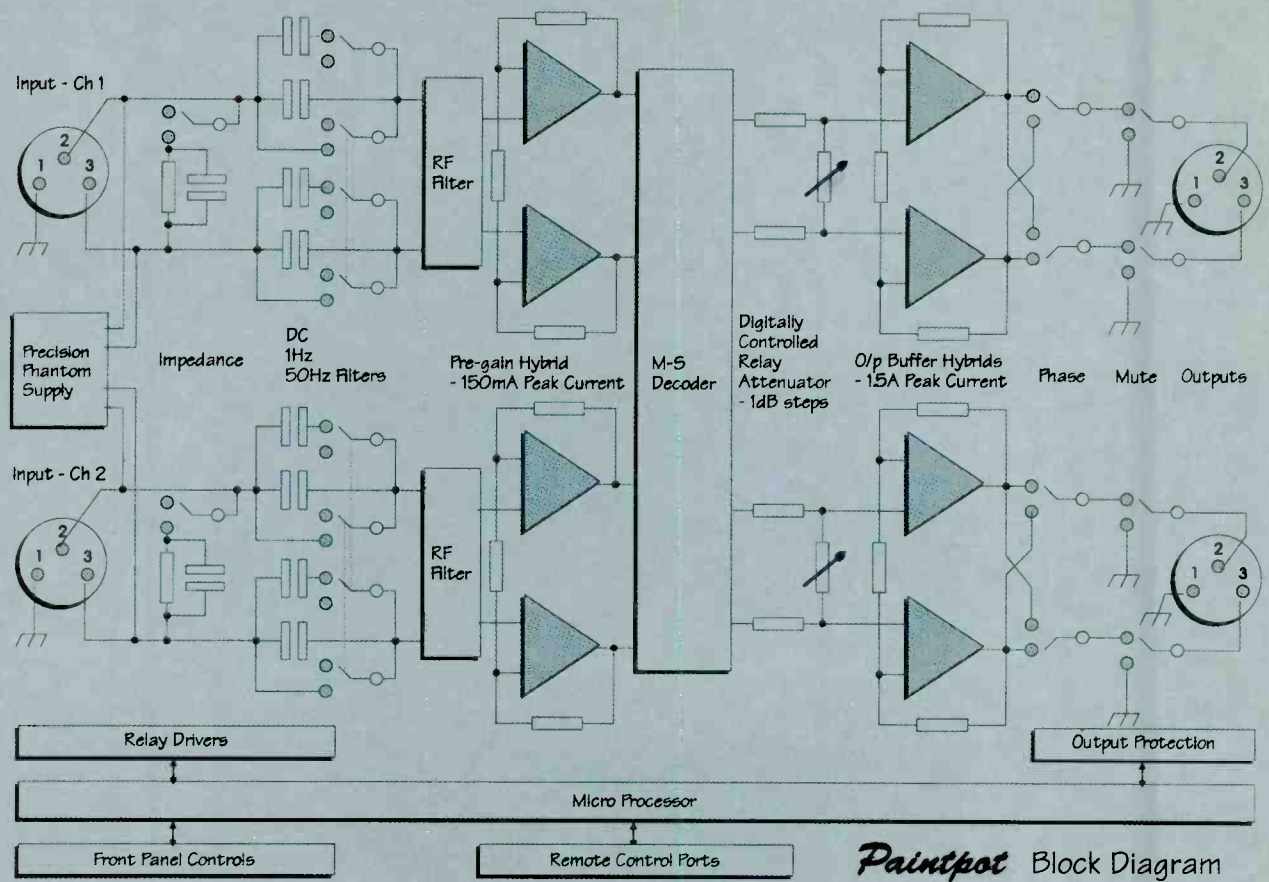## Sam Wise evaluates the Crookwood *Paintpot* dual mic preamp

65

Fig.1: Block diagram

Input - Ch 1

Precision Phantom Supply

Impedance

DC 1Hz 50Hz Filters

RF Filter

Pre-gain Hybrid - 150mA Peak Current

M-S Decoder

Digitally Controlled Relay Attenuator - 1dB steps

O/p Buffer Hybrids - 1.5A Peak Current

Phase Mute Outputs

Input - Ch 2

RF Filter

Relay Drivers

Output Protection

Micro Processor

Front Panel Controls

Remote Control Ports

Paintpot Block Diagram

by BS415, making electric shock very unlikely. The power lead is removable, connecting to the side of the unit by IEC mains socket. *Paintpot* has also been designed to meet proposed EMI-EMC regulations, which will hopefully benefit the user by being resistant to external magnetic fields, and also will not interfere with other items of audio equipment in use.

All screws are well retained, using nyloc nuts or loctite to prevent unexpected disassembly.

A leather carrying handle on the top completes the unit. This is folded down just like a paintpot handle when the unit is in use.

Paintpot literature emphasises the use of high quality, carefully selected components throughout the unit, which inspection verifies is the case. The manual is basic, but well written and helpful to purchasers of what is essentially a high performance but simple specialist product.

Operation and performance

Except for the mains switch which is on the bottom to prevent accidental operation, all controls are on the top panel, making for easy operation when stood on the floor or other surface.

All controls are mechanical switches, which are read by an internal microprocessor. This in turn is used to

switch relays to perform the actual electrical switching. This has several advantages. First, the unit can be remote controlled by an RS485 serial link, secondly, the relays can be placed optimally on the circuit layout to give the best audio performance, thirdly, microprocessor control allows for intelligent control of the switching— reducing

clicks and other unwanted behaviour.

Fig.1 is the block diagram of the circuitry, clearly revealing the simple and purist basic circuitry. The circuit is essentially a balanced instrumentation amplifier which is DC coupled. This means that the low frequency response is excellent, and performance has a good chance of ▶

MANUFACTURER'S SPECIFICATION

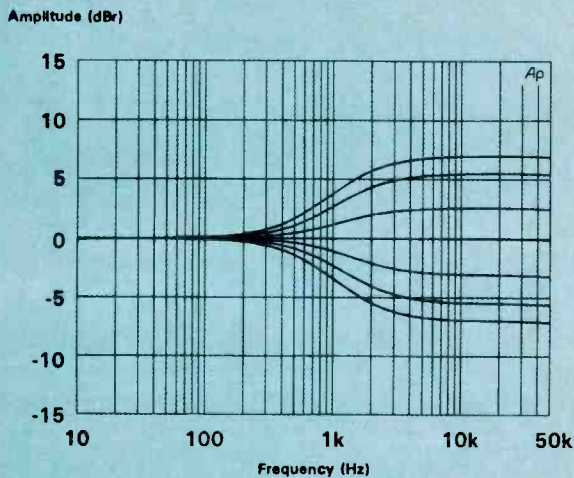|	
Inputs:	**Electronically balanced and floating**
Impedance:	high=10kΩ in parallel with 470pF low= 2kΩ in parallel with 3.3nF
Input clipping point:	+16dBu (Pregain at -20) -2dBu (Pregain at 0) -20dBu (Pregain at +20)
	Common Mode Rejection Ratio: >60dB (20Hz to 20kHz, any gain)
	Phantom power: 48 volts
Outputs:	**Electronically balanced and floating**
Impedance:	<50Ω
	Output clipping point: +27dBu (600Ω to 100kΩ load, 20Hz to 20kHz)
Frequency response:	better than ±0.03dB (DC to 20kHz, any gain, ref. 1kHz) -3dB at 500kHz, 1st order response
Channel balance:	within 0.05dB (any gain, DC to 20kHz)
Gain accuracy:	±0.5dB, any gain, DC to 20kHz
Filters:	AC(1Hz): -3dB at 50Hz, ±2Hz
DC offset:	<2mV(after 20 seconds)
M-S rejection:	>50dB (20Hz to 20kHz, any gain)
Interchannel crosstalk:	<-85dB (20Hz to 20kHz, any gain) <-100dB (1kHz, any gain)
CMRR:	>60dB, 20Hz to 20kHz, any gain
TH+D:	<0.007% (600Ω load, 20Hz to 20kHz at +20dBu output, 0dBu in, 130kHz bandpass) <0.003% (100kΩ load, 1kHz at +20dBu out, 130kHz bandpass, any gain)
Equivalent input noise:	<128dBu (>30dB gain, 22kHz bandpass, shorted input)
Output protection trip current:	>300mA peak for 10ms

www.americanradiohistory.com

Fig.2: The effect of the TILT switch on frequency response

being excellent as well. The unit has no real ground, except at a tie point in the centre of a pair of feedback resistors, thus ground loop noise should never be a problem. A DC servo keeps the output centred to prevent any DC offset. A front panel switch selecting phantom power on introduces a high quality capacitor in the input, as does selection of the 1Hz or 50Hz filter positions.

Pregain sets the basic level of amplification to -20 or +20dB for the main input section. This then feeds a balanced passive attenuator giving 2dB gain steps calibrated from +24 to +44dB. This gives a total gain range from +4 to +64dB, including a central null position. In practice, the microprocessor is reading the switches, and disallows a gain change until the signal is at a zero crossing or below -30 dB in level, making most switching silent in operation. A neutral position in the Gain switch also allows Pregain to be switched in the same noise-free manner, enabling any gain level to be set while recording with little risk of detection. Accuracy of these controls is good, being within 0.3dB of setting at all gains up to 60dB. The 62dB and 64dB positions were not quite as good with errors of 0.5dB and 0.9dB respectively. Since this is not a measurement amplifier, these gain errors are not likely to affect users, and are better than what most people are using anyway.

The output amplifier of the unit is very well balanced and low in source impedance. When combined with the very large current drive capability, the unit is able to drive long cables with no quality loss. Short circuit detection is used to ▶
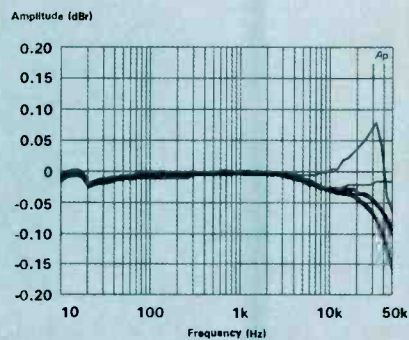
Fig.3: Frequency response with equaliser—OUT at various gain settings. Note full scale is ±0.2dB
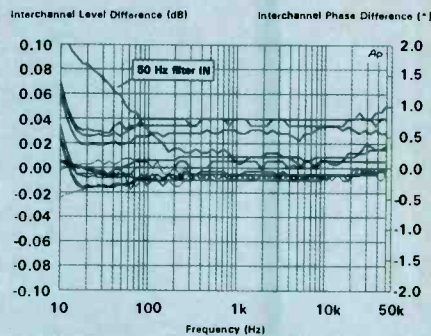
Fig.4: Phase and level difference with equaliser—OUT at various gain settings. Dotted lines are phase related
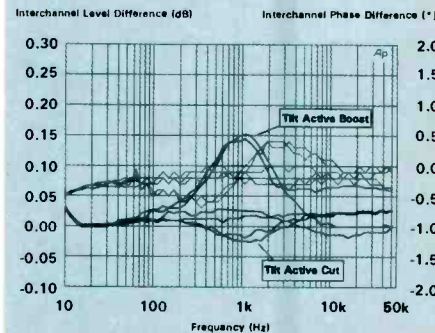
Fig.5: Phase and level difference with equaliser—IN. TILT control causes some small but negligible channel mismatch. Top curves are the maximum tilt positions. Dotted curves are phase

TABLE 1: INPUT CLIPPING LEVELS

PREGAIN SETTING	+20dB	0	-20dB
INPUT CLIP POINT	+20.1dBu	-0.1dBu	-18.6dbu

TABLE 2

	OUR MEASUREMENTS BEFORE THE WIRING CHANGE				CROOKWOOD MEASUREMENTS AFTER THE WIRING CHANGE			
	SHORTED INPUT		200Ω SOURCE		SHORTED INPUT		200Ω SOURCE	
GAIN dB	NOISE LEVEL dBu	EIN dB SOURCE	NOISE LEVEL dBu	EIN dB SOURCE	NOISE LEVEL dBu	EIN dB SOURCE	NOISE LEVEL dBu	EIN dB SOURCE
26							-94.0	-111.5
28								
30	-116.0						-91.5	-123.5
32								
34								
36								
38								
40	-84.7	-124.7	-83.4	-123.4				
42								

protect the output stage from potential self-destruction, with a warning led on the front panel should protection be activated. Unbalanced loads which short one leg to ground can be driven without damage or requiring any alterations to wiring or connectors.

Additional input controls are LF Filter which is switchable in three positions: DC, 1Hz and 50Hz. We do not have an oscillator stable enough below 10Hz to measure the last two position accurately, but the actual roll-off frequency in the 1Hz position is about 3Hz, and the 50Hz position is quite accurate. Roll-off rate is 6dB/oct Impedance sets a high or low input impedance of about 2kΩ in the Lo position, and 10kΩ in the Hi position. Phantom enables a standard 48-volt supply which has been carefully designed for low noise, via DIN standard value 6k8Ω carefully matched resistors.

The final controls on each channel are Phase and Tilt. Phase is self-explanatory, and just inverts the signal connections at the output. Tilt functions as a shelving boost-cut filter with a turnover point at about 1kHz. The purpose of this is modest correction to microphone HF response, while maintaining the signal integrity. The resulting curves are shown in **Fig.2**.

The controls on each channel are totally separate, enabling the use of the unit for dual mono as well as stereo applications. As detailed below, stereo matching is superb.

In addition to the channel controls, there are a few 'global' functions. The mains switch is on the bottom of the unit to prevent accidental operation, and to allow the unit to be maintained at its warmed up and therefore most accurately calibrated condition. A top panel POWER STANDBY switch disconnects power from internal items which consume energy, but do not need to be kept active for stability reasons. Mode selects Dual Mono and M-S functions. In either mode channel frequency response is excellent as shown in **Fig.3**, as is phase and level difference between as displayed **Fig.4**. Even the Tilt equaliser introduces almost no channel difference as **Fig.5** reveals. This truly excellent stereo matching will give confidence to any recording engineer requiring excellent matching in his stereo recording system, maintaining rock solid image position even if the controls are adjusted during a session.

M-S inserts a decoding matrix into the internal signal path of the *Paintpot* to produce left and right outputs from an M-S stereo microphone. Its performance is given in **Fig.6**, indicating that the difference function is working better than required for stereo recording purposes. The unit tested placed the M-S decoder circuitry before the attenuators, but on units from serial number 20 and up, this is relocated following the attenuators, allowing them to be used to alter image width.

Lastly RESET with adjacent led indicators clears the latched shorted output and unbalance warnings. Another pair of LEDs indicates a near clipping level on each channel. This operates at +20dBu, 6dB before clipping when the *Paintpot* is used as a stand-alone item, but can be adjusted from 0dBu to +26dBu when the remote is used.

More performance

The inputs are well balanced, having a common mode rejection ratio of better than 110dB at low frequencies, remaining better than 65dB up to 200kHz as shown in **Fig.7**.
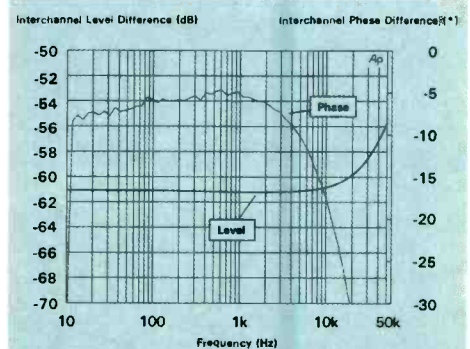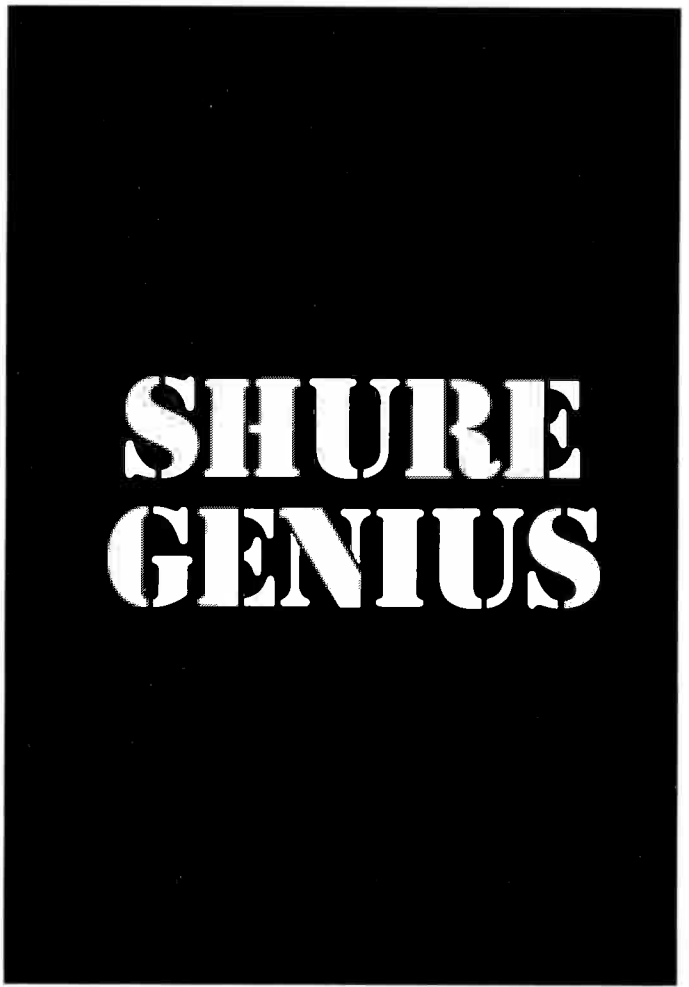

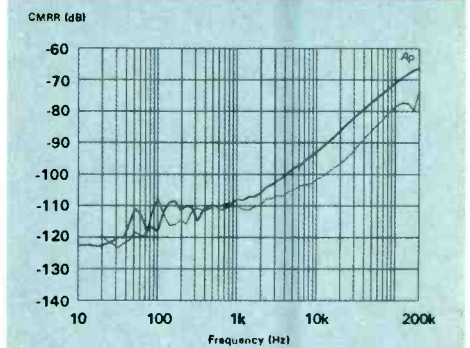**Fig.6: Phase and level difference with M-S active, showing M-S matrix performance**

Fig.7: Input common mode rejection. Gain 50dB. Both channels shown

SHURE GENIUS

Paintpot side connection panel

As shown in **Fig.3**, channel amplitude flatness is exceptional, varying no more than -0.04dB over the audio band when the unit is switched to DC. With the 50Hz filter switched in, this is maintained except for one test condition where a 600Ω source was input into the Lo impedance setting—here the error increased to 0.22dB worst case. Shocking—shockingly good that is.
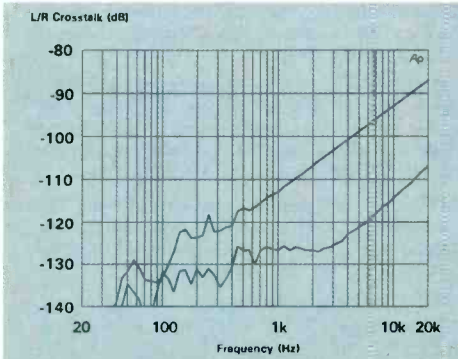
Again, looking at **Fig.4**, stereo channel level

Fig.8: Crosstalk. 1 channel driven at -5dBu with 30dB gain. Other channel terminated in 200Ω, 30dB gain. Level difference between channel outputs calibrated at 1kHz
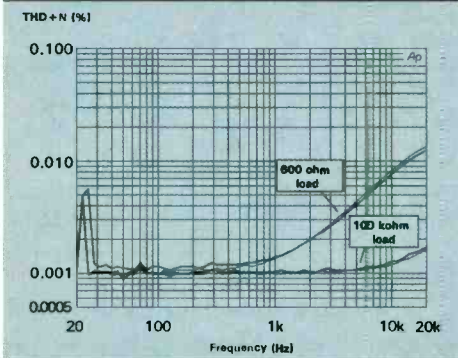

Fig.9: THD + Noise at mic input. Channel gain set at 20dB. Input level 0dB, output level +20dBu Bandpass filter 22Hz to 80kHz, rms rectifier

difference reaches a worst case of 0.04dB, with a phase error of 0.2°, except again where the 50Hz filter is in use, here the error is more than 0.08dB at 20Hz. How many of us can reliably even measure this? Tilt makes matters worse with a maximum error of 0.15dB. Seriously, these differences can hardly be seen, and will never be heard. Superb performance!

Input overload occurs at different levels according to the Pregain setting. See **Table 1** for the level at which 0.5% THD occurs.

Equivalent input noise (EIN), which is a good rating method for preamplifier noise, was measured on the unit supplied and was somewhat disappointing at medium and lower gain settings. At gains above medium -127dB or so is considered excellent. On return to Crookwood, this was investigated and found to be related to the positioning of cabling between the power supply and one of the internal PCBs. Moving the cable cured the problem and I am told that this is now incorporated in the standard production techniques. Both sets of results are given in **Table 2** below. All measurements are true rms, audio band limited 400Hz to 22kHz. A set of measurements 22Hz to 22kHz showed no difference in noise level.

Interchannel crosstalk performance is also good as shown in **Fig.8**. There is a significant difference in results depending on the channel stimulated, but the worst case is better than -110dB between channels at 1kHz. This is certainly more than adequate.

Lastly, we measured distortion at various levels and frequencies. The total harmonic distortion plus noise plot is shown in **Fig.9** for both channels under two different load conditions. The lower curve will be the more common result, the upper one representing typical performance with a headphone connected while placing a microphone. The

measurement was almost entirely noise up to 10kHz, where real harmonic distortion began to creep in at 0.003%. Again excellent performance. Intermodulation distortion was also practically nonexistent. Output overload occurred at +276dBu with a 600Ω load at 1kHz, dropping a little to +27.0dBu at 10kHz, both within specification.

Remote control

The remote control unit does not exist yet, though it is the reason why I came onto *Paintpot* in the first place, researching remote controlled microphone amplifiers to use at the front of digital audio 'snakes'. However, the remote will provide a number of features to the Paintpot. For one, gain steps will be 1dB, further smoothing gain adjustments. Up to 16 *Paintpot*s can be controlled from one remote.

Summary

Crookwood are a small company, but a serious one, and were very helpful to me when preparing this review. *Paintpot* is an impressive beast, which looks the part. For the recording purist, it is a must—allowing a very clean interface directly between the microphone and recorder. For the rest of us, it is probably a matter of how many microphones we need since 16 *Paintpot*s at £1,600 each would require a goodly sum of money to buy and a lot of hands to carry.

But I like the *Paintpot*. ∎

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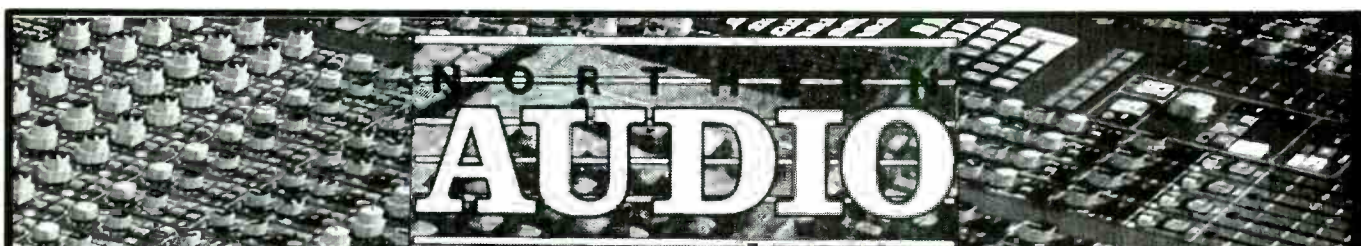PHONE HOLLAND (31) 3465.70670 • OR FAX (31) 3465.72707

Put several pieces of a jigsaw together and you might end up with a complete market opportunity that no-one yet seems to be exploiting.

The TV and video industry is getting excited about Video Dial Tone. The plan is to code movies to MPEG 1 standard, as used for Full Motion Video on CD, and give telephone subscribers the chance to dial-a-movie for delivery by phone.

Of course it is a lot less easy than people are now predicting. The data rate for VHS quality pictures and stereo sound is 1.5 Mb/s. At this rate, the signal will only travel a few kilometres down a twisted pair of telephone wires. High-speed delivery, with a 100-minute movie piped in ten minutes would put data rates up to 15Mb and cut delivery distances to a few hundred metres. But telephone engineers are now talking seriously about 1.5Mbit/s on twisted pairs, this gives us the first piece of the jigsaw.

The second piece comes from the trend towards even higher data speeds for fax and electronic mail modems. A speed of 9.6Kb/s used to be seen as the practical ceiling. Now modem makers are moving to 14.4Kb/s and on to 19.2 and 28.8Kb/s.

The third piece of the jigsaw falls into place as radio stations cut costs. Just listen to any news or talk show and check how often 'guests' and 'experts' are now working in the studio with the anchor presenter. It used to be routine. Now the presenter is in the studio, and the expert guests are on the end of a 'lo-fi' domestic telephone line.

Radio stations have only ever paid guests small fees but they have sweetened the pill by offering transport to and from the studio. Now guests are told there is no money for a car. With car parking difficult and public transport unreliable, most guests cannot justify offering their time for the money paid. So the station settles for a phone link, and the broadcast sounds less professional.

Now another piece of the jigsaw: commercial radio broadcasting in Britain used to be controlled by the Independent Broadcasting Authority, which had tight technical standards. Now that the Radio Authority has taken over from the IBA there is virtually no control over technical standards. Radio stations can do whatever their listeners like, or let them get away with.

Take Classic FM in London, they now make no bones about targeting
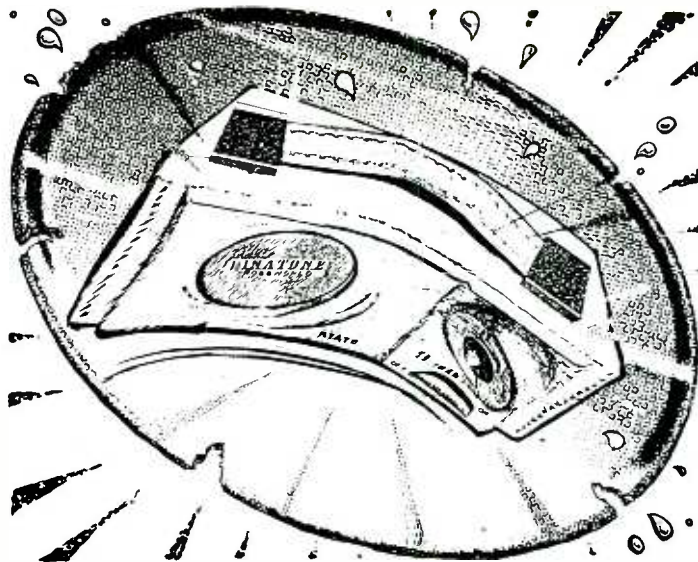
Barry Fox

Remote radio daze and discs begin to lose their shine

an audience which is listening mainly in noisy conditions on car radios and portables. So Classic FM use very heavy compression, with multiband squashing of classical music down to a rock station range of around 15dB. This pleases the majority who hear a loud signal and upsets only the minority who are listening on a high quality home hi-fi. CFM also rely far more on telephone line links than most listeners would ever dream.

Eight of ten of CFM's live concert outside broadcasts are carried by ISDN link, because it is far cheaper than hiring one of British Telecom's analogue music lines. It costs only £400 to install an ISDN line, with a quarterly rental of £84 and then normal telephone call rates for use. The single-line pair can support a data rate of 128Kb/s, in both directions at once. A double-line doubles data rate to 256Kb/s.

Currently CFM use Musicam to code stereo down to 128Kb/s, but plan to move to 256Kb/s with APT-X, the 4:1 compression system developed by Audio Processing Technology of Belfast. Thanks to the Radio Authority's light touch, CFM are already using APT-X to store music jingles and advertisements on computer discs instead of tape. CFM will soon start using ISDN to broadcast from home studios.

Put the jigsaw together and you can see the need for a low-cost audio compressor and modem that radio journalists and regular contributors can plug into the standard socket of their existing, analogue twisted-pair telephone line. This unit would work in mono only, converting speech from a fair quality boom to around 20 or 30Kb/s and squirting it down the

phone line like a fax.

The radio station would demodulate, decompress and route into the studio mixing desk. The result would be a near-broadcast-quality mono feed into the broadcast signal. To save bits, the remote guest need only hear the studio feed in low quality.

With the necessary compression and high-speed modem chips available to manufacturers, it should surely be easy for some bright-spark company to put together a reasonably priced, easy-to-use package.

London entertainment listing magazine *Time Out* (24th November, 1993) told how CD singles pressed by the PDO plant in Blackburn were 'rotting away' and extrapolated that not only might 'millions of CDs' be affected, but CD-ROM discs used by businesses to store computer data might also be at risk.

The facts do not justify this dire extrapolation, and only a few thousand ephemeral pop singles are likely to be affected. But other pressing plants are likely to find that they have similar pigeons coming home to roost.

Philips built the Blackburn plant to press 12-inch video discs, but demand was small so it began pressing 5-inch CDs. Until 1988 all were packaged in plastic jewel boxes. Then the record companies started releasing 'singles' on CD and cut costs by using cardboard sleeves. Although the standards set by Philips specified sleeve size they did not mention material quality. In late 1989 PDO's Technical Services Director Dave Wilson found that some discs in board

sleeves were refusing to play after a few months, while others were fine.

Analysis of the sleeves showed that some were made from processed paper called solid sulphite board, and did not affect the discs. Other sleeves were made from untreated wood pulp. This was releasing sulphur which ate through the protective lacquer on the label surface of the CD, reached the metal reflective layer underneath and ate into that, too. Consequently, cheap players immediately refused to play the disc while more expensive players with more powerful digital error correction circuitry played the disc perfectly, only failing after several more months when the holes had grown larger.

PDO found that if a drop of methylated spirits is put on the board surface, the material turns clear to reveal a pulp of free fibres if the board is untreated.

As soon as the effect was identified, in late 1989, PDO worked with Philips to set a standard for CD sleeve chemistry. By then the record companies were starting to use plastic jewel boxes, because of the perceived low value of card.

PDO say it is very difficult now to put a number on the discs at risk. Record companies often only ordered the minimum number of copies of a CD single, which is usually 1,000. Of these they gave away some to radio stations, sold a few hundred and then junked the unsold remainder. PDO believe that only a few dozen titles were packaged in pulp sleeves. So the total number of discs at risk is in the thousands rather than tens of thousands.

The record companies are likely to have supplied similar sleeves to other pressing plants, and time could show these to be at risk, too. PDO is one of the few plants in the world to use silver (instead of aluminium) as a reflective material for CDs. This is because PDO began in the early-1980s pressing video discs and the only method of coating then available was the wet silver process used for making mirrors. Most other CD plants use the more modern process of sputtering aluminium. Although the type of reflective metal used may alter the time it takes for erosion to occur, if the sleeve material is releasing sulphur it will eventually make its mark.

Record collectors with CDs in cardboard sleeves can now check the material with a meths tests and then put any with pulp sleeves into plastic jewel boxes. ■

ILLUSTRATION: CARL FLINT