April 1991 $5.00 £2.00

STUDIO SOUND

AND BROADCAST ENGINEERING

A LINK HOUSE MAGAZINE

MIXING CONSOLES

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Focusrite

The Focusrite Studio Console has been introduced to the world of music recording to meet the aspirations of recording engineers seeking perfection in all that they do. Whether recording in the analogue or digital domains, the means with which the audio signal reaches tape is of paramount importance.

The microphone pre-amplifier and summing amplifiers must be completely transparent whilst the equalizer should enhance the sound in precisely the way desired. The rest of the console must offer similar levels of perfection and also ease of use throughout the recording, overdubbing and mixing process.

The Focusrite Studio Console successfully addresses all these objectives. Based around the Focusrite transformer coupled mic – pre and equalizer, the console offers such features as 48 bus routing 12 mono and 2 stereo sends and 3 stereo buses. Optional features include left-centre-right panning for mixing to picture.

200 kHz bandwidth throughout the console and ultra low-noise balanced buses ensure optimum phase coherence and noise performance. Transformer balanced insert sends and line outputs ensure electrical isolation from external sources.

The uniuque facility of global horizontal switching and other group status controls accelerate initial setup. The choice of GML or SSL moving fader automation systems offers familarity without compromising the sonic performance of the console.

For those recording engineers who seek perfection, and for the studios that serve them, we offer you the excellence of the Focusrite Studio Console.

Metropolis Studio, London

Designed for Perfection

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Soundcraft Sapphyre

ULTIMATION™

THE 'ULTIMATE AUTOMATION' SYSTEM

ULTIMATION is an entirely new concept in console automation, designed to let engineers choose the type of system most appropriate to the task in hand.

It can work as either a dedicated VCA system, a dedicated moving fader system, or in a way that combines the best features of both systems. The engineer is free to decide.

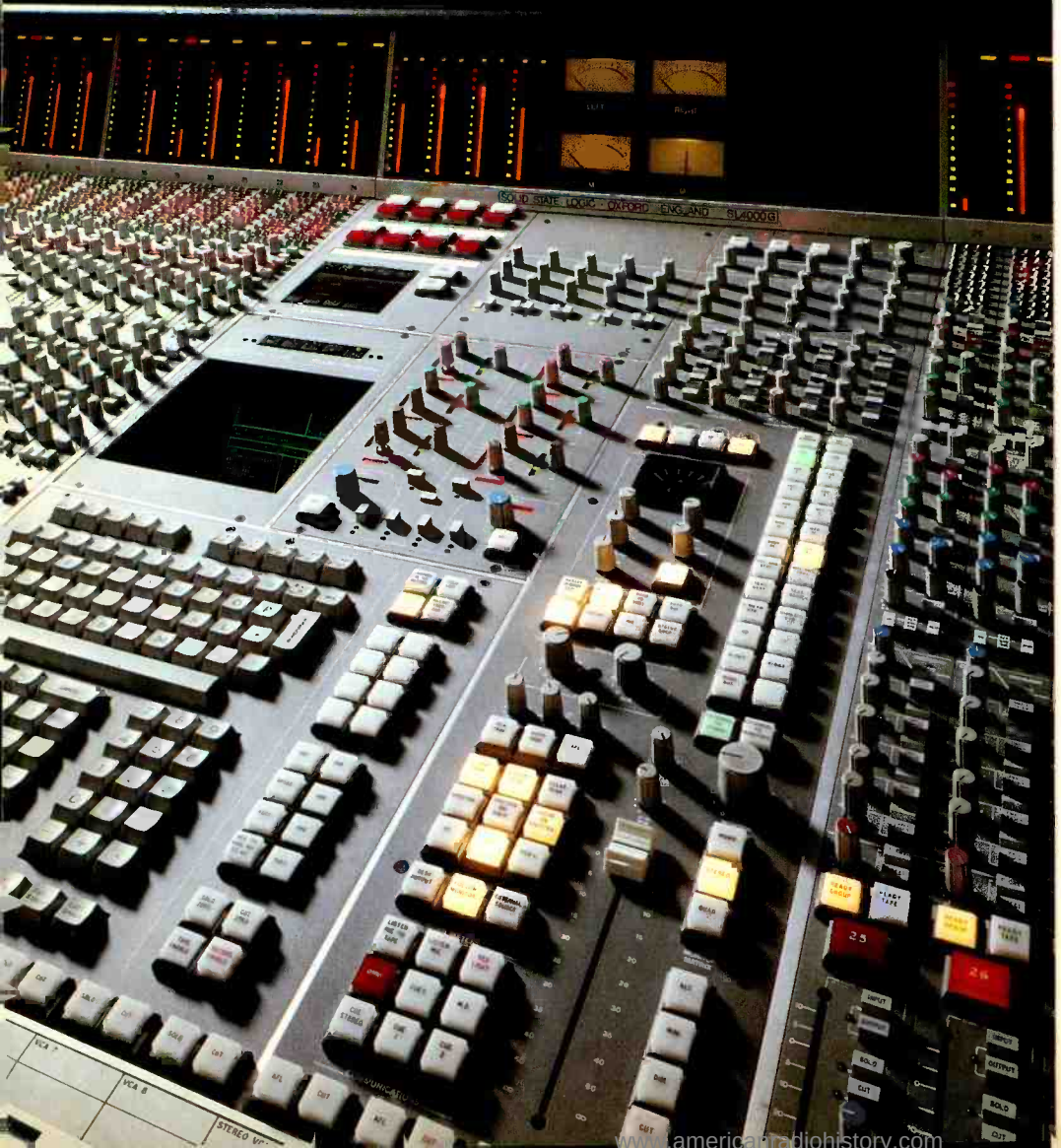
By using SSL's unique dual signal path circuitry, ULTIMATION really is the ultimate in console automation.

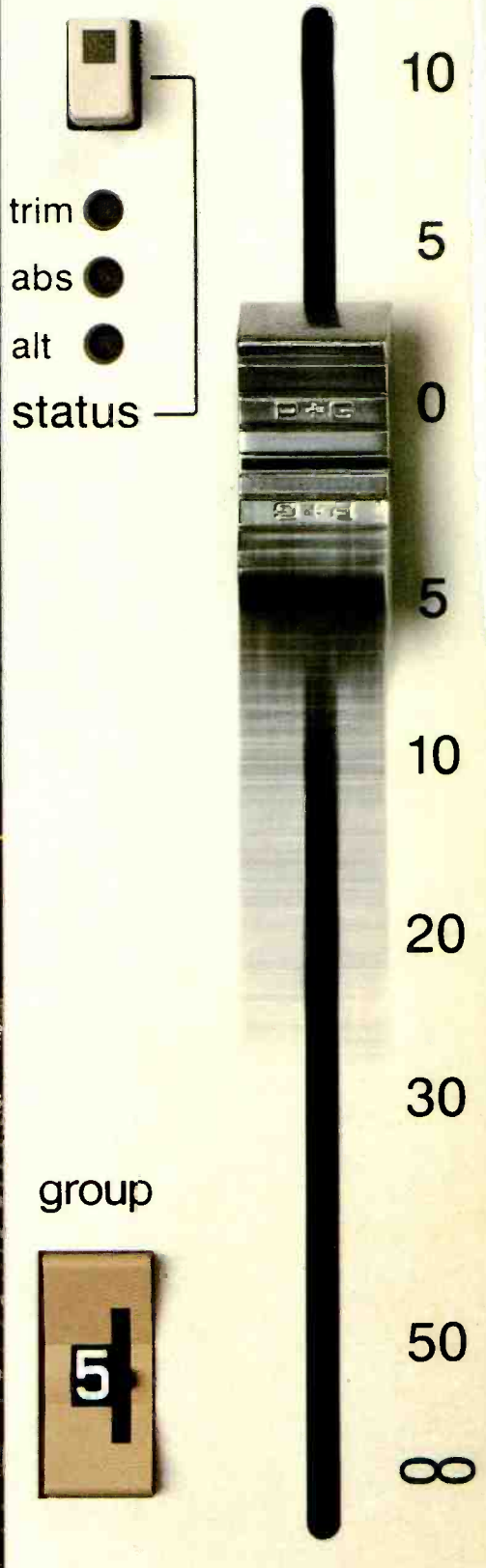
Main features include:

- ▶ Works as standard G Series VCA system, or as a full moving fader system
- ▶ Dual signal path technology allows *combined* operation, providing the benefits of both systems
- ▶ Automatic selection of gain element most suitable to task
- ▶ Reads existing G Series mix data

In keeping with SSL's commitment to system compatibility, ULTIMATION reads all existing G Series mix data, and can be retrofitted to any console using the G Series computer.

At last studios can provide an automation system to suit any client, yet retain the familiarity of the world's most respected console system.





Solid State Logic

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ABC
MEMBER OF THE AUDIT BUREAU OF CIRCULATIONS

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(ABC audited)

No fixed mix

It was on this page a few years ago that I was trying to keep an open mind on the subject of physically smaller consoles. I was aiming to look at some of the technical and commercial forces behind console developments. On one side I contrasted the physical dimensional challenge that a monster console can present to an engineer working on it, together with the acoustic problems that such a structure can present within an increasingly accurate monitoring environment. In contrast I looked at the possibilities of compact consoles using vastly different sized control surfaces and the real and perceived problems this could present to the working engineer. When that was written, there was not really a great deal of choice or variety of approach but, as you would expect, technology has allowed many further possibilities.

The Neve *DSP* consoles still represent the most radically different approach to large console design closely followed by the Trident *Di-An* but, of course, neither of these were small consoles (although they could have been if there had been a demand). There have been more recent implementations of similar concepts and this approach still remains viable.

When I was writing before we were seeing the first of the computer mixing systems where there was no physical console other than a graphic on-screen representation together with a QWERTY keyboard and mouse. Although much beloved by the marketing departments this totally 'virtual' approach was deemed to be of rarified interest even allowing for the incredible amount of control such systems promised. I sense that currently there is little interest in handling more than half a dozen signals on screen at any one time and these normally only in conjunction with a workstation type application.

More recently we have seen the next steps being presented by both the virtual and physical 'camps'. Taking the latter first, we saw the ill-fated Novation console with its compact assignability, SAJE *Memory*, the Sony *VSP-8000* and so on. We have seen the centralisation of facilities and the increasing integration of a visual display to enhance facilities or even as an essential part of the operation (beyond just showing track info and routing). The Yamaha *DMC1000* featured in this issue is no exception; using a blend of assignability, dedicated controls and elaborate LCD displays, it is a particularly interesting combination of design approaches.

If we then turn our attention to the on-screen computer mixers and their next stage of development, what do we see? Well they seem to see their next development direction as incorporating some physical control mechanism—such as a bank of faders and knobs! Examples of the approach include those from NED, Symetrix and WaveFrame.

With these developments we see what appeared to be two quite diverse development directions actually becoming far closer and maybe nearer to defining a console type for the future that is both instinctive and versatile without alienating either approach. In fact it would appear that the gulf has been bridged!

I would like to suggest another direction that may remain to be physically implemented but is apparently quite possible. The SSL *ScreenSound* uses a screen, pen and graphics tablet approach that fits very well with the picture-related aspects of the audio control but what is not so widely known is that when part of the ScreenNet network multiples of units can be run in parallel—up to 56 channels. The Yamaha *DMC* console previously mentioned can be cascaded with additional units make far larger numbers of channels onto a central bus. The concept of console expandability acquires a new meaning. Traditionally a studio has charged a client for control usage by the hour and this at its most basic has been for the room and the console has to be of sufficient size to handle the maximum number of tracks expected. Could such an approach to console expandability be used as a way of charging for the size of console provided? For example, could we see the basic studio rate be for a console capable of handling 24-track applications with more channels costing more? Or is this just a non-starter? Whatever your feelings, the possibility would appear to be there.

Keith Spencer-Allen

Cover: Yamaha DMC1000. Photography by Roger Phillips

A MEGA NEW STUDIO CONSOLE

(IN EVERY AREA BUT THE PRICE)

THE WORD MEGAS–FROM WHICH THE MODERN WORD 'MEGA' IS DERIVED–MEANS, LITERALLY, 'GREAT'.

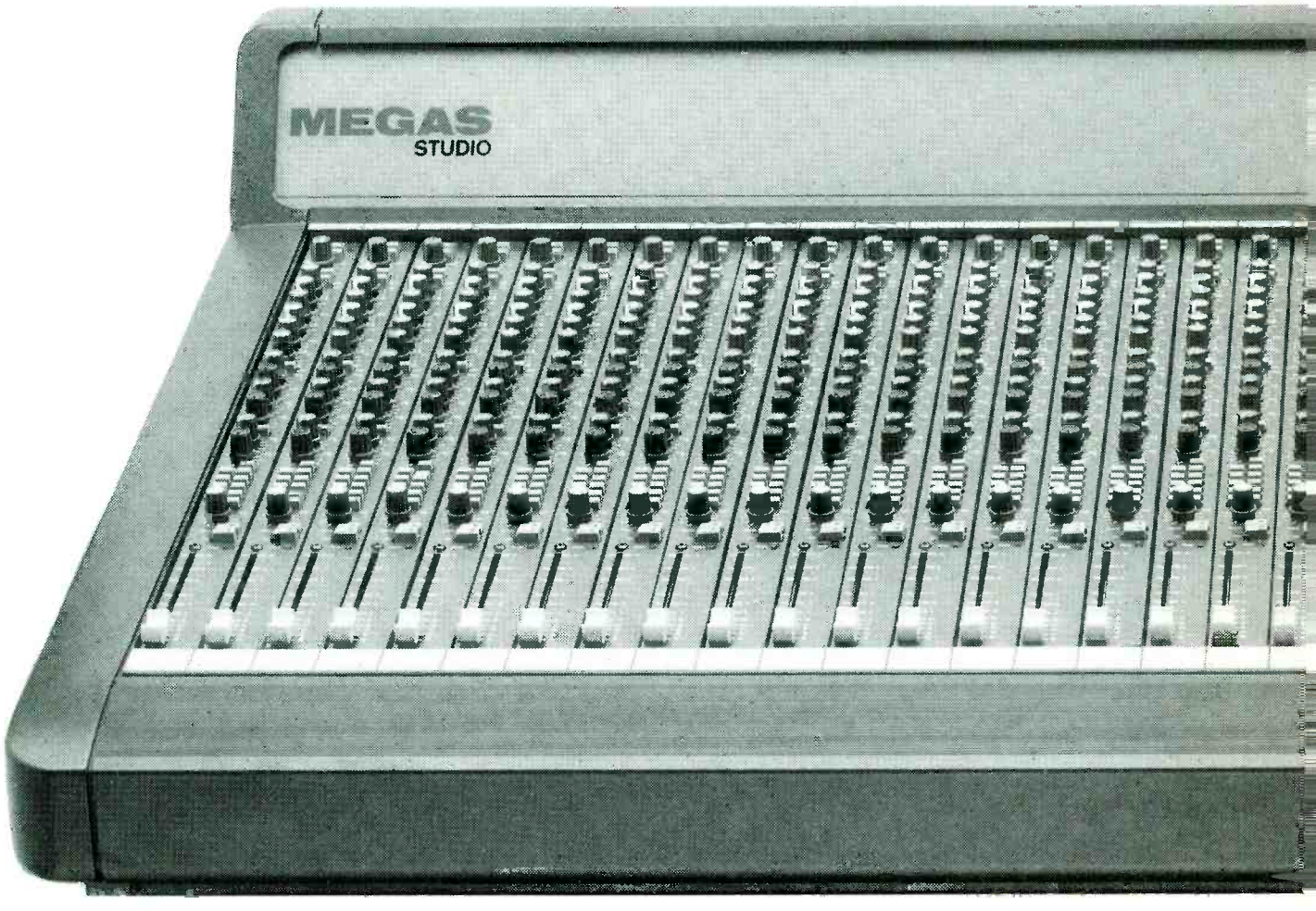
AND 'GREAT' IS EXACTLY WHAT OUR NEW MEGAS RANGE OF CONSOLES HAVE BEEN DESIGNED TO BE.

MEGAS IS THE PRODUCT OF BOTH ADVANCED AUDIO DESIGN TECHNOLOGY AND PRACTICAL THOUGHT; COMBINING UP-TO-DATE COMPONENTS WITH MODERN MANUFACTURING METHODS.

THE RESULT IS A RANGE OF LOW COST CONSOLES THAT PROVIDE AN IMPECCABLE DEGREE OF AUDIO QUALITY TOGETHER WITH UNMATCHED STANDARD FACILITIES AND FEATURES.

FOR EXAMPLE, THE MEGAS STUDIO RECORDING CONSOLE IS FITTED WITH MIDI CONTROLLED MUTING AS STANDARD, AND CAN EASILY BE INTERFACED WITH A SEQUENCER TO PROVIDE AUTOMATED MUTES.

THE STUDIO CAN BE SPECIFIED IN EITHER 16 OR



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24 GROUP BUSS FORMATS. FOUR DIFFERENT FRAME SIZES CAN ACCOMMODATE A COMBINATION OF MONO OR STEREO INPUT MODULES PLUS UP TO TWELVE DUAL GROUP MODULES, A COMPREHENSIVE MASTER MODULE AND AN INTERNAL PATCHBAY. IN FACT, IT'S EXACTLY THE MIXER THAT YOU NEED IT TO BE.

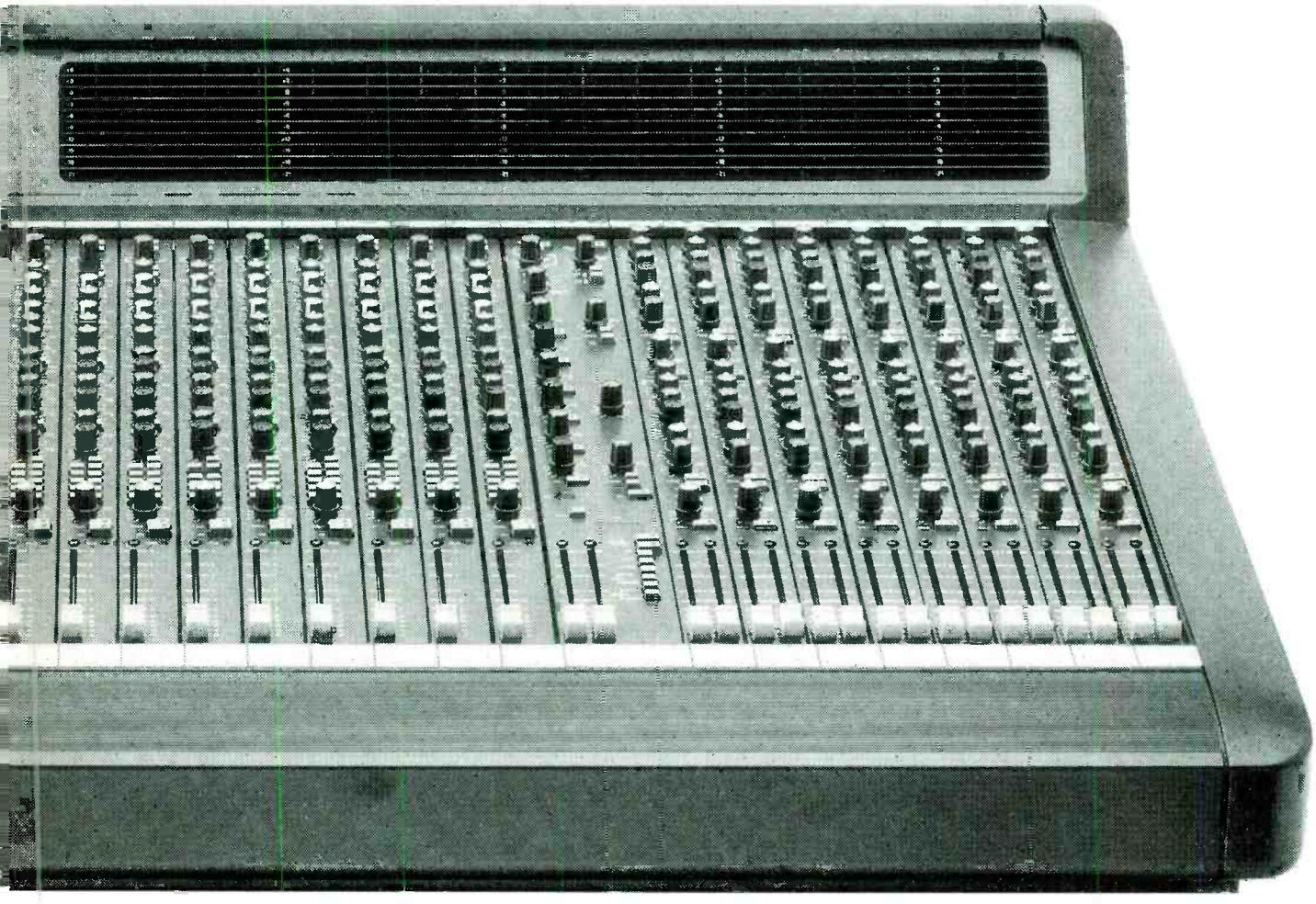
WE'VE PAID SPECIAL ATTENTION TO AREAS YOU'LL NOTICE VERY QUICKLY.

SUCH AS OUR NEW EQUALISER DESIGN THAT NOT ONLY PROVIDES VERSATILE AND TRANSPARENT CONTROL OF THE AUDIO SIGNAL BUT ALSO SOUNDS, WE THINK YOU'LL AGREE, AS SWEET AS THOSE ON CONSOLES COSTING MANY TIMES MORE.

AND A REMARKABLE NEW SWITCHING TYPE POWER SUPPLY UNIT THAT OPERATES AT LOW TEMPERATURES FOR HIGHER EFFICIENCY AND FAR GREATER RELIABILITY. EVEN THE HEADPHONE SOCKET HAS BEEN POSITIONED TO BE EASY TO REACH AND YET OUT OF THE WAY.

IN SHORT, THE MEGAS STUDIO IS THE PERFECT SOLUTION FOR THE TWO GREAT PROBLEMS IN LIFE; CREATIVE NECESSITY AND ECONOMIC REALITY.

MEGAS
STUDIO



TECHNICAL SPECIFICATIONS: NOISE -127.5db, CROSSTALK >90db, CHANNEL MUTING >90db, INTERCHANNEL >80db.

Ampex job losses not due to recession only

Ampex's announcement of job losses in Europe and the USA have been blamed mainly on the downturn in both economies. In the UK, however, the purchasing inactivity of the main ITV companies before the awarding of new franchises has been seen as a secondary reason behind Ampex's decision.

In Europe, 93 people have been made redundant 45 of them at the UK Ampex headquarters in Reading, Berkshire. Reductions have been made in all divisions but mainly sales and service. Some of the job losses have been achieved by redistributing marketing support to other Ampex regions and redirecting resources to product development.

Ampex claim that this re-organisation together with recent capital investment by Ampex owner Ed Bramson places the company in a strong position, not only for current operations but also for longer term R&D projects. Terry Radford, vice-president of sales at Ampex Europe has said that a number of key product developments will be announced at NAB next year which will "re-affirm Ampex's commitment to the industry".

Elliott Bros form new sound contracting company

Bruce and Guy Elliott, and Mike Novak have formed a new company, Novak Electromusic Ltd, in response to "changing market demands, and the increasing requirement for a more controlled approach to sound contracting". Elliott Bros & Novak Electromusic will be operating from Greenwich, and will specialise in sound contracting.

The service will include all aspects of design, room measurement, product specification and acquisition, project management, installations and commissioning. The design engineers, headed by Mike Novak, have full AutoCAD and AcoustaCAD facilities.

Bruce Elliott, managing director, commented, "We are moving with the times. We have combined our proven style of system engineering and management with the considerable expertise of Mike Novak in sound contracting." Additional personnel to join the new company are Jon Trotter who has been appointed installations manager. Elliott Bros & Novak Electromusic, 33 Greenwich Market, London SE10 9HZ, UK. Tel: 071-293 1443. Fax: 071-293 5856.

News from the AES

Our most recent event was the AES British Section Conference **Will You be Legal? – Implications of EC Directives for Audio and Video Engineers** held on Tuesday, March 19th, 1991. This covered the effects of legislation to be implemented on January 1st, 1992, and the Chairman was Allen Mornington-West.

This legislation affects manufacturers, designers and installers of professional audio and video equipment and systems. It was intended to reveal the extent and scope of the legislation as well as discussing the strategies of coping with both the legal and engineering consequences of the European performance standards involved. Ignorance of the law is, we are advised, no defence.

This was an important conference, and there will be a set of papers available in due course.

Our next lecture will be held on Tuesday, 9th April and will be given by Henrik Raunkjaer of Bang and Olufsen, where he is Senior Manager of Research and Design. He will be talking on the subject of **Multi-room Audio and Video Systems**. It is now possible to listen to sound and watch television around the home without having separate units in each room. Mr Raunkjaer will describe Bang and Olufsen's basic philosophy and then go on to indicate how the functions that they wanted to achieve were implemented with the chosen technology. He will also attempt to give some insights into the future of these systems.

The lecture will be held at the ITC (formerly IBA), 70 Brompton Road, London SW3. The ITC is opposite Harrods and Knightsbridge Underground, between the Nationwide Anglia Building Society and Boots. The evening starts with coffee at 6.30pm followed by the lecture at 7.00pm.

For further details on the Conference, or on any other aspect of the AES, please contact: **Heather Lane, AES British Section, Lent Rise Road, Burnham, Slough, Berks SL1 7NY, UK. Tel: 0628 663725. Fax: 0628 667002.**

Audio engineering courses

Sound Practice, a team of six audio engineering specialist trainers have designed a new series of modular courses. The first *Audio Engineering Practice* covers core material essential to anyone who has to maintain and align an audio system. The second *Audio Update* is designed to give engineers the confidence to deal with advanced audio technology. The courses run from June 1991 through to March 1992 and bookings are taken on a first come first served basis.

Contact: Dave McClure, Training Manager, Audio Engineering Unit, BBC Wood Norton, Evesham WR11 4TF, UK. Tel: 0386 45123, ext 229.

Literature received

- APRS have published the 1991 edition of their reference book, the **Guide to Recording in the UK**. The guide gives comprehensive details of over 100 sound recording and post-production facilities and mobiles plus, for the first time, pressing and duplication facilities. Copies are available price £7.00 plus £1 p&p from APRS, 2 Windsor Square, Silver Street, Reading, Berks RG1 2TH, UK. Tel: 0734 756218.
- An 8-page, fully illustrated catalogue describing **Simpson Electric's Mercer line of test equipment** is now available. For a free copy of the catalogue contact Mercer Electronics, 859 Dundee Avenue, Elgin, IL 60120-3090, USA. Tel: (708) 697-2275. Fax: (708) 697-2272.
- The British Radio & Electronic Equipment Manufacturer's Association (BREMA) have published **The Setmakers**, a 464-page book celebrating the inventions of radio and television, with over 500 photographs. The book was written by Keith Geddes and is available at £12.45 plus £2.50 p&p from John O'Neill, tel: 0737 355240.

Exhibitions and conferences

April 8th to 10th Cable & Satellite Exhibition 1991, Olympia, London, UK.
April 15th to 18th NAB, Las Vegas, USA.
April 26th to 28th MIDI Music Show, Novotel Hotel, Hammersmith, London, UK.
May 18th to 19th SPARS Technical conference on digital audio workstations, Orlando, FL, USA.
June 5th to 7th APRS, Olympia 2, London, UK.
June 13th to 18th International Television Symposium, Centre des Congres, Montreux, Switzerland.
June 25th to 27th Multimedia 91 conference & exhibition, Olympia 2, London, UK.
July 10th to 12th Pro Audio Asia 91, World Trade Centre, Singapore.
July 10th to 14th International Music Show, Olympia 2, London, UK.
September 8th to 11th Light & Sound Show 91, Olympia 2, London, UK.
September 15th to 18th Vision & Audio 91, Earls Court, London, UK.
November 29th to December 1st Broadcast India 91, Nehru Centre, Bombay, India.

MIKE MYERS RECOMMENDS "A SOUND INVESTMENT"

With over 22 years in the music business, using every kind of studio set-up there is, Songwriter/Producer Mike Myers hardly expected to be 'amazed' by any new piece of recording equipment. Yet that was his response to the AKAI Digital Multi-track System (A-DAM) and DD-1000 recorder/editor which he and partners Jim Dooley and Danny Saxon have installed at their north London studio.

Mike, who has worked with the Shadows, the Nolans and the Dooleys and more lately Billy Ocean, now produces film and commercials music as well as recording a number of up-and-coming American artists.

"When we were setting up the new company, Ad-Trax, we were naturally interested in A-DAM because of the relatively low price," recalls Mike, "but I expected a trade-off in quality - either a slight sound degeneration or other inherent problem. I was also somewhat nervous of a new tape format."

These fears all proved unfounded and Ad-Trax now boasts a 24-track A-DAM System.

"It's an exceptionally well thought-out and reliable system, dropping in and out precision is superb and you can rehearse drop-ins without committing yourself. The equipment also interfaces with all other makes of machine, which is a big advantage."

Ad-Trax used the A-DAM System for two radio commercials which were placed in the top five finalists of the Independent Radio Advertising Awards.

"We had a lot of favourable comments about the high quality of those ads. I can understand people being wary of A-DAM's 8mm format but no one who tries it will be disappointed and it's a third of the price of any other comparable system."

Optical disk recorder

Ad-Trax has also invested in AKAI's DD-1000 Magneto Optical Disk Recorder which is ideal for mastering, editing and post production. These various applications offer powerful features which are both simple and effective and, combined with the advantage of a removable disk, put the DD-1000 in a class of its own.

Mike Myers describes the DD-1000 as the word processor of the recording industry.

"The machine uses a non-destructive editing system so no matter how many edits you make, the original is always there. The finished work can be downloaded onto DAT tape, freeing up the optical disk for re-use."

The DD-1000 has the huge advantage of being portable. One of the first jobs Mike Myers used it for was at Capital Radio to record a David Jensen voiceover on a pre-recorded commercial, using the 4-track replay and merge facilities to great effect.

"Within an hour we had recorded, edited and transmitted the commercial via satellite to independent stations across the country. The DD-1000 is great for voiceovers as you can shorten and elongate them. We've also used it to edit in chorus vocals by triggering the machine at the required point via MIDI. It takes only minutes to do."

Post-syncing and film music work is another use to which the DD-1000 has been put.

"We can record sound effects on optical disk and feed them onto the soundtrack at SMPTE hit points. Foreign language dubbing can be done in the same way. We are waiting for a time-stretch facility on the machine as well as a remote control which are both due out from AKAI soon."

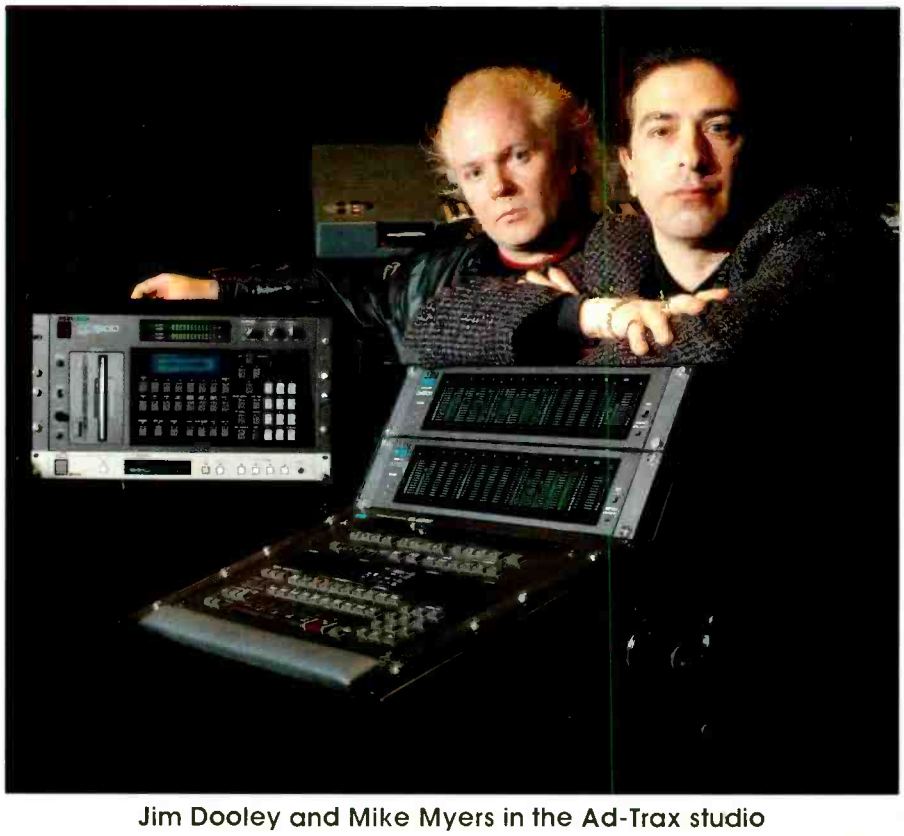
Highly cost-effective

Ad-Trax has established itself in a highly competitive market in only six months and attributes some of its success to the decision to invest in AKAI equipment.

"It's been highly cost-effective for a set-up like ours which is involved in commercial music for radio, films, TV and the record industry."

A 12-track A-DAM digital multi-track recording system costs £15,695 and can be expanded to 24 or 36 tracks. The DD-1000 is available at £7,750.

More information from AKAI Digital Division on 081 897 6388.



Jim Dooley and Mike Myers in the Ad-Trax studio

AMS move US office

Six months after the purchase of AMS by Siemens of Austria, AMS's US office has been closed and all operations have been transferred to the Neve North America office in Bethel, Connecticut.

AMS will draw upon the resources of Neve in the areas of administration, accounting, sales/service and advertising/promotion. AMS will relocate their corporate offices from Petaluma, CA, to Bethel, CT.

AMS senior sales manager John Gluck has been appointed president of AMS North America. Gluck commented, "The move to Connecticut and the sharing of Neve North America's well-established marketing and support resources represents a major boost to AMS activities in this extremely important market. This step is an indication of the many changes made possible by the audio video group's purchase of AMS and illustrates our intent to provide our customers with the highest standard of support."

Also relocating to Bethel are Graham Murray, AMS technical services manager, and Stuart Hirotsu, product support engineer. Both come from the Petaluma office. AMS sales/service will be located in the Neve Los Angeles, New York and Bethel offices. A Chicago sales/service office will be established with Doug Ordon as AMS sales manager.

In-brief

● Cambs, UK: **Cambridge Digital complete Finnish training courses:** Cambridge Digital have recently completed a series of training courses in Digital Audio Technology for the Finnish Broadcasting Company YLE. David Pope, MD, and consultant John Watkinson presented a series of courses designed to guide analogue engineers from fundamental digital audio principles up to a high degree of competence in the subject.

● San Francisco CA, USA: **100th Sonic System sold:** Abbey Road studios, London, have ordered another digital audio workstation from Sonic Solutions in San Francisco. This will be the third Sonic System delivered to Abbey Road and the 100th system delivered by Sonic Solutions to record companies around the world.

● North Hollywood, CA, USA: **ASC authorised by Neumann:** Audio Services Corporation have announced that they are now an authorised Neumann microphone service centre. Ron Meyer, ASC's engineering manager, recently completed a one week training course at Neumann's Berlin factory. The training, directed by Stephen Peus, development engineer for Neumann, covered Neumann's entire microphone line. ASC are expanding their services to the recording and music industries. Audio Services Corporation, Service Department, 10639 Riverside Drive, North Hollywood, CA 91602, USA. Tel: (818) 980-9891. Fax: (818) 980-9911.

Contracts

● As part of a major expansion plan Sounds Interchange in Toronto, Canada, have invested in a **Neve** *VR48* console. Sounds Interchange are a large audio/video production and post-production facility in Toronto. They have also ordered 16 **Audio Kinetics** *ES.Lock 1.11* units. The ESbus units are being installed in four new studios.

● BBC Radio Projects have taken delivery of an **AMS** *AudioFile Plus* for Radio 1's Steve Wright show. The 8-output unit was chosen for its multitrack facilities and reliability record.

● BBC Television have recently bought six **Shure** *VP88* stereo microphones for their outside broadcast units based at Kendal Avenue, London. The new mics will be used primarily for sports OBs including test cricket, Wimbledon tennis and golf.

● **Clive Green & Co** are to supply a 70-input *Cadac E*-type console for the New York production of *Miss Saigon*.

● **DDA**'s French distributors Regiscene have sold their first two *DMR12* recording consoles to Free Son studios and Coppelia studios, both in Paris. Also TF-1, France's largest private broadcasting organisation have bought a *Q* series console. The 24-input matrix version was bought for floor use in live situations.

● Orinoco, a London recording studio, have opened their new-look control room for business, following the installation of a 60-channel **Neve** *VR* console fitted with *Flying Faders* automation. The installation has gone hand-in-hand with an extensive development programme, which has included the refurbishment of the main studio, the building of a new DDA-based 24-track programming suite, and the addition of a digital editing facility.

Neve VR60 installed at Orinoco, London

● **Adams-Smith** have announced that ABC Television in New York and Fox Television in Burbank, CA, have each ordered a second *2600 E-A/V* audio editor. The unit now contains Super-Sync, which includes Cross-Lock, allowing synchronisation and editing of mixed timecode rates, and Vari-Lock, which synchronises and edits off-speed material.

● Sound Developments, a UK audio and video post-production company, have installed a **DAR** *SoundStation II* digital audio production system in their London facility. The system is operating as the digital audio recording component of a unique on-line autoconforming system that Sound Developments have developed in collaboration with Andy Newell of DJC Software and DAR's engineering staff.

● Jeff 'Skunk' Baxter, Julian Lennon and Joe Perry of Aerosmith have all recently chosen **Allen & Heath** mixers. Baxter has installed a 32-input *Sigma* in his studio at Village Recorders.

● **Pro-Bel**, Reading, Berkshire, have supplied the video and stereo audio switching system for Scottish Television's new eight-camera outside broadcast vehicle. The Pro-Bel equipment is a high density 32×32 video and dual audio system complete with a selection of standard panels and under monitor displays.

● Latest deliveries of the *FRED* tape editor from **Lyrec** include 25 units to Radio Free Europe; 10 to Swedish National Radio; 15 to Radio Nacional Espana; 13 to Danish Broadcasting; and 10 to Radio France.

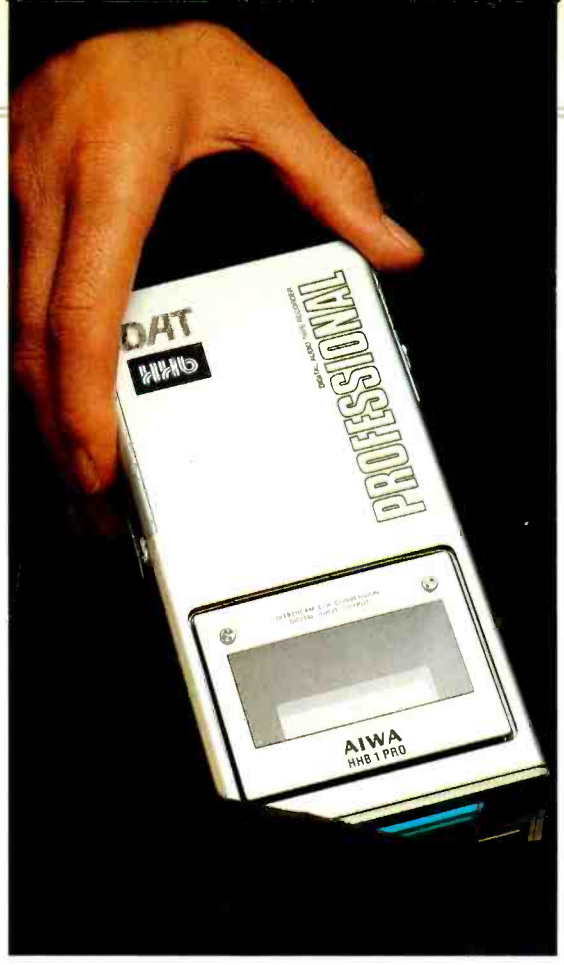
● Recent contracts for **Audiomation**'s new *Uptown 2000* moving fader console automation system include Studio 20 in Tienen, Belgium; Scanax Recording studio in Oslo, Norway; and A-String recording studio in Taipei in the Republic of China.

● **DAR**'s sale of the *SoundStation II* digital audio production system to Hebei TV marks the first installation of the DAR system in China and the first disk-based digital system to be used in all mainland China.

● Japanese audio equipment distributors Otaritec have announced the recent sale and installation of a 40-input **Trident** *Vector 432* console to Vivid Sound Corporation in Tokyo.

● **Hill Audio**, Kent, UK, have delivered their largest console yet, a 48/48/48 *Concept 8400* series with Audio Kinetics *Reflex* automation, to Future Sound Studios in Soest, Holland.

● **d&b audiotechnik** have recently installed a complex and unusual audio/video installation in Stuttgart. The project was undertaken for Allianz, one of the world's largest insurance companies, and involved d&b in providing audio within a room that itself could be configured in different ways. The system had to be invisible when not in use, and the set-up had to configure automatically and optimise the sonic performance. The system included 12 *E1* speakers installed in the ceiling that could be lowered and turned into position under master control.



Every audio professional knows that the DAT format is ideal for portable recording. But at HHB we believe it need not cost the earth.

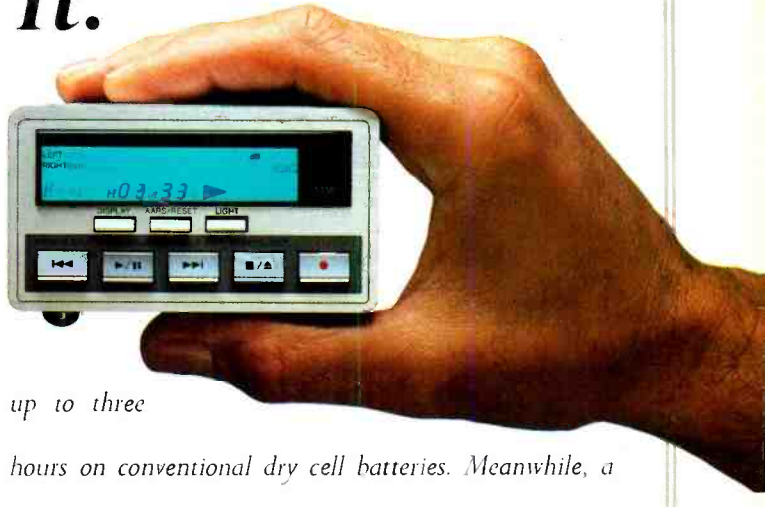
That's precisely why we've joined forces with Aiwa to design our own professional DAT portable – the HHB1 Pro.

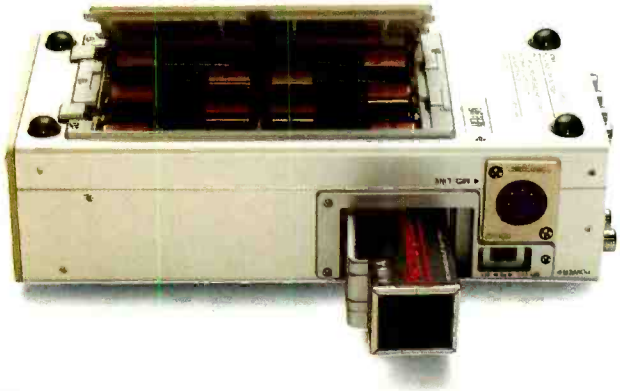
In spite of its compact dimensions, the rugged HHB1 Pro offers a wealth of features for the professional user. A single 5-pin XLR switchable mic/line input allows stereo recordings in the field, while audio quality is assured thanks to the latest single-bit oversampling conversion technology. Of course, AES/EBU as well as SPDIF digital interfaces are provided as standard. And because the Pro's informative LCD display can be illuminated, monitoring in low-light conditions could not be more convenient.

The HHB1 Pro stripes tape with 'absolute time' information as it records. So whenever you insert a recorded cassette, you can see precisely where you are on the tape. With Sony's PCM-7000 range of studio DAT recorders capable of editing to absolute time as well as time-code, you can be confident that your HHB1 Pro will function as their ideal low-cost acquisition partner.

The HHB1 Pro records for up to three hours on conventional dry cell batteries. Meanwhile, a multi-voltage transformer and a NiCad battery pack – together with a selection of useful professional accessories including a wired remote controller – are supplied as standard. Since it weighs in at under £1,000 and less than a kilogram, picking up an HHB1 Pro from the world's number one DAT centre just couldn't be easier.

A professional DAT recorder that goes easy on your pocket. And in it.

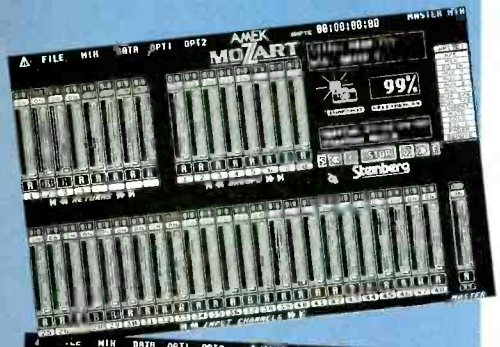


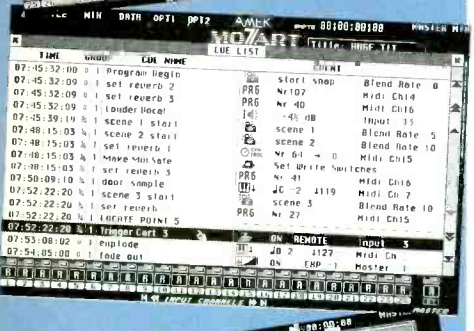


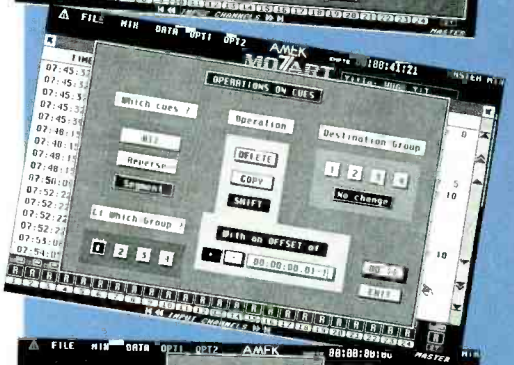
HHB COMMUNICATIONS LIMITED, 73-75 SCRUBS LANE, LONDON NW10 6QU PHONE 081-960 2144 TELEX 923393 FAX 081-960 1160

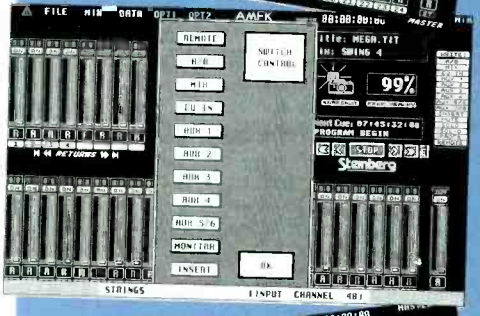
MOZART &POST

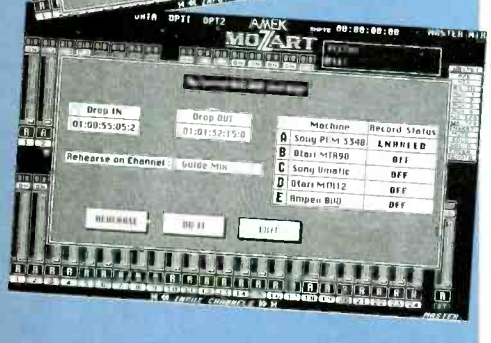
"For the music mix, we locked two 24-track machines with Dolby SR, a video machine, a four track, and a MAC computer to one of our two MOZARTs. MOZART is a very SMPTE-orientated mixing console" Andy Waterman, The Bakery, LA, quoted in Post magazine, November 1990.

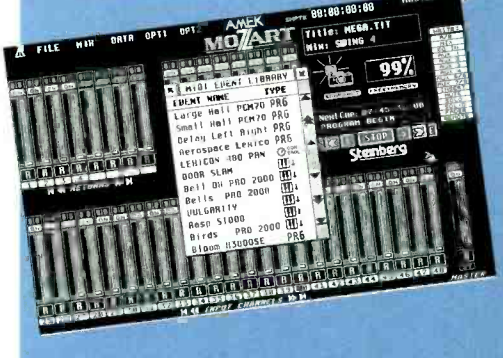












Nowadays, time is of the essence. Real time costs money, and the ability to work in the artificial world of SMPTE time earns money. AMEK's MOZART was designed with both sorts of time in mind. Computer-assisted mixing is now commonplace but systems of real sophistication were, until the arrival of MOZART, restricted to consoles in the highest price range.

Due to rapid developments in technology, computing power is now inexpensive and AMEK, in conjunction with STEINBERG, has developed advanced software for console automation which is unequalled.

SUPERTRUE allows the fader and 15 switches per input to be automated with real-time or off-line control of the functions. Move the faders, press automated switches such as Aux On/Off and Eq In/Out- all moves will be recorded as you make them, to quarter-frame accuracy. Fader and switch moves can be edited – or created – off-line.

The heart of SMPTE control in MOZART is the automated mixing system. Extended real-time control is provided via the Cue Sheet. Cue Events include MIDI functions, FADER level changes, SNAPSHOT load and blend, SWITCH and MUTE changes, READ / WRITE / UPDATE mode selection and MACHINE CONTROL locator points. SUPERTRUE also allows you to draw your own fader curves on the Fader Jobs screen. Curves can then be inserted into the Cue Sheet and triggered on/off between two SMPTE points, allowing fades of any shape and duration to be created.

MOZART was designed as an open-ended system. The new autolocator hard- and software expansion allows total control of multi-machine audio and video systems. The Locator panel fitted into the centre of the console provides Master Transport Controls with a full range of find, start and cycle functions. The machine control window in SUPERTRUE allows offsets, location times and pre- and post-roll parameters to be set up for multi-machine operation. Another window allows automatic, quarter-frame accurate punch in/out recording with a full rehearse function.

Finally, powerful routines provided in SUPERTRUE's Mix Processing and Cue operations screens allow extensive copying, moving and time-shifting of fader, switch and mix information.

MOZART gives you total synchronization flexibility whatever the various sources you are working with - multitrack, samplers, hard-disk recorders, video tape, computers - any realistic combination.

With its AMEK and Rupert Neve-designed audio systems, and an advancing digital control technology, MOZART is the best choice for all time.

AMEK

Head Office, Factory and Sales: AMEK Systems and Controls Ltd., New Islington Mill, Regent Trading Estate, Oldfield Road, Salford M5 4SX, England. Telephone: 061-834 6747. Telex: 668127. Fax: 061-834 0593.
AMEK/TAC US Operations: 10815 Burbank Blvd, North Hollywood, CA 91601. Telephone: 818/508 9788. Fax: 818/508 8619.

Digital compact cassette

Philips chose the January Consumer Electronics Show in Las Vegas for the second of their Digital Compact Cassette (DCC) press functions. This time the company produced a working prototype and it was very much a 'hands-on' affair.

Little new information emerged during the formal presentation, much of it repeating what was imparted at the Eindhoven launch in late 1990 (see 'Business', *Studio Sound* March 1991). But the panel of Philips staff, plus a representative from Tandy (the first hardware manufacturer to offer their full support), was as open as could be expected during question time.

The most important task was demonstrating that the format worked, which was done with a standalone player, first with the digital tape and then with a conventional analogue cassette. The unit itself looked exactly like a conventional mid-to-upper level hi-fi cassette deck, differing only in that it featured minor controls for programming and a display, which offers more information than can be culled from an analogue tape.

While it is impossible to judge sound played in an (enormous) unfamiliar room and through an unfamiliar system, the sound was of acceptable 'hi-fi' quality. Philips stressed that DCC is not aimed at the high end.

What was made emphatically clear during the launch were the marketing intentions. While consumers are showing greater and greater displeasure at the proliferation of incompatible formats in all fields—hi-fi, computers, games consoles and the rest—backwards compatibility is only part of the reason why DCC looks like having a bright future. The key issue concerning the marketing is DAT. While many of DCC's enemies conveniently forget that Philips markets DAT hardware, the Dutch giant quickly points out that DCC is not a rival for DAT—without coming out and saying that DAT is a commercial disaster. Philips spokespersons calmly suggested that DAT will survive as a professional, semi-pro and 'audiophile' format, as it is already established in those sectors.

The backwards compatibility is the initial sales pitch that will make DCC appealing to the vast base of cassette users around the globe. Cassettes are not the hi-fi standard; they're the standard of the man, woman and child in the street. And DCC will allow anyone who owns a library of cassettes (said to be 50 to 60 per household around the planet) to use them with the new hardware.

Philips also pointed out that recording was not a prime consideration, showing that the vast majority of cassette mechanisms sold since the early 1960s are playback-only. So when it emerged that the first player will *not* record onto analogue tapes, they were able to gloss over it quite convincingly.

This, we were told, was merely a response to the complication of fitting a record head into the already-crowded space of a DCC head/transport section. Future models will incorporate an analogue recording facility once the problems have been solved. (Could it be that Philips doesn't want analogue recording facility because it perpetuates the older cassette's existence?) Whatever the case, the analogue cassette portion of the DCC unit is there primarily to facilitate the transition from old to new.

Naturally, the software support was emphasised, this being the key to DCC's success; it was implied that this was only possible because Philips has made SCMS a mandatory non-optional feature of the DCC specification.

Philips have suggested that the first machines in 1992 will cost around the same as upmarket analogue tape decks. At present prices, this could mean a range of £400 to £800 ($800 to $1600). Shortly after the launch of the first standalone decks, a personal hi-fi version and an in-car unit will be released. We were shown one of the three personal hi-fi prototypes the company have in their laboratories.

While DCC will undergo further development before its launch, DAT, it seems, will remain as a pro product. Philips sum it all up with: DAT should be regarded as a modern equivalent, not of the compact cassette but of open reel tape, which in its day was the preserve of the pros and the hobbyists, while DCC will reach everyone else.

DAT's supporters are unlikely to recognise this. Would anyone like to wager as to which DAT manufacturer is the first to release a DCC-fighting 'twin deck' DAT recorder with the second transport being an analogue cassette section?

Ken Kessler

Specification
Frequency range: 5 Hz to 22 kHz @ 48 kHz; 5 Hz to 20 kHz @ 44.1 kHz; 5 Hz to 14.5 kHz @ 32 kHz
Dynamic range: >105 dB
THD: >92 dB
Wow/flutter: 'quartz crystal precision'
Sampling frequencies: 48, 44.1, 32 kHz
Audio bit rate: 384 kbit/s
Error correction system: C1, C2 Reed Solomon
Modulation system: 8:10 (ETM)
Pre-emphasis: optional
Recording time: 90 min with 120 min under development
Tape type: (video) chrome
Tape width: 3.78 mm
Tape speed: 4.76 cm/s
No. of tracks: eight digital audio; one auxiliary code
Track width: 185 μm
Track pitch: 195 μm

Philips DCC prototype at Las Vegas

Canon form Canon Audio

Canon have entered the world of consumer audio electronics with the establishment of Canon Audio Ltd, a new UK-based international company responsible for Canon audio product development, manufacture and marketing.

The company are initially based at the University of Surrey Research Park premises of Canon Research Europe, which has been responsible for the origination and development of their first product—a wide imaging stereo hi-fi speaker system due to be launched later this year. Managing director of Canon Audio is Andy Szeliga, formerly MD of Harman Audio.

The speakers are to be manufactured under licence by Sussex-based Audio Electronics Ltd and Canon Audio will oversee their introduction and export to markets worldwide.

Address changes

- **Cambridge Digital,** audio consultants, have moved premises to 7 Miller Close, Offord D'arcy, Cambs PE18 9SB, UK. Tel: 0480 812201. Fax: 0480 812280.
- **New England Digital** are moving to Unit 5, Sundial Court, Tolworth Rise South, Tolworth, Surrey KT5 9NN, UK. Tel: 081-335 3356. Fax: 081-335 3533.
- **Arup Acoustics** are to open a third office in the UK. The address is St Giles Hall, Pound Hill, Cambridge CB3 6AE, UK. Tel: 0223 355033. Fax: 0223 61258,

People

- **Audiomation Systems,** Co Durham, UK, have appointed David Pope European head of technical marketing support for their *Uptown* moving fader console automation system. Pope already has his own company: audio consultancy Cambridge Digital.
- **TGI,** Hampshire, UK, have announced the appointment of Alex Munro as managing director of TGI subsidiaries, Tannoy and Mordaunt-Short. Munro was previously managing director of KEF Electronics.

CES, Las Vegas, a report

With Saddam's deadline mere days away and a recession in full swing, the 1991 Winter Consumer Electronics show in Las Vegas was not a crowded nor particularly jolly affair. Neither was it bursting at the seams with innovatory products to tempt music lovers. Nintendo and Sega, the main players in the computer games consoles war, enjoyed more press than anyone else, which says a lot about the future of home entertainment.

A trend that must be recognised by audio manufacturers is multimedia, possibly the one concept that will allow pre-recorded music to stay healthy in the next century. The marriage of computers, video and audio, made so much easier and viable through digital technology, is off to a shaky start, even though **Commodore** launched CD-TV at the show. But CD-TV is not the international standard for the concept Philips' CD-Interactive (CD-I) promises to be. But to Commodore's credit, it will have hardware in the shops first.

The concept of interactive educational software, business programs and games will introduce an audience so far immune to high quality sound; to digitally encoded music. However widespread the ownership of CD, it has a long way to go before enjoying the market penetration of the lowly cassette. But advances in this area were limited to mere refinements or frills, with multi-disc players in profusion and separate D/A converters and transports being the most noticeable manifestations.

Harman-Kardon showed a pair of multidisc players of the top-loading variety, which were the first to boast bitstream technology. The **Madrigal** *Proceed* CD library employs a 100-disc tray like that of a slide projector, but mounted vertically like a Ferris wheel. Part of the package is a comprehensive remote control with an LCD panel read-out supplying track listings, music categories and so on to make selection easier.

Audio Access, though, left far less to be desired. That company's player employs two vertical banks of 50 CDs each, with the CD transport mounted on a gantry in-between. In addition to greater simplicity, the AA alternative is based on a field-tested German-made CD jukebox for reliability. But the icing on the cake is their computer interface (IBM or Apple *Mac*), which gives the owner a graphics display including a full-colour representation of the sleeve-art, full track listing and icons shaped like a CD player's controls. Touch the screen over the corresponding icon and the player goes into that operational mode.

Although not intended to rival either DAT or DCC, **Kenwood** demonstrated a 'budget' version of the £20,000 two-box CD-Write Once recording system shown to the European press last autumn. The *LZ-13* is a single chassis alternative, which, at first, looks just like a conventional up-market CD player with a few extra buttons or knobs. But this baby can record digital and analogue signals on write-once blanks such as those from **Taiyo Yuden**. This would be of interest mainly to professionals because of the high cost of blank discs—still in the £20 region. This unit, though, differs considerably from the two-box unit in that it lacks mixing and editing facilities and can only be used to copy completed recordings. This does, however, mean that it's still perfectly suitable for the more practical task of producing short runs of urgently-needed discs. In this respect, even at £20 per copy the discs are more viable and cost-effective than a run of conventionally pressed CDs, usually only available in minimum quantities measured in the hundreds.

California Audio Labs showed the finished version of the *System 1* digital processor first seen in Chicago. This unit is notable for the use of plug-in modules allowing the user to opt for 18 bit, 20 bit or MASH processing, with the useful option of—when two modules are fitted—A/B comparisons of the various types of technology.

Other new converters include the first-ever **Mark Levinson** processor, also from Madrigal. Called the *No 30* Reference Digital Processor, it employs 20 bit 8× oversampling and plug-in upgrades. **Theta** took the bold step of releasing a bitstream converter, unusual for American companies who still prefer multibit. The DS Pro Prime also offers optical output in addition to co-axial. These details, along with a price of around $1250 (£625), make it a more 'entry level', mass-appeal model than the existing units.

Audio Research, valve amplifier manufacturers, released their first converter, the *DAC-1*. Entirely solid-state, it features three separate power transformers—one each for digital input/filtering stages, for the digital converter and the analogue output stage. Five discrete regulators ensure a clean supply to each stage while two linear regulators are dedicated to the digital input stage and the auto muting circuit. **UltraAnalog** supplied the digital converter, a full 18 bit, 8× oversampling design for servicing 32, 44.1 or 48 kHz sampling frequencies. The high-current, all-discrete analogue section employs J-FETs at the input and MOSFETs at the output. A built-in AC line filter further guarantees clean power to all three transformers and none of the front panel switches are in the signal path. Phase inversion takes place in the digital domain. All three sets of inputs accept co-axial leads, while one can be switched to optical and two of the co-axials accept BNC connections. Three sources can be installed at once, with selection from the front panel.

VTL, launched the Reference Digital-to-Analogue Converter, a dual-mono valve design using UltraAnalog DACs and 20 bit, 8× oversampling technology. This, too, features three inputs, the selection of the three established sampling frequencies, balanced or unbalanced connection, seven separate power supplies and eight triode wide-band valve analogue amplifiers.

Another valve amp builder, **Cary Audio Design**, showed a processor-only version of *CAD-5500* preamp-cum-converter, shorn of the preamp facilities to create a cost-effective D/A converter.

Krell showed the finished version of *Stealth*, first seen in June. This is the company's entry-level converter with 1 bit D/A technology, balanced or unbalanced operation, two proprietary star-earthed, double-regulated power supplies, four digital inputs (*XLR*, TOSlink fibre optic and two RCA co-axial), digital record out and digital tape loop. Also shown was the *CD-DSP*, Krell's first transport with built-in DSP software-based processor. This is based on the *MD-2* CD transport chassis and it offers three digital inputs to allow it to serve as a digital operations centre.

The tweakers haven't run out of toys for the hobbyists, with the green pen, for painting a CD's edges, now accepted as 'normal'. The main credibility stretcher at Las Vegas was a device consisting of a *Plexiglas* pipe on a stand, with loops of fishing line suspended from the top. This accessory, called a *Floater*, enables you to keep your speaker cables from making contact with the floor, without having to resort to hanging them from the ceiling. The price is four for $49 (£24), but you'll probably need a dozen or more to do a complete system.

And if that's not wild enough, then how about the **Tice** digital clock? Plug it in anywhere and it improves the sound of your hi-fi. Cost to you? Only $350 . . .

Ken Kessler

NAMM, Anaheim—a show report

Just as the October, 1989, AES Convention in New York was rocked by the Northern California earthquake that occurred the day before it started, the January meeting of the National Association of Music Merchants in Anaheim, California, seemed as if it might pale to insignificance in light of the belligerencies that had begun in the Gulf 2 days earlier. Certainly during set-up those booths with video monitors got more than their share of attention.

Four days long, the Winter 1991 NAMM show was the longest ever, as well as the largest. One great surprise came on the first day, when the organisation's board of directors announced the cancellation of the summer meeting. When the musical instrument business was centred around the Midwestern United States, the summer NAMM show was the major industry event, but in the years that manufacturing has gravitated toward the Far East, the California show has gained importance. Usually held in Chicago during June, the summer show this year was scheduled for New York City in August, which few were looking forward to, and so many considered its cancellation would be a form of euthanasia.

Consoles and automation

The parent company Conneaut Audio

Devices, well known for their rackmount mixers, late last year acquired **Musically Intelligent Devices**, makers of the *MegaMix* automation system, and have jumped into mainframe consoles in a big way. This show was their first opportunity to display their new direction. Their CAD *Maxcon* automated consoles will be available in widths up to 109 inches, with 144 inputs (288 at mixdown).

Allen & Heath were showing their new *Spectrum* desk (see elsewhere in this issue).

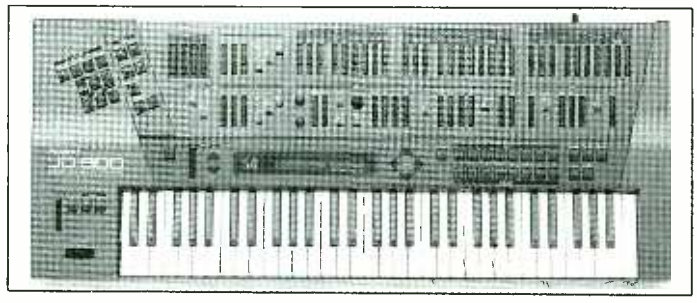
The Russ Jones Marketing Group, who distribute **Steinberg** software in America, were showing the low-cost *Niche* Mix Automation System, fondly known as the 'Fader Monster', a control device designed by Michael Stewart to accompany the Niche ACM MIDI-controlled fader box. The new controller contains 16 slide faders with mute buttons, five system buttons and a jog/shuttle/data entry wheel. The faders can operate in absolute, relative or immediate modes, and can be automatically nulled. Fifteen nested groups can be configured and 32 snapshots of faders and mutes stored on board. All controls send standard MIDI commands, so the box can be used in a wide variety of systems.

System mnagement, synch and software

J L Cooper Electronics were showing a clever device for improving the user interface of computer-based sequencing and editing systems. The *CS-1* has 11 programmable function keys, a four-button 'diamond' and a jog wheel, as well as jacks for two footswitches. It is designed to connect to the ADB bus of an Apple *Macintosh*, so the computer sees it as just another keyboard and mouse, but it puts the controls for programs like *SoundTools* or *Performer* into a much more usable form. All the functions are completely assignable under software and it is compatible with standard *Macintosh* macro programs like *QuicKeys*. Versions for other computers using MIDI or RS-232 communications will be available soon.

The company were also showing two inexpensive SMPTE-to-MIDI timecode converters and 'smart' FSK generators: the *PPS-2* and the *Sync Link*, which includes a *Macintosh* interface.

Anatek, best known for their 'Pocket' line of MIDI accessories, were showing something completely different: a mass-storage module



expressly designed for digital audio that combines a 640 Mbyte fixed hard disk with a DAT deck. The DAT portion can store up to 1.3 Gbytes on a removable 60 min tape for archiving. The device includes an SCSI2 interface.

The company also have two new programmable patchbays. The *SMP-7* handles seven MIDI inputs and eight outputs, with 64 patch memories, and it reads and writes SMPTE timecode. The *SMP-16* adds a 16×16 MIDI-controlled audio patchbay.

Midiman were showing the *Syncman Pro* SMPTE-MIDI timecode reader/generator/converter, with some unique features. It can 'spot lock' to a videotape containing just a SMPTE 'header' a couple of seconds long, following video sync after the SMPTE goes off. This allows users to lock to a videotape without sacrificing an audio track. It can also record up to 768 MIDI 'hits'—notes or controller changes—with their respective timecode numbers and replay them to, for example, trigger samplers for Foley work. The same hit list capabilities can be used to control punch-ins and -outs on a remotable tape deck.

Lone Wolf, makers of the *MediaLink* fibre optic-based Local Area Network (LAN) for distribution of MIDI, audio and video signals, were showing new network management software for Atari, IBM and *Macintosh* computers. The software can create a 'Virtual Studio' in which MIDI devices on various MIDI cables in different studios can be connected together merely by drawing lines on a screen, while the *MediaLink* hardware takes care of the physical connections.

Opcode Systems were showing a preliminary version of their *Studio V* MIDI system controller, featuring 15 MIDI inputs and outputs for 240 channels, internal storage of set-ups, separate audio and SMPTE inputs, and comprehensive realtime data mapping facilities. No delivery date is set yet. The unit will run under a new *Macintosh*-based operating system Opcode is calling 'Open MIDI System', or OMS, which is a MIDIQ-based LAN, providing compatibility with Apple's *MIDI Manager* system software and other existing MIDI 'super-systems' but with greater capability than these.

Opcode also announced that their universal patch librarian program *Galaxy* will be upgraded to include editing modules for a wide variety of synthesisers.

Along the same lines, **Dr T's Music Software** were showing a new *Macintosh* version of their *X-oR* universal synth editor/librarian. Without getting into detailed comparisons of these two programs, users might be interested in the fact that *Galaxy* will not support Oberheim's *Matrix* or *X-pander* synths, while *X-oR* will not work with the Kurzweil *1000/1200* series of keyboards and modules.

Synths and samplers

Roland's most interesting entry was a retro keyboard with something not seen for a number of years: plenty of knobs and sliders. The *JD-800* looks very much like an updated *Jupiter-8* and features discrete controls for pitch and volume envelopes, filters, LFOs and tone mixing. It includes 108 digital waveforms in memory, seven on-board effects, 24 voices and six-part multimbral operation. Roland were also showing the *S-750*, a version of the *S-770* top-of-the-line sampler but without an internal hard disk or AES/EBU digital inputs and outputs.

Korg were showing a rackmount version of their *Wavestation*. The *Wavestation A/D* has twice the PCM sample memory—4 Mbytes—of its predecessor, as well as expanded effects and wave sequencing capabilities, plus it has stereo analogue audio inputs, so real sounds can be manipulated by the unit's waveshaping and modulation circuitry, for vocoding, etc.

Peavey are becoming major players in the synth market and introduced a number of modules based around their *DPM3*. The *DPM SP* sample playback module will play up to 16 voices of 44.1 kHz samples, on 16 MIDI channels, through four audio outputs. Its internal memory is expandable to 32 Mbytes using standard SIMMs. It includes a floppy-disk drive, and it can send and load samples over SCSI. A companion A/D device for recording samples, the *SPM SX*, will be available. A new version of the synthesiser, the *DPM 3se*, is designed to control the modules and has 1 Mbyte of onboard RAM for editing samples.

A welcome trend among synthesiser manufacturers is drum playback modules. In the past, if you wanted to sequence with drums, you had to buy a drum machine, complete with unnecessary pattern-programming controls. Roland and Kawai recently introduced patternless percussion modules and two new units were being shown by E-Mu and Alesis.

E-Mu's *Procussion* contains 1,000 sounds in 4 Mbytes of ROM (expandable to eight), culled from the *EIII* sound library. Up to 32 sounds can be played at one time and four sounds at a time can be combined and crossfaded using velocity or controllers. The unit has six audio outputs.

Alesis's *D4* is on the other end of the price scale but has plenty of useful features. There are 400 editable sounds in ROM and four audio outputs. In addition, six audio trigger inputs are provided for accurate substitution of sampled drum sounds for live or recorded drum tracks.

Tape decks

Tascam announced the *MSR-16S*, the Dolby *S* version of their popular ½ inch 16-track deck. It costs more than the earlier version (which has dbx noise reduction).

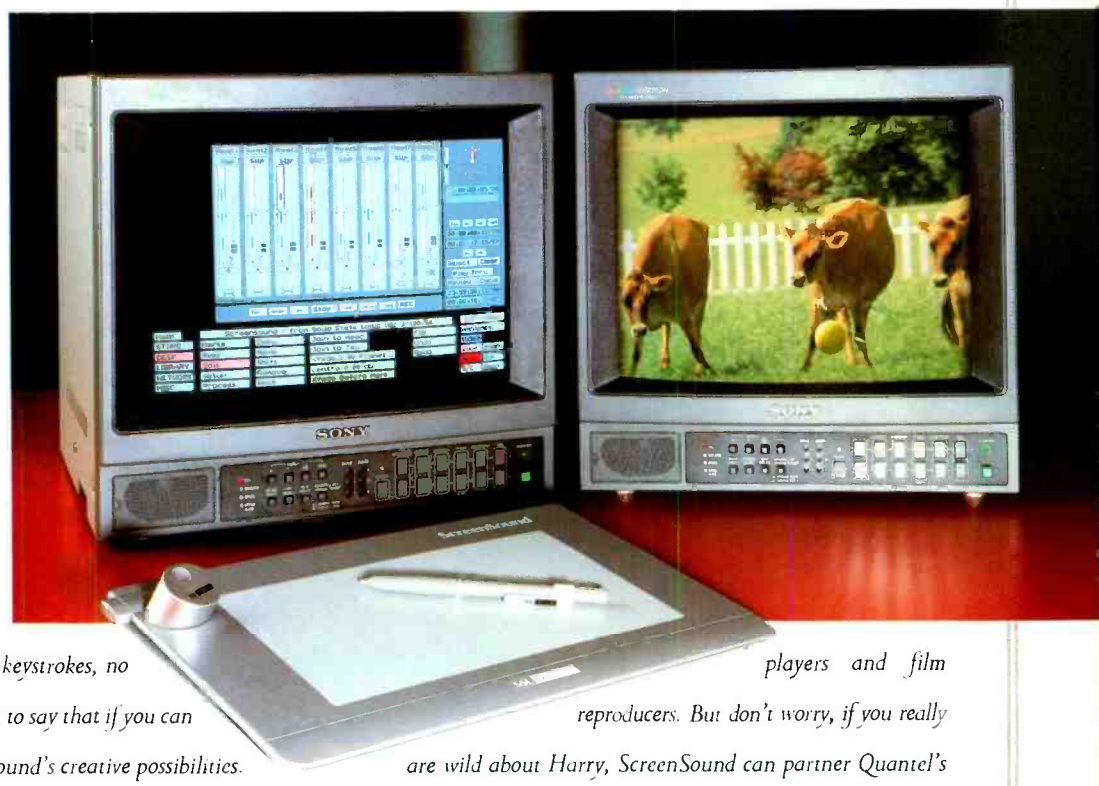
Alesis's *ADAT* was without doubt the most talked-about new item at the show. It is an 8-track digital recorder, with −10 or +4 analogue inputs, recording at 48 kHz on standard S-VHS cassettes at about 3× normal speed (45 minutes on a *T-120*), all at a very reasonable price. It comes with a remote control capable of linking two units together with a proprietary synchronisation scheme, which is accurate, according to Alesis, to a single sample. An optional big remote control (known as the 'BRC') will be able to handle up to 16 machines (128 tracks) with the same accuracy and provide SMPTE and MIDI timecode lock-up as well. AES/EBU digital I/O is planned for the future.

Paul D Lehrman

You don't have to be wild about Harry to be bowled over by new generation ScreenSound.

You don't have to run a digital video environment to make full use of ScreenSound's powerful new capabilities. Following an intense period of R&D, ScreenSound is now fully equipped to perform as the essential digital audio command centre for film and video.

As the name suggests, ScreenSound is all about editing audio, visually. Music, dialogue and effects can be edited, polished and laid back to film or video with sub-frame accuracy. Meanwhile, all this is achieved without incurring the generation losses that have always afflicted analogue audio. It couldn't be easier to use, thanks to a highly intuitive user interface that brings operator and system closer together. Simple, gestural movements of the cordless pen on ScreenSound's tablet provide control over all functions. No fiddly keystrokes, no mousey manouevres. At HHB we like to say that if you can hold a pen, you can explore ScreenSound's creative possibilities.



With SSL's new 'SoundNet' technology, archiving and uploading are instant procedures. You can even share and copy work between multiple ScreenSound command centres or play back as many as 56 channels of digital or analogue audio. Audio tracks appear on screen as "reels" of tape that can be edited, timeslipped, and crossfaded. Gain and pan position can be automated against timecode while an 'audio scrub' facility permits accurate manipulation of edit points. Thanks to powerful search and sort routines, ScreenSound helps you maintain an accurate record of sound clips held within the system, while a WORM optical disc subsystem can provide massive off-line storage. With the 'SoundNet' option you can obtain instant on-line access to as much as 48 hours of audio. Naturally, a unique degree of operational speed and production efficiency are principal system benefits: ScreenSound can control up to sixteen VTRs, videodisc players and film reproducers. But don't worry, if you really are wild about Harry, ScreenSound can partner Quantel's system to provide operators with unparalleled levels of control.

Of course, ScreenSound comes from Solid State Logic, one of the world's most respected pro-audio manufacturers. But in the UK, new generation ScreenSound can be found only at HHB. And since HHB is widely recognised as Britain's leading source for digital audio technology, you also access the best expertise and the finest service in the business. So contact us now for a demonstration or to receive more information on ScreenSound's expanding facilities.

SSL DIGITAL

HHB COMMUNICATIONS LIMITED, 73-75 SCRUBS LANE, LONDON NW10 6QU PHONE 081-960 2144 TELEX 923393 FAX 081-960 1160

Still from 'Anchor Butter Cows' used by kind permission of The Moving Picture Company and Saatchi & Saatchi.

Saturn 624 multitrack

Saturn Research have introduced a mid price 2 inch 24-track tape machine based on the established *824*. The *624* uses the same chassis design as the *824* as well as the same transport, heads and audio electronics and so, is closely related. Saturn say that cost reductions have mainly been made be replacing the *824*'s digital alignment system with conventional trim pots as well as in other non critical areas.

The *624* also has a new standard remote although an optional autolocate remote is available.

Saturn Research, Unit 3A, 6-24 Southgate Road, London N1 3JJ, UK. Tel: 071-923 1892.
USA: Redwood Marketing, PO Box 270007, Nashville, TN 37227-0067, USA. Tel: (615) 254-7400. Fax: (615) 242-5774.

Rack Macs

Current Music Technology is a US company specialising in rackmount and ruggedised versions of Apple *Macintosh* products for pro applications. An extensive range of new products has recently been added. The *Macintosh Classic Support* is a modification of the existing *Mac'n Rak* chassis to support the *Mac Classic*.

For mounting *Mac II cx, ci, LC* and *SI* there are two alternatives—a rugged chassis with sound damping, additional cooling and a further hard drive bay; or a specialised shelf without the other features for fixed installation. For the *Mac II, IIx* and *IIfx* a rackmount chassis is available with extra cooling and vibration damping. CMT are also offering rackmount versions of IBM compatibles.

Current Music Technology, 146 Paoli Pike, Malvern, PA 19355, USA. Tel: (215) 647-9426.

Gefell from the East

In 1943, before the Iron Curtain was erected, Georg Neumann founded a company in East Germany, Microtech Gefell GmbH. For over 30 years Gefell was the exclusive manufacturer of broadcast condenser mics for the Eastern Bloc. Following the reunification of Germany, and with Neumann's assistance, The Gefell range of mics has been brought up to western audio standards.

So far there are three models available and these are, the *MV692/M93* omnidirectional, *MV692/M70* cardioid and the *MV692/UM70* switchable omni/cardioid/fig-of-8. We understand that these mics will also be very competitively priced.

UK: FWO Bauch Ltd, 49 Theobald Street, Borehamwood, Herts WD6 4RZ. Tel: 081-953 0091.
USA: Gotham Audio Corp, NY.

Allen & Heath Spectrum

A&H have launched a console designed as a smaller version of the *Saber* and includes most of its features. The console is available in 16-, 24- and 32-input frames with the possibility of adding a further eight channels to all sizes. The meter bridge can be repositioned following extension so that it remains central. All inputs are available on remix and there is full EQ on all inputs. There are six effects sends, a stereo effects return and four mono returns. Inputs are switchable +4/−10 dB individually and each channel has a direct output. There is a choice of bargraph or vu metering.

Allen & Heath, 69 Ship Street, Brighton, Sussex BN1 1AE, UK. Tel: 0273 23346.
USA: Allen & Heath, 5 Connair Street, USA.

The line and mic inputs, as well as recording outputs, are transformerless designs which use proprietary, laser-trimmed, thick-film resistive networks. This provides highly accurate tracking of the output stages, which clip only at levels that exceed +27dBu.

Available in configurations from 24 to 50 inputs, the Series 34C may be equipped with Otari's DISKMIX™ *Moving Faders* or DISKMIX VCA automation systems. And you may choose either high resolution LED metering or VU mechanical metering.

Quality construction, superb sonic performance, reasonable cost, and Otari worldwide customer support make the Series 34C the perfect choice for the recordist, and for the studio owner. Call your nearest Otari regional office for information.

Otari, Inc., Tokyo (0424) 81-8626
Otari Corp., California (415) 341-5900
Otari (UK) Ltd., United Kingdom (0753) 580777
Otari Singapore Pte., Ltd. (65) 743-7711
Otari Deutschland GmbH., West Germany 02159/50861-3

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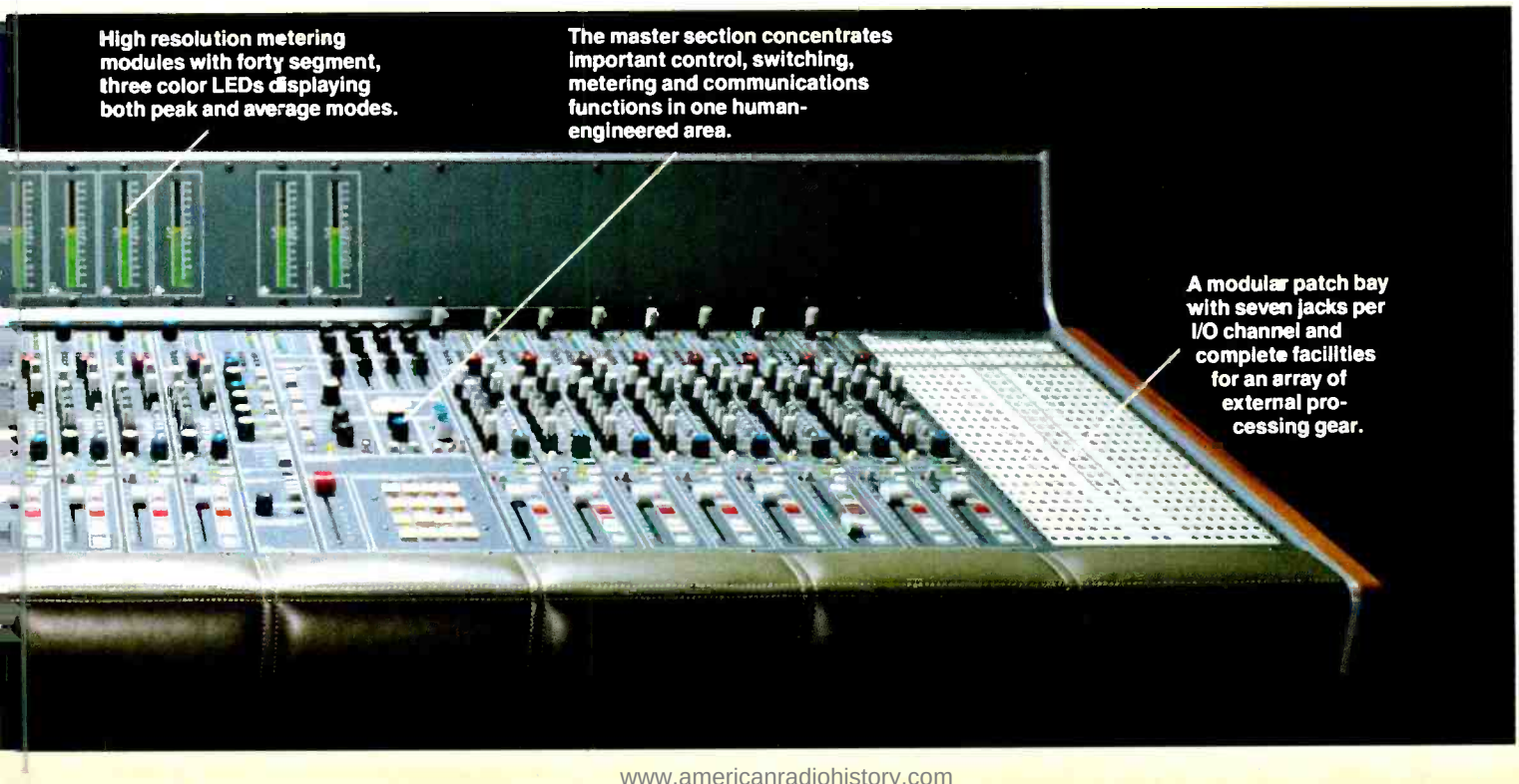

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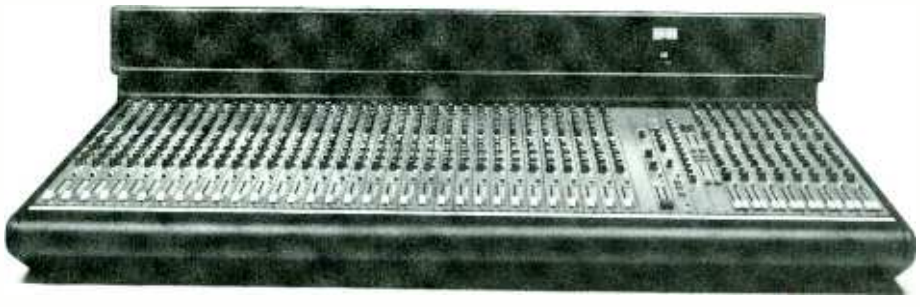


Concept Series Modular Mixing Consoles

Among the standard options available on the **Concept** series of mixing consoles are choices of: Eq type, frame size, subgroup type, number of aux sends, type of metering, degree of mute and fader automation. Special requirements can also usually be accommodated. The consoles can be configured for film or sound recording, theatre applications, broadcast and live sound reinforcement. You have the freedom of choice over the facilities you want. One thing is always standard however – the transparency of sound and technical excellence that the Concept series has become renowned for.

Hill
AUDIO

Hill Audio Ltd., Hollingbourne House Hollingbourne, Maidstone, Kent ME17 1QJ England. Telephone: (0622) 880555 Telex: 966641 Hill G Fax: (0622) 880550)
Hill Audio Inc., 5002B N. Royal Atlanta Dr, Tucker, GA 30084 USA. Telephone: (404) 934 1851 Fax: (404) 934 1840 Telex: 293827 HLAD

Azonic Acoustical Analysis Service

Azonic Inc, a US manufacturer and distributor of acoustic control products has introduced an acoustical modelling and analysis program. The Azonic Acoustical Analysis Service (AAAS) is a three-dimensional modelling program intended to maximise acoustic control within the client's environment. The program can simulate the use of Azonic noise and reverb control products within the chosen space to meet desired acoustic needs. The customer uses a Sound Analysis Kit to collect the data needed. This includes a starter gun, tape recorder and a detailed questionnaire requiring information about physical dimensions, constructional materials, usage, etc. The gun is recorded firing in different specified room locations and all the information sent to Azonic who analyse the recordings together with the written data. The completed analysis provides graphs and recommendations for treatment to achieve desired performance. Azonic are claiming substantial cost savings due to analysis before starting a project as well as a guarantee on performance to within ±3 dB if the instructions are followed.

Azonic Inc, 1610 East Cliff Road, Burnsville, MN 55337, USA. Tel: (612) 894-2230.

DDA updates

DDA have produced a Mute and Solo MIDI interface for the *DMR12* console featuring up to 10 mute groups, 128 console patches and a solo-in-place facility. All mute data can be recorded via the standard MIDI link to an external sequencer or MIDI recording system. The system is known as MUSIC (MUte and Solo Interface Computer).

DDA have also announced a collaboration with Uptown and Audiomation to enable use of the Uptown moving fader automation system in the *AMR24* and *224V/232* consoles. Uptown have designed custom panels so that the faders will fit into the consoles fader wells. DDA are developing an interface that will allow all the switch data of the *DCM224V* or *DCM232* to be stored and edited on the same computer used for the Uptown interface. The Uptown system can also provide machine control using a connection to the Adams-Smith *Zeta 3* synchroniser.

DDA, Unit 1, Inwood Business Park, Whitton Road, Hounslow, Middx TW3 2EB, UK. Tel: 081-570 7161.

USA: Klark-Teknik Electronics Inc, 200 Sea Lane, Farmingdale, NY 11735. Tel: (516) 249-3660.

WaveFrame software updates

WaveFrame have announced updates to their system software (*Version 4.09*) and operating system (Microsoft *Windows 3.0*), which they say, will significantly enhance the system's multitasking performance as well as allowing faster operating speed and flexibility. The user interface has improved with the screens updating more smoothly and looking sharper. Timecode values can now be selected and moved around the screen for instant editing and location purposes. In the disk recorder a new Cue window can be opened offering the user 10 timecode boxes to speed the setting up of cue lists. There are also software enhancements to the mixer permitting modified gain structures and a new mixer preset system that allows individual or grouped channel values to be stored for recall. Other software enhancements with *Version 4.09* include a more efficient user interface for sampling, an improved back-up and restore interface and a soundstore database printing utility.

WaveFrame Corporation, 2511 55th Street, Boulder, CO 80301, USA. Tel: (303) 447-1572.

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

E-V dynamic mic

Electro-Voice have launched a new cardioid dynamic mic known as the *RE38N/D*. This mic is similar to other E-V N/D series mics but offers an extended HF response and variable EQ switching offering 16 different possibilities (eight LF positions and two HF). E-V say that the vibration isolation eliminates the need for external shockmounts. The pivot arrangement for holding the mic has detented positions to prevent slipping in use.

Electro-Voice, 600 Cecil Street, Buchanan, MI 49107, USA. Tel: (616) 695-6831.

UK: Shuttlesound, 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX. Tel: 081-640 9600. Fax: 081-640 0106.

Calrec RQ series

Console manufacturer Calrec have announced the *RQ* series of rackmounting units. The first is a mic/line preamp *RQP3200*, which incorporates a 4-band parametric EQ with variable HF and LF filters plus a compressor/limiter with an expander/gate section.

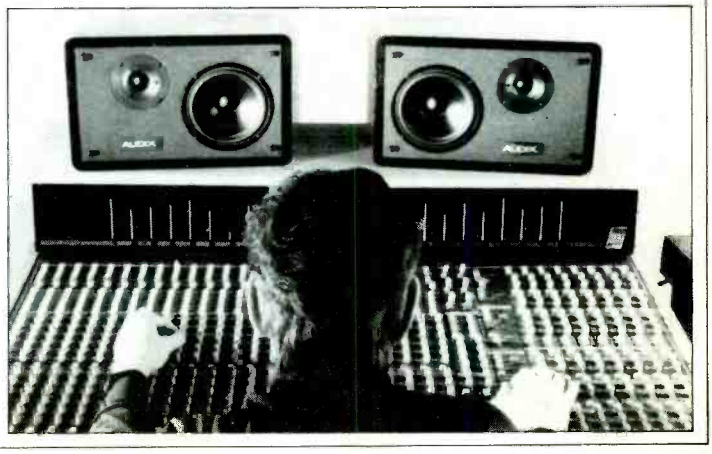
The second unit in the series is the *RQD6400*, which is a 1U twin stereo compressor/limiter with voice-over control, separate compression and limiting functions with high resolution 20-segment gain reduction bargraphs. More units will be added to the range thoughout the year.

Calrec Audio Ltd, Nutclough Mill, Hebden Bridge, West Yorks HX7 8EZ, UK. Tel: 0422 842159.

US Audix monitors

The US Audix company has launched a 2-way nearfield monitor known as the *HRM-1*. The LF driver is a 6½ inch polypropyline unit while the HF is handled by a polyamid dome tweeter. The crossover is at 3 kHz with 24 dB/octave filter slopes. Audix claim that the use of a 1½ inch voice coil on the LF unit together with the vented cabinet design gives a power handling of 150 W and a −3 dB response of 48 kHz in a cabinet of 16×10×7¼ inches. Audix have used a Neoprene faceplate to cover the front cabinet surface to reduce cabinet diffraction problems and the speaker pairs are component matched and supplied as symmetrical sets. The *HRM-1* is available in Natural Oak or Black Oak finishes.

Audix, 5653 Stoneridge, Pleasanton, CA 94566, USA. Tel: (415) 463-1112.



Carver can handle it!

Whatever the size of your venue Carver can handle it.

Carver amplifiers are in use at the largest stadia in the world and at the smallest clubs.

There is a new range of six models from the PM-120 offering 40 watts per channel into 8 ohms up to the PT-1250 offering 465 watts and weighing an unbelievable 11lbs.

Carver professional amplifiers are renowned for their pure sonic excellence, portability and the ability to produce awesome power.

Whether your audience numbers a hundred or a hundred thousand, Carver won't let you down

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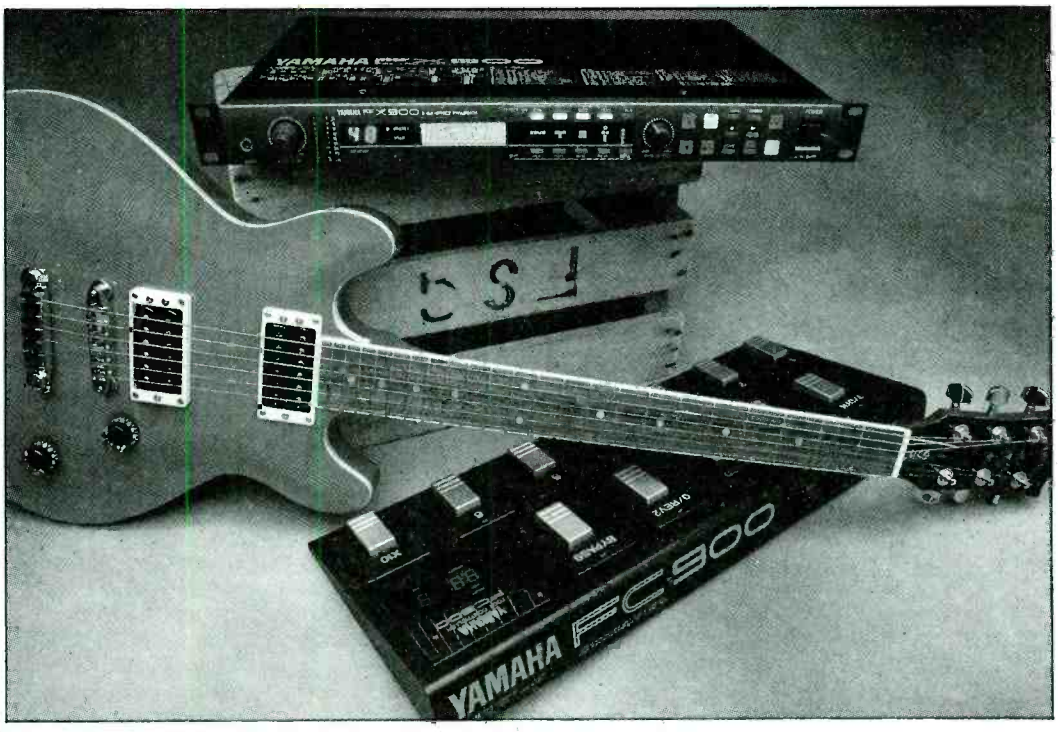
Sole UK Distributor: HW International, 3-5 Eden Grove, London N7 8EQ. Tel: 071-607 2717



Yamaha effects for the player

While Yamaha's Digital Systems package has been attracting a lot of attention with the integration of its *DMC1000* digital console, *DRU8* digital recorder and *DTR2* DAT machine, the company have also found time to release a number of products for the musician market.

The *FX900* Simul Effect processor allows seven programmable effects to be processed in stereo in 100 factory and 100 user memories. As applicable to a guitar rack as to a studio, the effects provided include compression, parametric EQ, reverb, early reflections, modulation, distortion and wah effects. Flexibility is ensured with the inclusion of realtime MIDI control of parameters. A/D conversion is $\Delta\Sigma$ with 18 bit, 8× oversampled DAC.

Downmarket from the *FX900* is the *EMP100* multi-effects unit offering early reflections, reverb, pitch shift, delay, chorus, flange and a number of combination effects arranged in 100 factory and 50 user memories with 20 kHz bandwidth and 16 bit quantisation. Interestingly the unit resurrects the concept of the tap tempo footswitch for delay time setting.

Building on Yamaha's longstanding relationship with bass players through their highly regarded range of bass guitars and in line with what other manufacturers are doing (see Korg story), a version of the company's *FX500* effects unit has been remodelled for the four-string player in the guise of the *FX500B*. Sharing many of the original's features and performance, the *FX500B* has been optimised for the task in hand with 30 factory presets combining overdrive, compressor, EQ, modulation and reverb/delay. Bassists can also indulge in realtime MIDI manipulation of two parameters per patch and store configurations in 30 user memories.

Rackmount Wavestation enhanced

Korg continues to dominate and innovate in the keyboard market with the release of a rackmount version of the *Wavestation*, which has been enhanced to include many popular features.

Sharing all the features of its keyboarded brother, the *WS/AD* (analogue-to-digital) can additionally incorporate sounds received at its inputs into existing presets as waveforms, or process sounds through the unit's internal circuitry. An obvious example of the latter would be the transformation of a vocal input into vocoder-type effects. Korg expect deliveries to start in June/July and are clearly maintaining their position as one of the few synth manufacturers to add value consistently to their existing products in a manner that is immediately palpable to the user.

With a long tradition of building outboard processors, Korg can be accused of having been a little quiet in this area of late aside from their ventures with the *A3* guitar multi-effects unit. The balance would seem to have been redressed with the launch of five units.

The *A5* standalone effects unit offers chains of five effects optimised for use by guitarists (*A5GR*), bass players (*A5B*) and keyboard players and vocalists (*A5FX*) in 30 presets with 44.1 kHz 16 bit sampling. Intended to replace the aforementioned *A3* guitar multi-effects unit, the rackmount *A2* offers 97 chains of effects in a stereo in and out configuration. Finally, the top of the range 2U *A1* will also find applications in the studio and permits seven stereo effects to be combined in any order in 20 custom chains and sports the added bonus of *XLR* connections.

Studio Sound's Music News is compiled by Zenon Schoepe

MIDI on view

Monitoring a MIDI datastream to isolate a problem is no longer the exclusive realm of hex-maniacs with the arrival of Datastream's MIDI *Viewport*. Featuring a MIDI in and out and powered by four AA batteries in a unit the size of two cigarette packets, a 20×2 LCD shows incoming data in hex and, more importantly, plain English. The last 64 bytes of data received are held in memory and can be scrolled through at will. A test function is additionally included that emits note C3 on all channels for checking network connections and MIDI cables can be tested by bridging across the unit.
UK: Digisound, 16 Lauriston Road, London SW19 4TQ. Tel: 081-946 0467.

Rolls rolls in to rock

Started by ex-DOD employee David O Di Francesco, great things are expected of American company The Rolls Corporation who are placing a distinct emphasis on the requirements of the guitarist who wants to keep abreast of technology.

The *HA32* is a full function 2-channel belt pack amplifier in a hardened steel case with easy access effects switches mounted on top of the unit. Features include chorus with regeneration, level and tone controls, aux output and input, and a stereo headphones output. The unit is powered by 9 V batteries or an AC adapter and can be injected directly into a console courtesy of a built-in speaker simulator.

The *MP45* MIDI tube pre-amp sports the resonance of 12 *AX7* tubes with stereo chorus effects, 5-band EQ, tube gain and a presence control. All settings are programmable and can be recalled via MIDI.

The strong need for MIDI pedals among guitarists would seem to be satisfied by the company's *MP80*. Banks and patches can be selected and recalled by a double tap on the 15×5×1½ inch unit, which also houses two LED displays. The foot pedal can be phantom powered.
The Rolls Corp, 7023 South 400 West, Midvale, UT 84047, USA. Tel: (801) 562-5628.
UK: Trevor Cash International, 1 John Oliver Building, 53 Wood Street, Barnet, Herts EN5 4BS. Tel: 081-449 5566.

Comment

The 'real time' Gulf War makes this page unusually difficult to produce with printing lead times meaning news may well be swept aside by the pace of events. In early February, the entertainments industry was already feeling the effects of war as reports of cancelled or postponed shows multiplied. And it comes at the time when European sound reinforcement business is normally confirmed for the peak spring/summer touring season. ML

Dire Straits postpone

Dire Straits' mammoth world tour, planned to start in Dublin in June and finish in Spring 1992, was one of the first major schedules to suffer postponement because of the Gulf War.

Ed Bicknell of Damage Management said the first dates would not take place until August at the earliest, and that key production crew have been retained "on standby". He explained the decision had been taken reluctantly and cited concerns over venue security, air travel, insurance costs and "the difficulty of predicting the situation in 6 months time."

Ironically, just days before the announcement, Britannia Row were confirmed as SR company for the tour after an evaluation by Straits' representatives of Turbosound's *Flashlight* SR, MSI and Concert Sound systems behind closed doors at the Brixton Academy.

Tour news

This section was compiled from information available up to February 5th.

• **Audiolease** recently finished a tour with EMF and now have 7 weeks with Motorhead (UK/Europe). The SR system, comprising 24 *MSL3*s and 16 subs, is interesting, said Steve Sunderland, "because Lemmy's vocals will be intelligible for the first time"—with a vocal processing 'chain' consisting of Drawmer gate, K-T graphic and a compressor. With 120 dBA out front that's some feat.

• **Britannia Row**'s confirmed tours included Robert Cray, Aswad, David Lee Roth's stadium dates (using 80 Turbosound *TMS-3*s), Cliff Richard's gospel tour, Gloria Estefan and James Last. Dire Straits postponed, and Whitney Houston's visit was also in doubt because of the Gulf crisis. This year's Glastonbury Festival—a regular Britannia Row event—was cancelled by farmer/promoter Michael Eavis. He apparently threw in the towel after remnants of the 'hippie convoy' declined to leave the site.

• **Clair Brothers** had European tours by Bob Dylan, Sting and Paul Simon on their roster. Greg Hall, commenting on the Gulf Effect from Clair's US HQ, said: "There's concern about transportation and security. But so far there have been no changes in their plans."

• **Electrotec** UK's Ian Calendar said the company has Megadeth coming to the UK and Europe in March, and major shows by Rod Stewart from April. Rod's tour is being sponsored by Pepsi—who also backed his hit duet with Tina Turner (*It Takes Two*) as well as Tina's 1990 shows.

• **Showco** are bringing over their *Prism* system for George Michael's Wembley Arena dates in March. ZZ Top's European shows are also on their agenda.

• **SSE**'s Chris Beale said smaller firms are suffering a secondary impact of the Gulf crisis as US acts cancel big tours—hire rates are being slashed as companies try to employ their idle assets. However, SSE has a healthy roster with Iggy Pop and Dream Academy now and, from March, the RTE Irish Music Awards in Dublin, Big Dish, Little Angels, AC/DC, UB40, MC Hammer and White Lion. Another 14 *MT* systems are on order plus TAC's new *SR6000* console.

• **Star Hire**'s mobile *Contain-A-Stage* concept—a trailer which unfolds into a covered stage complete with lighting, SR and power supply—has its third successive year on the Bailey's Irish Cream 18-date tour of shows at UK stately homes, and a contract for 27 similar venues with the 'Fireworks and Laser Symphony Concerts' series.

News roundup

• **Martin Audio** used the Frankfurt Music Fair for the official launch of two products that surfaced as prototypes at last year's PLASA show.

The *F1* combines many of the company's flagship *F2* system attributes in a compact two-box system, which, say Martin, is ideal for touring and fixed installations "which do not require the interchangeable rackmount facility of the *F2* top box". An *F2*-compatible flying system is a retrofit option.

Inside the 'top box' are a 12 inch mid/horn and 1.4 inch exit titanium compression driver, together giving 65° horizontal dispersion. Two high efficiency drivers are mounted in the ported LF enclosure. The new *PM3* bi-amped monitor has a 15 inch LF driver, mid-range horn and 1 inch compression driver. The 'quasi-trapezoidal' cabinet is designed for corporate presentation, music playback and high-level stage monitoring, say Martin—adding that "the design of wedge monitors is something only a handful of PA speaker makers take responsibility for."

• High technology on the dance floor is afoot with **Stage Accompany** UK's latest installation at Cairo's nightclub in Swindon. Employing *Enhanced Ambisonic Surround Sound* processors from AGM Digital Arts (of Ely, Cambridgeshire) and Stage Accompany's *SANet*—a PC-based serial control network commanding *SA* processors, amplifiers and parametric EQ—the complex system feeds a matrixed array of Ambisonic *SA Performer* range speakers.

• **ANT Telecommunications** in Slough have announced the *Radio PA* system. Technical details were not supplied for this new wireless PA distribution system, although ANT say the receiving speakers can be "literally miles apart" from the central operating console—and that speakers can be activated singly or in groups. It's aimed at the public address/safety market but we will be interested to see whether the system's audio performance opens potential sound reinforcement applications.

• Thai cheer: **Celestion** say Thailand's Fong Nam orchestra were delighted to discover their UK concerts this year will use an identical sound reinforcement system to their own rig in Thailand—a Celestion *SR*. The orchestra say the *SR* is the only system that can faithfully reproduce its traditional instruments, such as the glass xylophone, as well as modern instruments and samplers.

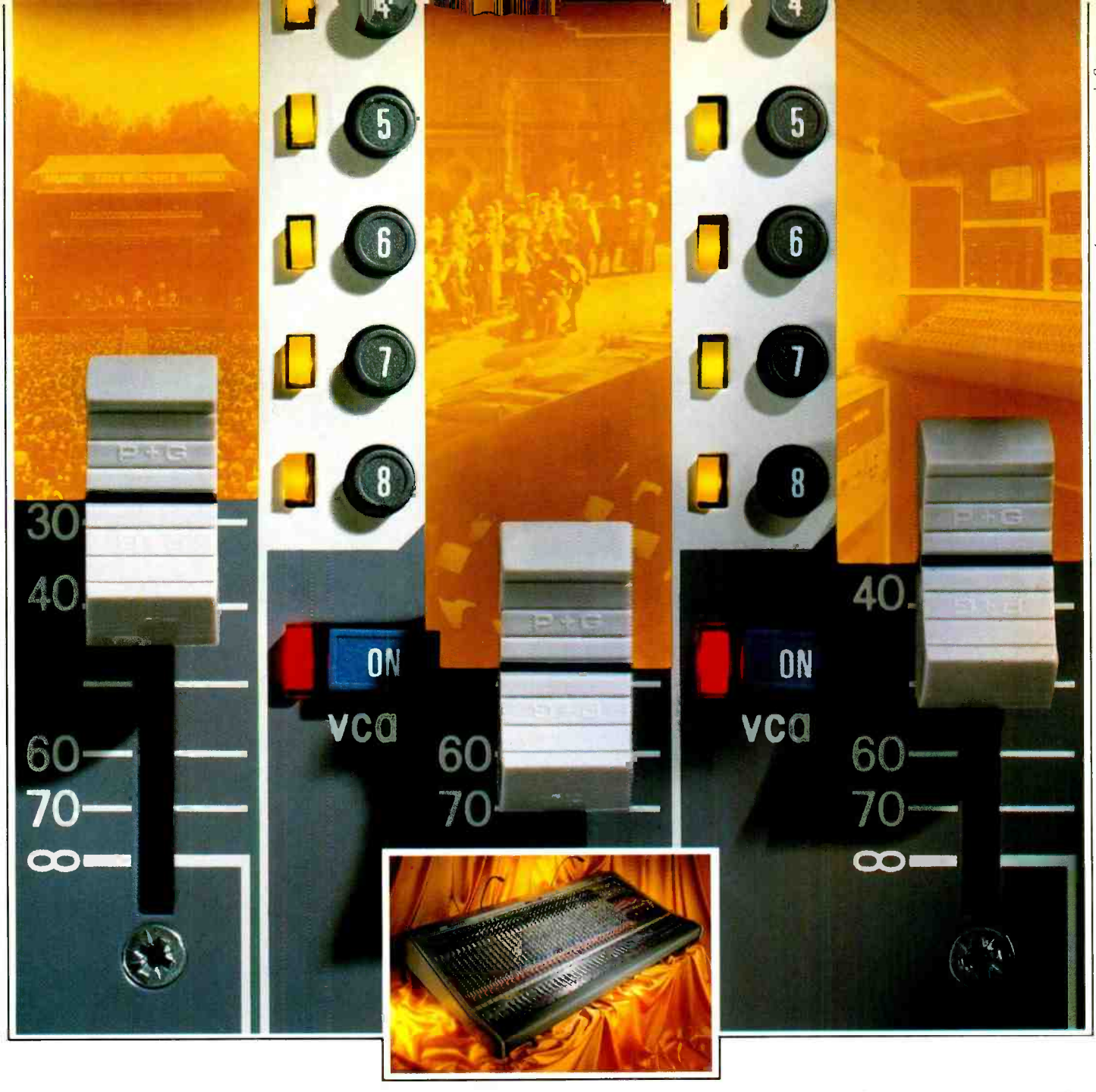
• **Yamaha**'s new *PM1200* console has had two UK theatre debuts—on 'portable' duty between control room and FOH at the 1000-seat Marlowe Theatre in Canterbury and in 24-channel FOH role at the Sadlers Wells Theatre.

• **Theatre Projects** have launched an Atari *ST*-based MIDI computer control system, designed to "dramatically improve co-ordination between sound and lighting technology, including Vari-Lites". The likeliest buyers are long-run theatre productions.

• **Nexo** introduced two products at the Frankfurt Music Fair. The *TS2400* is the first and smallest sample of a new professional touring speaker range—with larger and more powerful long-throw enclosures to follow. Nexo say large-scale applications have been targeted with new technology drivers including a 3 inch diaphragm neodymium HF unit, consideration of different flown coupling configurations and truck-friendly enclosure sizes. The system uses a dedicated TD controller unit. Alongside it is the new *L-Sub* range of subwoofers—the *LS2000* and *LS1500*—providing servo-controlled low and very low reinforcement in tandem with the *L-Sub TD Controller*. The latter controls variable phase, delay, 12/24/48 dB/octave slopes and integral test signal generator.

• **Backroom Rentals**, the specialist power supply and control systems hire firm, is now the sole focus of Laurence Dunnett's attentions. Until late last year Dunnett ran the company in parallel with his co-partnership in Electrotec UK. He has now decided to back 'growing demand' for his expertise with complex PSU systems—uninterruptible supplies being one of his specialities. His equipment will power the Pet Shop Boys' forthcoming live dates. With its lavish use of MIDI controlled sequencers, effects and samplers it's an example, he says, of technology-dependent performances demanding truly secure power supplies.

Studio Sound's Live Sound news is compiled by Mike Lethby



Photograph of O.B. Vehicle courtesy of N.O.B. Holland.

Every performance needs the Midas touch

XL3 LIVE PERFORMANCE CONSOLE

The new Midas XL3 Live Performance Console is a remarkable step forward in console design.

Created by Midas and Klark-Teknik, it combines in one console front of house and monitor mixing for live performances in sound reinforcement, theatre, major installations and broadcast.

For each input channel, no less than 18 sends can be individually routed to 16 fader controlled outputs which function either as sub groups or aux send masters assignable via 2 VCA output groups into stereo masters. In addition, any input can be assigned directly to stereo masters and 8 VCA sub groups. That's why the XL3 is the only console that can give you a 40 channel front of house mix one night and 40/18 channels of monitor mix the next. And if you need further outputs, the Midas XL88 external 8 by 8 line level matrix mixer provides the perfect answer to signal distribution in any live environment.

Add to this outstanding versatility, a crystal clear, logical control surface, plus Klark-Teknik's renowned electronic design and the result is a console with superb sonic performance that's pure joy for the engineer.

Call Midas today and experience the Midas touch for yourself.

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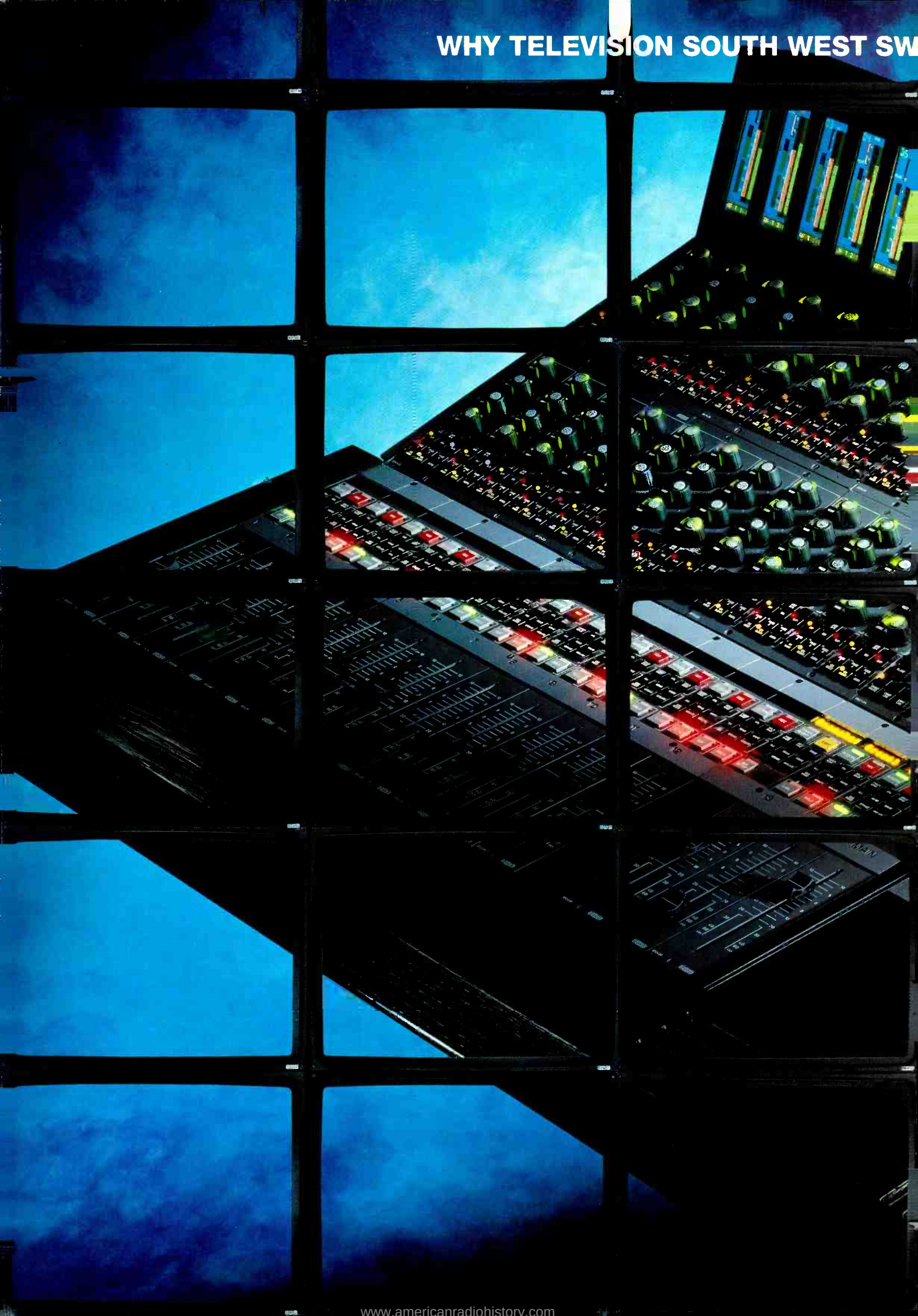
Tel: (0562) 741515 Telex: 339821 KLARTK G Fax No: (0562) 745371.

Klark-Teknik (Singapore) Pte Limited
7500A Beach Road, 04-308 The Plaza, Singapore 0719. Tel: 293 9736 Fax No: 293 9738.



"We were so impressed by the Logic 1, we felt we had to have one for the Dubbing Suite" That was the first impression of Chris Woolf and Paul Roberts, engineer and operator at Television South West.

After 6 months heavy use at TSW they are convinced the AMS Logic 1 is the finest mixing console available. *"Far more powerful than you would believe from its size."*

The Logic 1 combines analogue control surfaces with digital sound technology. Easy to operate, the console interfaces with the AudioFile, labelling and routing digital sound profiles. According to Chris, *"It is simply not possible to distort on the mixing busses."*

The power of full dynamic automation, unique to AMS, gives operators like Paul total repeatability in real time. The cleanliness of digital equalisation confirms the superiority of the Logic 1.

The AMS Logic 1 is quite simply the finest mixing console available.

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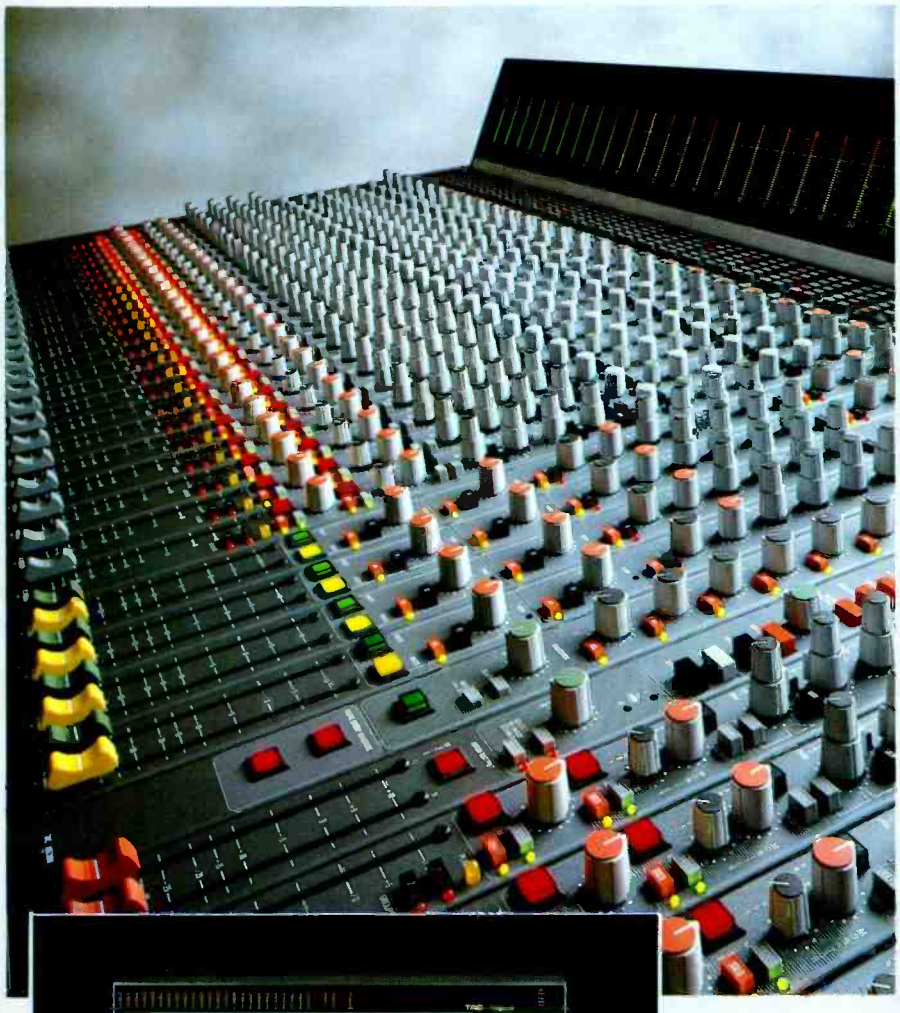

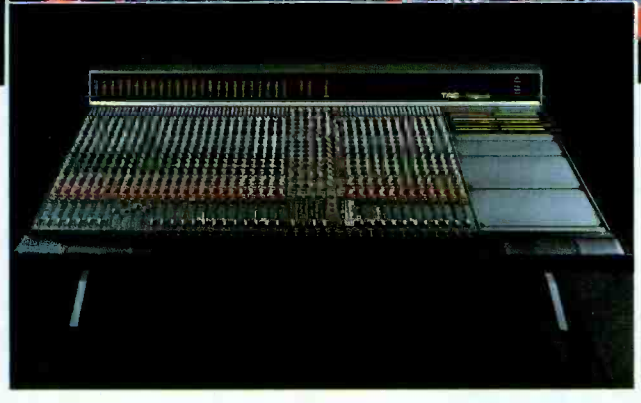

You need an edge. The world is competitive and you're creative but hard-pressed. You need a console that allows your creativity to shine through without compromising your finances.

Building consoles to meet that specification is what TAC does to perfection. We deliver an unbeatable combination of performance, features, construction and price.

We know console technology. The MAGNUM is proof positive and is perfect for multitrack, video post and broadcast production. Look at this checklist for comparison:

CONFIGURATION: 24-buss in-line console with dual-input technology, allowing each channel to have two independent signal paths.

EQUALIZER: the powerful 4-band semi-parametric equalizer can be split between channel and monitor paths, giving 72 inputs with eq on mixdown in the 36 input chassis.

AUXILIARIES: 8 Auxiliary sends – four mono and two stereo – can be fed from channel or monitor signal paths. Sends 7 & 8 can also be routed to the 24 busses for extra effects feeds. 8 effects returns, with sends to all 8 Auxiliaries, are provided.

SOLO/PFL SYSTEM: MAGNUM has a Master Status controlled in-place solo switch on both channel and monitor paths. When de-activated, solo switches provide a full Pfl facility. MAGNUM also has two independent Mute groups.

AUTOMATION: channel mutes can be automated via MIDI; C-MIX fader and mute automation system can be fitted.

CHASSIS: welded steel with a pcb-based bussing system and an expandable 288-point TT jackfield.

Some of the many other vital facilities include channel fader reverse, 30-segment LED meters with peak hold, optional stereo line inputs, 8 audio subgroups and comprehensive control room monitor system.

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TAC

DIGITALLY CONTROLLED ANALOGUE MIXING 10 YEARS ON

Console designer Richard Swettenham examines design development over the last 10 years and considers the current validity of a digitally controlled analogue mixing console

In 1981, the author presented in a US journal[1] a general picture of the possibilities of a logic and voltage controlled analogue mixing system with a control surface containing only DC voltages and digital signals. All audio was contained in separate racks. The article showed that the whole of the control information might be memorised and read back by a computer in step with timecode, providing automation of all functions. In the 10 years since then, it is perhaps surprising that such systems have not come into more general use. This article examines why this is so, and presents once again the possibilities opened up by such systems.

It is not certain when the concept of the fully assignable mixing console began but the user demand for such a thing probably originated in the BBC around 1978. At this time fader automation in music mixing consoles, after a slow build up over about 10 years, had become solidly established. Originally all the fader information had been stored on one or more tracks of an audio recorder but the limitations of this led to the recording, instead, of a much simpler timecode, which was used to synchronise mix information stored in computer memory with the movement of the tape. This use of timecode, learned from television, where the electronic 'frame' derived from video, enabled any kind of event to be related to the time reference of the primary soundtrack. As the desire for more tracks in music recording had grown, and post-production sound sweetening had become more usual in television, synchronising of tape recorders engaged the attention of manufacturers and led to the availability of units to lock anything to anything, audio or video, so long as a timecode track existed. At the same time, arrangements for static memorising of console track assignments had come into use, implemented by relays and then by solid state switches. Putting all this together, channel mutes, changes of routing, and triggering of external functions such as start of tape machines and changes to outboard effects, could all march along with the music to an accuracy of a frame (40 or 33 ms) or better.

Beyond fader automation

At the end of the '70s it was a great source of debate among recording engineers, and one which has never really ceased. How much automation and memory is really useful in the mixing operation? Is it worth carrying it further than faders and mutes? How much is worth paying for? Of course, the ability to recall routing set-ups for items and sessions was a useful time saver but how much more?

There had been a few attempts to produce an automated equaliser for consoles but the limitation of usable circuit techniques and components meant they attracted little acceptance. Blocks of remotely controllable equalisers were offered by NTP of Denmark and Param in Germany, but were not totally satisfactory, and did not integrate with any master time reference. The technology was not really ready.

The gap was temporarily bridged by SSL's introduction of *Total Recall*, which showed a video screen image of where each physical control had previously been set, enabling them to be reset manually. It was a further time saver in setting up and resuming previous work in progress, perhaps in another studio but it only scanned and logged the setting when it was told to, and did nothing else.

Given a conventionally constructed modular console there was really no alternative, other than lots of little motors turning the knobs.

Digital dreams

By 1981 Calrec had supplied a prototype mixer to the BBC for location use, with full remote operation of EQ, mic gain and other functions, but it did not go into operational service. Nothing was yet offered on the commercial market. Probably the brakes were being held on by the 'wait and see' attitude over what would come out of the fast advance of digital audio tape recording. Would this and the forthcoming CD make it essential for top level mixing desks also to be digital? Would they really be 'better' and make everything analogue look outdated? Would digitally controlled analogue merely be a short term interim solution? The author remembers being asked in 1978 by a studio owner due for a new console, "Ought we to wait for digital?" The reply was that he would certainly wear out at least one new conventional desk before needing to face that decision—and so it has proved.

A fully digital-audio mixing system can certainly offer benefits in rationalising the control surface, and this was advanced as a main feature in its favour, but it opens up a very much larger area of unfamiliar territory in the actual signal processing. EMI Studios already had such a system in operation before 1981[2] and development of the Neve DSP console was underway, but cost, complexity and unfamiliarity made such systems alarming to all but the largest and best funded users. But the main advantages gained in the area of control surface compactness and ergonomic layout are equally available in a digitally controlled analogue system without nearly as much added complication.

Bigger is not better

By the beginning of the '80s a fairly strong case was established that for the 'big studio' at least, some quite radical changes were needed in console design. With the demand for more channels, the desk had become just too big. Arm reach got more and more difficult, and heat output, which had dropped to nothing when valves were banished from consoles, was now back at alarming levels. Consequent flow of hot air full of dirt particles meant more frequent cleaning and replacement of expensive switches. Favourite outboard items could no longer be fitted in the desk and control rooms had become progressively larger to accept the 'monsters', while acoustic designers got unhappier at this large obstacle deflecting the sound. Quantity production techniques did something to slow the rise of cost but made maintenance more difficult and mostly made custom options out of the question. This is written in the past tense but very little improvement can be seen.

The DSP did not transform the situation; like Concorde it flew as designed, proving it can be done, but like Concorde no ordinary mortal can contemplate using it and it serves only the most affluent clients. More recently, these users' attention has moved back towards digitally controlled analogue and this technology, in the final form developed by Calrec, then part of AMS, has proved itself well in several years service in TV studios and in perhaps the most critical of all areas, live television outside broadcast. This, of course, is using equipment semi-custom built to exacting user specifications, and hence still extremely expensive. But performance and reliability are no longer in question. Fear of computer crash has proved largely illusory. As in aviation, critical elements can be doubled or tripled if considered really necessary for on-air security.

In the recording studio world, however, progress is very much more limited. By 1985 it appeared that product using the same principles for this market would soon appear from several sources but only two designs were launched. Both were aimed at the top end of the market and they cannot be said to have succeeded commercially. It is notable, however, that both offerings were

constructed in the Big Impressive Console style, with all the electronics in the console itself. So any real advantages over the heavy 'industry standards' available to those who could shop at these top market prices may not have seemed clear or worthwhile; they may have been seen as gimmicky.

Taking stock

Those recording and video-post users who did grasp the potential benefits still longed for proven and affordable hardware. In October 1985, Solid State Logic produced a discussion document *The Future of Audio Console Design*[4] in which the technical options were carefully examined. This may have frightened some competitors with how difficult it all was and SSL reached the surprising conclusion that they would devote their own research to the fully digital solution. This view may have been strengthened by the arrival of many digital audio editing systems, and hard disk audio storage. Simultaneously the US journal *Recording Engineer/Producer*[5] ran a very full survey 'The Virtual Console' of design ideas and how the manufacturing industry was thinking.

It looked then as if the next year or two would bring lots of exciting and competing product and that those who were not already designing might be too late. But if some felt then that digitally controlled analogue audio might have a short market life, even now in 1991 the situation is virtually unchanged; the same arguments apply and the user need has not been met. The great leap forward just has not happened. Some affordable product is just starting to emerge, in a very limited way, but it appears that except for the broadcast network 'battleships', nobody has quite got it right yet[6].

It is valid to question whether users have yet fully appreciated the possibilities of the technique, or made judgements simply based on the initial impression given by products so far presented. So with apologies to already converted enthusiasts, it seems time to expound the principles and some of the options once again.

Objectives

The objectives of console design are what they have always been: to provide the most manageable tool to be used in an instinctive way by creative people working at speed under pressure, so they can do credit to their own talent and that of musicians and actors. They do not include the creation of something that suggests a fictional spaceship, or fills the onlooker with wonder that any one person can comprehend and manipulate it all. Every control function must justify the space it occupies in the general layout. The total control area must be reachable, all displays instantly comprehensible.

The benefits of the assignable technique are:

- smaller and lighter control consoles for the same number of channels
- improved ergonomics of the work surface
- affordable custom options
- greater possibilities of display of useful information
- improved environment for, and serviceability of, active electronics
- memory storage, instant recall and update of all control functions
- possible reduction in number of faders

Control surface

From the days when every major studio or broadcasting organisation planned new consoles according to their own methods of working, to be built either in-house or from detailed drawings, the ideal was to have a fully custom built console. But even when budgets did permit, available technology limited the ideal layout. There was no escaping the necessity for every control to be repeated on a per channel basis. User and designer agonised over whether each control was really necessary. Custom builders created new modules on request but their location was always a compromise. To be competitive without dozens of options, quantity builders put all the features cost would allow in every channel, crowding panels to the limit.

Once every control does not have to be repeated channel by channel, great freedom follows. Organisation of priorities can give its full benefit. When mixing stereo, the ideal is to make and hear changes while still seated on the centre line. So the best layout is that which puts the most often touched controls within easy reach, and the most important visual indications straight ahead. Moving out symmetrically from the centre line are controls less often adjusted, and second priority meters and lights. Furthest out are set-up controls, which are unlikely to be touched or looked at during a take, though bright warning lights at the extremes of vision will still attract attention.

An example panel layout organised in this way is shown in **Fig 1**. This example is based on music recording. Specialists in live television or concert sound may like to consider how they would modify it for their own way of working.

Controls are grouped with respect to the centre line, so that having completed the set-up and input assignment and grouped channels under the VCA faders, everything other than re-adjustment of individual faders can be reached without moving up and down the console. Two identical EQ panels enable the settings of two channels to be under the engineer's hands for manipulation during a take—this could also be arranged to operate identically upon several channels; all others having been set individually and memorised, changing on timecodes if so programmed. On accessing channels, they appear alternately in EQ panels 1 and 2, with numerical indication of the channel they refer to. All other per-channel controls refer to one accessed channel at a time. Aux sends control and display level settings to (say) eight aux mixes from that channel or overall gain of eight aux send groups.

Monitor select & level refers to overall speaker monitoring; Track monitoring to the monitor mix of individual tracks, which may be done with one set of controls, stepping along the channels/tracks with Access buttons (simulating an in-line console) or by a full set of non-assignable monitor faders with mutes and solos, simulating 'separate monitoring'. The dedicated buttons can include selection of what appears on the VDUs, as discussed under 'Displays'. If EQ panel 2 is considered unnecessary, the space could be given to a QWERTY keyboard for writing to displays. This could otherwise slide out from under the control surface when needed.

It is wildly unlikely that there will ever be full agreement on the style and layout of assignable control surfaces but the beauty of the principle is that any panel arrangement can readily be connected to standard data capture cards beneath the panels, which are polled by a main data collection processor. Hence one manufacturer may easily offer control surfaces optimised for music recording, video post-production or concert sound. It is even economically possible once again to produce 'custom specials' at reasonable price uplift, as only the surface controls and software need be changed; the rest remains standard.

As in the personal computer world, special data input devices may be supplied by third party sources for use with machines using standard operating systems; various front ends and software enhancements could be marketed for an established system.

When each fader is no longer related to a module directly above, the pitch of the faders can come down to the minimum comfortable, say 25 or even 20 mm, bringing more channels within easy reach and reducing overall console length. All that has to exist directly above each fader is a small vertical row of buttons; 'Access' or 'Call', which brings all controls of the corresponding channel on to the assignable panels above, Mute, Solo, Automation, Write and Read. There could also be one 'soft' knob, designated at set-up time to be, say, Aux Send 1 or Pan, so that it may be manipulated instantly without touching 'Access' first. But this is the only case where change of function is fairly safe. Anywhere else on the control surface a knob or button which can 'become' something else, so that an indicator must

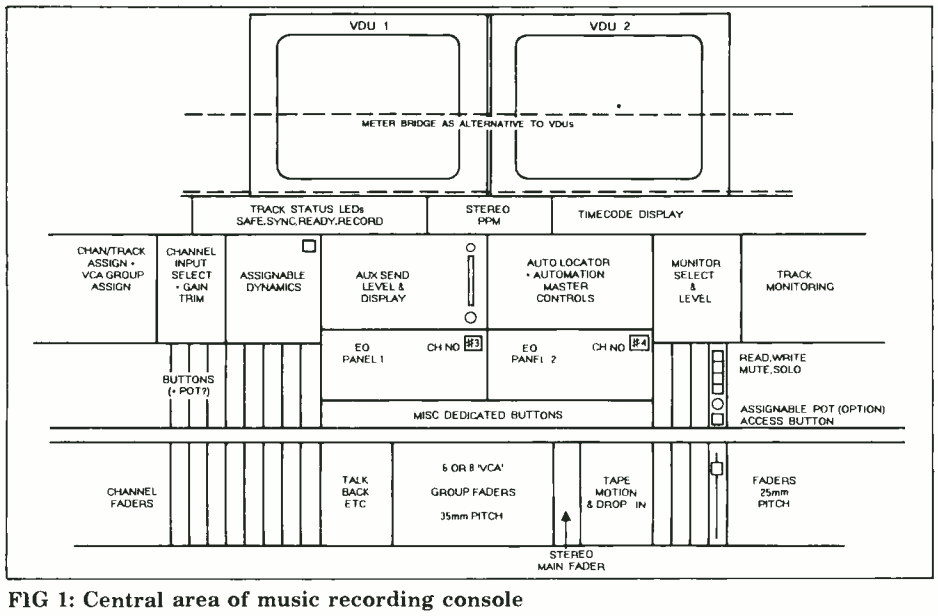

FIG 1: Central area of music recording console

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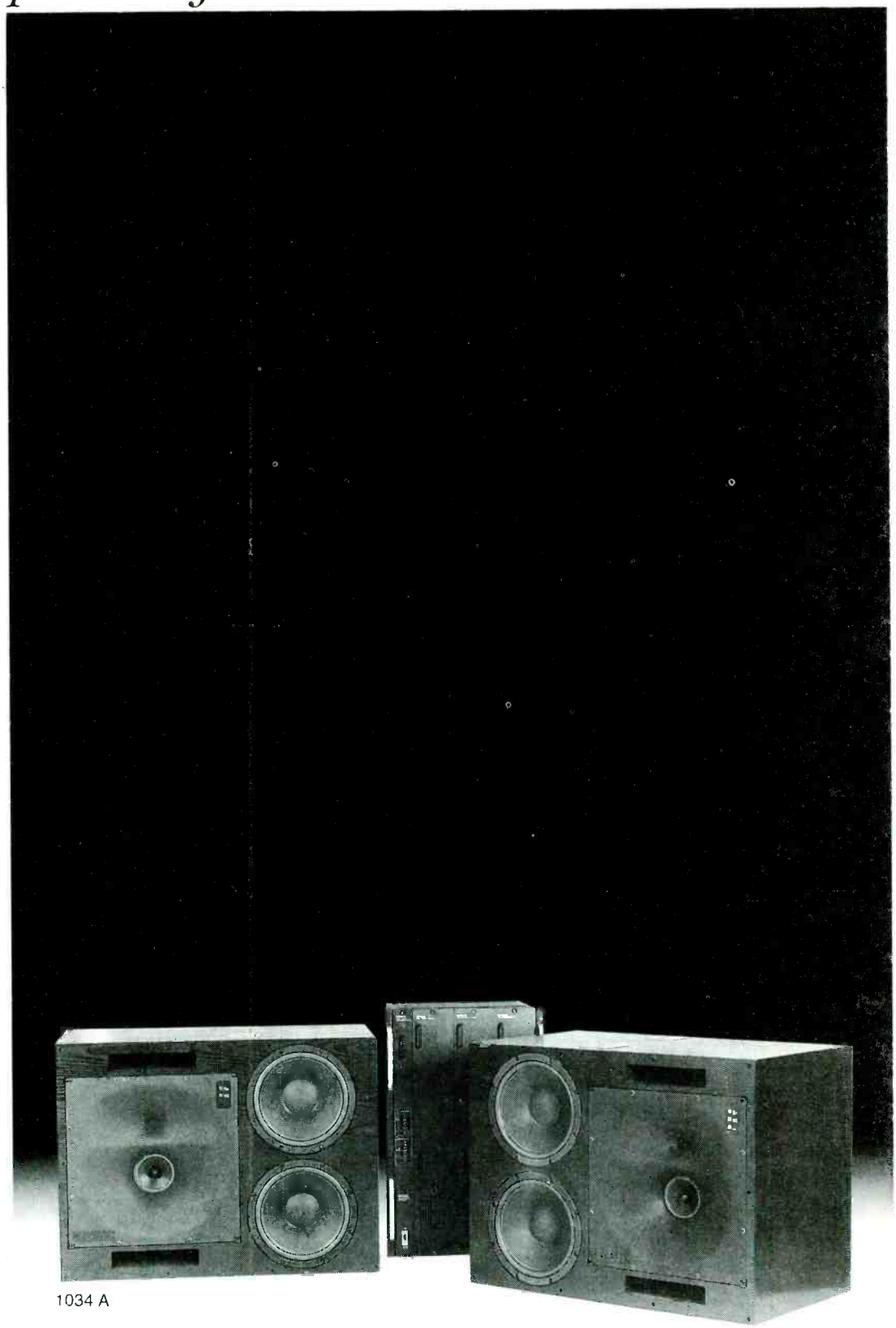

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be looked at to see which it is, is most dangerous and sure to cause mistakes.

Displays

On creating a situation where a control panel addresses more than one channel we need to see immediately the whole status of that channel, and the effect of the changes we make. There is a choice whether to show this around or above the control being moved, on some kind of legend display unit, on a video screen or in more than one place. Personal preferences come into this but the essential is that every indication must be obvious in meaning and easy to read. The height of a vertical LED column meaning 'quantity' is generally accepted but is it the best way to convey frequency, or pan position? For the LEDs to present a 'pointer' round a knob is more instinctive than a column. And how best do we show what a compressor-expander is doing? In general the worst choice is to make the operator read numbers, except for channel numbers and timecodes. As to level metering, horizontal rows of real vu's and ppm's have given place to bargraphs, saving overall width but a horizontal row of anything was always a compromise dictated by number of tracks, and it is a little surprising to see it carried over into assignable designs when alternatives exist.

With current computer practice the video display is capable of representing virtually anything, and looking at many things in rapid succession, combined, or superimposed. As a display of track levels, it has been in use as long as fader automation and it is the centre of all off-line editing systems. One screen may get crowded but within the width of a very small console two screens will fit, with a huge amount of varying information available to be called up by channel access or dedicated buttons according to what is important at that moment. For example, the left screen may carry time of day, timecode and machine status at the top, levels of channels readied for recording (rather than all 32) in the middle, mute flags and message boxes at the bottom. The right screen may show settings of channel accessed: routing confirmation, EQ shown as response curve or knob pictorial (and even numbers), dynamics settings, effects send levels, pan, etc. On top of this one may call up take sheets, edit decision lists, table of all track assignments or 'Desk Help'. During a playback, one could display all track levels or automated fader levels across both screens. Or for a final mix, incorporate the currently available display that shows four 'real' BBC ppm's on screen. Major warnings may be flashed on top of the display, such as 'one minute tape left'. The possibilities are wide open for exploration.

Mix data recall

Writing and recalling settings for all knobs and switches is the same process as for faders and mutes; there is simply a great deal more of it to be scanned and updated in the same unit of time. Of course, we do not necessarily have to scan everything all the time; sections of the control surface may have write/read switches so that what will not be changed need not be rewritten. With the techniques now available we can reconsider the old question, 'How much automation is useful?' Much play has been made with the term 'total dynamic automation'. This would logically imply that if each and every variable control and switch in every channel is being moved simultaneously, by an unlimited number of human hands, the automation will be able to log all this and reproduce it in the Read mode without detectable delay. It may seem a desirable goal and it is feasible, but to be totally dynamic requires a speed and data capacity that pushes the processing hardware to its limits and hence increases cost to handle it reliably. It may be that users who have asked for this have not fully thought through its use. Given a large conventional control surface where any control at all might be moved at any moment, total automation might be thought desirable. One may ask, however, 'Having found the right starting setting and put this in memory, how often will any control other than a fader ever be moved in a gradual manner during a take? And if it is, will such moves relate to more than one or a pair of channels?' Most unlikely, and with the fader moves taken care of by automation it is little hardship to perform such changes manually at each mix attempt. Is it worth a level of complexity and cost just to avoid this? In any case, in assignable working, where control panels address one or two channels at a time, the only way to programme simultaneous gradual changes, if actually required, would be to build up the moves in a channel by channel manner on successive passes of the tape—an extremely time consuming process better accomplished on an off-line editing system.

All systems so far have provided a 'snapshot' change facility where, at a programmed instant read from timecode, any number of controls may jump to new settings. With video, this will happen in relation to picture cuts but by a facility called crossfade, which is easier to provide than dynamic update, the change may be stretched out from instant to gradual over some seconds. So through the length of a musical item, for example, one could move controls and mark a 'snap' every so many seconds and then have the system crossfade the change over the same time period. Exactly as for fader automation, the amount of data (set-ups and mix attempts) held for instant access in the console is simply related to the amount of RAM provided.

Mixes may be written to hard disk either as a periodic dump or when instructed, and provision to make floppy disks will usually be provided.

Audio electronics

Another major benefit of 'remote audio' is the chance to optimise electronic circuit layout, construction and serviceability. This ideal becomes much more achievable upon losing the restrictions of leading the audio to and from panel controls. A vast saving of long wiring runs and use of screened cable also follows. The 'console plus rack' arrangement is totally familiar in television studio practice. The audio circuitry can be on well ventilated but also well screened PC boards whose layout can be totally logical for the shortest signal paths. The size of these boards and their connectors means that standard electronics industry rack and packaging systems can be used, combining well established reliability standards with cost saving over special metalwork. Board layout will minimise any possible digital interference with audio, using separate connectors for control data input.

There are obviously choices in physical arrangement between a 'one channel equals one large PC board' layout, and the breaking down of a channel into functional blocks. Within reason, it does not matter a great deal how much rack space is required per channel. There may be a case for an active equaliser or dynamics control card to be piggy-backed on a channel routing card, with the address, etc, decoding chips on the main card. This would make possible a choice of EQ cards, or their omission where not required.

There is a strong case for removing microphone amplifiers to the studio, in well shielded units, still with control and indication extended from the console. Mic signals then enter the racks at full line level and all channel inputs are equal and uncommitted. In many cases the number of channels used for microphone may be minimal.

It is easy to arrange shielding, air filtering and heat extraction for racks, which may (and desirably should be) in a separate cubicle. The maintenance advantage is obvious; faulty cards may be exchanged at the rack during operations without alarming production personnel.

With suitable patching a doubtful element may be bypassed via a spare channel, then brought out on a extender board or plugged into a test position. Tape recorders will be fed on multipin connectors from the rack and further multicore cable will run to patchbays for insertion of outboard effects devices. Units capable of being remotely controlled may reside at the rack position; those requiring manual control will call for audio pairs extended back to rack trolleys near the console.

Sending the data

The various possible ways to send the data are fairly standard computer communications technology, not nearly as sophisticated as digital audio, so there is little to be secretive about. In some early designs control information was serialised down very high speed fibre optic links on the basis that it might have to run hundreds of metres. This is very rarely a requirement and it has also become evident that some audio will usually need to be returned to the control position for outboard equipment and monitoring, so the number of cable ways is not a significant factor. It is also a matter of convenience whether transmission is bidirectional, or whether return of displays and indications is by a separate path. Duplicated cables may be necessary for security on location but not in a permanent installation.

A major design choice is whether the data collection processor in the console is a relatively 'dumb' device, which merely transmits an encoding of knob movements and switch closures ignorant of what they 'mean'. If so, the interpretation of these into commands to be sent to the audio cards, and stored in RAM for recall under time control, is then done by a 'serious computer' at the rack location, where megabytes of hard disk may also reside. Alternatively, the serious work may be done by a powerful processor under the console, with relatively simple transmit-receive circuitry at the rack. Obviously this choice determines the nature of what is sent down the link.

No one transmission method seems outstandingly preferable but the chosen method should be applicable to systems of any size, and in outside location use, to a system of varying size, ie one to four audio racks. Therefore, the command structure should be able to control not less than 128 audio channels, plus groups, masters and auxiliaries, each with the number of variables and switches found in a really comprehensive present day music console, plus the return of all necessary status and display

LISTEN

data. It must not 'run out of' space and speed but must remain valid and usable for relatively simple systems down to 24 channels or even less. Fader movements and mutes should be transmitted with a response time of ¼-frame and one frame for everything else.

The idea of 'plug-in' front ends implies a need to standardise the interface, the physical connections and the code by which instructions pass from console to rack. Unless the SMPTE or AES are unusually quick to define and declare standards, there is the danger of a proliferation of proprietary methods, from which in time a *de facto* standard will no doubt be established by commercial clout and units in the field. Some makers may say, "We don't want *their* box to plug into *our* system," but the computer industry has demonstrated that sooner or later it will, so perhaps a lot of wasted energy may be saved by sensible dialogue. **Fig 2** shows a generic arrangement of data collection and transfer.

One step further?

So far it has been generally held that there still has to be a fader dedicated to every input, that whatever else is assignable, the fader holds as it were the identity of the source, and can be moved instantly without touching anything else first. (Opponents of assignable mixing argue one might want to touch any control instantly—but to identify correctly by eye and grab the right knob in the conventional forest can hardly be quicker than to touch an Access button. It is possible too, at arm's length, to reach out to the wrong fader!)

The more that planned track-laying and mixdown, aided by automation and moving faders, have become standard, the less weight the '96 physical faders' argument holds. With VCA grouping, free combinations of channels are brought under control of six or eight centrally located faders and these combinations can be changed as a mix proceeds. Any individual fader interventions will most likely be made as a result of hearing the mix in the Read mode. So it seems reasonable to Call channels to be manipulated in one pass of the tape into a limited number of faders. That number is wildly unlikely to exceed 12! Electronic labels are provided above the faders in some designs and these can change at pre-arranged time points, 'letting go' of one channel and accessing another, with unchanged levels till the fader is moved. Or much more simply, a limited run of faders may have say three illuminated buttons in the writing strip. Three source names are written on or beside them; the lighted one is the one accessed. On touching another, control is taken of another signal. Engineers may not yet be ready for this approach but the means are inherent in the assignable technique.

How virtual can you get?

Since the above mentioned article[4] was published the term 'virtual console' has been loosely applied to any kind of system that separates control functions from audio circuitry. More correctly perhaps it should refer to a system that presents an image of something that does not exist. This is the case in the digital audio console, in which operations are being performed upon a model of audio signals inside a computer.

But another 'virtual' approach has appeared more recently, in which the real control surface has disappeared altogether, to be replaced by a picture of a console on a screen, which is manipulated by a mouse or trackball. Faders are moved by pointing an arrow, clicking on the one to be moved and moving the arrow on the picture. By similar moves faders may be grouped together. By clicking on an 'access button' an equaliser panel picture appears, and the 'knobs' can be pulled round with the pointer. This is most ingenious but has the snag that one has to place the pointer quite accurately on a small image, click and move the right amount, and more seriously that only one fader or group can be touched at once. Like many ideas that derive from a computer approach, it forces the operator to do everything serially, in a 'select, then execute' manner. This must inevitably slow down operation and will probably rule it out for any kind of realtime use.

VDU edit stations are proposed on which, in principle, one may prepare off-line set-ups and control sequences to be run later on the real console system. But unlike video, where still frames are meaningful, one can hardly do this without listening to the effect of what one is doing, therefore, access to the real audio system, or at least some channels of it, is still needed, so it is hard to see that efficiency is really improved.

Personal computer-based systems with mouse and keyboard as the controlling elements are admirable for off-line use when unlimited time is available, typically for a musician to manipulate his material in a step-by-step, modify and listen manner, with some possibilities that would otherwise call for very ambitious hardware but not being usable in realtime these are essentially editing rather than mixing systems.□

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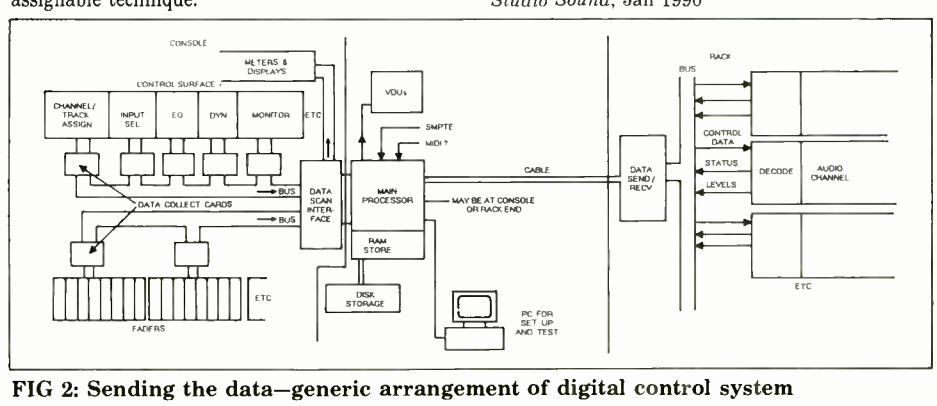
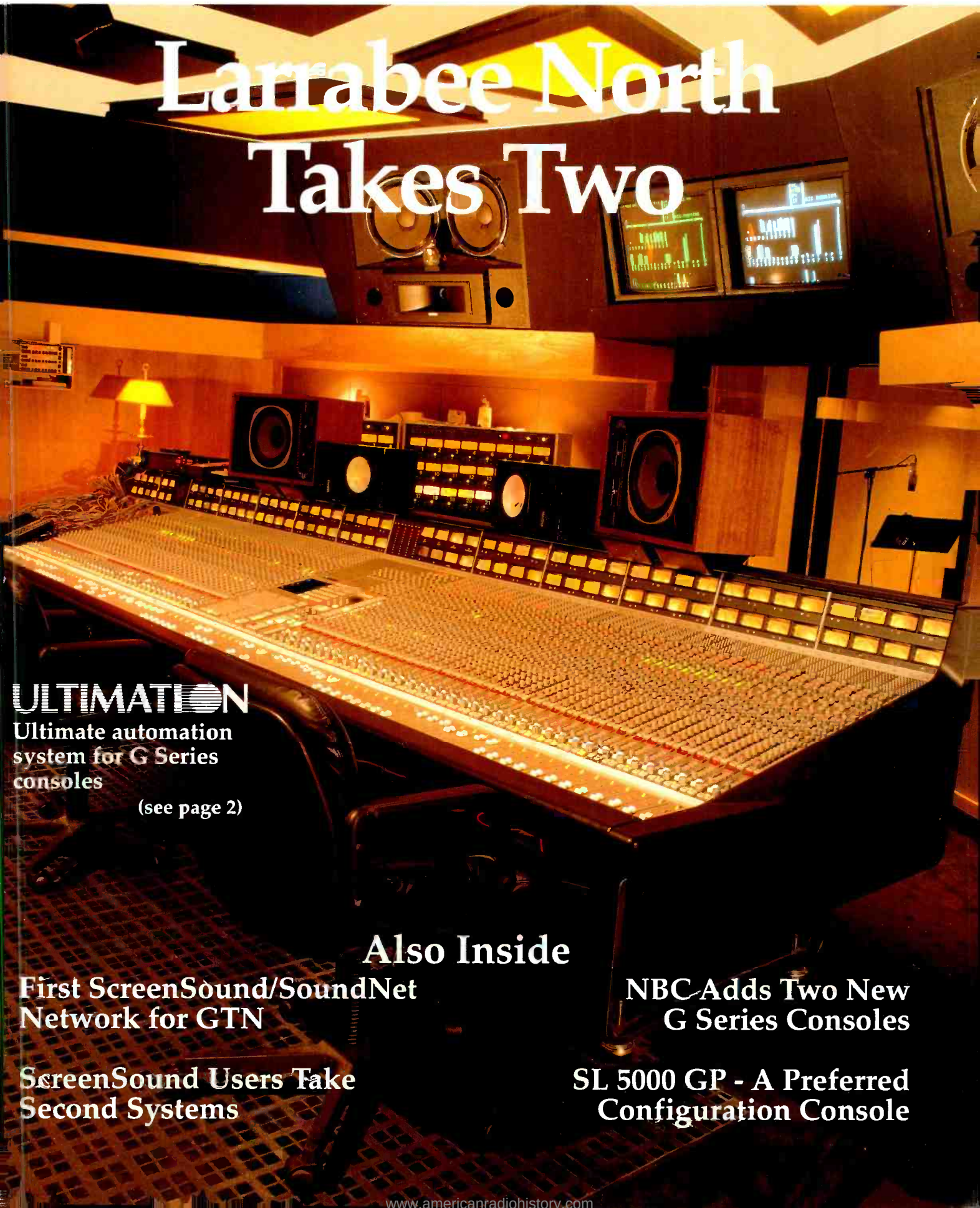


FIG 2: Sending the data—generic arrangement of digital control system

Spring '91

Solid State Logic



Larrabee North Takes Two

ULTIMATION
Ultimate automation system for G Series consoles

(see page 2)

Also Inside

Ultimate Automation System for G Series Consoles

There have always been two schools of thought amongst audio engineers about the type of automation system that is most desirable.

Moving faders have the advantage of giving the engineer a constant visual reference to the level of a particular channel. For subtlety in mixing though, most engineers agree that VCAs allow far greater control in making the minute adjustments which effect the character of a mix.

Now SSL has introduced a new automation system with three modes of operation. Called 'Ultimation' (for Ultimate Automation) the system can work as either a dedicated VCA system, a dedicated moving fader system, or in a way that combines the best features of both systems. The engineer is free to decide.

These advances are made possible by SSL's unique dual signal path circuitry. Ultimation uses a VCA element as well as an analogue path through the fader wiper.

Under processor control, the most appropriate audio path is selected according to the operational needs of the engineer. When writing or playing absolute fader levels, the audio passes through the wiper of a servo-controlled fader. On touching the fader, the audio is automatically routed through the VCA element to allow a standard SSL real-time Trim update.

Ultimation is an extension of SSL's G Series automation, able to read G Series mix data. It is available as a standard option on all new G Series consoles, and as a retrofit to any console using the G Series computer.

"For some time SSL has fitted moving faders to its post-production range of consoles, where the positional feedback of moving faders is better suited to operational requirements," says SSL Product Director, Chris Jenkins. "Our music clients, on the other hand, have been very vocal in their belief that music mixing sets different priorities for an automation system than those required for audio post-production."

"Now technological developments have enabled SSL to provide a system to suit all needs," Jenkins says. "We are delighted to offer studio owners what we believe is the ultimate automation system."

ULTIMATION

Main features include:

Moving Fader or VCA operation

Dual Signal Path technology combines benefits of both systems

Movement can be turned on or off

Automatic VCA take-over allows moving faders to trim

Works with existing G Series Computer

Uses existing G Series mix data

Simple thumbwheel grouping

Expanded software functions

Standard option on new G Series consoles

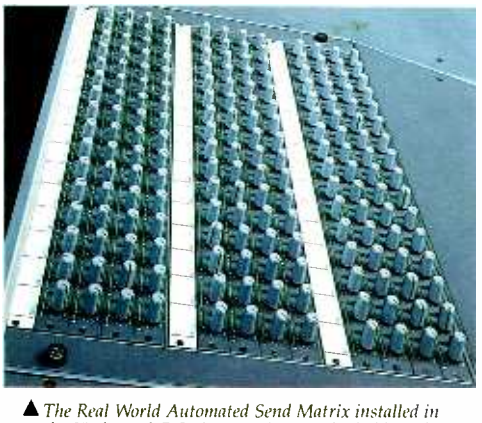
Fully retrofittable to any console using G Series Computer

Options Provide Mixers With Definitive System

The ability to offer a specialised service is increasingly important to recording studios. In an effort to help meet these changing demands, Solid State Logic is offering a wide range of options on any new G Series console. In some cases, these modifications are also available as a retrofit.

One recent addition to these options is the Real World Automated Send Matrix, a system designed by technicians at Peter Gabriel's Real World Studios. The unit, which can be built into the console, provides an additional four effects sends per channel, bringing the total available to eight mono and one stereo send.

An automation computer is part of the system, enabling each send on/off switch to be automated to timecode.



▲ *The Real World Automated Send Matrix installed in the 80 channel G Series console at Real World Studios, Box, England*

Current G Series options include:

Variety of frame sizes

45 degree or 90 degree console 'wings'

Mono/Stereo channels

Ultimation

Metering (VU or Plasma Bargraph)

Left, right or remote patchbay

Bell EQ or G Series EQ

Computer Subsystems:
Total Recall™; Master Transport Selector; Synchroniser Controller; Events Controller; Programmable Equaliser

Real World Automated Send Matrix
(4 additional effects sends per channel)

Larrabee North Opens For Business With Two SSLs

Keith Cohen is one of the leading mixers/producers who inaugurated Los Angeles' new Larrabee North recording studio earlier this year.

"Larrabee North is spectacular," says Cohen, who was mixing tracks for the new solo album from Susannah Hoffs (former lead singer of The Bangles). "With an incredible 80 input SSL, it's simply the definition of state-of-the-art. Because of the way I work, the SSLs, with their unmatched automation capabilities, are the only consoles that will do."

The new facility has recently completed a $2.5 million upgrade, featuring the addition of two of the USA's largest SSL consoles. The two consoles (one 80 input and one 72 input) both feature the Real World Automated Send Matrix. In addition, both consoles are equipped with G Series EQ modules as well as the optional SSL Bell EQ — answering a growing industry trend to pick and choose from both 'styles' of SSL sound treatment.

"My idea was to provide my clients with two additional mixing rooms, in what for many of them will be a more convenient location," says studio owner Kevin Mills.

"We've been in business for over two decades, and we've learned what people who almost live in studios really need. We've planned everything from the ground up — from the electrical and audio wiring to the acoustics and the outboard gear."

The Plant Adds Second G Series Console

The reason why artists like Michael Bolton, Mariah Carey, and Carlos Santana make their music at The Plant, in picturesque Sausalito, California, has a lot to do with the facility's approach to music making - that a studio should cater to its clients' human needs as well as technological needs.

The Plant's commitment to their clients' technological needs is apparent in the studio's addition of a second SL 4000 G Series console.

"It's my clients that wanted the SSL console. We get calls from artists asking us if we have an SSL," says Bob Skye, co-owner of the The Plant with Arne Frager. "We are flabbergasted by the consoles, they are the hub of all the operations here, as a console should be."

▲ *Michael Bolton 'experimenting' with the SL 4000 G Series console at The Plant, Sausalito*

The success of such advanced technology, combined with a comfortable atmosphere, is evident in the consistent stream of artists that have recorded number one records at The Plant. These have included Stevie Wonder, Huey Lewis, Aretha Franklin, Fleetwood Mac, and this year's Grammy nominee for Album of the Year, Mariah Carey. Recently, Michael Bolton has been working on his upcoming album with producer Walter Afanasieff, using the SSL consoles in both Studios A and B.

Commenting on The Plant's choice of SSL, Arne Frager says, "We wanted a console that would be in keeping with the philosophy of The Plant Studios, which is sonic excellence, ease of use and minimum maintenance requirements."

Saylor On Track With New G Series Console

Skip Saylor Recording has installed an 80 channel SL 4000 G Series console as the centrepiece of its innovative new mixing suite.

Since the installation, leading mixers and engineers have been quick to take advantage of the powerful capabilities of the new console in Studio A. Brian Malouf has been working on a Gardner Cole album for Warner Bros., and mixer and engineer John Potoker has been working with CBS's Martika, and the UK group, The Blessing, signed to MCA.

"Producers and engineers are ecstatic about our new G Series," adds Saylor. "We've created the ultimate mix experience with this console and its capabilities. This console will set a new standard in the field of mixing."

The deluxe mixing room, which opened in October, features an 80 channel SL 4000 G Series console, with G Series computer, Total Recall™, SSL Bell and G Series equalisers, and the Real World Cue Matrix Systems.

"I just had to have the best, and the SSL console is the best," says owner Skip Saylor. "The Real World automated cue sends make echo and effect capabilities almost double what they were before. You can do so much on each individual channel, that this 80 channel console runs as the equivalent of 100 channels."

▲ *Brian Malouf at the 80 channel SL 4000 G Series console in Skip Saylor Recording*

Dierks Celebrates Two Decades with a New G Series

Dierks Studios, one of Germany's foremost recording centres, is installing a second SL 4000 console at its Pulheim facility near Cologne.

Founded twenty years ago by self-trained engineer/producer Dieter Dierks, the studio has built up an impressive client list, including artists such as Sting, Prince, U2 and Stevie Wonder. Dierks already operates a 48 channel SL 4000, and the new G Series console, fitted with 64 channels, will be based in a studio currently undergoing refurbishment by Neil Grant of Harris Grant Associates.

Although Dierks is best known for its music recording and mixing facilities, it is also involved in audio-for-video post-production. The studio has both a four camera video mobile and an audio truck which are called in to handle major Outside Broadcast projects for TV. A recent major event was the mixing for TV transmission of The Rolling Stones' tour concerts.

Harry Braun, Studio Manager, comments: "The concerts performed during The Rolling Stones' *Urban Jungle* tour were recorded using our audio mobile. The concert held in Barcelona was mixed by Cedric Beatty and Benji Lefevre using our SL 4000 in Studio Three, and was then broadcast on Tele Cinqo in Spain and France."

"A further mix was produced for SkyTV, and once our new console is installed, we expect to be working on yet more mixes for the video, film and live album of the tour.

"As a studio with an international clientele, we place a lot of value on the fact that audio engineers don't have to get used to working on the new desk; SSL is the worldwide standard and every audio engineer is already familiar with the console. "

"The SSL's responsiveness and recall ability are points to recommend it, and the well-designed ergonomics and handling guarantee stress-free working. Naturally, the integration of the computer is important, and the flexibility of the EQ system is especially impressive."

ScreenSound Brings Music To Lynch's *Twin Peaks*

As David Lynch's *Twin Peaks* hit television series runs through its second season, music editor Lori Eschler has been doing her job better - and more easily- than ever before. She thanks ScreenSound, on which she performs all the show's music edits, for helping her meet this season's hectic production deadlines with ease.

"For the first season of *Twin Peaks*, I edited on 35mm full coat mag. However, since most of the show is tracked from a library of cues, I was constantly having to inventory and re-order transfers so that I wouldn't have to wait if I suddenly needed a cue," says Eschler. "When I saw the schedule for the 1990/91 season, I knew that I had to find a more efficient system for editing and accessing the music."

"ScreenSound is designed so that the physical motions of marking, cutting, splicing, shifting, cross-fading, etc. become almost subconscious, so that I now spend less energy on the manipulation process and more on aesthetic concerns," adds Eschler.

"ScreenSound lets me quickly create combinations of material while I'm working with pictures so I can alter the placement of a given piece to achieve the effect that I want. Of course, this can all be done on film or tape, but it's so much faster on the ScreenSound system that my imagination is the only thing that slows me down."

"I am continually exploring new techniques. We have recently begun mixing directly from ScreenSound and I have an additional system on the stage so that I am able to make changes in a matter of seconds that could sometimes take fifteen minutes or more when we were mixing from analogue tape or film," she says. "As this technology continues to evolve and become more accepted, the process of music and sound editing can become less encumbered with physical concerns, leaving more energy for creative endeavours."

▲ *Lori Eschler, Music Editor of* Twin Peaks, *at the ScreenSound system*

Audio Post Houses Double ScreenSound Capacity

As an indication of growing marketplace acceptance of Solid State Logic's ScreenSound, several leading post production houses have recently added second digital audio-for-video editor/mixers to handle increased customer demand.

Soundtrack Recording of New York, Image Express in San Francisco and Vancouver Studios of Canada are the first facilities in the world to feature two ScreenSound systems.

Soundtrack Recording Studios, the first audio post-production facility on the east coast to install ScreenSound, has purchased its second system to help meet its increasing post-production demands.

"We had already decided to add another hard disk audio system," says studio owner Rob Cavicchio. "The fact that ScreenSound was so well received by our clients and engineers made it an easy decision which way to go."

According to Cavicchio, New York producers and engineers are becoming more aware of the ScreenSound system and its powerful digital editing and machine control capabilities.

Chief engineer Bob Giammarco will work with the new ScreenSound system. He was able to train Soundtrack's staff engineers, Bill Higley and Pete Rincon, to use the system in a matter of hours. "It's by far the easiest system to use," explains Cavicchio. "Engineers who are using ScreenSound are more productive because they are very comfortable with the interface."

Detroit's Image Express also recently reaffirmed its commitment to advanced audio-for-video post-production by purchasing a second ScreenSound for use in its new San Francisco facility, which is scheduled to open later this winter. Like its Detroit counterpart, the new ScreenSound system will be used to complement the facility's EditDroid digital video editor.

"ScreenSound will be the charter piece of equipment in the new facility," says Image Express co-owner, Lee Lipner. "We are so enthusiastic about the device that we would never have considered opening the new facility without one."

Meanwhile, Canada's Vancouver Studios, which recorded last year's Grammy Award winning album *Torch & Twang* by k.d. Lang, is also diversifying into the audio-for-video post-production marketplace with two ScreenSound systems.

"ScreenSound proved itself on its speed and flexibility," says vice president of operations Bruce Levens. "The fact that the very top film and television studios are moving toward the system was another factor. So was the strong support we have received from SSL over the years."

▲ *Pete Rincon, Bob Giammarco and Bill Higley working on ScreenSound at Soundtrack Recording Studios, New York*

First ScreenSound Network Using SoundNet Goes To GTN in Detroit

Just three months after its debut, the world's first ScreenSound/SoundNet Network has been sold to General Television Network (GTN) in Detroit.

This integrated and advanced digital audio-for-video network is part of an overall facility upgrade that will make GTN one of the leading production centres in the Midwest.

The SoundNet sale is part of a large audio post-production package that also includes two ScreenSound digital audio-for-video editor/mixers, and a 40 channel SL 6000 G Series console specially modified for surround sound.

"We're building a new audio wing as part of a general effort to make GTN a complete, state-of-the-art facility. We've upgraded our film transfer technology, and continue to upgrade our graphics and editorial capabilities. ScreenSound and SoundNet were the logical way to take our audio into the digital domain," says audio supervisor Jay Scott.

"The flexibility of being able to pass the work around our various audio post rooms with SoundNet for music and mix is extremely powerful from a creative and work efficiency standpoint," says Scott. "Back-up and restoration procedures are an off-line function, which also saves our clients time and money."

Designed to work in conjunction with the company's ScreenSound digital audio-for-video editor, SoundNet allows up to seven operators to share and copy work without compromising system power and performance. SoundNet also provides central mass storage of audio and a database of all sound clips.

Howard Schwartz Puts ScreenSound to Work

Howard Schwartz has built his studio in New York on a very simple premise: Give the client the best talent and tools and make audio-for-video a comfortable and creative process.

The latest example of Howard Schwartz Recording's commitment is the new digital post-production suite, featuring ScreenSound and the most advanced assortment of digital audio and video recorders and outboard gear. ScreenSound is the first in an anticipated network of systems Schwartz will be installing, and ScreenSound is already doing extremely well. Apart from numerous television and radio commercials, including several for MTV, the studio has also used ScreenSound for dialogue replacement on the new Avenue Pictures/BBC release *Object of Beauty*, starring Andy McDowell and John Malkovitch, as well as on a recent episode of *Saturday Night Live* (see Page 7).

▲ *Howard Schwartz, owner of the prestigious Howard Schwartz Recording in New York, at the studio's ScreenSound system*

"The client reaction has been excellent," says Howard Schwartz. "Since many of our clients come to us from video post houses, they are extremely comfortable with the way ScreenSound looks and works. They are already familiar with the tablet and pen interface."

"Most hard disk systems come from the music end of the business. ScreenSound's advantage is that it was developed as a post-production tool from the very start."

Ease of use was a significant attraction for Schwartz who reveals that, within ten days of the system being installed, operator Roy Latham was already using it for all his sessions. Latham adds that clients positively respond to ScreenSound, "As soon as I give them a pen and let them play for themselves."

Schwartz has noted that clients especially enjoy going through the process with the operator. "There's so much more flexibility and room to be creative. They love seeing the operator physically move the audio by two frames; that was always one of the great promises of hard disk recording, and ScreenSound delivers."

ScreenSound for Holland's Leading Studio

Artisound, based in Amsterdam and one of Holland's leading post-production studios for commercials, has ordered a ScreenSound system which is to be installed in a specially rebuilt studio.

ScreenSound will be used for post-production on radio and TV commercials and for the production of A/V programmes.

Eli van Thijn, owner of Artisound says, "We looked at all the systems on the market, but ScreenSound was the most operator-friendly system. Also the client can see what's going on, and that is very important in this industry because clients like to be able to follow progress. With some other systems all they can see is an engineer fiddling around with all kinds of keyboards that they don't understand."

"Now, with ScreenSound, Artisound is ready to undertake the most demanding projects in digital post-production."

Artisound already has a 48 channel SL 4000 console in Studio 1, where commercials and songs written by Eli's songwriter/producer son, Eric, are mixed.

SL 5000 M Series Consoles Dub For Film & TV

Soundtrack quality is increasingly important to the success of films, video and television programmes.

A large number of recent motion picture releases and hit television series have recently been dubbed on the popular SL 5000 M Series consoles which are a feature of the world's leading film studios.

According to Don Bassman, sound director of 20th Century Fox's Zanuck Theatre, recent motion picture projects have included *The Hunt for Red October*, *Die Hard II*, *Ford Fairlane*, and *Predator II*. The dubbing theatre features one of the largest SL 5000 M Series consoles ever assembled, with a custom-designed, four-man, 80 input configuration with moving faders and G Series automation with Total Recall™.

"I decided to purchase the SSL console because I wanted the best. I had seen work on SSL consoles in the past and I knew what they provided was better than anything else in the marketplace — it was the only direction in which we could go."

Lucasfilm's Sprocket Systems, site of four SL 5000 M Series consoles, was recently responsible for the soundtracks of *Godfather III*, *Five Heartbeats*, *Avalon*, and *Henry and June*, according to General Manager Tom Kobayashi.

Meanwhile, Disney/Buena Vista has been using its 68 channel SL 5000 M Series console for work on numerous Stephen Cannell productions, including *Hunter*, *Wise Guy* and *21 Jump Street*. The console was also used to produce temp dubs for soon-to-be-released Disney features *White Fang* and *Run*.

At the Hitchcock Theatre at Universal Studios, *Ghost Dad* with Bill Cosby and *Bird On A Wire* with Mel Gibson and Goldie Hawn were both dubbed on the SL 5000 M Series console.

▲ *The four-operator SL 5000 M Series console at 20th Century Fox*

Forthcoming releases include *Delirious* with John Candy and *The Hard Way* with Michael J. Fox and James Woods. "The console sounds great and it has been extremely reliable," says Universal's chief engineer, Doc Goldstein.

NBC Adds Two SSL Consoles For On-Air Audio

▲ *Michael Delugg, audio consultant on NBC's* Late Night with David Letterman, *at one of the organisation's 22 SSL consoles*

NBC, in New York, has recently added SL 6000 G Series and SL 4000 G Series consoles to its family of SSLs. The broadcaster now operates a total of 22 SSL consoles in its New York and Burbank studios, and 2 outside broadcast units.

In the 6A control room, NBC recently installed an SL 4000 G Series console to mix live-to-air the music of Paul Shaffer and The World's Most Dangerous Band, as well as the many musical guests that are broadcast on *Late Night With David Letterman*.

In Studio 8G the SL 6000 G Series console was recently installed to mix live-to-air audio for the *Donahue Show* as well as special sports broadcasts.

"We purchased the new SSLs not only because they are familiar to us and lend themselves well to maintenance and training, but simply because the console's operating functions are fantastic," says NBC Technical Engineering and Maintenance engineer Jim Starzynski. "We consider the Solid State Logic console to be one of the few that can offer network-calibre performance and reliability."

"SSL also facilitates itself nicely to surround sound. In 6A, using the console's quad monitoring bus allowed us to patch in surround decoders and have it work just fine without any extra modification," he adds.

The *Letterman* show is broadcast in surround sound and is a high-profile example of the priority that NBC has placed on broadcast sound quality.

"Getting music to sound good out of a television set is a real challenge, and getting a live band to sound good on television is a double challenge," says *Late Night's* audio consultant, Michael Delugg. "The dynamics processing and the strength of SSL's EQ have helped me avoid a lot of problems that are inherent in a live music broadcast situation."

Delugg, whose engineering and production credits include work with such artists as Barry Manilow, Dionne Warwick and Paul Anka, says that engineering music for broadcast on television also means taking into account that many viewers will be listening to the music in a variety of formats, including mono, stereo, and surround sound. "It's a challenge to get the mix to sound good in mono and stereo," says Delugg. "And since *Late Night* is also broadcast in surround sound, I have to consider the music with regard to all three."

SL 5000 GP, A Preferred Configuration Console

The past few years have seen an increasing number of broadcasters choose and install SL 5000 M Series custom consoles for a range of challenging and critical tasks.

In each case the SL 5000 M Series has been chosen for its combination of power and ease of use in a professional, fail-safe package. The advanced modular technology used in the SL 5000 provides great flexibility, while maintaining smooth instinctive operation under programme pressure.

The success of these consoles and the acclaim of their operators has not gone unnoticed.

Clear patterns of console specification for practical broadcast production applications have emerged, and in response SSL is now offering a fully-specified version of the custom SL 5000 M Series, the SL 5000 GP. This Preferred Configuration provides, through production economies, the power of the SL 5000 M architecture with a faster delivery time and with significant financial savings.

The SL 5000 GP is particularly configured for television and radio production and on-air presentation, where routing flexibility and multiple output capability are essential for programme flow. It offers 24 mono and 8 stereo channels with full equalisation and routing to the main programme bus, plus an additional 12 stereo balanced mix busses that are used for subgroups and Independent Main Outputs. Options include SSL's Instant Reset™ and Total Recall™ computer systems, together with an extended auxiliary system and dynamic fader automation.

▲ *The SL 5000 GP, Preferred Configuration console*

Although the frame configuration is fixed, considerable flexibility still exists to vary the proportions of mono and stereo channels, or to reduce the number of channels leaving prewired positions that may be installed at a later date.

A full documentation package is available from SSL offices that describes the structure, configuration and performance of the SL 5000 GP in detail.

ScreenSound In Review by John Alberts

I recently used the SSL ScreenSound in a demo situation at Howard Schwartz Recording for one week. My goal with the demo was to do some relatively simple jobs with the unit. I did the assembly, sweetening, mixing and re-lay to D-2 for two half-hour specials of *Saturday Night Live* (European syndication), as well as the assembly and placement of sound effects for NBC's *How To Be Famous* (Spy Magazine).

I was extremely impressed with most aspects of the device. The human interface is designed in a way enabling one to really start working after only a two or three hour training period. The intuitive prompts guided me through the more complicated functions, and there is a kind of real-time feel about things which enabled me to work in a 'stream of consciousness' manner. The Store (hard disk storage for sound clips) is available at all times to preview any and all sound files on the various disk drives. The speed at which you can access this is amazing.

Using the companion TEAC recordable video disc as my video source, I noticed a hard-lock to video that I had never before experienced. Placement of sounds was effortless and always exactly on the desired frame. Also, while using the automation, I was able to do static fader moves on the exact frame of the scene change.

The pen and tablet system, in my opinion, has its limitations, but to me is preferable to an intensive keystroke situation. The automation works nicely, but of course is limited to one fader or one group of faders at a time. You can go back and write other faders, however. This automation does provide some interesting possibilities for panning (i.e., following the action with the pen across a stereo effect on two grouped faders).

Finally, I found the computer very fast, with no waiting time between windows. Also, there was never any loss of data (as I have found with other systems). Machine control is another plus, with ScreenSound's on-board synchronisaton system. I used it with a Sony DVR-10 and it was flawless.

I completed the second *Saturday Night Live* show in less time than I would have by conventional means. For fairly simple mixes, and for preparation of elements for bigger mixes (transferring to multitrack), I feel there is a real place for this device.

As SSL further develops both software and hardware (there is plenty of room for growth in the system), it will become even more attractive.

▲ *John Alberts is a Monitor Award winning audio-for-video post production engineer based in New York who regularly works at Howard Schwartz Recording. Last year he handled post-production for live musical performances of The Who, The Rolling Stones and Paul Simon*

Surveys Show SSL is Market Leader in Japan

Surveys published recently in Japan show SSL to be the market leader in audio recording and mixing consoles.

Two independent magazine surveys reveal that SSL consoles have captured 48% of the music recording and 24% of the audio post market in Japan.

One of the latest post-production facilities to choose SSL is Totsu A/V in Osaka, which has recently installed an SL 4056 G Series with Total Recall™ and an SL 4032. Takarazuka, Sound Atelier and Meijho ABA, the three main recording studios in Osaka, have also installed SSL consoles.

The dramatic increase in Tokyo land prices has led to many new recording studios being established away from the city. The newest of these so-called 'resort studios' is Moody Moon, an SL 4072 G Series equipped studio which opened last December in an idyllic country location 200km north of Tokyo. Other SSL equipped 'resort studios' include Music Inn Yamanakako, Kawaguchiko Studios, JVC Yamanakako, Marine Studio and Little Bach.

Meanwhile, in Tokyo, the waterfront area has been a very successful site for Smile Garage. Established five years ago on the fifth floor of an old cereals warehouse in the docks, this two studio facility originally had an SL 4040 E Series and an SL6048 E Series and has recently upgraded to two 72 channel SL 6000 G Series consoles with Total Recall™.

▲ *Moody Moon Studios features a 72 channel SL 4000 G Series console in a country location north of Tokyo*

apt-X™ 100 Data Compression Gains Ground

The digital audio data compression system introduced by SSL subsidiary Audio Processing Technology (APT) continues to be well received by manufacturers of recording and broadcast equipment.

apt-X™ 100 system technology has been adopted by a number of manufacturers in the UK, the USA, and also in Japan where APT is represented by SSL Japan. Applications where **apt-X™ 100** technology is used include satellite transmission, hard disk recording and cable transmission.

Charlie Day, Operations Director of APT comments that, "APT data compression has established itself as the number one choice. Sales of all three formats of the **apt-X™ 100** system are increasing rapidly. Many of these sales are for inclusion in new, innovative products due to be announced at either the AES in Paris or NAB in Las Vegas."

Further information on how **apt-X™ 100** is being used by companies around the world is featured in 'Database', the APT newsletter, available from the company's Begbroke office.

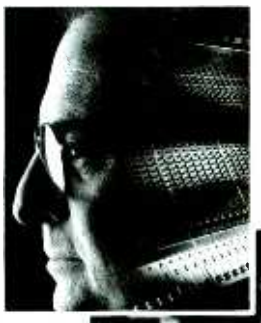
The Black Book

After more than a year in production, SSL has published 'The Black Book', a collection of superb black and white photographs of world-famous British and American record producers, engineers and musicians - all of them SSL users.

Inspired by the jazz portraits of the thirties and forties, each portrait is accompanied by a short biography, a discography and short quotes about music production.

The Black Book documents an unfamiliar side of contemporary music making for the first time, and is published by SSL as a tribute to the recording industry.

A limited number of copies are available through SSL offices.



◀ *Phil Ramone (left) and Dave Stewart (below), two of the leading producers featured in 'The Black Book'*

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"I need to automate my effects sends and EQ in"

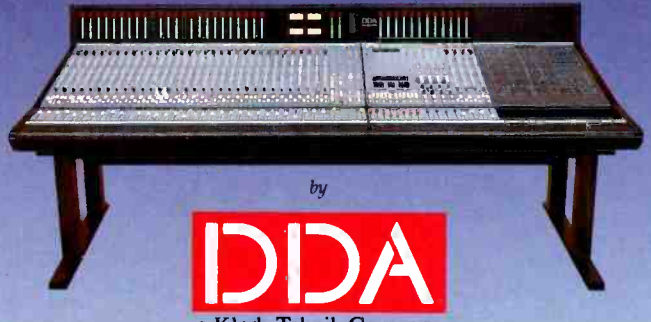
The central computer section of the DCM224V console enables 24 channel functions to be automated dynamically against timecode.

Auxiliary on/off, EQ in/out, and pre/post switching are among the facilities which can be automated for frame-accurate repeatability.

For a demonstration of this remarkably flexible console call DDA, or Stirling Audio Systems in the UK.

WE LISTENED

DCM224V



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YAMAHA DMC1000

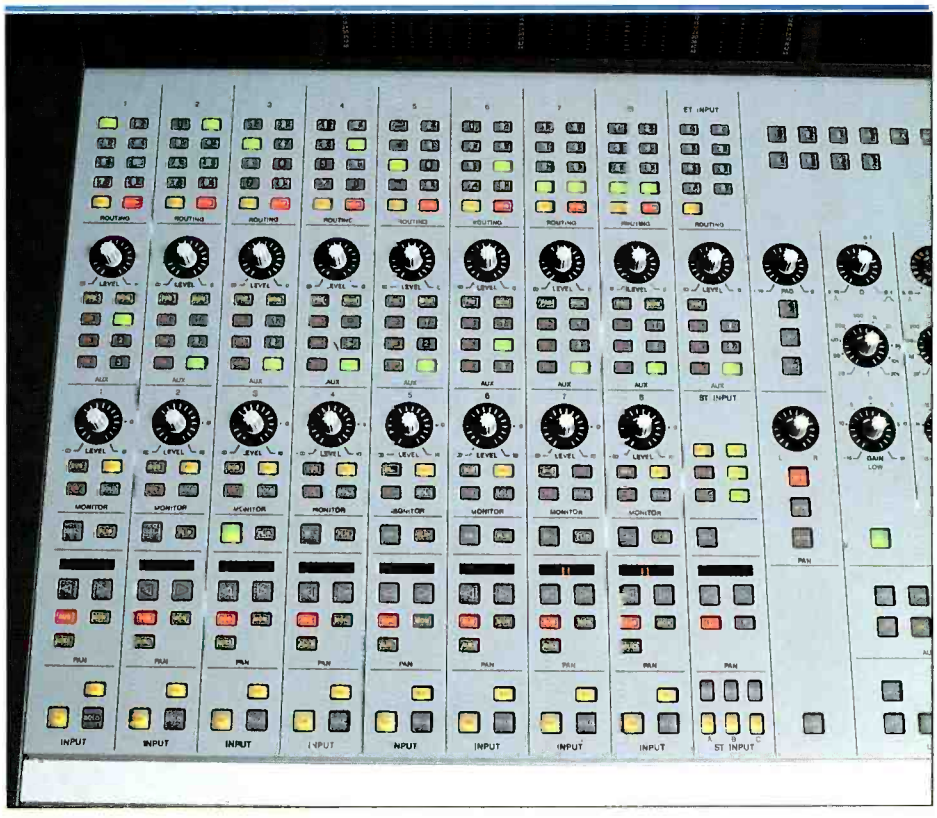
The latest digital console from Yamaha is described by Patrick Stapley

The *DMC1000* was launched at February's AES show, where it was shown along with production models of the *DRU8* digital recorder and the *DMR8* integrated digital mixer/recorder. This, the second generation of digital consoles from Yamaha, represents a large step forward; through the intelligent use of multifunction and assignable controls, as well as menu-based system parameters, a considerable number of facilities have been condensed into a relatively small space without overcomplicating the essentially tactile nature of the user interface. The console's compact size (770×909×363 mm) and sturdy construction, make it ideally suited for location work but it has applications wherever a comprehensive, fully automated, 8-track digital mixer is required.

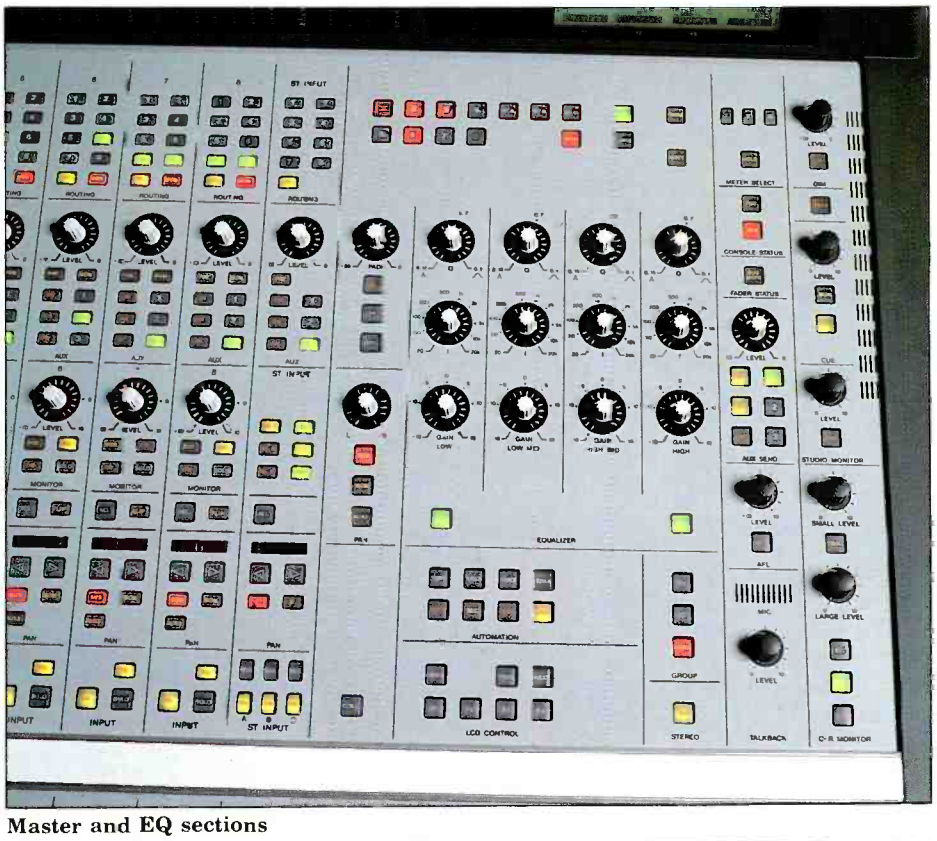
The control surface has a familiar appearance, looking almost modular in construction. It contains a mixture of dedicated and assignable controls, consisting of two types of rotary control and self illuminating buttons. The left hand side of the control surface is taken up with eight I/O channels and a stereo input channel—the main stereo bus fader is situated to the right of the console. At the top of each channel strip, in the traditional manner, is the group routing matrix: apart from the eight group buttons there is switching to the stereo bus and a MON button for assigning the monitor path to the groups. Below this is the auxiliary section with two mono and one stereo send, all of which are separately assignable to both channel and monitor paths and pre/post and on/off selection. A single continuously rotating control, surrounded by 16 illuminating segments, is used to control and display the levels for each selected send—the same type of arrangement is used on the auxiliary master section. Panning for the stereo auxiliary is assigned from the channel pan section, which also controls monitor and group pans: LEFT and RIGHT buttons move the image between 33 points, with a multisegment strip indicator displaying the relative position. Alternatively, panning is controllable from a central rotary control allowing the speed of the pan to be dictated manually.

The monitor section on each channel may be used as a tape return or as an additional input; either way its output is switchable between the stereo and group buses. Monitor to group bus assignment will automatically disconnect the channel input signal from the routing matrix. A BUS button on the monitor section is used for tape/group switching, and a FLIP button swaps the monitor rotary level control with the channel fader, allowing a monitor input to be controlled by a linear fader—levels automatically re-adjust as the two controls flip. Both channel and monitor paths have SOLO, CUT and EQ (IN/OUT) buttons. Two types of solo are available—a destructive in-place solo, or a safe solo with a selectable source (PFL, AFL, APL)—the solo status and level is globally controlled from a master section.

EQ controls are centrally placed in the console where they are assigned to a channel by either pressing an individual SELECT button, or touching a fader. The same continuous rotary controls used for the auxiliaries and monitor levels are used here. Once assigned to a channel the EQ can be toggled between the CHAN and MON paths allowing separate adjustment and display for each. Although the EQ is arranged into four bands (low, low mid, high mid and high) each shares the same frequency range (20 Hz to 20 kHz), gain (±18 dB in 0.5 dB steps) and Q (0.1 to 8.16). The only distinguishing feature, is that the low and high bands are switchable between peak and



Channels



Master and EQ sections

Equaliser section

Data parameter and memory control section

shelf. A RESET button will return all EQ gain controls to zero. In addition low and highpass filters, phase reversal and a digital pad (0 to 96 dB in 1.3 dB steps) are available on this central area of the console.

Depending on assignment, the 100 mm motorised faders will control channel or monitor levels, or attenuate the bus outputs. When the faders have been assigned as bus trims, a red LED flashes above fader 8 acting as a status alert. Another LED, positioned next to the 0 dB position on each fader, indicates unity gain. Provision is made for two moving fader groups and any fader within a group can exercise master control. There are also two links that group together other selected channel functions like EQ, panning, etc; both links and groups share a matrix of channel select buttons, which help simplify their creation, modification and deletion, as well as giving a clear visual indication of set-ups. A global function allows certain controls, such as FADER FLIP and AUX ON/OFF, to be globally operated from any single channel.

The stereo channel controls three stereo inputs—A, B and C—which may be used to return effects, including the two internal *SPX1000*-type processors, or as an additional six inputs. Individual SELECT buttons assign one stereo input at a time to the channel, lighting the respective A, B or C LED on the fader to confirm control. Channel controls will either operate in parallel or stereo depending on their function—ie Aux 1 will send a parallel output but Aux 3 will send a stereo signal determined by the pan, which is separately assignable between the left and right channel inputs. The stereo inputs along with channel and monitor inputs provides a total of 22 inputs for mixdown or direct to stereo operations; if the desk is configured for normal 8-track recording, there are 14 inputs.

Metering is divided into four separate sections—eight input meters switch between the channel inputs and six stereo inputs; another eight switch between tape and bus; four switch from the auxiliaries to the stereo cue and studio LS outputs; and two 32-segment meters follow source selection from the control room monitor section. Apart from the main stereo output, all the other meters are 12-segment. Their ballistics and peak-hold functions are user adjustable.

Analogue level controls are provided for the cue output, studio LS, talkback mic, nearfield monitors, main monitors and a headphone socket. The cue and studio LS share source selection between aux and mon—mon follows the control room monitor source, ie mix, external or cue. The control room speakers can be mono'd and dimmed to a user selected level, and three talkback buttons allow communication/ID to various destinations—this includes a proper slated output with a low frequency tone.

The functions so far described have all been controlled from either assignable or dedicated controls; another level of control is available for system parameters that in general do not require such frequent adjustment. Using the cursor keys, function keys and data wheel, a host of menus and parameter pages can be accessed and modified from the backlit LCD (64×8 characters). Just some of these functions include the configuration of inputs and outputs, control and assignment of the two *SPX1000*-type effects processors, MIDI control, oscillator, wordclock source, channel safe solo select, EQ graphic display, fader crossfade time, external 2-track source selection, insert positions, aux to cue select, emphasis status, timecode details, link parameters select, meter parameters, channel delay/etc. The delay facility allows each channel to be delayed by up to 370 ms in sample steps; if all the channels are delayed by 185 ms, they can then be advanced or retarded in relation to one another, opening up some intriguing possibilities for phase correcting multi-mic set-ups, as well as addressing MIDI and sampling delays.

Up to 64 snapshots of the console may be stored onto a memory card for loading in regularly used set-ups, to implement scene changes, or to instantly recall a working configuration. In addition the *DMC1000* offers realtime dynamic automation of all channel parameters, with storage to a 3½ inch floppy disk drive. Complete console set-ups including all menu-based parameters can be stored and recalled.

The rear panel has a selection of I/O connectors, allowing digital interface between formats—SDIF2, Prodigi, AES/EBU, Yamaha and S/PDIF2—any three simultaneously. Analogue I/Os are available for the auxiliary sends, the three stereo inputs, and the speaker and cue sends. Wordclock, MIDI and timecode sockets are provided, along with a remote RS422 port for external control from editors or computers (EFU M2 and CMX protocols) and cascade I/O connectors for linking together additional *DMC1000*s.

Yamaha also have some optional extras that interface with the console—these include the *IFU5A* and *IFU5B* 19 inch rackmount digital patchbays; the *A* is used for tape returns and the *B* for tape sends. Each patchbay caters for up to 32 channels and allows analogue-like crosspatching and paralleling, using bantam-type patch cords. The *IFU5A* will also patch and route word and bit clocks. Eight- and 2-channel 19 bit $\Delta\Sigma$ A/D converters and 8-channel 20 bit D/A converters are available; and there is an *HA8* mic preamp with eight phantom powered channels.

At the time of writing, the console was still, strictly speaking, in a prototype form: so certain aspects of its operation are better left for future consideration.□

Ed note: We have just been informed that production of the *DMC1000* started in February and that there have been some software changes over the model examined here.

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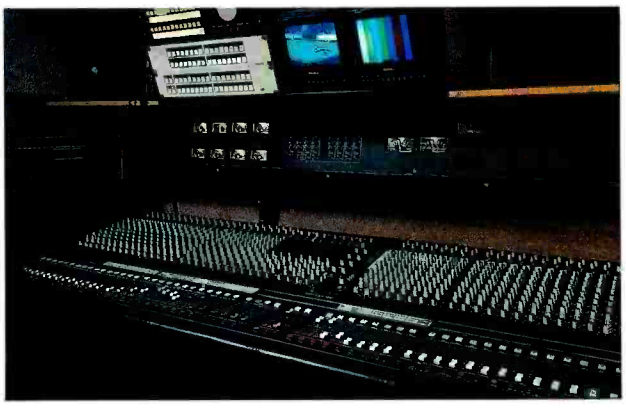

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MYSTICAL HUNTERS

Julian Mitchell reports on a small studio in Stuttgart, Germany, with big ideas on how the German music industry should develop

Stuttgart is Germany's motor-city. Everywhere you look you see Mercedes and Porsche as this is the home of those illustrious marques. A heavy-industry town doesn't seem to lend itself to creative music recording but Mystical Hunters studio has still found enough work to finance their single room facility.

The studio is in an old office block in the centre of Stuttgart. The only clue that there is a studio on one of the block's floors is the name on the intercom button. Once pushed, Mario Tauro, a naturalised Italian lets you in. Tauro is the owner, manager, house engineer and main producer of the studio but he started his career as a session musician. He was determined, however, to make a business out of Mystical Hunters and not let it become a hobby for his own projects. To that end he invested heavily in equipment before moving in. Much of his MIDI equipment, outboard and the Otari *MTR 90* were bought prior to the studio doors being opened to customers.

As a session musician Tauro had worked in many of the top German studios and felt that although the recordings being made were technically very good they weren't producing creative music to rival anything coming out of the UK and the USA. He surmised that the answer was to offer artists a more agreeable recording environment and put the emphasis on making people feel at ease. This feeling was a direct result of Tauro's days of session playing in large and sometimes impersonal surroundings. Here was the motivation to start the studio. He wanted to give German music a heart and an international market. As far as Tauro was concerned German music had everything except this international reference.

The facility

Mystical Hunters is on one level of the building and is small without being cramped. There is a main room as you walk in the front door which is by necessity a multi-purpose area. An artist's relaxation area, a bar and machine room. With the premium on space the Otari *MTR 90* had to be put out here, however, it's kept in an acoustic hood usually designed for professional office printers or other noisy equipment. This open plan area is split at the far end with a small office. The control room and performance area are both off this open plan area.

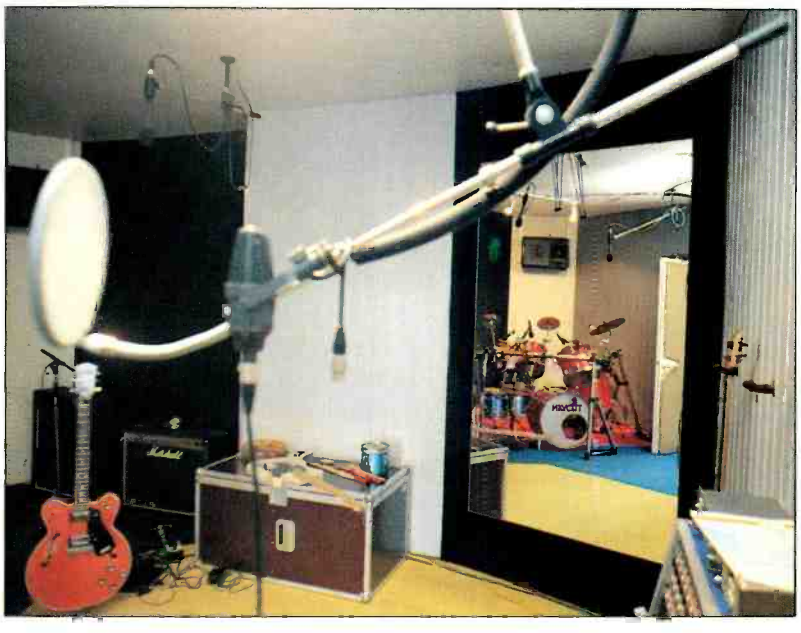
The control room, which measures approximately 12×10 ft is nearer the front door and features a Soundcraft *3200* as its centrepiece. The *3200* is a replacement for a D&R console that was originally bought for the studio. Directly behind the desk is an Atari *Mega ST* computer, which is usually running C-Lab software, mainly because of a special relationship that Tauro has with the main software developer at C-Lab. Behind the master keyboards and computer terminal, and to the sides are the racks of outboard equipment and rackmounted synth modules. In front of the desk the main monitors are Tannoy *FSMs*, with *NS10*s and Auratones for nearfield. Between the Tannoys and up is a TV monitor, which is used for some audio-for-video work.

Because of the studio's sub-street location the only window in the control room looks straight out onto a concrete wall and it's only by straining your neck that you can see daylight up above. It was through this window that they had to get the *3200* in. A glance at the size of the desk and the size of the window makes you wonder how they did it. Apparently it was done by getting a case-maker to build a case to the measurements of the console. The case fitted and by removing the meter bridge on the *3200* the desk just fitted with 1 cm to spare. The alternative was expensive, hire a crane and haul the desk over the building to the other side where the windows are much bigger.

The control room and performance area connect visually but not in the usual manner. There is no large sliding glass area here but a small glass area where the producer can just see the vocalist or other solo performers. The reason for this is the same for other studios with similar control room/performance area arrangements. Tauro feels that he gets a better performance if the artist feels he or she isn't being scrutinised too much. It's all part of the idea of getting the mood right and then capturing it. The performance area measuring approximately 10×10 ft is just smaller than the control room and like the rest of the studio was designed by Tauro himself. Acoustically he decided on a multi-surface room to offer as much choice as possible to customers. There's a dry wall and a mirrored wall near the studio's own drum kit, so you have effectively a 'live end dead end', arrangement with the drum placement pushed into a far corner and screened off for isolation.

Having designed how he wanted the studio to look and sound mainly through reading books about the subject and his previous experience, Tauro had local builders do the work. The studio was also wired by a local technician Heinz Schwitzel who also acts as Mystical Hunter's maintenance man. Harman Deutschland, who distribute Soundcraft in Germany, were initially wary of letting anybody else wire the desk in but were very pleased with the result and lack of noise.

The studio's performance area has proved very popular for Tauro. He is particularly pleased with the guitar sound he gets in there and finds that his idea of limited visual contact with the control room brings interesting results: "I firmly believe I



Fred says, if it's worth recording it's worth

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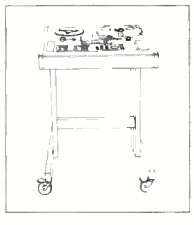
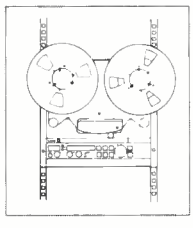
Table-top

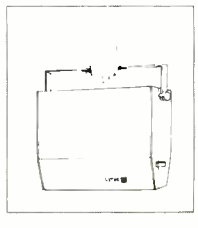
Console



Rack-mounted



Portable



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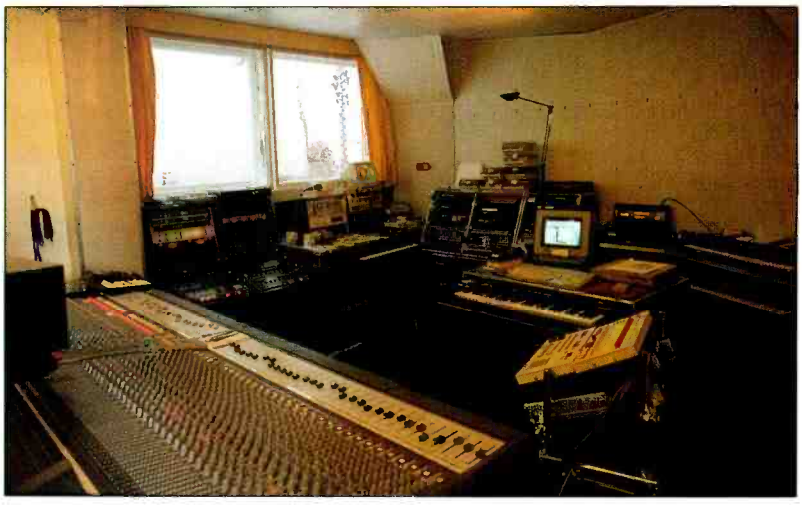
get an improved performance this way, it helps the artist explode creatively!"

Two worlds

Tauro discovered MIDI long after he had become a session musician. Mystical Hunters therefore operates happily in the worlds of both acoustic and MIDI recording. Tauro believes that with software improvements this kind of relationship is the way forward for MIDI. For example, the studio is finding that using real drummers is now far more preferable to quantised drum machines. Also that drummers are becoming more used to playing to MIDI as the software improves, Tauro: "If you have got a drummer who can play very exactly, not necessarily to a particular beat but to a certain 'groove' and he can keep to that 'groove', as the machinery's getting better with MIDI you can actually lock the machine on that groove as well, unlike before. That enhances the possibilities of drummers playing to MIDI."

The more equipment Tauro bought for the studio the more he realised the D&R console was in need of upgrading. He considered buying a Soundcraft *TS24* or an Amek *Mozart* but after advice from Harman Deutschland he realised he needed a certain number of inputs with EQ that an in-line console couldn't offer him. It was then that he became interested in the Soundcraft *3200*. In fact, to demonstrate how well the *3200* fitted into studios, Harman took Tauro to Shambles studio in the UK, who had recently bought one. After speaking with the people at Shambles and seeing the desk in use he agreed to invest in a 56-input *3200* with Yello Tec automation. The main reason he hadn't gone straight for a larger console in the beginning was really his own lack of engineering experience.

But Tauro took to the *3200* very quickly and had soon mastered the use of it. Tauro: "I soon felt that I had been using it for 10 years or more. It is not a desk that will bring in clients just from its name though, like SSL, but it is a high class console which is starting to find acceptance with lots of different



studios. I have given Soundcraft some feedback on the console and have asked for a slight modification which they are going to carry out. At the moment there are no LED input meters, which makes working with a big set-up of MIDI pretty difficult as there might be something wrong in the MIDI patching for a sampler or a whole set-up and you might get lost as to where the MIDI signals come into the desk. If for example you hear a snare drum from somewhere but you have no LED input metering you can't really find out where it comes from. It takes about 20 minutes to check all the channels, all the wiring and MIDI patching to find where the signal came from."

Mystical work

Mystical Hunters doesn't operate like a normal commercial facility. Tauro does have a rate for people to come in and use

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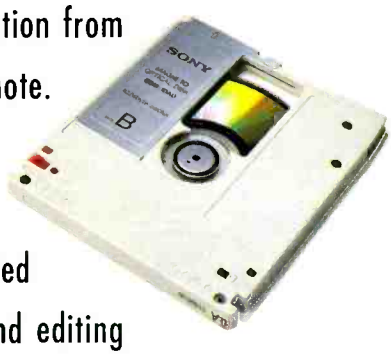

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DD1000
MAGNETO OPTICAL DISK RECORDER

Equipment list

Console: Soundcraft *3200* 32/24 32-bus/80-input remix with Yello Tec automation
Recorder: Otari *MTR 90*
Mastering: Telefunken *M21*, Otari *MTR 10* with Dolby *SR*, Tascam *DA-30* DAT, DigiDesign *SoundTools* with 1600 Mbyte hard disk
Monitoring: Tannoy *FSM*, Yamaha *NS-10*, Auratone
Effects: (include) Lexicon *480L/LXP1/LXP5*, Yamaha *REV7/SPX90*, *tc 2290*, Eventide *H-3000*
Microphones: (include) Beyer, AKG, Neumann, Shure
MIDI: (include) Atari *Mega ST 5*, C-Lab *Notator*, *Unitor* and *Export*; Cubase; Akai *S-900/S-1000*+DAC HD.
Various MIDI tone generators and guitars and drums

the studio with their own engineer and producer but he prefers to attract people who want to work with him. If they want to do that the rate goes down. Artists don't even have to bring in their own instruments as the studio holds a very good amount of guitars, synthesisers and drums.

"I think when you start with your studio you have the problem of having a certain level of equipment and a certain standard to bring people into the studio. Once you achieve that and people have been working in the studio and they are happy with the production, they don't come back for the equipment, they come back for you personally and your attitude."

Tauro is glad to get away from the star console idea that attracts customers by its name: "I don't think it's the console that's attracting the customers at the moment. I hope it's the place, the work and the atmosphere here. I can't say I've got a Soundcraft *3200* like I can say I've got a Solid State *G* series, and people start walking in. But they do find out pretty easily that from the quality side of things this console is as good as a Solid State and they are very pleased with it and look forward to coming back because they are pleased with the audio quality and the way it works. This isn't a console for people who think about Solid State mysteries, it's a console for people who want to do reasonable and high quality work. The console doesn't pretend to be anything else but an honest console, you can see straightaway what it can do and what it can't, and it's supported by the brand name Soundcraft, which is not that well known in the studio scene but is in the live/PA side. I think it's a special kind of person that goes for the Soundcraft console, it's not the ones that are looking for the gimmicks and all the lights, it's the ones that are approaching sound from a very different point of view, they are concentrating on the sound itself."

For someone without a sound engineering background Tauro was at first worried that he may have problems working with the console, especially as it is a split design and Tauro had only been used to the in-line design of the D&R. "I find it very logical and easy to operate because I have a clear section where I have all the signals that I'm sending to tape and bringing back from tape, and in the other section of the console I'm bringing all the MIDI gear. It's a very transparent set-up and I don't have to do a lot of repatching and getting lost in the signal flow. The logical structure of the console, compared to an in-line console like the D&R console I had before, is forcing you to have a logical set-up from the very beginning, with the in-line I was tending to randomly route things and bring them in and out of the console because there was no logical separation between one side of the console and the other."

Tauro's ambitions lie in two different directions, technically he wants to bring in more hard disk recording equipment and so his alignment and recording on analogue multitrack and digitally can be enhanced and improved. His career ambition is to attract some British artists over to his studio as the UK is the place where, he thinks, the most creative work originates. Most important to Tauro, however, is to keep the idea of a relaxed, informal atmosphere in the studio as this is how he can bring the best out of people. As Tauro says himself there are only three things that he is looking for in the music and people that come to Mystical Hunters, "Style, heart and power!"□

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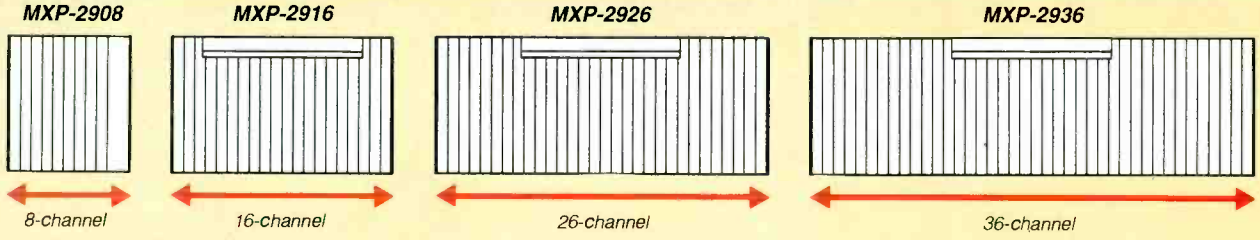

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FLYING FADERS

The Flying Faders automation system was a joint development between Neve and Martinsound. Patrick Stapley takes a practical look at the system

Flying Faders is Neve's fourth generation of moving fader automation. The company developed *Necam*, the world's first moving fader system, back in 1977, which was followed by *Necam 2* in 1982 and *Necam 96* in '84. *Flying Faders* has been with us since autumn '88, and in excess of 100 systems have been sold worldwide with the highest percentage going to America and Japan. Unlike previous systems, the design is not wholly proprietary but a combined effort with Pasadena-based company Martinsound Technologies. Neve's product manager, Roger Cameron, explains the tie up.

"The original core technology for *Flying Faders* had been developed by Martinsound some time ago. We saw what they were doing and thought that the general concept and direction they were going in was good and looked interesting. At this stage, however, there was little in the way of user interface apart from the fader design, so we bought the technology rights and gave them our user interface specifications and detailed instructions on how we wanted the system to be adapted and produced to suit our requirements. So essentially, Martinsound developed the system in conjunction with our specifications leaving us the technology rights to manufacture, produce and develop the system in the future as we see fit. The manufacture of *Flying Faders* is now completely taken care of by our Scottish factory."

Cameron is quick to point out that *Flying Faders* and *Necam* are very different systems both in terms of software and hardware: "I am not even sure whether one can start to compare the two, the overall philosophy, resolution, speed and general performance of *Flying Faders* is so vastly superior."

For example the faders, which Martinsound developed, are based around a P&G conductive plastic track and have a top to bottom time of 100 ms—twice as fast as *Necam 96*. The resolution of the fader is also greatly improved being divided into 4096 digital steps providing $^{1}/_{10}$ dB accuracy throughout the scale.

Apart from the fader, the other components of the system comprise a customised Hewlett-Packard dual processor with 2 Mbytes of RAM, a 40 Mbyte hard disk and built-in 3½ inch HD floppy drive; this connects to the servo rack via fibre optic cabling. There is an independent power supply, a high resolution 14 inch Hewlett-Packard VGA colour monitor and an Adams-Smith *2600* to provide master machine control. On-board system control is from a dedicated keyboard panel, a Global Master Module and a mouse/trackball.

Although initially designed to interface with the *V* series, a conscious effort has been made to ensure compatibility with other manufacturers' consoles as well as previous Neve designs. It's interesting to note that Metropolis Studios in London have added *Flying Faders* to their recently installed Focusrite console.

Operation

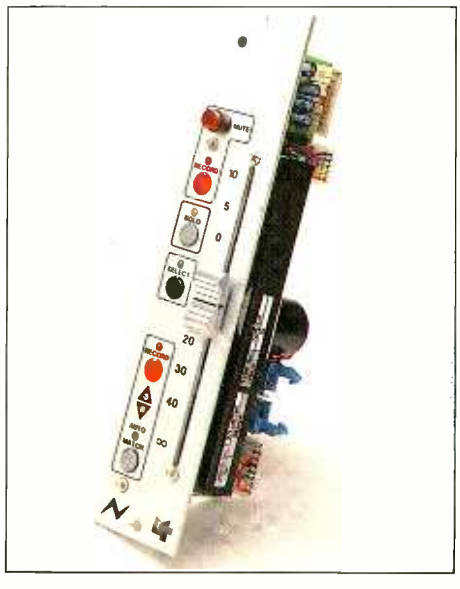
It is difficult not to make comparisons between *Flying Faders* and *Necam* but one of the very good points about *Necam 96* from the beginner's point of view was the ability to operate the system at a basic level very quickly without having to study masses of information; the same is true for *Flying Faders*, which, to quote Neve, "When used in its simplest form can bring automation to the engineer by the press of a single button." By virtue of the fact that most of the manual exists in various Help windows, the user can build up his expertise gradually without having to wade through pages of text to find relevant instructions.

To the left of each fader is a column of buttons and indicators arranged in four sections. At the top is an illuminating MUTE button, a mode/status LED and a RECORD button, which is responsible for toggling through mute modes and mute status. Below this is a SOLO button with associated LED. Next is the SELECT button and its LED, which is used in conjunction with the RECORD buttons to individually set or interrogate mute and fader status; it also plays a part in the system's comprehensive grouping facility. The last section controls the fader and contains a RECORD button and LED providing similar mode and status switching to the mute section; also in this section is a double-arrowed null indicator and a MATCH button plus its LED, which is used to access two level matching modes.

These controls are duplicated on the Global Master Module where they allow certain functions to be set across the board from single point switching. This module contains an additional column of buttons that enable/disable the automation itself, the grouping assignment and the status assignment. PLUS and MINUS buttons serve a dual function in incrementing/decrementing through mixes and globally trimming faders.

Status for faders and mutes can be set independently either globally or locally using the SELECT button along with the appropriate RECORD button. There are three statuses: Ready Record, Safe and Isolate (RSI). Safe allows replay only of recorded data and Isolate totally removes channels from automation. Ready Record is subdivided into two operational modes, which also can be set independently between faders and mutes: Touch Record and Lock Record. In Touch Record the fader or mute will play back the previous moves until it is touched/pressed down, at which point it switches to absolute replacing moves with the current data. Once released the fader will return to read previous information but at a relative position unless it is properly nulled, the mute will retain its updated state until the next opposite mute event. Once it is pressed down the MUTE button causes it to wipe out previous events and to write a new event on release.

Lock Record differs from Touch Record in that once the fader or mute has been released it no longer replays previous moves but continues writing its present level/state to the end of the mix. A third mode—Auto Match— exists purely



for faders; in this mode the fader will automatically return to the null point at a specified glide time once it has been released. Level matching can alternatively be activated at any time during updates by pressing the Match button on each fader—glide times for Match and Auto Match are separately programmable. All modes can be set at any time during the mix locally or globally using the RECORD buttons. Touch Record is the system default on selecting Ready Record status.

In addition to faders and mutes, up to four channel buttons can be automated and in the case of the *V* series these are the CHANNEL MUTE, MONITOR MUTE, EQ IN and INSERT IN. These buttons are switchable between Touch Record and isolate only.

So to start a basic automated mix, the faders and mutes are set to provide an initial setting. The RUN button on the Global Master, is pressed on (faders and mutes assume Touch Record mode) and the tape is put into play. As soon as timecode is received the computer takes a snapshot of all fader and automated switches; this Initial Preset is used in subsequent passes as a starting reference. Fader moves and cuts are performed, the tape is rewound to the start ready to play back or update. This first pass is automatically stored in RAM as Mix 1/Pass 1; subsequent passes will appear as Mix 1/Pass 2, Mix 1/Pass 3, etc, and unless deleted, every pass will be saved to RAM. The current pass always forms the basis for the following playback/update unless a different pass is selected, in which case a branch appears and a new mix number is issued. For example if my last pass were Mix 1/Pass 10 but for some reason I wanted to return to Pass 5, the resultant mix from this update would be Mix 2/Pass 1. All this information can be viewed in the Mix/Pass window (Mix Tree), which shows a graphic representation of the mixes stored in RAM and their hierarchical relationships to one another. Of course, as with all automation systems it is advisable to back up mixes periodically to non-volatile storage, and this can be done to either hard or floppy disk.

Grouping

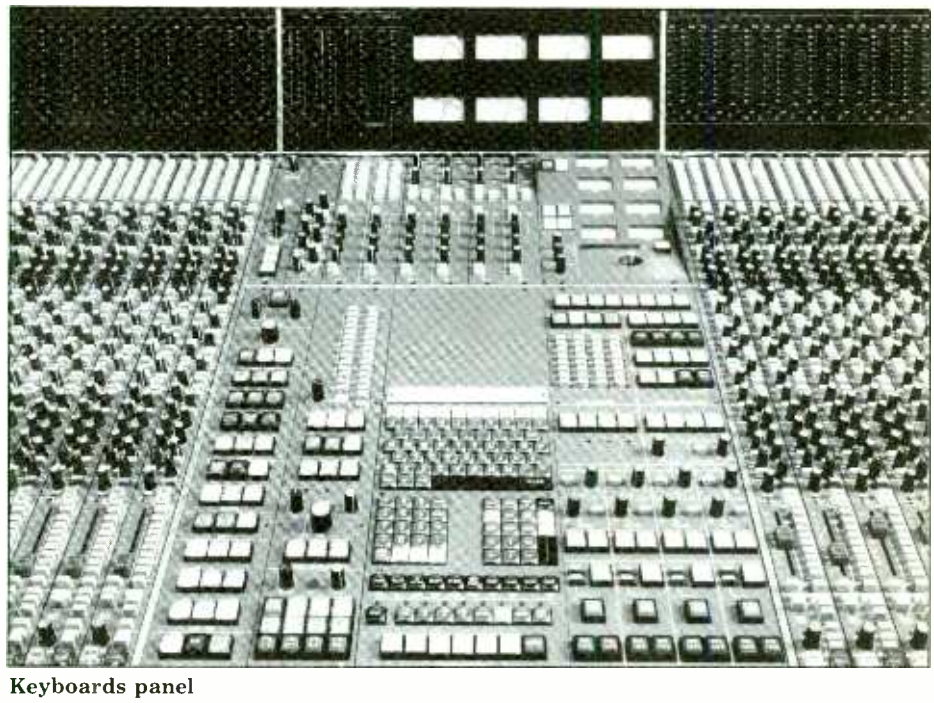
The grouping system on *Flying Faders* has introduced some new features that are unique to moving fader automation. There are two types of group available in the system: Group, which operates like a conventional VCA group with an independent fader acting as master, and link, which is similar to the original *Necam* arrangement where all faders and cuts within the group display master control. Although mutes are included in Links and Groups, channel switches and solos are not; RSI switching on a Group Master has no effect on its slaves but mode switching on a Link fader affects the rest of the Link.

The nature of the systems grouping can create a complex hierarchy with Group Masters coming under control of other Group Masters and Faders being part of a Group and a Link. Although this offers enormous flexibility it can also lead to confusion; to help the user, an Interrogation facility is provided that shows each channel's status either in relation to a nominated channel, or on a global basis. The various statuses are displayed by the Select LED and Null LEDs on each channel.

Because dynamic Group data is only written and replayed from the Group Master, none of the slave channels will retain Group moves or mutes once the Group Master is removed. The Coalesce function has been designed to get round this problem by copying all Group Master data to their respective slaves, it also allows the user to include/exclude any submasters that may exist. This Group Master data combines with any individual channel data, freeing the slaves from master control. Links on the other hand, because each channel is in effect a master, has no need for this type of facility.

Another useful Group facility is Suspend; if the Group Master fader has been set too high or too low, it can be moved using Suspend mode to a new level without affecting its slaves. Individual fader settings in Links are also adjustable by touching any fader in the Link along with the one that is to be reset; by removing touch from the first fader the second can be adjusted without affecting the rest of the Link; once the new level has been set, the fader is released rejoining the Link at the adjusted level.

The Group function provides a method of trimming a fader that is already moving. Due to the fact that the record modes cause the fader to switch to Absolute Write when touched, it is impossible to trim a continually moving fader successfully without losing some dynamic data. However, if the fader is slaved to a Group Master it can be trimmed up and down relative to its original moves, and if desired the Coalesce feature may be used to superimpose the Group Master moves onto the slave faders thus allowing the group to be disbanded.



Keyboards panel

Trim

Flying Faders offers a number of 'on- and off-line' Trim facilities. By calling up 'Trim Faders' from the Mix Menu, it is possible to enter a trim offset for selected faders between two timecode points. The Trim in and out points have separately programmable crossfade times of up to 999 frames. Of course, the whole duration of the mix can be trimmed on as many faders as required and by as little as $^1/_{10}$ dB. Like other screen-based functions, the operator has the choice of entering commands directly from the keyboard, the Global Master, or by using the mouse/trackball. There is also Manual Trim, that enables the engineer to run the mix from any point and adjust a fader to the desired relative level, once the tape is stopped, a new Pass will be created with the fader trimmed by this offset for the entire mix.

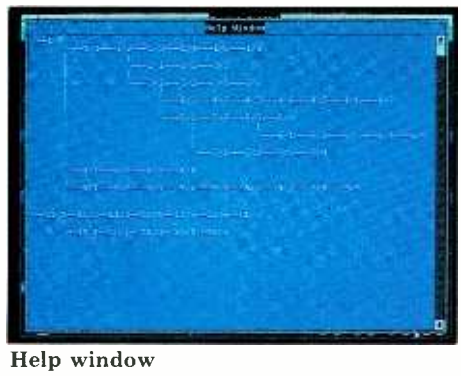
Menus and screens

The main screen gives details of the current project, Mix/Pass number, locate From and To information, the previous and next Label (user-defined cue point), the amount of free RAM, the null Glide times for each matching mode, and timecode readout. Transport control is directly available from this screen as well as from the keyboard. Other features like Safety Nets, where channels automatically switch to Safe either side of a user-programmed window, and Local Zero, where the timecode is given an offset to provide a zero reference, are also accessible from here. At the top of the screen is a Menu Bar used to access the seven main menus: System, Mix, Options, Transport, Lists, Gangs and Help.

Flying Faders incorporates Microsoft *Windows* as part of its user interface; when a menu is opened, operations requiring more than a single button press, ie Trim Faders, will produce a dialogue box containing all the relevant operational parameters. This is another example of how the system simplifies operation by avoiding complicated sequences of button presses; and to speed things further, the more regularly used dialogue boxes, such as Label, Mute, Keep, Load, etc, can be accessed directly from dedicated function keys.

To give an idea of how this system works, I will run through the procedure for storing a mix from RAM to disk—whichever Mix/Pass is displayed on the main screen will be saved to disk. Either the Mix menu is selected followed by the Keep Mix entry, or the KEEP key is pressed; this brings up the Keep dialogue box. All mixes are comprehensively filed under Client, Project, Title and Mix, but if no entry is, or has been made the computer will substitute Workclient, Workproject, etc. and give each subsequent mix a new number. Notes about the mix can be written, its locate times changed and its destination confirmed (hard or floppy disk). Once the desired information has been entered, the Keep button on the screen is

Flying Faders main screen



Help window

selected and another dialogue box will appear to confirm the mix has been kept. When a mix is kept to disk the following information is saved along with it: Fader Moves, Mute Events, Channel Events, A/B Cut Events, Groups and Links, Mix Notes, Locate Times (including pre-roll), Reel Start and End Times, Auto Record Times, Label List, Channel ID List, Glide Times, Smart Keys, Timecode Type, and Local Zero Position.

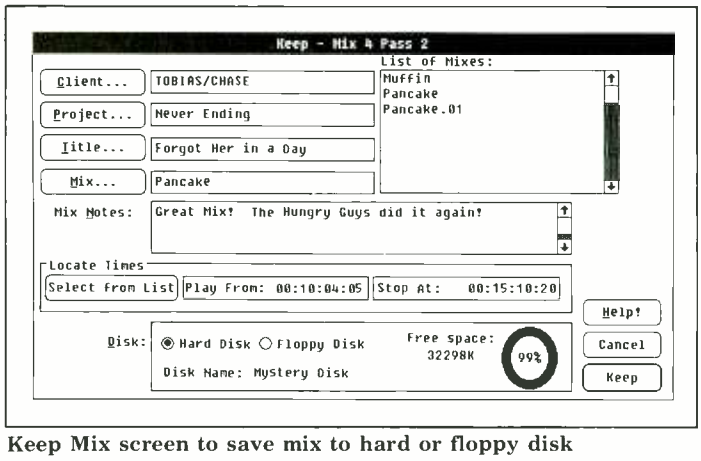
The Mix Menu contains a number of other commands and dialogue boxes: Load Mix, Delete Mix, Show Mix Statistics (which displays the channel numbers with fader and event data, as well as the timecode points between which all these moves have occurred, plus the memory size of the current Mix/Pass chain), Show Mix/Pass Tree, Clear Mix Memory, Trim Faders, Merge Channel Data, Copy Channel Data, Erase Channel Data, Select Mix/Pass, and Run On (switches automation on).

Merges are performed extremely quickly and may be performed using all or selected channels. There is also the option to include/exclude faders and/or mutes and/or channel buttons, and a choice of 'splice' mode; Absolute or Relative. Absolute causes the faders to level match from the left side of the splice to the right at a specified crossfade time; Relative causes the faders to assume an offset on the right side of the splice that matches the levels on the left side. By the provision of Merge From and Merge To times, inserts are easily achievable.

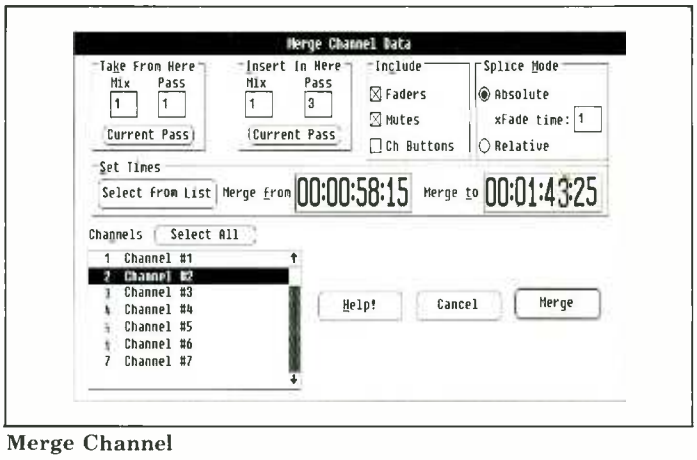
The Channel Copy facility, which allows data from one channel to be copied to one or more channels, shares a number of similarities with the Merge function. The source channel can include/exclude fader, mute and channel buttons, and there are Absolute and Relative modes. A fader's moves can be duplicated to itself in a different Mix/Pass, or to other channels in the same Mix/Pass. A timecode offset enables channel data to be copied to other parts of the mix, providing a useful facility when working on programme with repeated sections—the present drawback being that only one source channel is catered for at a time.

From the System Menu, floppy disks are formatted and named, the entire system is resettable as for power up, faders can be put through a 20 second calibration procedure to ensure they meet with the system spec, and various statistic and diagnostic windows are accessible. If fitted, the Recall system for Neve *VR* consoles is called up either from this menu or from a dedicated key on the console.

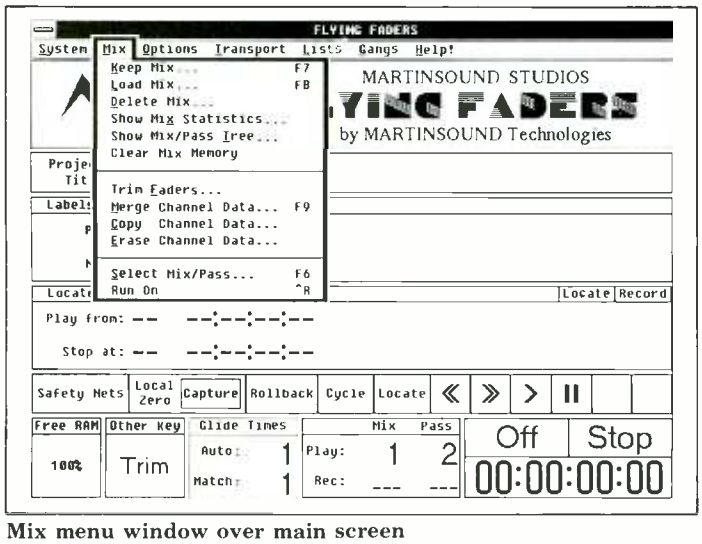
The Options Menu provides access to a number of additional facilities. Smart keys, a hangover from *Necam 96*, enable a string of commands to be memorised and executed by a single key press. Another facility that was available on *Necam 96* but has become more elaborate is Text Pages. These have been designed to keep general notes and a simple wordprocessor including Cut and Paste functions is incorporated; Text Pages are filed separately to Mix Notes which are specific to mix data. The Timecode dialogue box allows selection of frame rate and allows the user to mask hours and frames for all the code displays. If the incoming code doesn't match the set code, a message will appear telling the user that the timecode is not set properly and shows the frame rate of the incoming code. The SHIFT button on the Global Master doubles as a timecode indicator and will light if good code is being read but flashes to indicate poor quality or faulty code; it goes out altogether if code is not present. The Local Zero facility mentioned earlier is set from this box. Also available from this menu are the dialogue boxes for setting Glide Times, Safety Net Times and for switching any or all of the console's



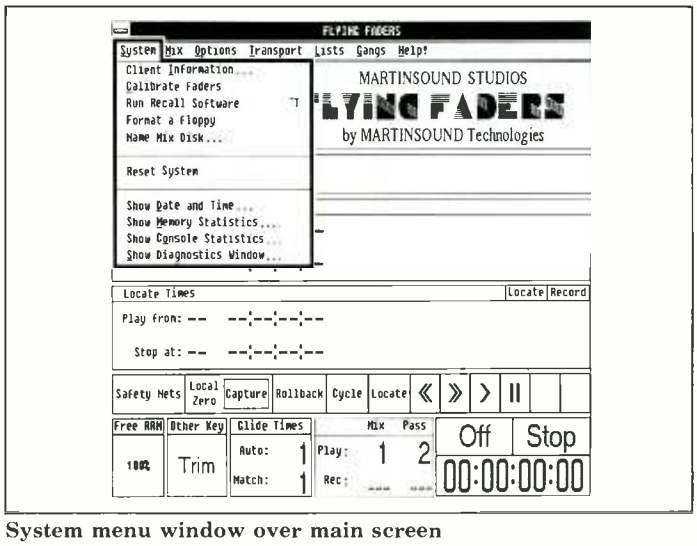
Keep Mix screen to save mix to hard or floppy disk



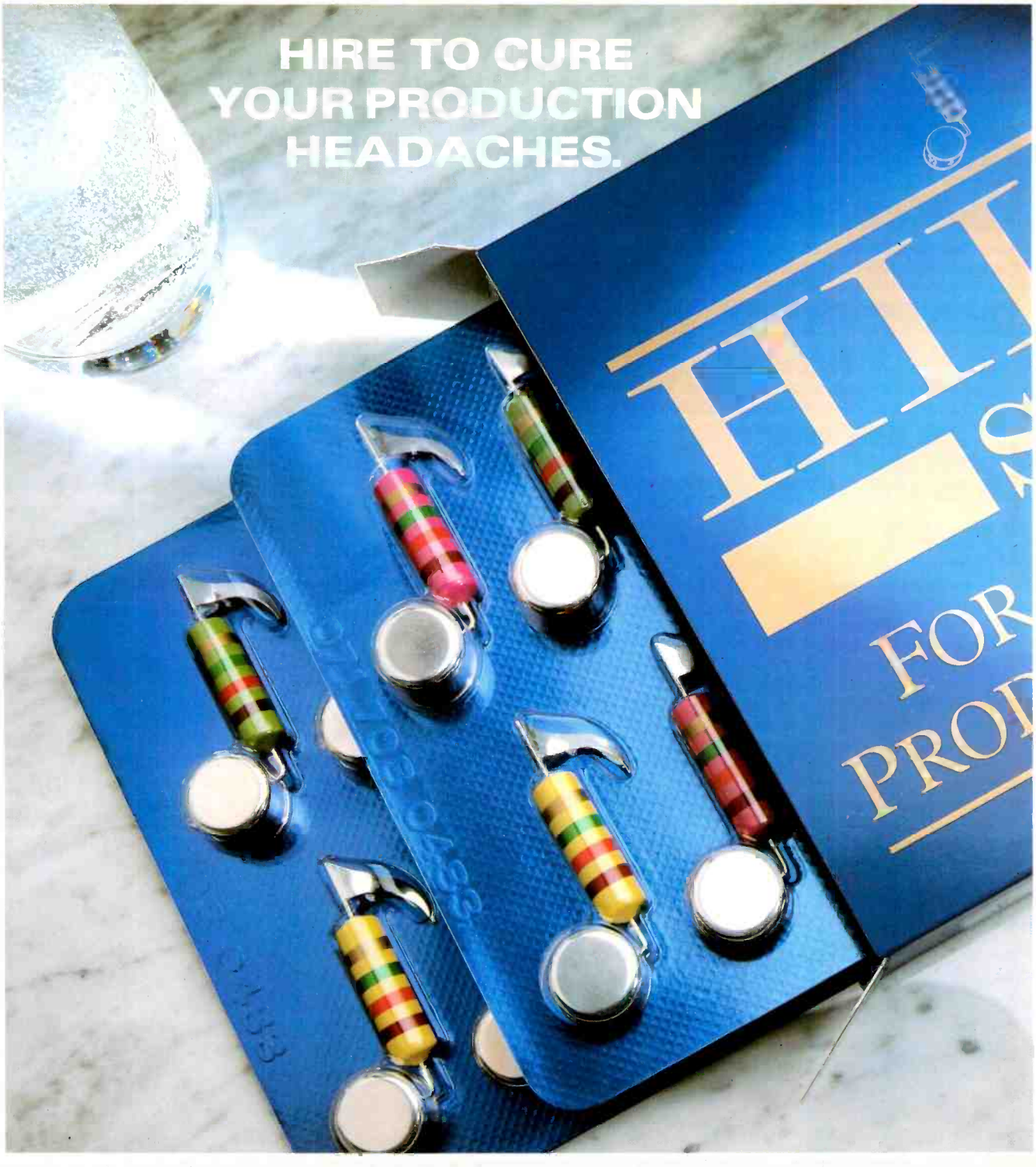
Merge Channel



Mix menu window over main screen



System menu window over main screen



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PIONEERS IN AUDIO ACCESSIBLITY

MIXING CONSOLES

The following list of mixing consoles introduced over the last year has been compiled from information available to us at the time of writing. We would welcome details of omissions for inclusion in future surveys

Soundtracs Megas Mix

ADT: *5MT* production system is the latest version of the ADT recording console and is available in frame sizes of up to 56 input channels. The console is fully modular and features five different modules.

Allen & Heath: A 24-track version of the *Saber* console with patchbay, metering and monitoring for 24 channels. There is now capacity for 36 tie-lines for external patching via multipin connectors. This version of the *Saber* is available in frame sizes of 28, 36 and 44 inputs and 28 and 36 with a patchbay.

The new *Spectrum* desk from the *Saber* line for recording and sound reinforcement. Available with up to 32 main inputs, plus 16 additional line inputs and four mono and two stereo effects returns.

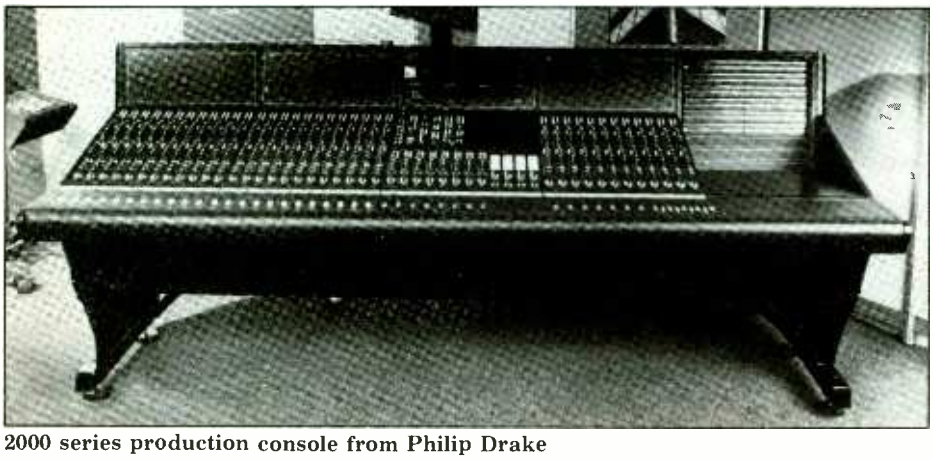
Amek: *Hendrix* is based on the technology and concepts used in the *Mozart* system. The standard configuration is 40 in-line, dual-path input channels with eight stereo returns. A version of the Amek/Steinberg *Supertrue* automation system is included in the console as standard equipment, with realtime automation of eight switches per input channel. Inputs feature 4-band parametric EQ with swept pass filters, 24 bus routing, 12 auxiliary buses, and multimode panning, which allows 3-channel stereo with the possibility of using one of the sends as a Surround source.

API: The *Discrete* series was designed for the customer to configure to his requirements. The idea was to make a single model but within that to make it as flexible as possible. The console is available in sizes from 32 channel upwards.

CAD: *Maxcon* automated consoles are available in widths up to 109 inches, with 144 inputs (288 at mixdown).

DDA: *Profile* is a 24-bus mixing console, designed around a similar concept to the *DMR12* desk. *Profile* features 56 modules in a 2.4 metre frame, offering 32 mic inputs with 24-track monitoring, all modules have 4-band fully parametric EQ, 10 auxiliary buses with a direct aux out feature and 24-bus routing. This concept allows monitoring and mixdown to be performed from the same position on the desk. An automation package is offered with full colour graphics, VCA faders, noise gates and cue list editing.

DMR12 uses a multiple input design. Every input module has 4-band EQ, eight auxes and routing to all the multitrack outputs. All inputs have identical EQ and channel features. The *DCM224V* is a variation on the *DCM232* designed for post-production. Enhancements include an extended EQ with five bands and two filters on the input module.



2000 series production console from Philip Drake

D&R: *Avalon* is a 32-bus recording console with 4-band parametric EQ, eight aux sends per module, channel or monitor routing, three mute groups on monitor and channel with local mutes, optional patchbay and automation ready. The *Marilon* series of consoles has the same basic features as the old *Dayner* series but with the addition of 24 buses, extended EQ features, slide fader in the monitoring section and optional patchbay.

Eela Audio: *S440* Broadcast Control Centre is a user configurable broadcast desk, achieved by using a custom single chip microprocessor in each module programmed for most common types of work in a broadcast studio.

Euphonix: *Crescendo* digitally-controlled analogue console system features a different approach to signal routing. Dynamic and snapshot automation includes control over faders, sends, EQ and routing, as well as headphone and talkback.

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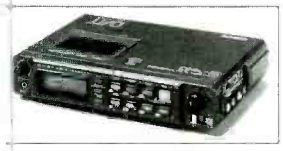

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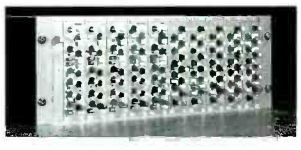

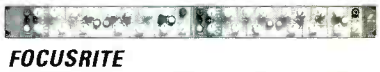

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Focusrite: The *Studio* console's design has been drawn from the Focusrite *ISA* module products and combined with a logic control system in a new open architecture. The architecture is a mixture of the traditional split and in-line formats.

Midas: *XL3* sound reinforcement console concentrates on providing more versatility for applications in front-of-house, monitor and theatre environments.

Neotek: *Encore* post-production console is derived from Neotek's *Elite*. The *Encore* is available with comprehensive machine control, integrated moving fader automation, and 4- or 6-track formats. *Encore* desks are designed to individual order and are available in versions for one to three operators, with up to 96 inputs.

Neve: *44* series broadcast console designed for the increase in stereo radio and television, but later will be used for video post-production with developments including editing and VCA options.

Otari: *Premiere* has been designed specifically for the needs of feature film and TV post-production. The console is available with no maximum number of modules and can be configured to accommodate from one to three engineers. Each module features dual line inputs, 4-band parametric EQ, sweepable high- and lowpass filters, LED input level meters with peak indication, 4-channel panning, eight aux sends with independent level and mute controls. Automation is available in the form of the *Diskmix 3/Film* moving fader system. Otari's Sound Workshop *Series 54* is a 24 bus audio console designed for production, mixing, music recording, broadcast and post-production work and is available in configurations from 24 to 60 dual-path input modules featuring ten aux sends per modules and four band EQ. Also available in a modified version specifically adapted to cinema and multi-channel video post-production is the *Series 54*/Film console. Here the mix path pan control on each module is converted into a 3-channel with an independent centre. The master section allows complete 4-channel monitoring.

Philip Drake: *2000 Production* console has been designed especially for production for stereo TV. The console allows choice from a selection of modules such as mono or stereo inputs and multitrack in-line types.

Ramsa: *WR-C900* sound reinforcement console is available in two frame sizes (32- and 20-input). Provides four groups, left/centre/right main buses with 3-channel panning.

Sound Creators: *VMX* series monitor console designed to provide add-on monitoring where the studio console has run out of monitor channels, or as an effects submixer.

Soundcraft: *Delta 200* has been followed by *Delta 8, Delta Venue* and *Delta Monitor* consoles. The *Delta 8* is an 8-bus recording console in a split format. It incorporates eight groups and either 20, 28 or 36 inputs. The *Delta Venue* is available in four frame sizes, 16-, 24-, 32- or 40-channel.

Sapphyre is an in-line design with an I/O module giving access to dual signal paths. Each module incorporates individual noise gates and switchable 4-band EQ. The *Sapphyre* is available with 24, 32, 40 or 48 inputs. The console has been aimed primarily at the 16- and 24-track market and is capable of driving a 32-track tape machine.

Spirit Studio and *Live* consoles. The *Studio* uses an in-line format and is available in two frame sizes, 16/8/2 and 24/8/2. Use of the four stereo effects returns gives a possible 56 inputs available on the larger framed *Spirit Studio*. Each channel has a 4-band EQ section with fixed HF and LF controls with two swept frequency mid bands. The console has eight groups, with 16-segment LED bargraph meters and a subgroup switch for each pair of groups, which routes their output direct to the stereo mix bus. Above the groups are the four stereo effects returns each with a 2-band EQ section. The *Spirit Live* console is available in three frame sizes, 8/3, 16/3, 24/3. The console provides an additional mono output alongside the main stereo faders, which can be used as a completely independent mono output, or can be set to give a mixed sum of L and R stereo output signals. This could be used for an L-C-R loudspeaker set-up or for creating separate vocal and instrument PA systems.

Soundtracs: *Sequel* sound reinforcement console bridges the gap between the *SPA* console and the *MX* range. The console is available in 24-, 32-, 40- and 48-input versions. The input channels feature a 4-band parametric equaliser with highpass filter.

Megas series comprises the *Megas Stage, Studio* and *Mix* consoles. The *Megas Mix* is a general purpose console for various applications where a maximum of four group buses are sufficient. The various frame sizes can be loaded with three different types of input modules and a maximum of two dual group modules providing four audio groups. The *Megas Stage* is a dedicated sound reinforcement console. The *Megas Studio* is a dedicated recording console with 16 or 24 group

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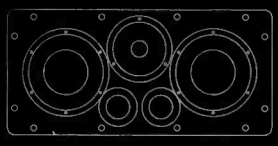

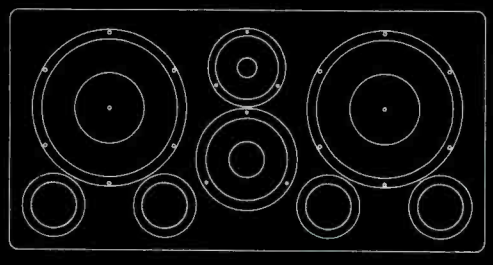

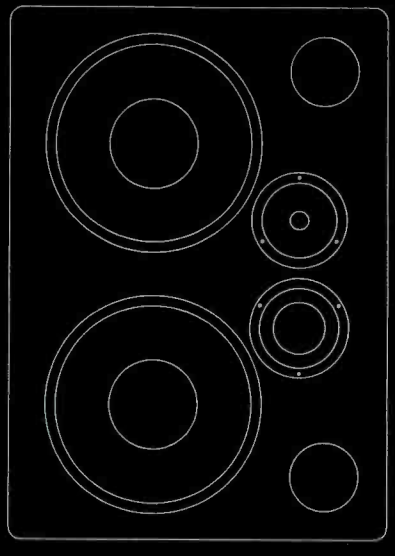

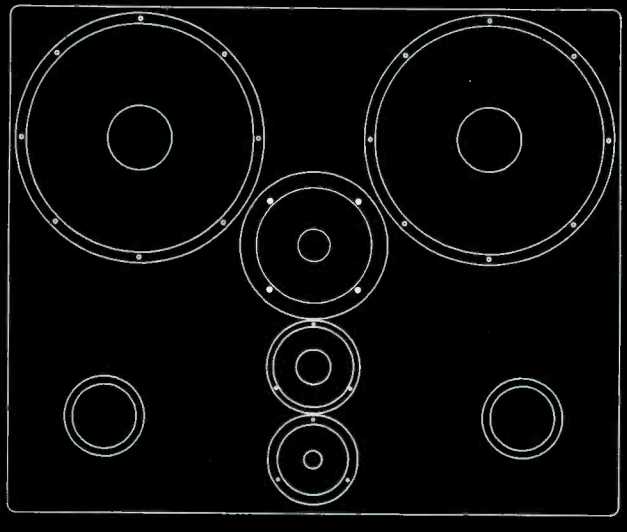

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Soundcraft Sapphyre

buses with or without patchbays. The variety of frame sizes can be loaded with mono and stereo input modules plus up to eight dual group modules (16 bus) or 12 dual group modules (24 bus) to provide 16 or 24 group returns respectively. MIDI mute automation is also fitted.

Quartz 24-bus in-line design with automated mute on all channels and auxes. The on-board MIDI muting system is an extension of that used on the *PC* MIDI console. Frame sizes for 32 and 48 modules are available with the 48-channel size being only 2.2 metres wide.

Studer: *D920* digital broadcast production console, which processes digital and analogue signals, switched together with an existing analogue console free of feedback.

TAC: *Magnum* 24-bus recording console. Featuring 4-band EQ, optional MIDI muting and fader automation systems.

B2 4-bus console has been designed for smaller sound reinforcement and video post-production applications and features electronically balanced inputs, busing and outputs, 4-band semi-parametric EQ, auxiliary sends and four stereo effects returns.

Tascam: *M-3700* series of consoles, an automated console based on the established *M-3500* series of multitrack recording systems. The automation allows control of fader level, channel, monitor and aux mutes and EQ on/off.

Trident: *Vector 432* in-line design features four matrixed stereo buses, 32 group outputs and 16 externally and internally triggerable automute groups that can be channels, monitors and aux mutes in any combination.

Yamaha: *DMC1000* digital console features full automation and a fairly conventional type of control surface. It has 14 input channels, eight mono and three stereo, with eight monitor input channels that can be mixed onto the main stereo bus giving 22 inputs. Internal processing is provided by *SPX1000*s.

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MONITORING SYSTEMS THE TORTUOUS WAY AHEAD PART 12

Phil Newell continues his series

Perfect monitoring conditions will not exist during my lifetime although I am not anticipating my demise for three or four decades yet. Systems are very diverse in their construction and performance, this diversity being a function of the points of compromise chosen by the individuals concerned with their design and use.

On many occasions I have been called to studios to try to improve the sonic compatibility between the large and small monitor systems. After a certain degree of adjustment, especially when using the music being recorded in the studio at the time of the adjustments or the producer's favourite CD, a reasonably high degree of compatibility can usually be achieved. In almost all cases, this adjustment is made by some form of amplitude correction to one or more drive units in one or other of the systems. Unfortunately, despite the compatibility that may apparently have been achieved, a different type of musical programme may well reveal alarming differences in what had previously been judged compatible.

The key to the above observation is in the fact that the adjustments made to improve the systems' compatibility had been amplitude adjustments to the drivers of one or both systems, whereas amplitude is only one of the properties that define the characteristics of a system. A different type of music may be more revealing in terms of phase characteristics or non-linear distortions, such as harmonic and intermodulation. Further, it is quite possible that the amplitude adjustments may have compensated for system differences in the most dominant region of the first piece of music, whereas they may have disrupted a sensitive range in the subsequent piece. The interaction potential is infuriating. It is not possible to define the most important parameters or to what degree their errors can be tolerated as, dependent upon drive signal, the priorities for a subjectively 'natural reproduction can shift within wide limits.

In general terms, phase is of less importance to steady state type programme, such as sustained reverberant organ chords or bowed violins, but as music is generally composed of very complex waveforms, all the above distortions, both linear (amplitude and phase) and non-linear, have differing degrees of subjective effect on the signal. Indeed, the very phase responses that would be of minor effect on the sustained organ chord would be more noticeable on more transient signals, whereas the transient signals would tend to mask other problems such as resonances which the organ chord could excite in a most objectionable way. Moreover, Toole and Olive state[1] that on an instrument such as an organ, the detection or otherwise of minor amplitude irregularities could be entirely dependent upon the amount of reverberation on that signal.

Over 150 years ago, Ohm, and later Helmholtz, carried out experiments that appeared to show that the ear was 'phase deaf'. Their experiments had been carried out on sinewaves, which are very steady in nature and not representative of normal musical programme, but nonetheless, their work is still quoted in some circles as implying the relative unimportance of phase responses in terms of audibility.

Thresholds of perception

Phase responses have an enormous bearing on the waveforms of transient signals, changing considerably the way the shock of the transient excites the ear. Again, an argument had been put forward implying that if waveform distortion changed the envelope only within the integration time of the ear, then the effect would not be perceived, but once again our perception systems seem to be far more subtle than often previously believed. In general terms, the concept of the effect of the integration time states that if the ear samples sounds in periods of time of length x, then it would matter not whether 10 units of a sound occurred simultaneously, or whether each was separated by a period of $^1/_{10}$ x, or even any combination in between; the ear would still perceive 10 sound units within that time window of length x, so the perception of any combination would be the same for all cases. Controlled experiments have been performed that seem to bear this out, yet like the work of Ohm and Helmholtz on phase deafness, the experiments did not relate to the true complexities of everyday musical programme.

For every experiment which proves an effect inaudible, there is a set of circumstances that clearly show it to *be* audible.

Michael Gerzon published a series of papers, articles and discussion documents that dealt with attempts to define limits of the audibility of various distortions and combinations of distortions[2]. He pointed out that some very low level effects, which are by no means always subtle in perceptual terms, cannot be measured purely because we do not yet have the equipment or techniques. Gerzon's article quotes some of Harwood's work from the '50s at the BBC. The implications of this work were that if severe colouration could be detected from resonances 40dB below the main signal, then that −40 dB signal (1%) represented an accuracy in terms of amplitude of ±0.1 dB, and in phase of ±0.6°. Gerzon goes on to suggest that a factor of 10 improvement on these specifications would still produce audible colouration on some programme material, that is, accuracy to ±0.01 dB in amplitude and ±0.06° in phase. In support of this he cites work by Roger Lagadec at Studer in the early '80s who detected audible colouration caused by amplitude response ripples in a digital filter at levels only just above ±0.001 dB.

Work is currently being undertaken, notably in Canada, which is leading towards the conclusion that for a true sense of 'being there', phase, and to a slightly lesser degree amplitude response, must be maintained down to a frequency of 0.001 Hz. Clearly, we can hear neither 0.001 Hz nor 100 kHz, yet our tactile senses respond in these regions. Conduction from the cheek bones at 100 kHz has been detected as significant and 0.001 Hz, which represents one cycle roughly every 16 minutes, is in the 'weather frequency' band where one 'feels it in the bones' that a change is on the way. Much more work still needs to be done on the integration by the brain of the audible and tactile sensations but it is of little doubt that they are very important in terms of 'natural' reproduction[3]. Largely on instinct, for many years I have advocated electronic crossover and amplifier responses from DC to 100 kHz and nothing has recently come to light to cause me to change my mind.

At very low frequencies, phase does seem to play an important part, possibly to a greater degree (if you can compare apples to bananas) than amplitude. Gerzon once again claims that relative phase accuracy down to 15 Hz is probably more important than amplitude accuracy down to 5 Hz in the naturalness of perception; but here, once again, integration of the senses plays an important role. Many studio designers will have come across rooms with say a ±3 dB ⅓-octave pressure amplitude response at low frequencies, which clients may feel are more natural or accurate than another room with say a ±1½ dB pressure amplitude curve (frequency response). Phase differences aside, there is a 'does the room look like it ought to sound like that?' input. In this case, if our general subconscious experience of day-to-day life has taught us to expect certain amplitude irregularities from certain sizes, shapes or finishes of rooms, then the brain can frequently 'hear through' the room response by superimposing an inverse of the anticipated response and perceiving an 'accurate' subjective sound. Blindfolding may well reverse the general consensus of which room was considered to be more subjectively accurate. When digital filtering and phase/amplitude correction becomes further advanced, this is one problem that will surely have to be addressed in more detail. As there is much promise in the field of digital phase correction of low frequencies, an expansive field of research is now opening up in 5 Hz to 100 Hz region.

The great valve debate

Ironically, while we seem to be moving towards the criteria of 0.001 Hz to 100 kHz frequency bandwidth, ±0.05° phase response from 10 Hz to 40 kHz and a ±0.01 dB pressure amplitude response over the same range, until such 'perfection' is achievable, there is a certain element of 'the wider you open the window, the more dirt blows in'.

For many years I viewed valve amplifiers rather in the same light as vintage cars: nicely made and nice to have but not really suited to driving in today's conditions. Although I would still not advocate valve amplifiers for monitoring use, as I doubt that such technology will lead us much further down the road to improved systems, the affection some people have for them may well stem from their 'limiting the amount of dirt which can blow in'.

There are definitely circumstances under which objectively 'worse' can lead to subjectively 'better' when more essential detail is left exposed, even at the expense of conventional 'accuracy'. Emile Ford has been exploiting this concept for over 30 years with his 'Livoteque' high efficiency sound reinforcement systems. They have anything *but* a linear amplitude response, seeking to deliver the essential detail only above the ambient noise and room acoustics. The masking of low level information by other low level signals can be an important factor in perceived quality where that secondary low level signal is not part of the original signal. In other words, there are situations where 'improved' specifications can lead to a deterioration in perceived or preferential concepts of accuracy until a much lower point is reached in the generation of spurious distortion products, where the benefits of further reductions will once again become apparent. For example, a simple transistor radio may well have alarming non-linear distortion figures yet sound tolerable for everyday use; its poor response bandwidth rendering benign many of its more annoying characteristics. Improving its technical performance by opening up the bandwidth may render the unit most unpleasant to listen to; a further improvement in technical performance by reducing non-linearities may be required before the unit could once again be considered as subjectively acceptable as in its original state. There is also a hideous convolution of the different distortion products, which conspire to mask each other in a very programme-dependent way, such that it cannot be said in any general terms whether ±1 dB in the pressure amplitude response is any more or less important than 1° in phase accuracy, or than an extra 0.1% of harmonic distortion. Even within the harmonic distortion, no absolute, independent of programme, ratio exists in terms of the relative importance of the individual harmonics.

Low level awareness

The sensitivity of the ear to low level effects has again been dealt with at some length by Gerzon[4],

not only in terms of perceived colouration, but also in the context of very low level or even subliminal mixing of instruments in a musical balance; especially where such low level signals either have a positive effect on the 'feel' of the music or would be conspicuous by their absence. On the same subject, Stanley Lipshitz reported that on one test to determine the audibility or otherwise of the insertion of a capacitor into a test circuit, one listener was consistently scoring a very high degree of accuracy in detecting its inclusion. Much hard searching revealed a buzz, produced by the switch and wiring detecting airborne interference when the capacitor was switched in; this buzz was 80 dB below the programme. Screening of the circuitry removed the buzz and rendered the insertion of the capacitor undetectable. The above highlights not only the extreme rigour with which subjective listening tests must be carried out, but also the probability that many subjective differences deemed to be audible in cables, amplifiers and other links in the monitoring chain may not be inherent, but due to outside interference. An example of such circumstances would be where an amplifier was said to be more 'gritty' than a seemingly inferior amplifier with reduced bandwidth. It may well be that the wideband amplifier was subject to more radio frequency interference in the test set-up; removing the test to a more screened location may well have reversed the subjective preference.

The brain notwithstanding, what the ear alone can respond to is worth note. It has a range of hearing of around 120 dB, which represents a power differential from the quietest sounds to the threshold of hearing of 10^{12} or 1 million, million times. At the threshold of hearing, 0 dB, the distance of movement of the eardrum is around $^1/_{100}$ of the diameter of a hydrogen molecule, or about $^1/_{10,000}$ of $^1/_{1,000,000}$ inch. After exposure to a bright light, the eyes take time to recover the ability to see in low light levels: after exposure to a loud sound, the ear/brain also takes time to regain its sensitivity to low level sounds. In the latter case, however, after exposure to a burst of sound at 120 dB, the ear can recover its 0 dB sensitivity at the threshold of hearing in around ½ second[5]; in other words within the reverberation time of an average domestic living room. It has also been shown that some people who cannot hear a 20 kHz tone can under some circumstances detect the effect of amplitude changes of as little as 0.1 dB at 20 kHz.

In the first paragraph I mentioned the gross imperfections of current monitor systems. In the light of the sensitivity of our sensory systems, the performance of any currently available loudspeaker is certainly grossly imperfect; the wild advertising claims of many manufacturers do nothing to further help the industry in general to understand just where we need to be going: nor how far we still have to go. Being so far away from perfection, it is little wonder that arguments rage over which systems are 'the best'. In current reality it is merely a case of choosing a system, the imperfections of which are most subjectively benign with regard to the programme material being recorded and which also offend as little as possible the sensibilities of the individuals involved in the recording and reproduction process. As the variables are so great, it is inevitable that a wide range of monitor systems will exist in order to provide a 'best fit' compromise to each individual set of circumstances.

Further development prospects

The technology is currently in a position where we may be required to go backwards subjectively until we reach a much greater degree of objective, measured accuracy that will allow us once again to move subjectively forwards. In other words, the window is currently opened sufficiently wide to allow in a great deal of subjectively confusing dirt. Given that we are looking for accuracy via bits of metal and cardboard, somewhat arbitarily glued to chassis and coils and mounted in a less than specific magnetic field, it is little wonder in the light of the previous paragraphs that we have not achieved perfection.

The listening rooms themselves also limit severely the advance towards accuracy of reproduction. For the foreseeable future, the background noise of listening rooms will always be superimposed upon the background noise of recording rooms. The room's own acoustic properties will also superimpose themselves upon the reproduced signal to some degree.

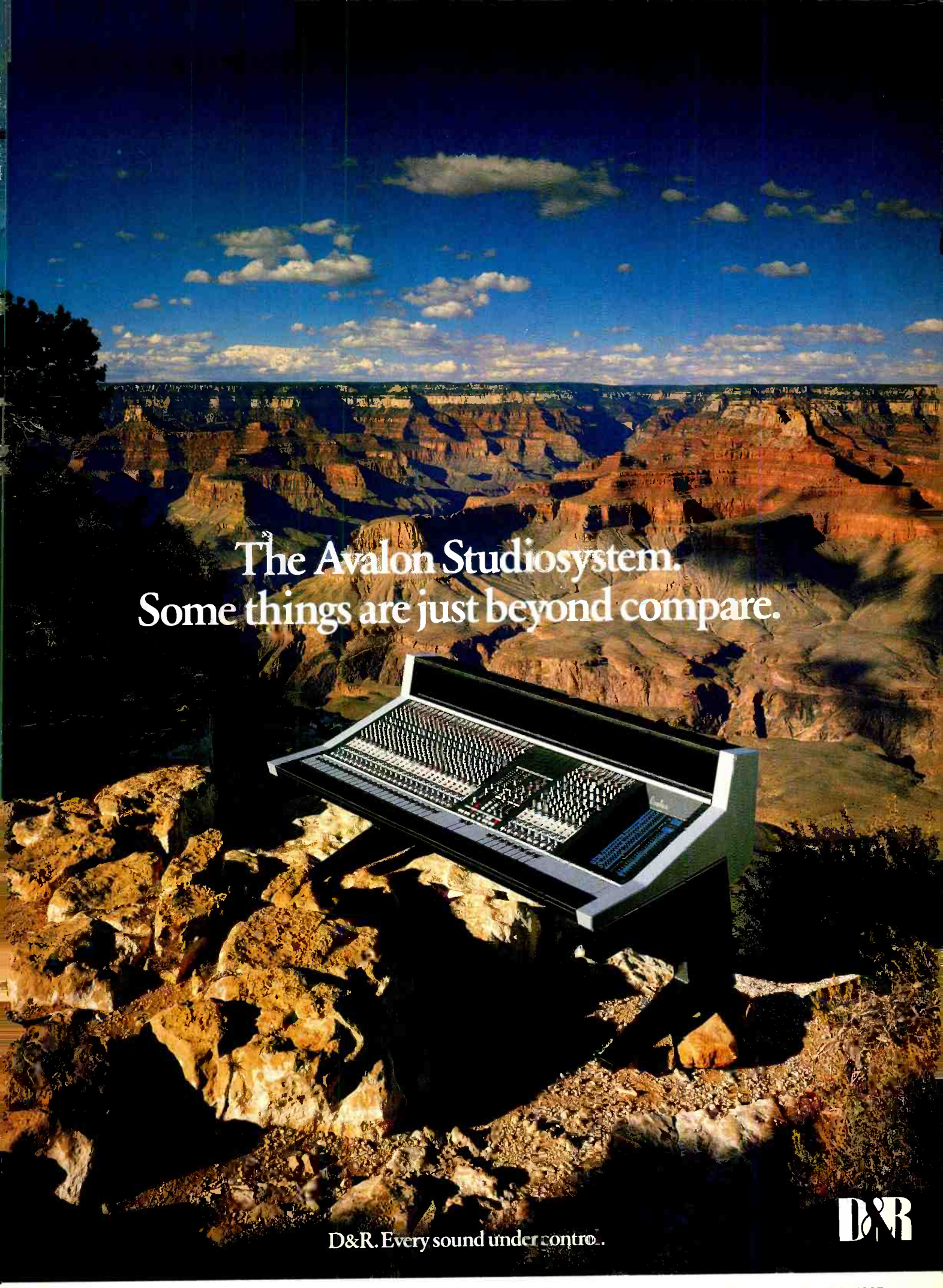
Given today's technology, we cannot even make vague claims towards true accuracy. In the near future, digital signal processing may lead the way forward to more controlled performance but even here, we do not yet know just how far we will need to go before we can claim sufficient subjective improvements, nor at just what level of imperfection reduction this subjective improvement will begin to occur. Certainly the low frequency phase accuracy problem seems to be tailor-made for attack by digital filtering methods.

At the moment, all recorded music is intended for playback over loudspeaker systems of one sort or another, including headphones. Our current great distance from audio perfection would imply that we must bear in mind the object of the exercise, which in turn means that we should choose our monitor systems as we would any other tool for a job: we should select the tools we feel are the most efficient while enabling us to achieve our objectives to the greatest possible degree. This implies compromise, which necessitates a variety of products to choose from.

So where does that leave us? To stop seeking perfection, no matter how elusive it may be, would be defeatist.

If you cannot equalise a Shure microphone to sound like a Neumann, or a Neumann to sound like a Schoeps, then why expect to be able to equalise two monitor systems to any degree of compatibility? Only to a limited degree on a narrow range of sounds can any reasonably accurate sonic compatibility be achieved. Across the board sonic compatibility is still way out of our reach! It was the enormous ravine we still must bridge in our search for perfection that drew me to write the opening sentence of this discussion: 'Perfect monitoring conditions will not exist during my lifetime'.□

1 Toole F E and Olive S, 'The modification of timbre by resonances: perception and measurements', *JAES*, vol 36, no 3, pp 122-142, March 1988
2 Gerzon, Michael, 'Why do equalisers sound different?', *Studio Sound*, vol 32, no 7, pp 58-65, July 1990
3 Khanna, Tonndorf and Queller, *Journal of Acoustical Society of America*, vol 70, 1981
4 Gerzon, Michael, 'A question of balance', *Records Quarterly* magazine, vol 2, no 3, pp 48-53, 1987
5 Viemeister, Neal F, *An overview of psychoacoustics and auditory perception*, AES Preprint, Sound of Audio, 8th International Conference, Washington DC, USA, 1990



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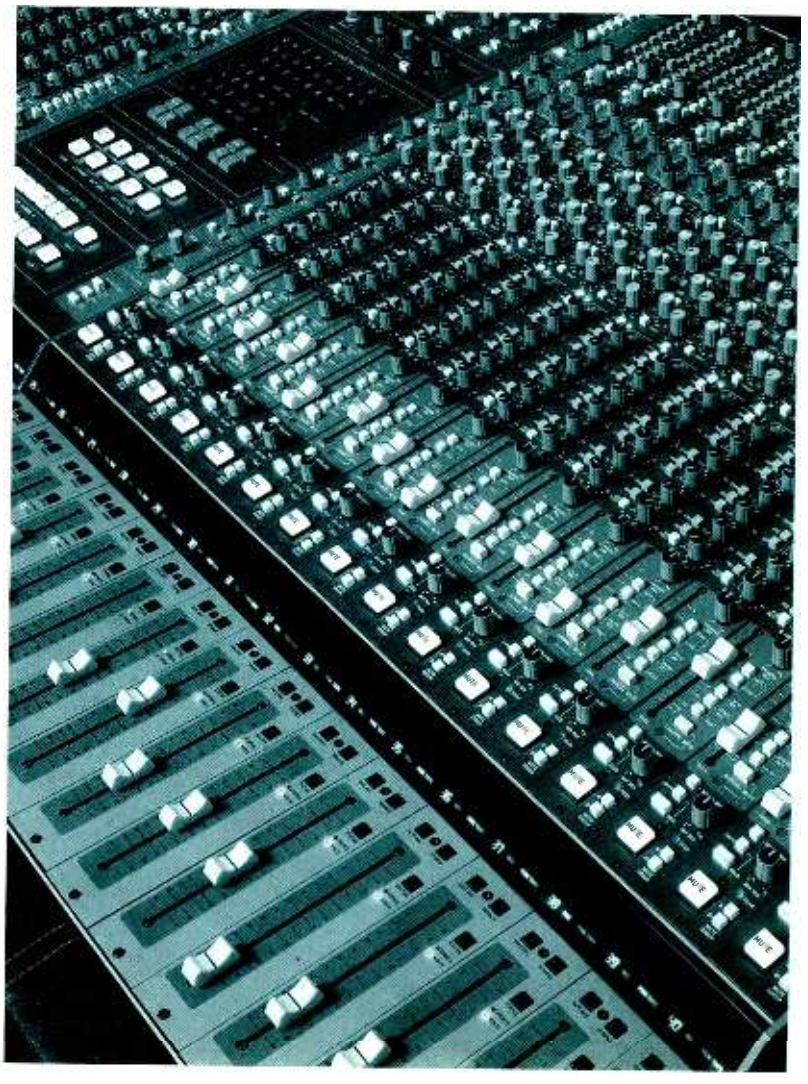

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Barry Fox

DCC: Mono fools

The cat is now firmly among the pigeons. While some Philips spokesmen will talk only of DCC—the new Digital Compact Cassette designed to supplant DAT being co-developed with a 'big Japanese hardware manufacturer'—others have confirmed that DCC was co-developed with Matsushita (Panasonic/Technics). And Matsushita is a co-licensor with Philips, like Sony on CD. Also like CD, royalties will be payable on both hardware and software. The amounts are not yet agreed.

Until now all the Japanese manufacturers, with the backing of MITI the Japanese government's trade ministry, have kept a solid front on DAT. The increasingly bitter rivalry between Sony and Matsushita is doubtless a contributory factor to the split on DCC. Sony is far more heavily committed to DAT than Matsushita.

Although, like Matsushita, Philips will sell DAT decks (bought in from JVC) the Dutch company has over recent years lost confidence in DAT.

"The DAT conference which developed the system and set the standard, was for technical engineers only. The music industry was never involved at that stage. They picked the wrong recording density and the wrong tape. In my eyes this was the main mistake," says Philips' Consumer Electronics Division project manager Gijs (Jerry) Wirtz:

"DCC was specified in reverse. We told the engineers what we wanted. The system had to be digital, giving digital sound quality and digital features. The cassette also had to be thinner and more appealing.

"Our targets set the standard for DCC," explains Wirtz.

Philips wanted to use chrome video tape, which means a minimum recorded wavelength of 1 micron. The tape speed had to be the same as for analogue compact cassettes, 4.76 cm/s. The sampling frequencies had to be the same as for DAT. But the stationary heads could not be of expensive precision design with very tiny gaps. With standard cassette tape width, this set a practical limit of eight data tracks and one auxiliary track, per stereo pair. This in turn created a requirement for 4 bit code.

Philips looked first at ADPCM (Adaptive Delta PCM, as used for CD-I). This system encodes only the difference between samples but does so in 4 or 8 bit PCM words. Quality is inevitably compromised. Then Philips looked at sub-band coding. This technology was already being developed, for instance as a Eureka project for radio broadcasting. The technique, now refined and known as Precision Adaptive Sub-band Coding, or PASC, takes advantage of the masking effect of the human ear on which noise reduction systems like Dolby depend.

The sound is sampled and digitally filtered into 32 bands. Each band is separately coded. The number of bits used to code each band varies depending on the signal content, up to 15 bits. This creates a bit pool of sufficient capacity to average at 4 bits. A small amount of dither—random noise—is added to the recorded signal to conceal coding errors. Raw stereo data rate is 384 kbit/s. The final data rate, for stereo, with error correction, auxiliary codes (eg for indexing) and main channel subcode data capable of displaying up to 400 text characters a second on a screen, is 0.77 Mbit/s.

The smallest recorded wavelength is 0.99 microns. The DCC tracks are recorded with a width of 185 microns but read out with 70 micron heads, to create tolerances for tape weave. I tried tapping the cassette before playback and it worked perfectly.

The heads are made like integrated circuits, by lithographic techniques, on a single silicon wafer. The electronics for many strips are grown on a large circular wafer of silicon, which is later diced into individual chips.

The first growth layer creates a double strip of magnetoresistive heads, for reading both analogue and digital signals from tape (changes in magnetism vary the resistance conductor). Then a second layer of magnetoinductive heads is grown on top for digital recording. Completely separate heads are needed for analogue erasure and recording, so this feature may well be excluded from decks, for cost considerations. No erase heads are needed for digital recording.

The head chips are used in pairs, to provide electrical switching for auto-reverse. Although this is instant, the tape still takes a finite time to reverse direction. So there is a brief but noticeable gap in the sound.

The cassette has spool holes on only one side and does not flip over. These holes, and the tape path, are protected by a slider that opens only when the cassette is loaded in a deck. The two sides of the tape are called 'sectors'. The auxiliary track carries track information and a search head will scan the auxiliary track during fast wind. Access will be faster than for conventional compact cassette, because the search head scans both sectors of the tape at the same time. But search will not be as fast as DAT.

The same chip set will work either for DCC or digital audio broadcasting with 2 or 2½ bits. The DCC chips will also recognise a double play mono cassette, ie a cassette in which each stereo channel contains a different mono recording. This will double the playing time to 180 minutes for re-releases of old mono recordings.

Musicassettes are made (at 64× speed) by master and slaves fitted with new heads.

My main reservation is on stereo playing time; so far only 90 min with 120 min promised from thinner tape (9 instead of 11 microns). And this is split in the middle by a gap while the tape direction reverses. Although only a fraction of a second, it is noticeable unless pre-recorded software is timed to match programme breaks with tape reverses. The gap makes DCC far less suitable for time-shifting radio than DAT.

Why didn't Philips provide a half speed option to double playing time, with 2 bit coding accuracy? This would give quality comparable to digital radio, and thus be ideal for radio recording or in-car entertainment. Wirtz says this would block off access to existing tape mechanisms, which only play at the standard 4.76 cm/s speed.

Personally, I believe this is a bad decision. Philips could easily provide a half-speed option on DCC, as with VHS video. This would give domestic users the choice of standard or long-play recording. Pre-recorded musicassettes would, of course, all be at standard speed, as with video.

As DCC will not go on sale until Spring 1992, there is still time for Philips to think again. Those with long memories will recall how Philips originally designed CD as a 14 bit system.

Philips acknowledge that if DCC tapes are copied then errors and interpolations may multiply, but they claim that none of the sub-bands is treated as empty and effectively 'gated'. Gating, and unmeasured amplitude modulation effects similar to those heard from noise reduction systems, are potential weak links in any sub-band coding system. Philips believes that in designing DCC by ear, rather than by mathematical formulae, the effects have been made inaudible.

Dolby Labs reckon they have managed to put both analogue and digital soundtracks on the same 35 mm cinema film print. If so, then this knocks the Kodak digital-only system out cold. Unless, of course, there is a technology exchange deal between the two companies.

The Dolby system mates nicely with the spread of the Nicam digital stereo TV system. In the UK there are now nearly a million TV sets and VCRs with Nicam decoders in use.

And you can be sure that last Christmas most of them were tuned to ITV for Eddie Murphy's *Beverly Hills Cop II*. Nearly 13 million people watched it on Christmas Day evening.

The pictures were surprisingly poor quality, with nasty smear on motion. And although it's a stereo film, the Nicam soundtrack sounded suspiciously like mono to me. The clue was that Eddie Murphy wasn't swearing.

The transmission originated from Thames TV in London. I phoned Thames and asked whether it had really been in stereo. I was obviously the first to raise any query and was assured that *Cop II* had indeed been in stereo because "All our films are in Nicam stereo".

I persisted, and insisted Thames check again. Sure enough the answer came back that at the last minute the station had realised that it could not broadcast Eddie Murphy's language on Christmas Day evening and in a panic had only been able to get hold of an airline version of the film. This was an NTSC (1 inch C-format) tape, which was converted to PAL for transmission.

And yes, the airline version was in mono.

So 13 million people saw a rotten quality NTSC conversion and at least ¾ million of them thought they were listening in stereo when actually the sound was mono. And not a single person (other than me) complained.

The worrying thing is that the people at Thames may now think they can get away with similar liberties in the future, on the principle that if no-one notices, why bother.

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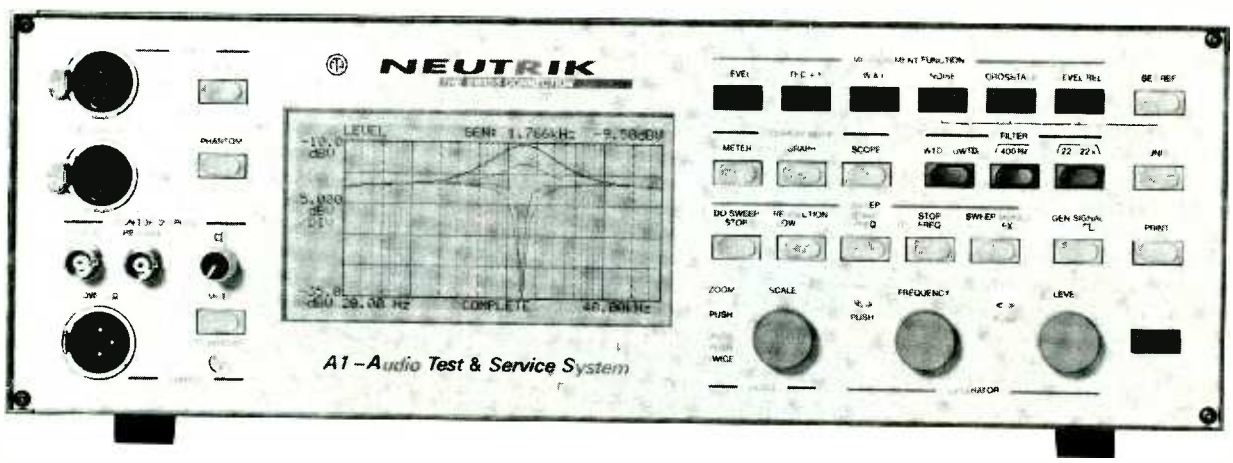

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frequently come in later than you do, acting as though they actually own the place rather than you! But you think, "Well, I actually have a good bunch working here and I'll give them that."

Your secretary marches into your office and announces that your travel agent refuses to book you into Detroit for an AES conference for under £1,500 for room and air. You knew the air fares had gone up with the petrol crisis but this was ridiculous. Have they replaced JP4 with French champagne? Not to mention nearly £100 per night to stay in Arson town. The murder capital of North America. What will the AES do next? Hold a mini-conference on battlefield audio in Baghdad, Iraq?

It now seems to be time for you and you alone to tackle the airlines. You call British Airways and discover that there is no easy and inexpensive way to get to Detroit. So you call an American airline. They tell you that you would have had to have booked last October for this February to get the cheap seats. However, at any fare level you can bring an acquaintance for free. You tell them that you don't have any acquaintances that much into pain that they would want to jet to beautiful downtown Detroit in February, for a little taste of Michigan winter and other Detroit fun. You tell the airline that if bitter cold is your thing, why Moscow is quite beautiful this time of year. They have no sense of humour and tell you that they don't fly to Moscow...please to try Aeroflot. Then finally, you discover that you can get a budget ticket to Toronto on a Canadian carrier and then fly a 'puddle-jumper' to Windsor in Canada for a song and Windsor is right across the river (and border) from Detroit. This is the right answer but it requires the airline reservationist to sort through over 200 different fare basings and, incidentally, to have you wait while they do it. Since the proposed airfare expires at 6 am Toronto time on that exact day, you have no choice but to sit on the phone and wait. In the meantime your secretary has gone to see her NHS doctor about an infected nose earring, so you have to answer the incoming phone lines as well. This entire process takes over 2 hours and leaves you feeling like a limp dish-rag. Some fun, Huh!

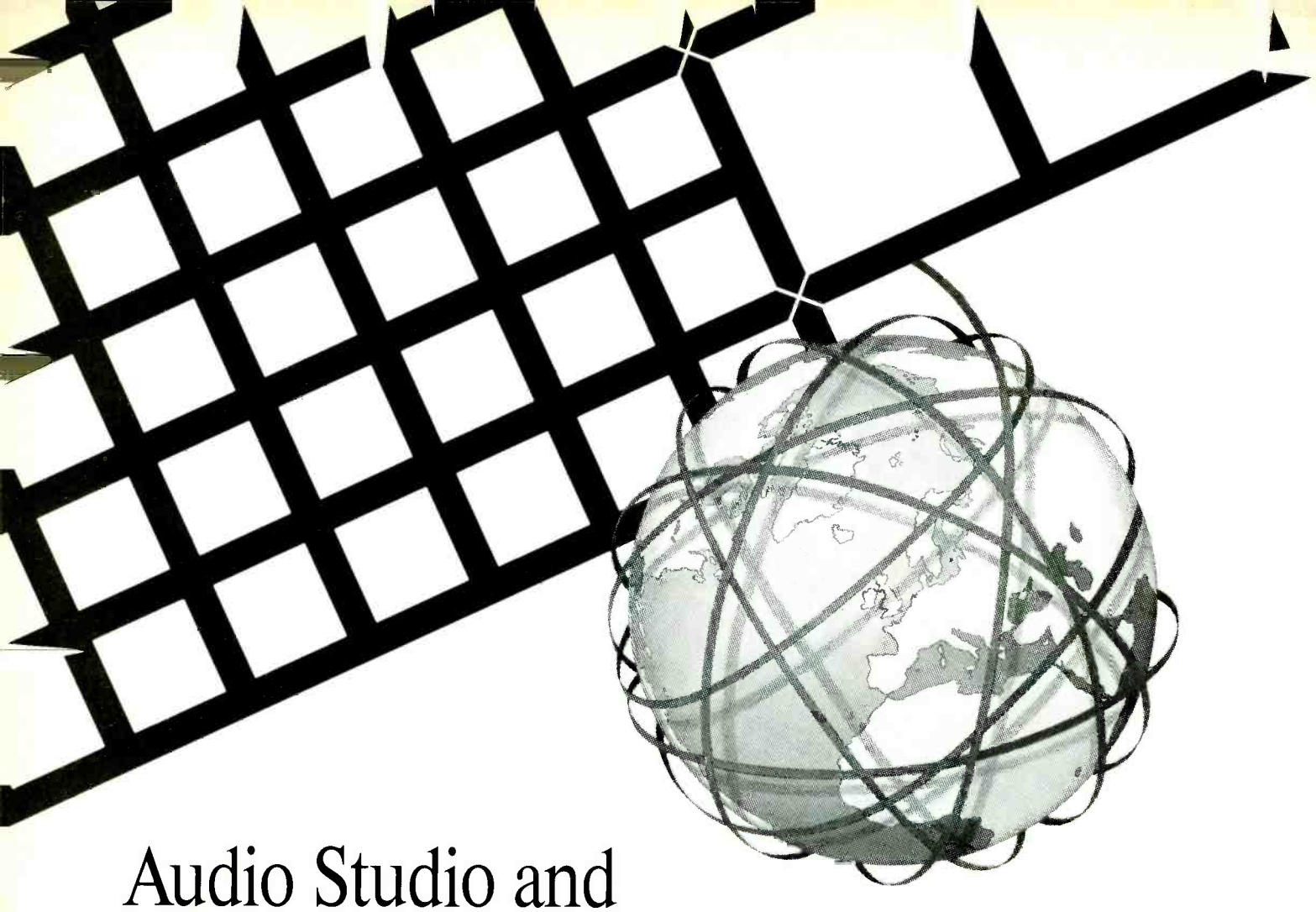
Just about the time you finish with your travel reservation excitement in walks your chief engineer, Ian. He tells you, as calmly as he can, that since the big console in Studio Three was made by a company that changed hands after bankruptcy, the commitment to service has been good but that certain parts have gone from availability. In this particular case, the frame and legs have developed deep cracks due to improper casting of the metal. Since the frame holds all the modules and since the whole mechanical set-up is now out of true many of the modules no longer make proper contact at their connectors. This is a grade one disaster, since Studio Three has been your most popular and productive facility. After much real anguish and 14 telephone calls, you find that your initial hunch is correct and the metal frame and legs can be replaced by wooden ones. In addition, the module connectors can be replaced with 'floating' cable connectors that will not depend on console mechanics to sustain connections. It will cost several thousand quid but that is a lot different from the £10,000 that Ian had been quoted for the metal parts. Who knows, if our solution is indeed successful, it might be just wizard enough to merit an article in *Studio Sound*.

After Ian leaves, Mick walks in and announces that the electronic parts supply house has placed your studio on 'cash only' status. It seems they have done a credit check on you and found that you and your studio staff have company and personal credit cards extending hundreds of thousands of pounds of credit. You call the parts house. It seems that your credit is not bad but rather that there is too much of it extended. You call the credit bureau and discover from a male airheaded twit that the bureau has refused to remove old credit cards cancelled by the bank after it upgraded all the existing cards to Gold status. You call your solicitor and explain the situation. You then sit and wait in a state of suspended animation. Finally, your solicitor calls back. He has put together a conference call of the bank officer and the solicitor for the credit bureau. All has been fixed. But at what price to your nervous system.

Now, as we leave our North London studio owner to continue his hopefully otherwise uneventful day, we must stop and take note of his daily travails. It is clear that he has had a number of stressful happenings. It is equally true that he has remained functional and has handled most of the situations well and has, in the end, resolved all of them. The goal, however, is to handle stress so that it does not overwhelm the individual. Of course, some stress-filled situations cannot be made to go away.

An American studio owner tells this harrowing story: "I had just been to an expensive stress reduction clinic. I learned all the meditation techniques to eliminate stress. I was working late one night at the studio. My complex is within a larger building. All the toilets are interconnected and flush from common cold water sources. They all drain through common sewage piping. As I was about to leave, I answered the call to nature. Upon completing my ablutions, I flushed the porcelain facility. I was then treated to a waterworks show of such spectacular nature that it was quite overwhelming. The overflowing toilet began to flood the room. Unfortunately, my tape archives were just below the toilet, on the next floor and I knew that the slab was pierced by pipe and cable holes. I grabbed a pitcher for drinking water and proceeded to bail virtually all night. I called the night security number but was told that building management had left instructions not to call plumbers before 8 am. No amount of holding my fingers together or imagining myself in Acapulco could relieve the angst of the situation. Fortunately, my tapes were dry, which is more than I could say for my shoes or my psyche."

Here are some stress reduction techniques that will help to make every day more enjoyable as



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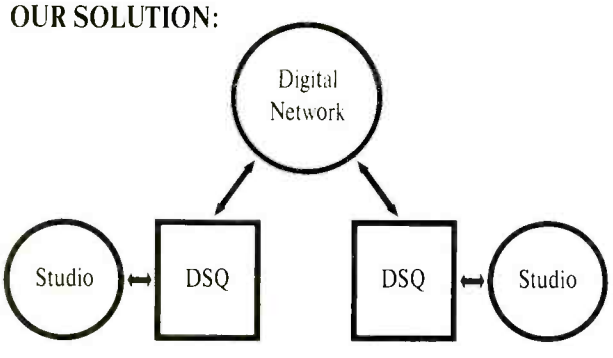

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stressful situations are made more manageable:

Learn to pick your battles: It frequently is better to ignore certain occurrences that might otherwise stress you. Dining out is a good example. If you order a meal and it is served differently than you had intended, it may be better to ignore a problem rather than create a stressful situation. Obviously, if you had ordered a hamburger steak with broccoli and you receive corn, it is not worth the aggravation to complain about the change—if you like corn! In other words, if a vendor promises you a part for a digital tape recorder via Federal Express or Data Post and it is sent by the ordinary mail, that is fine. As long as you got the part in question promptly, a negative response is not worth your time or stress. Save your 'stress' energies to focus on situations that cannot be easily resolved or ignored.

Anger really hurts you...not your intended target: It is virtually catastrophic to allow any situation to move you to anger. It is especially unfortunate if you allow that anger to build and fester. That kind of stress makes you less productive and frequently produces emotions that mask appropriate response in a business sense. In addition, the process of growing and nurturing the anger will affect the human body negatively. Anger causes negative changes in the vascular system impacting blood pressure and even changing certain of the hearing physiologies. Increased secretion of acid to the stomach occurs along with a host of other physiological changes. Remember that living well is the best revenge.

Use praise, not poison, to deal with service providers: When a second or third party makes a mistake in some transaction that involves you and/or your business, deal with that situation with kindness and humour. The person you will probably be addressing is frequently not compensated properly and is subjected to more stress during their working day than you or I might experience in a full week. It does seem important to remember that this person you might be screaming at, actually controls some portion of your daily existence via their keyboard. How much better for them to think well of you. Stable and pleasant behaviour also prevents you from creating more stress for yourself over an already vexing problem.

Minimise your reaction to negative daily occurrences: Allow yourself to absorb without rancour the progression of daily blows from such gentle and caring providers as British Rail and Amtrak, General Telephone and British Telecom, London Transport and the Boston 'T', the US Postal Service and the Royal Mail, the Inland Revenue and the Internal Revenue Service, etc. These 'Tse Tse flies' of daily life in our busy society will not go away, so learning to ignore them removes a significant amount of unecessary stress.

Fix what you can—but know what you cannot: Learn to accept that certain stressful situations will not be resolved with one phone call, or overnight, or perhaps by any of your efforts at all. What we all must do is learn to disengage from such situations, after we have taken all logical steps to deal with the problem. Once a problem has been identified, we must fix it or learn to ignore or tolerate it. A good example of this is the delivery to your desk of a good rumour that your landlord is going to sell the building that houses your studio. The rumour goes on to say that the buyer of the property will be a particularly unctuous developer who intends to displace your 'diskery' with a yuppie 'eatery'. After you have talked to your landlord, who vociferously denies any such thing...what else can you do? You can also try to get more information from local banking and real estate sources and you can begin to search for alternative space. But ultimately, having done all that, you must ignore other implications until something changes or you receive more information.

Resign from the clean desk brigade: Do not create your own internal stress by setting unrealistic goals for yourself. A frequent problem for many successful individuals is to try to 'finish' all pending business and tasks, even if the timeframe for such evolutions is not really at an end.

A mix engineer, much sought after for film work, commented: "I usually force myself to do a full day's business after I have finished a day on the console. I will read scripts, scores, sound effects cues, etc, several days or even weeks before I have to tackle that particular project. It really freaks me out that I do that, even though my goal is to 'build' a number of work-free days."

It is unfortunately true that we cannot close our books each business day on all our pending options nor can we guarantee that our days can be compartmentalised with 'work' days and 'play' days. It is much less stressful to do what we can and balance our days so we don't create unnecessary stress with unnecessary internal deadlines. Have no fear about your future sources of stress; plenty will be sent your way without making more yourself!

The point of all this is to learn to manage your own business and personal activities rather than letting them manage you. Of course, stress management techniques can be useful. Simple solutions to stress are also useful. Use your lunch time to take a walk or play handball. Exercise remains the most successful method of stress reduction for many individuals. But the bottom line remains: what works best for you? Each of us will know instinctively when stress is under control. Whether you are a fourth engineer (talk about stressful) or a studio owner, you must control some of the pressure in your life. Many people talk about how not all stress is bad; the classic phrase is, 'Only the dead are stress-free'. But like most over-simplifications, that little 'canon' ignores the fact that some stress reduction helps to make us more effective in our audio business lives as well as to help us live longer.□

Akai A-DAM

A technical report by Sam Wise on the A-DAM digital audio multitrack

Akai's *A-DAM* (Akai Digital Audio Multitrack) has been on the market for about 18 months now, the ground-breaker in 'budget' digital audio multitracks. As with most things, 'budget' is a relative term. If your price range centres on products like the Tascam *238* 8-track cassette machine, then *A-DAM* is expensive. However, 24 tracks of *A-DAM* at a retail price of around £30,000 is less costly than a moderately priced 24-track analogue recorder with noise reduction, making it an attractive item for many studios.

A basic *A-DAM* system comes in three rackmountable units: *DR1200* 12-track transport, *DM1200* 12-track metering system and *DL1200* programmable autolocator/remote control. The *DL1200* provides for synchronous operation of up to three *DR1200* units, allowing construction of 12- to 36-track recorders. The system can be expanded from an initial 12 channels at any time.

The tape format is currently unique in the industry, using a rotary head helical-scan Video-8 based recording format. These tapes have the advantage of small size and low cost, being only a little larger than a standard size compact cassette. Recording time is currently limited to 14.5 minutes but this is equivalent to the duration of an analogue multitrack running at 30 in/s, which is the practice of many mid-range studios who cannot afford Dolby *SR* and first-line studios requiring extended HF response. Analogue tape cost of £70 for a 10 inch spool of 2 inch tape compares to £18 for the required two Video-8 cassettes for a 24-track format, making running costs low.

Aside from the performance advantages of 16 bit recording, the owner of 24 tracks of *A-DAM* gains a further benefit over any other current format. Few analogue studios and even fewer digital studios have the capability to make a back-up copy of the multitrack's master tape, and of course an analogue copy will suffer some degradation. The *DR1200* 12-track *A-DAM* transport units include the capability for direct digital transfer, making back-up master production trivial and without quality loss.

Manufacturer's specifications

Recording format: proprietary *A-DAM*, 16 bit linear, double Reed-Solomon Code error correction
Digital audio channels: 12
Sampling frequencies: 44.1 kHz, 48 kHz
Recording time: approximately 15 min (depending on sampling rate and national video format)
Tape varispeed: ±6%, 0.1% steps (only at 44.1 kHz)
Search time: 90 s or less
Allowable tapes: 8 mm video cassettes with 13 μm thickness (P6-90 in NTSC regions, P5-60 in PAL/SECAM regions)
Heads: three each digital record and playback; one each aux/control analogue record/playback; one each full-track erase; one each aux erase
Wow and flutter: below measurable limit

Digital audio channels

Frequency/response: 20 Hz to 20 kHz, +1.0 dB/−1.2 dB (at 44.1 kHz sampling)
S/N ratio: 90 dB or greater (with emphasis on)
Distortion: 0.05% or less THD (standard level=full scale −15 dB)
Crosstalk: 80 dB or better (1 kHz, 44.1 kHz sampling)
Crossfade time: at 44.1 kHz: 11.6, 23.2, 46.4 or 92.8 ms; at 48 kHz: 10.6, 21.3, 42.6, 85.2 ms
Analogue input/output: nominal +4 dBm, 600 Ω input produces −15 dB meter display
Maximum input/output level: +19 dBm @ 600 Ω is 0 dB
Input impedance: 10 kΩ, balanced
Output impedance: 10 Ω, balanced
Meter: 15-segment peak reading with 1277 or 319 ms release time

Analogue audio channel

Frequency response: 100 Hz to 14 kHz, ± 3 dB (aux channel)
Audio input level: nominal −10 dBV produces 0 dB meter display
Maximum input: −2 dBV produces +8 dB/OVER meter display
Input impedance: 50 kΩ
Output impedance: 100 Ω
Meter: 12-segment peak reading with 1,277 or 319 ms release time

Dimensions

DR1200 (transport): 310×425×530 mm (whd) basic unit; 332×483×630 mm (whd) including handles and connectors. The unit is 7U rackmount
Weight: 36 kg/79 lb
Power consumpton: 300 W/12 channels, powers *DL1200* and *DM1200*
DL1200 (control/locator): 310×425×70 mm (whd) basic unit; 310×483×80 mm overall with rack ears. The unit is 7U rackmount but requires about 1U more to clear the connectors
Weight: 9.7 kg/22 lb
DM1200 (meter): 88×425×150 mm (whd) basic unit; 96×483×150 mm (whd) with connectors
Weight: 5 kg/11 lb

DR1200 Record transport unit

The *A-DAM* system looks professional, making a statement that Akai intends the *A-DAM* to be considered a serious studio product. The *DR1200* recorder transports are large and heavy, requiring 7U of rack space at a depth of over 600 mm to clear the rear panel mounted connectors. The *DR1200* should not be housed in the simpler studio racking systems but will require a robust 800 mm deep housing.

The top of the *DR1200* contains an access panel for head cleaning (more about this later). This must be readily accessible for daily use but is hidden when the unit is rackmounted. Therefore, the *DR1200* must be mounted on heavy duty slides in a rack that will not tip forward when the transport is pulled forward.

Front panel

Large handles assist with pulling the unit forward in the rack and protect the front panel from damage, while an additional bezel prevents accidental operation of the power switch. A smoked perspex cover with cutouts forming protective recesses for the five transport control pushbuttons also covers various LED indicators. The top open/close button is self-explanatory. TAPE VIEW turns on a light inside the unit,

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illuminating the tape cassette so it can be seen reflected in an internal mirror. The remaining buttons select the sync mode. Without going into detail, these allow *DR1200*s to be mutually synchronised, controlled by one or more *DL1200* control units and/or to be SMPTE timecode locked as master or slaves to an external system. The sync mode switches must be set prior to power-up since they are only read by the internal microprocessors at that time.

The tape tray close button causes the tape to load to the rotary head mechanism, allowing quick starts during a session. The tape remains loaded during wind functions allowing for fast and accurate synchroniser chase control. If the transport is not used for a period of time, the tape is automatically unloaded to protect it from excessive wear due to rotary head action.

The rotary head carries three record and three playback heads, providing off-tape monitoring. There is an additional fixed head providing analogue record/playback, principally to allow timecode reading while spooling.

Tape reliability

In addition to the few operating controls, the front panel includes various helpful indicators. These show sampling frequency and emphasis setting, sync mode, tape load and run status, and warnings about tape quality, unformatted tape and record protection. If a tape shows a serious error level, recording will cease and the tape will be ejected from the drive. During tests and a demo period in a local studio, this happened several times with a particular brand of tape. Replacing this with Maxell or TDK tapes produced no further errors. Checking with a long term user, revealed similar experience: no failures during a year of operation once a suitable tape type was found. Akai UK recommend and sell Maxell tape while insisting that Sony tape should not be used. This criticism of Sony tape has no bearing on its performance as intended—for video recording—where it is highly regarded, but as digital audio tape it is currently not to be recommended.

The track encoding method used on the *DR1200* is optimised to reduce the risk of irrecoverable drop-outs due to tape damage. It also allows direct track-to-track copying, timed punch-in, and off-tape monitoring, which are essential to practical studio use. It will not tolerate the razor blade, however, unlike more expensive formats.

All *DR1200* functions are controlled by the *DL1200* and will be discussed in that section of the review.

Rear panel

For table-top use, the rear of the *DR1200* is protected by four large feet, lifting connectors clear of obstructions. The upper part of the rear panel contains two large fans. These have thermostatic speed control, and in our application did not cause irritating noise. Transport reliability makes it possible to locate the recorder remotely if lower noise levels are required.

In the panel centre are high quality gold plated D-type connectors which are used for all system intercommunication, link-up to an external synchroniser, and digital audio connections between *DR1200* units allowing 12-channel direct digital copying.

A further D-type labelled ERROR RATE is used during servicing to ensure correct performance. The SMPTE code track input and output are, disappointingly, on 6.35 mm jack sockets and operate at domestic levels; while all 12 audio inputs and outputs are *XLR*s set up for professional operating levels. These are balanced and wired pin 3 hot, contravening the AES standard. This will usually make no difference in practice in a balanced system.

Internal construction

Opening the top access plate reveals the tape transport mechanism and various internal connections and printed circuit boards. A further internal cover protects the head, guide, load and tape drive assemblies. This is held in place by a simple latch, which is clearly designed for routine use. The manual advises that the heads are cleaned daily and before any tapes are formatted.

Also visible here is an elapsed time indicator, which looks like a 1¼ inch fuse with a 0 to 10 mercury thermometer inside it. Adjacent to this, 0 to 3,000 hours is printed on the PCB, corresponding to the 0 to 10 'thermometer' reading. Although not mentioned in the manual except in passing, this is presumably Akai's estimate of head life, or at least allows them to access what is really happening when units come in for service.

The internal mechanism, while being a precision piece of engineering, is robustly built, including a die-cast aluminium chassis for strength and stability.

An attempt to examine the internals of the *DR1200* to assess serviceability led to the conclusion that, while things are relatively easy to get at, component level repairs and head assembly replacement may be best left to the Akai service agent. This is indeed what the manufacturer's recommend.

DM1200 Meter unit

A brief introduction to the *DM1200* 12-channel meter unit is in order next. It is a conventionally sized 2U high rackmounting unit, which is shallow enough to mount in a purpose-built remote trolley, or even within a section of larger mixing consoles.

Most connections are digital. A single multicore cable from the associated *DR1200*s carries level information from all 12 digital channels on up to three machines, the aux channel and power for the metering system.

An LED ladder array is used for each meter. The digital channels have 15 segments ranging in level from −55 to OVER (peak level). All LEDs are green except OVER, which is red. Located beneath each channel meter is a red LED indicating Record Ready (flashing) or On Record (steady). The analogue channel has a 12-segment meter ranging from −20 to +8, where +8 is peak recording level.

Digital machine overload characteristics are anything but subtle. But, to my knowledge, there has not yet been any conclusive published research into the audibility of transient digital overloads, that would help define the required meter characteristics. European and BBC type ppms were, of course, specified considering the audibility of overload in analogue recording systems. These currently specified characteristics may or may not define meter ballistics suitable for digital recording.

Meanwhile, every digital machine seems to have different meter characteristics and the *A-DAM* is no exception. The meters are sensibly peak reading, but without a specified attack time. As digitally driven LED meters they can be, and are, more or less instantly responding. To assist the user in observing this instant response, the decay time is switchable between 1.3 and 0.3 secs. In addition, peak hold can be enabled in either AUTO mode, with a 1.5 sec display time, or CONTINUOUS mode, which retains the highest peak continuously—useful for level setting.

A nice touch here is that because the meters display the final digital signal, they show the effects of emphasis. This results in higher readings than the console meters for programme material with large high frequency content.

One *DM1200* can be used to provide metering for up to three *DR1200*s since it includes a switch to select the required machine. But, at the price, unless space is cramped, most purchasers will want a *DM1200* for each *DR1200* they buy.

Machine control

This seems the obvious place to describe the *DL1200* machine control/autolocator unit. It will be impossible to go into detail here but the ease of use of this machine certainly impressed me and the staff at the studio where we did our user testing. We had no trouble learning to use the *DL1200* operating modes to good effect. Panel layout is sensible, separating buttons and indicators into functionally related groups. All the usual manual machine control, autolocate and synchroniser functions are available.

During the recording process the record start time (IN POINT) and record finish time (OUT POINT) are automatically stored in the machine's memory. They are then available for repeated use, without any special preparation. The *DL1200* has 105 tape locator memories, five of which are available on a single button press. AUTO PUNCH allows the machine to enter record automatically at a selected IN POINT, and return to play mode at selected OUT POINT. REHEARSAL mode allows trial punch-ins/-outs to be made without actually recording anything, allowing an engineer to check timings and an inexperienced performer to experience overdubbing without risk. TRANS PUNCH makes it possible to digitally copy one track to any number of others while punching-in and out automatically, ie, you can record several versions of a section, then edit the best bits down to one track digitally. This can then be inserted within a previously recorded take. Everything can be done in practice before taking the plunge.

PLAYBACK MUTE function mutes selected tracks automatically on punch-in and unmutes on punch-out, removing the need to mute the console monitor manually when replacing voices or instruments.

On punch-in and punch-out, crossfade is automatic and is performed by digitally mixing the original and new versions of the sound over a selected time interval. This can be set to 11.6, 23.2, 46.4 or 92.6 ms.

CHANNEL SELECT buttons allow channels to be placed in record ready condition, prior to record or punch-in. Indicators above these buttons indicate RECORD READY (flashing red), on RECORD (steady red) or PLAYBACK (green), along with a yellow DELAY LED, which warns that a digital delay is in use on that channel. Only 12 buttons and indicators are provided, with a machine select switch allowing their use with up to three *DR1200* units.

The memories of the *DL1200* hold not only locate position data but also information on four complete selections of CHANNEL SELECT status. This helps to overcome the disadvantage of sharing the 12 RECORD MODE selectors across several machines. An ALL CH PB button does allow all 36 channels to be put into playback at once.

At the end of a session, all the above data, and more, can be automatically recorded into a Table-of-Contents (TOC) area of the tape. Then when the session restarts at a later date, all channel selections, in/out points and locator positions are readily available. Other information written in the TOC is the tape length, pre-roll time, post-roll time, channel delay times and crossfade time.

Formatting tape

Before use in recording, the Video-8 cassettes must be formatted. At this time the sampling frequency and emphasis setting (on/off) must be decided, since this information is written into the TOC area of every tape, and can affect tape speed.

In a system with multiple *DR1200* units controlled by a single *DL1200*, tape can only be formatted in the master transport, however, SMPTE code striping can be done at the same time. All tapes must be striped if external SMPTE sync will be required, since in this mode the slave *DR1200* transports are apparently locked to SMPTE code rather than the internally generated digital clock.

Synchronisation of multiple DR1200s

As mentioned above, it is possible to run up to three *DR1200* units directly in sync, providing, in effect, a 36-channel recorder. The sync of these units is digital and to an accuracy much better than can be achieved with SMPTE code locking of multitracks. Indeed, as the measurements given later show, the phase accuracy between tracks can be better than that across a single multitrack head assembly.

SMPTE code can, however, be used to lock the set of *DR1200* units to a video machine, or any other timecode controllable unit, either as master or slave. By this means *A-DAM* systems can be joined to increase the 36 tracks directly available.

If the *A-DAM* system is the master in a timecode system, varispeed on the *A-DAM* can be used to give up to ±6% speed variation in 0.1% steps when recording at 44.1 kHz only.

Using the digital recording system to advantage, the *A-DAM* allows each track to have an offset delay adjustable in 0.1 ms steps. The range is up to 60 ms for 48 kHz sampling and 66 ms for 44.1 kHz sampling.

Measured operation

Compared to a typical analogue recorder, what does one say? Generally speaking digital machines are a bit boring to measure. This one is no exception, while some interesting features did show up.

The input impedance and output impedances of the digital channels are confirmed at about 10 kΩ and 100 Ω respectively. The meter −15 dB point corresponds to an input level of +4 dBu across all channels, with a level variation at 1 kHz of only ±0.1 dB. As with most digital systems, the peak level is equal to the clipping point. This occurs at meter 0 dB, corresponding to an input level of +19 dBu. This is a sensible figure that will extract the best overall signal to noise ratio from virtually all associated equipment. Common mode rejection was good as shown in **Fig 1**, with corresponding performance from other channels.

The input of the analogue aux channel is specified to operate at a nominal level of −10 dBV, or about 12 dB below the digital channels. However, as a SMPTE code track, this was operated at nominal professional line levels with no ill effects. This is confirmed by the graph shown in **Fig 2**, which indicates that frequency response does not vary as level increases even though there is some level compression. The aux channel is unbalanced and input/output impedances conform to specification.

Digital channel frequency response is well within specification as shown in **Fig 3**, with a total measured variation of −0.8 dB/+0.6 dB relative to 1 kHz. This was consistent across all

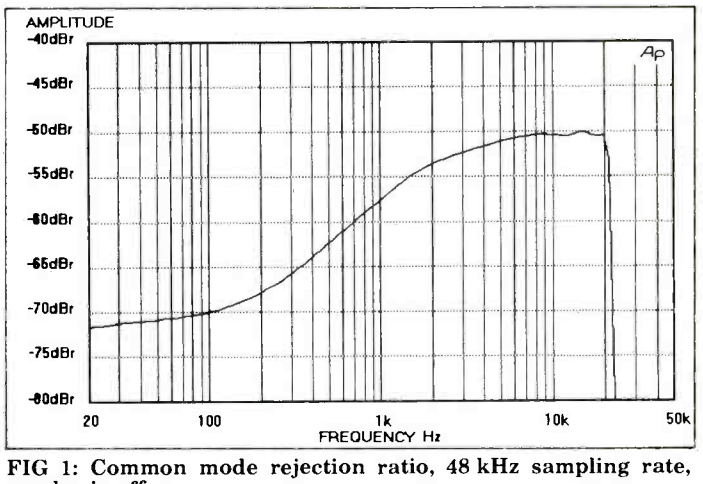

FIG 1: Common mode rejection ratio, 48 kHz sampling rate, emphasis off

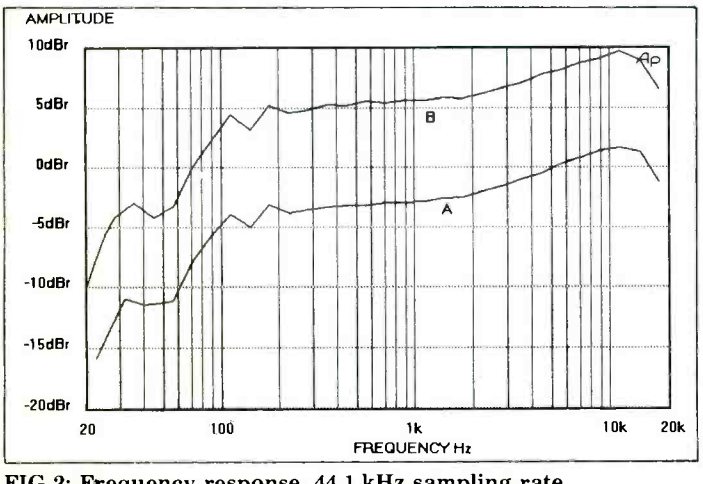

FIG 2: Frequency response, 44.1 kHz sampling rate A=0 dBu (−2 dBV) input level; B=+10 dBu (+8 dBV) input level

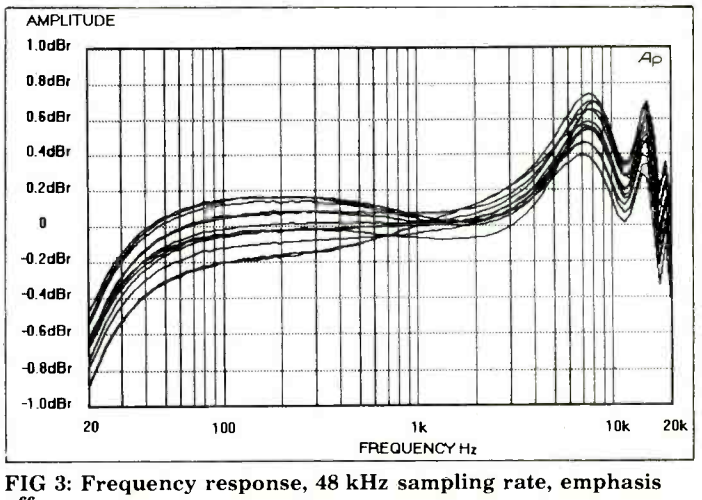

FIG 3: Frequency response, 48 kHz sampling rate, emphasis off

channels and speeds with little difference with emphasis on or off. Out-of-band response was brick-wall-like as shown in **Fig 4**, giving little opportunity for aliasing problems. **Fig 5** shows the squarewave response, which is predictably spring-like, having an overshoot of about 20% and a fairly slow decay rate. The frequency of oscillation is about 20 kHz, making it largely inaudible (except to bats), though the golden-eared may hear the onset and certain loudspeakers might produce audible subharmonics.

Phase response is also as expected, producing the usual group delay above 16 kHz (ie high frequencies are a bit late) due to the sharp filter cut-off. This is shown in **Fig 6**. Phase difference between channels is superb, much better than analogue machines, with differences produced mainly by component tolerances. **Fig 7** shows a worst case example between tracks on one machine, with virtually no phase difference between 100 Hz and 10 kHz. This figure also shows matching between channels on two internally synchronised *DR1200* machines, which in this case is better than that on one machine alone

The aux channel performed as specified, with a 2 dB boost at 14 kHz and 2.5 dB cut at 100 Hz when driven at −10 dBV. At specified peak level of −2 dBV, the LF response deteriorated to that shown in Fig 2 (lower curve), while a further 10 dB of input resulted in essentially the same frequency response while compressing in level across the band (upper curve).

Wideband noise is more or less as specified, the more or less referring to an imprecise specification of 90 dB below clipping. **Fig 8** shows the effects of emphasis, giving substantially better performance above 2 kHz but reduced HF recording headroom.

Various forms of distortion were measured. These are generally noise limited up to near the peak recording level. THD+N measures 0.03% at the specified level of −15 dB referenced to peak level, which is less than the specified 0.05% and much less than an analogue machine. CCIR tests using a pair of high frequency tones show the effect of emphasis, with rapidly increasing distortion for signals higher than −10 dB reference to peak level. This does, however, show up on the meter as an overload condition and is not a machine deficiency. It is up to the user to decide to use emphasis or not depending on expected programme content.

Modulation noise of one of the better channels is shown in **Fig 9**, this being typical of other digital recorders measured. **Fig 10**, however, shows a relatively poor performance, with noise modulation adding 5 dB of noise in some circumstances. The cause of this can be seen in **Fig** 11, which shows the converter linearity errors. For all channels in the machines tested, these increase below −80 dBr, but on one channel the errors rapidly reached 8 dB. The central curve of the three is typical of 16 bit converter systems measured to date, while those on the *A-DAM* were generally not as good as those seen on 2-channel DAT machines. This is the one area where the machine seems to require attention. But, can you hear this? In some instances, in quiet conditions, the modulation noise may become audible, but in most multitrack applications, this will probably be inaudible.

Accessory items

None of the following items has been tested. In order to allow digital transfers, Akai provide the

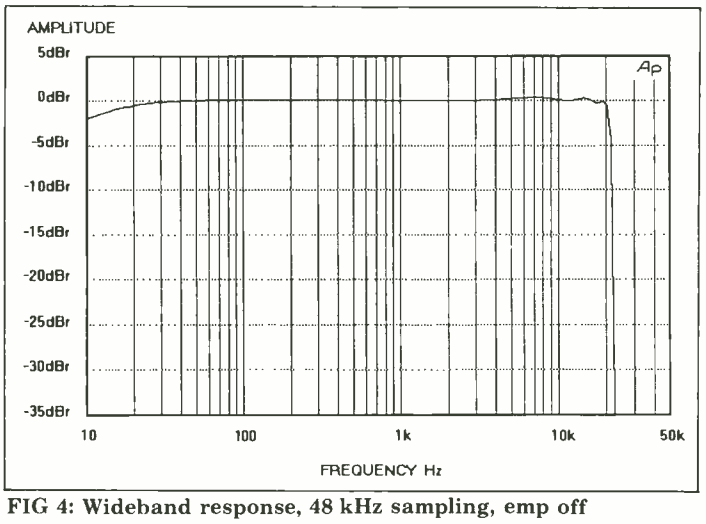

FIG 4: Wideband response, 48 kHz sampling, emp off

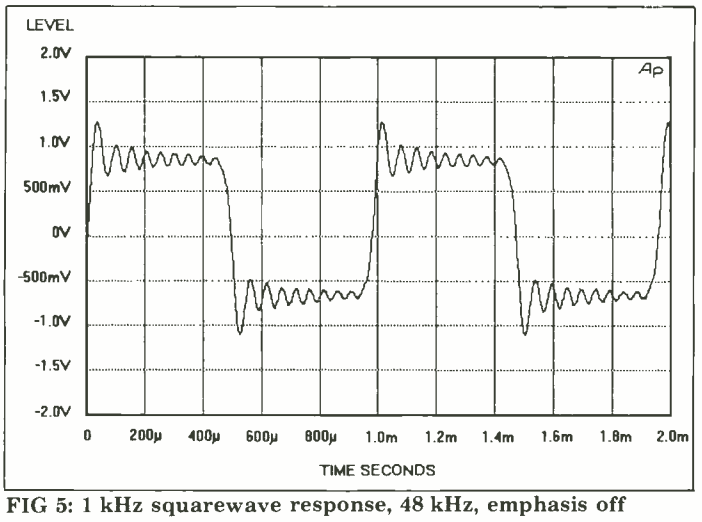

FIG 5: 1 kHz squarewave response, 48 kHz, emphasis off

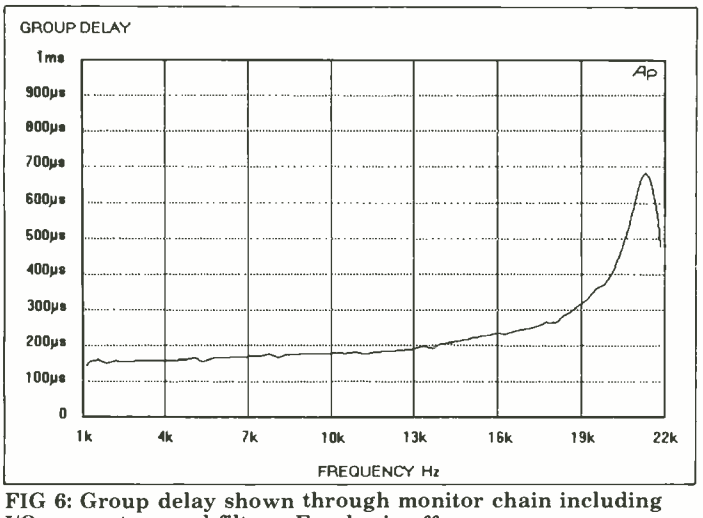

FIG 6: Group delay shown through monitor chain including I/O converters and filters. Emphasis off

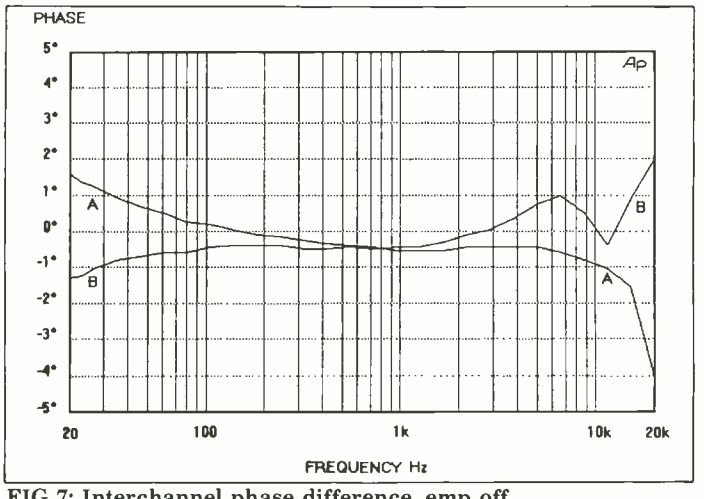

FIG 7: Interchannel phase difference, emp off
A=ch 1 to 8, machine 1, B=ch 1 machine 1 to ch 1 machine 2
NB: intermachine error is less than between tracks on one machine

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DIF1200. This is a 1U rackmounting unit having a digital connection to the *DR1200*, and AES/EBU and SPDIF connectors. It allows transfers to be made from a 2-track digital system onto any selected channels of the connected *DR1200*. Akai also market the Klotz digital *AFC112*, which is designed to transfer 12 channels at a time to or from either Sony or Mitsubishi digital multitracks. This is supplied with both types of connecting leads within the package price. Rumour has it that various means are under development to extend tape record time with no current user compatibility side effects.

Comment, Akai Digital UK: Readers may be interested to know a few additional points regarding the tape situation. Current running time is between 14.5 mins and 17 mins depending on the tape and the selected recording frequency. Akai will introduce new tape in June that offers an extended recording time in excess of 20 mins. We normally specify that 60 min tapes are used (P5 or P6); the *DR1200* will automatically reject any thinner type. With current software any errors will be memorised in the stick memory while the *DR1200* continues recording. We have no knowledge of any user experiencing a 'serious', ie audible, error. The machine automatically corrects all minor errors caused by dust or tape drop out.
NB The *DR1200* **will** accept NTSC P6-90s giving an extra 1 minute playing time with existing software. These tapes are available in the UK through Akai. The PAL P6-90 will, however, be rejected by the machine.

User comments

A phone call to producer Mike Hedges, who has used an *A-DAM* for 12 months, brought the comment "for the price it is stunningly good". He commented on the ease of use and flexibility of the *DL1200* machine control unit. Using an Adams-Smith synchroniser and setting up the *A-DAM* to slave, he is also impressed by the follow and lock-up speed. He says it is just "there". Also, long term users have commented on the inaudibility of punch-in/-out on this machine when used for rock music. (No classical users have been questioned.)

The users I spoke to found no audible defects in the converter systems, even though measurements found them less than the best. One user with access to a more expensive digital machine could hear no difference between them, although his comment is not statistically valid. I presently have no measurements from other digital multitracks for comparison.

Akai UK tell us that 60 machines have been sold to date. Once users learn to keep them clean, off the floor and to use recommended tapes, he says that problems are few. Apparently quite a number of these are being used in mobiles, or carried from studio to studio with no ill effects.

Conclusion

The Akai *A-DAM* is a high quality product by a manufacturer with a stable history and good technical support. It is easy to operate, while incorporating many useful features. Aside from installing better converters, or setting them up more carefully, there is little to suggest in the way of improvements. It would be nice though to have some compatible products available from other sources.□

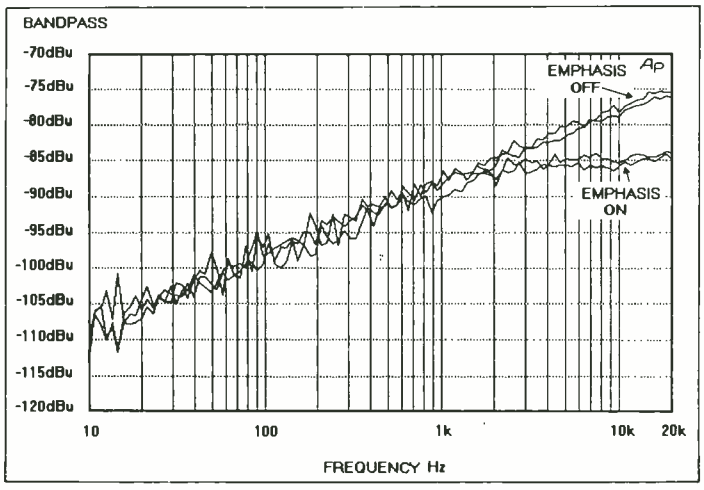


FIG 8: ⅓-octave noise spectrum 44.1 and 48 kHz sampling

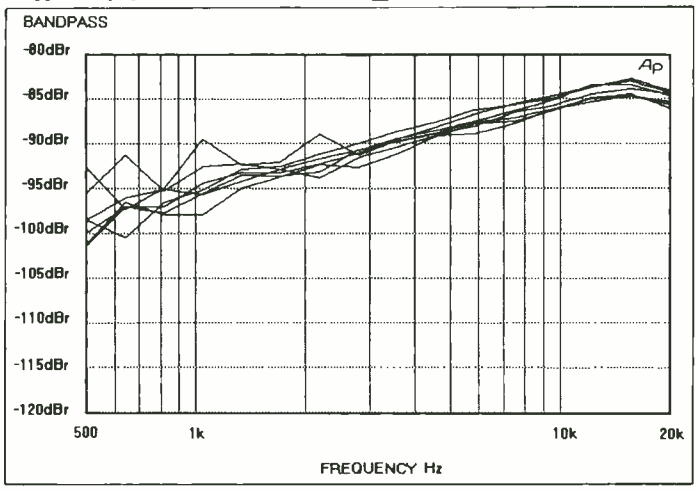


FIG 9: Modulation noise, machine 1, ch 1 48 kHz, emp off Modulation noise is difference between upper and lower curves

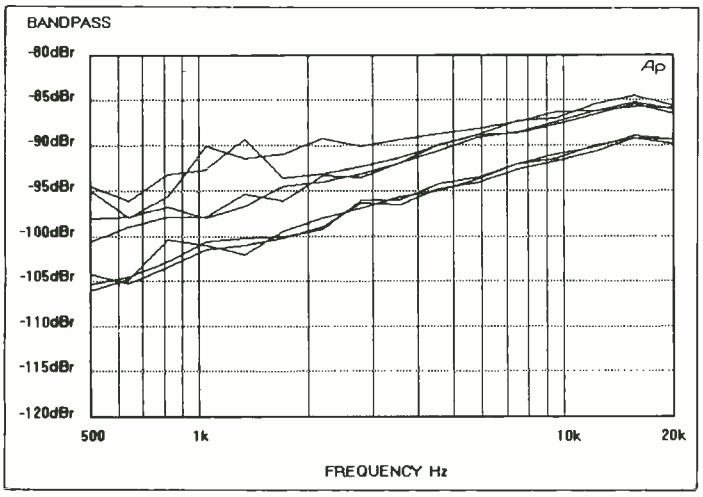


FIG 10: Modulation noise, machine 2, ch 8 48 kHz, emp off Modulation noise is difference between upper and lower curves

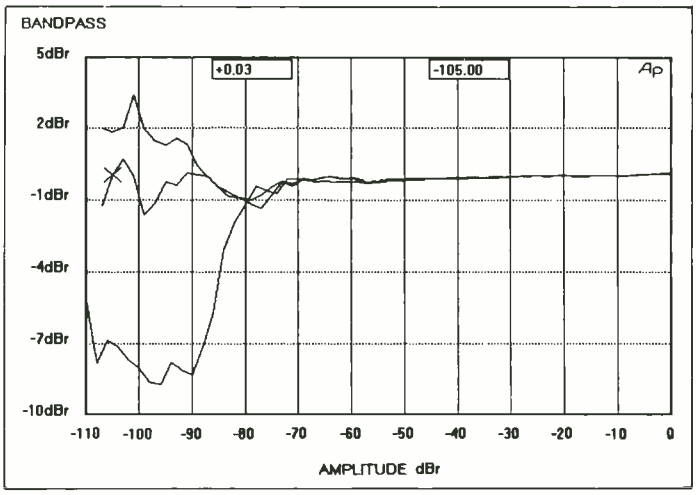
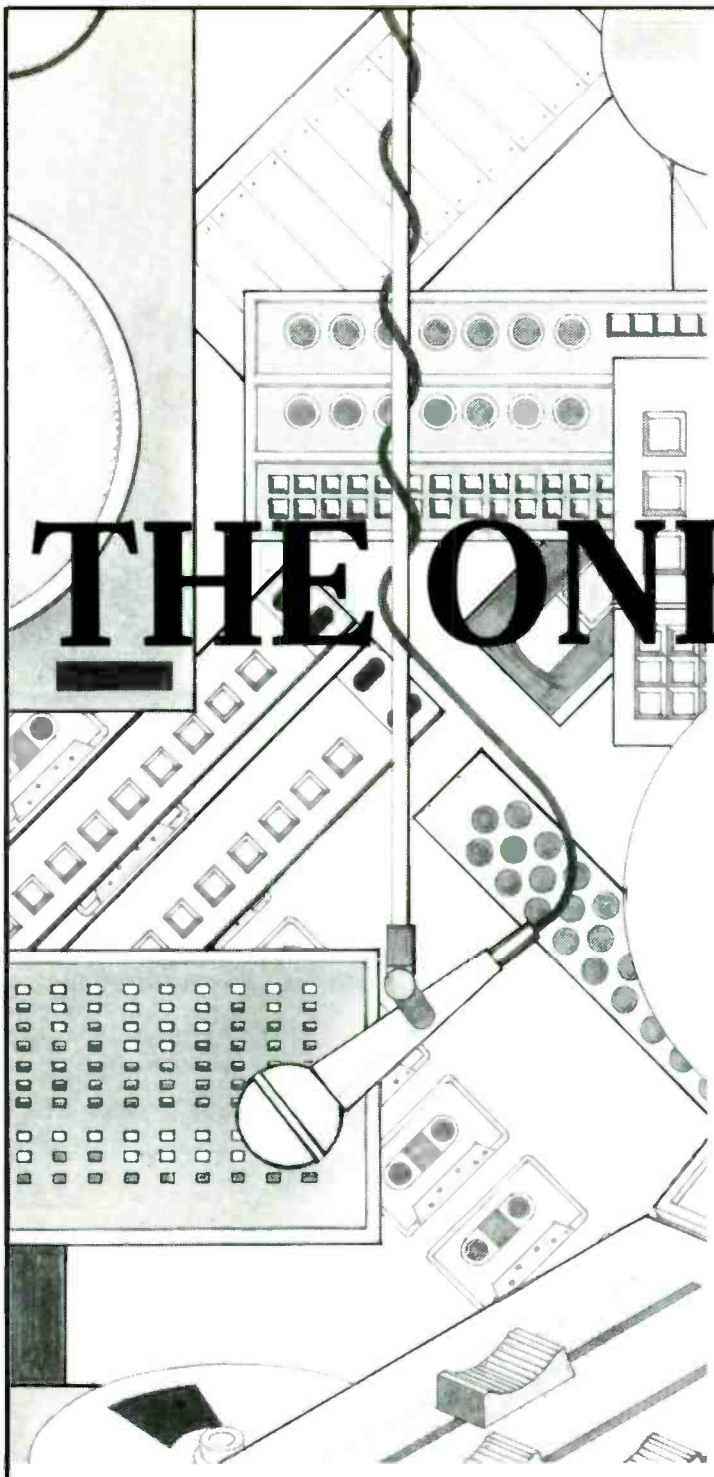


FIG 11: Input/output linearity best and worst tracks from machine 1, 48 kHz sampling rate emphasis off



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