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# STUDIO SOUND

AND BROADCAST ENGINEERING


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## SIGNAL PROCESSING DYNAMICS

CONTENTS

# STUDIO SOUND
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STUDIO SOUND
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# Cassette master quality

In recent months, this column and indeed the whole industry has been preoccupied with economic matters, and quite understandably so. There has been very little of any technical consequence to break through the aims of simply remaining in business—from month to month or even week to week.

Our business ultimately exists by the sale of the end product—music software or programming for film or broadcasting. The success of sales of either determines much more. Despite what we sometimes say about record companies, their competence and wisdom, they are charged with indirectly supporting large chunks of our business. With this in mind, I was rather disturbed to hear a report on BBC radio that the International Federation of the Phonographic Industry (IFPI) were claiming that 25% of all recorded music sold was counterfeit including the compact disc and cassette formats. The obvious question regarding any claim like this is—how do they know? But even if they are out by several hundred percent it is still a very large amount of sales lost to the legitimate industry and revenue that is not available for re-investment within our business.

The cassette is obviously the most vulnerable to piracy. It is cheap, easy to duplicate and widely accepted as a format throughout the world. In the less developed areas of the world, the cassette is the main music medium and it is in these areas that the degree of piracy has frequently prevented any legitimate record industry from developing. Such a matter should be of considerable concern to all of us.

The same radio report also stated that in the UK about 1% of product was counterfeit and this achieved by a heavy programme of investigation and co-operation with local authorities and presumably at some considerable cost. In other developed countries the situation is not quite so good. One move that would surely be sensible would be to increase the perceived or real value of the legitimate product. To this end, a number of record companies have installed digitally based duplication systems where the master is loaded into a digital store bypassing the analogue master. This has proved to be a significant advance in cassette audio quality. An article in *Studio Sound* as long ago as November 1983 identified the duplication master as a weak link in the signal chain. But life is not quite that simple.

Returning from a recent ITA conference in the US, Carl Snape, editor of *One to One*, our sister duplication magazine, told me that a hot topic among duplicators with digital equipment was that they were being supplied with unsuitable masters. These digital systems maintain the dynamic range and frequency response of the master but because of the tape used for the duplication itself, engineers are left with the choice of reducing the record level (not acceptable), compressing the signal or rolling off the HF energy. They quite rightly feel that these are artistic decisions and not something that should be left to them; rather that the supplied masters should incorporate the necessary changes for the cassette medium. They have an excellent chance of being right.

I subsequently spoke to a few UK mastering houses to see the situation from the other side. I asked how frequently they were required to produce cassette masters. Their answer was that with a few exceptions, this is something that they do very rarely, normally just providing a second copy of the CD master (although maybe title re-ordered). The record companies have little interest in paying the extra cost for a proper cassette master.

If you consider the cassette by far the major music medium it would seem negligent of the record companies not to provide a correct master for their prime market. Returning to my opening points and the subject of counterfeit cassettes, are the record companies not missing a major weapon in combating piracy? Maximising the quality and value of the legitimate cassettes surely is a major step towards this aim. And surely the creation of a separate master would seem to be a legitimate charge for the world's prime music medium.

**Keith Spencer-Allen**

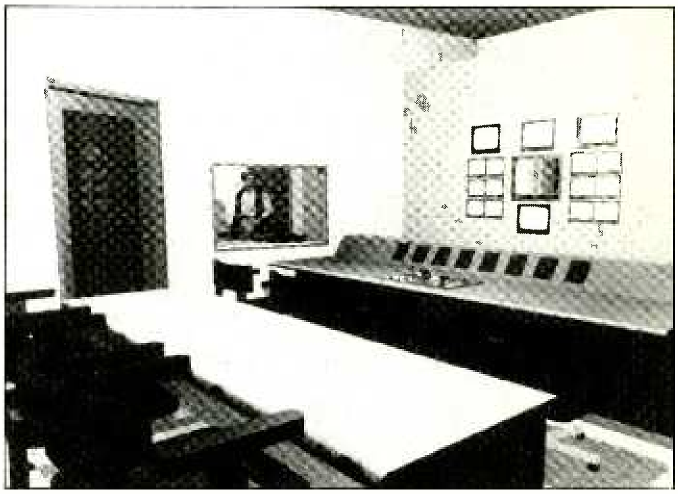
**Cover: Digital dynamics control in broadcasting—digital Optimod**

# Revox form UK company

Revox of Switzerland have decided to form their own UK company apart from their UK distributor FWO Bauch.

Bauch will still supply Revox broadcast products but will not be involved with the distribution of the hi-fi side of the company.

It was sales to the domestic market that led to the move. Brian Whittaker of Bauch comments, "Revox are putting an increasing emphasis on the high-end hi-fi market and they wanted to have more centralised distribution in that area. Bauch people will still be involved with Revox."



**CAD recording by Russ Berger Design Group of one of three control rooms at JC Penney's corporate US headquarters. This is part of the design process of BDG enabling a client to 'see' the finished appearance of a project prior to construction.**

# Shure open European office

Shure Brothers, USA, have established a wholly-owned subsidiary to achieve a stronger presence in Europe. Shure GmbH will be based in Heilbronn, Germany, and will be under the direction of managing director Berthold Burkhardtsmaier.

The office will serve the entire European Community as well as growing markets in Eastern Europe. Shure GmbH hope to play a vital role in providing key teleconferencing products and services for the 1992 single European market. Shure GmbH, Lohtorstr 24, 7100 Heilbronn, Germany. Tel: 07131 83221. Fax: 07131 627229.

# Agencies

• **Audio Kinetics** have appointed Paris-based Distribution de Systemes Professionnels (DSP) as French agent for the full range of Audio Kinetics synchroniser products and component automation systems. Distribution de Systemes Professionnels, 12 Avenue du Maine, F75015 Paris, France. Tel: 1.45.44.13.16. Fax: 1.45.44.36.34.

• Soundmaster have announced their North American distributorship for the UK-based **Raindirk** audio mixing console, and for French-based **Optifile** console automation. Soundmaster International Inc, 900 Hampshire Road, Westlake Village, CA 91361, USA. Tel: (805) 494-4545.

• **DynaudioAcoustics** have now appointed five more main dealers to handle their complete range of monitors. Gothenburg-based Tonkraft AB will handle sales in Sweden; New Musik, in Aarhus, will be responsible for sales in Denmark; and Ideal Systems, Dublin, will represent the range in Ireland. Hilton SARL will handle sales in France and Stirling Audio will handle demos in the UK.

• The Belgian signal processor manufacturer **Apex** have appointed Britannia Row Sales as sole UK distributor for their graphic equaliser range. Britannia Row Sales, 35 Britannia Row, London N1 8QH.

• **AMS** have announced three new distribution agreements in Europe and the Far East.

Malaysia: Transtel Technology, 23-B Jalan 25/24, Taman Mayang, 47301 Petalung Jaya, Selangor, Malaysia. Tel: 60 3 7031 336. Fax: 60 3 7033 048.

Singapore: Team 108, 41 Genting Lane, Singapore 1334. Tel: 65 74 89 333. Fax: 65 74 77 273.

Denmark: SoundWare A/S, Engoften 10, DK 8260 Viby J, Denmark. Tel: 45 86 88 89. Fax: 45 86 14 86 11.

• **Digital Audio Research** have appointed VIDI Video Digital as distributors of their *SoundStation* in Germany. VIDI Video Digital. Tel: 61.51.86033. Fax: 61.51.86032.

• **Pro-Bel** have appointed Video and Broadcasting Consultancy as their Scottish agents. Video and Broadcasting Consultancy Ltd, 214 Leith Walk, Edinburgh EH6 5EH, UK. Tel: 031-553 6029. Fax: 031-554 9272.

• Shuttlesound have added **Rane** to their list of pro-audio brand names. Shuttlesound, 4 The Willows Centre, Willows Lane, Mitcham, Surrey CR4 4NX, UK. Tel: 081-640 9600. Fax: 081-640 0106.

• FWO Bauch have announced their appointment as UK distributors for the *408 OMX* optical multitrack recorder/editor, manufactured by **Augan Instruments** in The Netherlands. FWO Bauch, 49 Theobald Street, Borehamwood, Herts WD6 4RZ, UK. Tel: 081-953 0091. Fax: 081-207 5970.

# Address changes

• **London Sound Centre** have moved to 24 Cadiz Street, London SE17, UK. Tel: 071-252 5800. Fax: 071-252 5811.

• **White Instruments** have moved to 1514 Ed Bluestein Boulevard, Austin, TX 78721, USA. Tel: (512) 389-3800. Fax: (512) 389-1515.

# Exhibitions & conferences

**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** PLASA Light & Sound Show, Olympia 2, London, UK.
**October 4th to 7th** AES New York, Hilton Hotel and Sheraton Centre, New York, USA.
**October 16th and 17th** The Playback Show '91, RDS Industries Hall, Dublin, Eire.
**October 17th to 21st** Mediatech 91, Milan Fiera, Lacchiarella, Italy.

1992

**March 24th to 27th** AES Convention, Vienna, Austria.
**October 2nd to 5th** AES Convention, San Francisco, CA, USA.

# Audio and video training

Campus AV are running a series of audio and video training courses at the University of Surrey, UK, beginning in July this year. A whole range of courses from introductory to advanced level covering both operational and technical aspects is available.

Campus AV was formed by David Pope of Cambridge Digital Audio Consultants and more details are available by telephoning 0480 812201.

# People

• **Spaceward**, Cambs, UK, have announced their new management structure. Steve Martin has now been promoted to general manager; Mark Lister is UK sales manager; and Paul O'Brien overseas sales manager.

• Andreas Koch has been appointed to the position of vice-president and general manager at **Studer Editech** of Menlo Park, CA, USA.

• Andy Wild has been named VP sales and marketing at **Euphonix**, Palo Alto, CA, USA. Wild had served as VP Western Operations at SSL USA since 1984.

• **Music Lab**, London, UK, have announced the arrival of Rik Picton who joins the Hire Division as general manager. Picton comes from 10 years at London Sound Hire.

# HHB DIGITAL

ISSUE 6
SUMMER 91 ◆ AUDIO ◆ TIMES ◆ ☎ 081 960 2144

**NEW LOOK LISTENING ROOM**
A visit to our Scrubs Lane premises is incomplete without experiencing HHB's brand new Listening Room: an acoustically-treated space with a full choice of active monitors by ATC, where customers can critically evaluate the very best products available from a wide range of manufacturers. Popular demonstration subjects include the revolutionary Yamaha DMC-1000 digital mixing console, Eventide's UltraHarmonizer range, valve processors from Summit and the latest generation of Apogee convertors.
**Call now to make your appointment**.

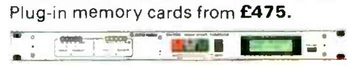
**SUMMIT AUDIO**
HHB is now the sole UK source for the full range of classic valve signal processors from Californian manufacturer Summit Audio. All Summit products are hand-built from selected components to deliver a uniquely musical sound that remains as popular as ever – especially in the age of 'clinical' digital. Alongside the TLA-100A Tube Levelling Amplifier (shown here) and TPA-200A Dual Tube Preamp are two equalizer designs: the EQF-100 Full Range Eq and the dual-channel EQP-200A. And remember: 'valve' is really pronounced 'toob'. TLA-100A: **£995**.


**YAMAHA DMC1000**
We're the nation's number one source for Yamaha's stunning new console. A 22-input digital audio mixer with timecode-based moving fader automation, instant recall of all front panel settings and powerful on-board DSP including 4-band parametric digital channel EQ. Yamaha has won the race to produce a full-function all-digital mixer that can interface directly with digital multitracks of all formats, hard disc recording systems, PCM-equipped VTRs, CD, DAT and digital signal processors. Touch-sensitive motorized faders and continuous rotary controls allow mixer moves to be automated against timecode during mixdown and subsequently edited. All parameters can be controlled via either MIDI or RS-422 for compatibility with video edit controllers. Equally at home in music recording or audio-for-video environments, we believe the DMC1000 represents an extraordinary development in digital audio.
**From £18,500**.

**DIGITAL AUDIO RESEARCH DASS-100**
'DASS' stands for 'Digital Audio Synchronising System', but there's far more to the DASS-100 than the name might suggest. Conceived as a 'problem solver' for the modern studio, the DASS-100 allows digital devices of all formats to be interfaced successfully in the digital domain. The spectrum of possible applications is vast, ranging from CD preparation and mastering to audio transfer between digital multitracks, hard disc recorders, D1, D2 & DX VTRs, CD, DAT, digital consoles and signal processors. Basic features include digital format conversion, sample rate conversion, gain adjustment, mixing, addition or removal of emphasis, DC offset removal, synchronisation to word clock and delay. Quick and easy to use, the DASS-100 is a must in any serious digital facility. **£7,995**.

**FOSTEX G24S**
24 tracks on 1" tape plus ultra-quiet Dolby S noise reduction, a removable front control panel that doubles as a remote with an in-built 10-point autolocator, MIDI function control and an on-board chase synchroniser option all make the G24S a formidable proposition. Brilliant user ergonomics and impeccable construction help ensure that the G24S is a real contender when it comes to choosing a studio multi-track. **£7,330**.

**SOLID STATE AUDIO FOR VIDEO**
Klark-Teknik's DN735 can record and play back short passages of stereo audio in perfect sync with other devices (notably VTRs) via externally applied SMPTE timecode. As such, it can augment any VTR with two fresh audio tracks, greatly simplifying stereo edits and crossfades. 20 seconds is standard, up to 175 seconds with additional memory cards. The 1u, 19" rack-mountable DN735 can be controlled manually, remotely, or via serial RS422. A snip to the audio-post specialist at **£3,550**. Plug-in memory cards from **£475.**



**APOGEE**
Here at last, the new generation of Apogee convertors offer startling audio quality. Both stand-alone units can help extract optimum performance from your existing digital hardware without substantial reinvestment. Simply the best convertors money can buy. AD500 **£1,195**
DA1000 **£1,595**.

**SONY STEREO MICS**
To partner your DAT portable, HHB offers a choice of stereo condenser microphones from the Sony range. The popular ECM-979 (shown here) and ECM-959 both represent extraordinary value for money, while the ECM-MS5 is built to tackle the most demanding applications. We also stock a wide selection of mics from other manufacturers, including the new VP88 from Shure. Sony ECM-979: **£210**.

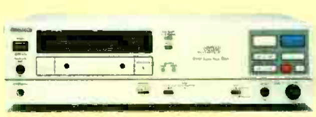
**MORE NEWS FROM EUROPE'S DAT CENTRE**
We're the world's leading supplier of DAT recorders to professional users. And we back all our DAT products with the best advice and service support in the business. Call us first to discuss your precise application requirements.

**AIWA HHB1 PRO KIT**
HHB's own groundbreaking professional portable with A-Time record capability is partnered with the Sony ECM979 stereo condenser mic to deliver an unbeatable ENG and location recording package. **£1,250**.

**SONY DTC1000ES 'PRO'**
Another HHB exclusive, the 'PRO' takes all the features of the industry standard, best-selling DTC1000ES, while adding a 44.1kHz digital record modification, balanced analogue XLR connectors and a rackmount kit as standard. **Unbeatable value at £1,195**.

**SONY TCD-D3**
We now have limited quantities of the world's first DAT Walkman. Buy the TCD-D3 from us and you also tap into Europe's finest service back-up. **Great value at £425**.

**SONY DTC-55ES**
Thanks to its superb performance and comprehensive function control, the DTC-55ES continues to provide audio professionals with an ideal low-cost alternative to conventional pro units.
**Now just £468.**



**PANASONIC SV3900/SV3700**
The new SV3900 from Panasonic can be controlled by either the SH-MK390 wired remote controller or via the unit's comprehensive serial interface ports. Other features include comprehensive indexing functions, SCMS status indication and error rate display. The SV3700 offers similar performance without wired remote operation.
SV3900: **£1,250** SV3700 **£950**.

**SONY PCM-7000 SERIES**
HHB has the full Sony range of professional 4-head recorders, options and remote controllers on demonstration. Featuring timecode, precision electronic editing and synchronisation, the PCM-7000 Series kicks DAT firmly into the nineties as the Number 1 choice for broadcast audio and post-production applications.
**Call now for price details.**

**SUMMER SALE BARGAINS**
HHB is offering a number of selected new and ex-demo items for sale at greatly reduced prices.
Call for further details.

| | |
|---|---|
| **Akai DR1200** 12-track Digital Recording System | **£8,950** |
| **Akai DD1000** Optical Disc Recorder | **£6,295** |
| **Akai S1000PB** Playback Sampler | **£1,195** |
| **Tascam MSR24** 1" 24-track tape recorder | **£5,295** |
| **Roland S770** Sampler | **£2,795** |
| **Roland SDE-3000A** Delay | **£549** |
| **Yamaha SPX1000** Multi-effects | **£750** |
| **Klark-Teknik DN360** Dual Graphic Equaliser | **£995** |
| **Wellard** Powered Monitors (pair) | **£795** |
| **Sony DXC-M7PK** Camera Kit (with lens) | **£6,200** |
| **Sony PVM-1320** Colour Monitor | **£995** |
| **Aiwa HDS1** DAT Portable | **£395** |

**– All prices exclude VAT.**

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

## Contracts

• The BBC have recently purchased a DAR *DASS 100* multifunctional digital audio interface system for their Pebble Mill Studios in Birmingham, UK.

• Video Post & Transfer have installed four sets of **Tannoy** *System 215 DMT* studio monitors at VP&T's new digital video/audio post-production complex in Dallas, TX, USA.

• Paul McCartney has invested in a Neve *VR72* 60-input console with *Flying Faders* automation and recall. Other recent Neve installations include Frank Zappa, Glenn Frey and Iron Maiden.

Ardent Studios in Memphis, TN; Pyramid Sound Studios in Ithaca, NY; and Village Recorders in Hollywood, CA, have all recently bought Neve *VR* consoles.

• **Focusrite Audio Engineering** have announced the sale of a 72-input *Studio* console to Ocean Way Studios, Hollywood, CA, USA.

• CTS Studios, London, have placed an order for a 60-channel Neve *VRP* console with *Flying Faders* automation and recall.

• **Audix Broadcast** have won a major contract to supply equipment to Radio Sokoto the newly built regional state radio station in Nigeria.

• **Lyrec** UK have just delivered a number of *Fred* editing systems as well as their first major delivery of *Frida* tape recorders to the BBC World Service at Bush House, London.

• **FWO Bauch** have announced that their System's Group have been awarded two contracts by the BBC's Transmission Engineering Department. The first is for the refurbishment of the OB room at one of the BBC's transmitter sites. The second contract is for the construction of a radio link vehicle.

• **Roland**'s Pro Audio Group have announced the sale of two Roland *Sound Space* processing systems to Larrabee Studios in Los Angeles, USA.

• Recent **AMS** *AudioFile* sales include a 4 hour *Plus* ordered by Edertrack Studios in Bilbao, Spain.

• Palace Recording Studio, Milan, have bought an **SSL** *SL 4048 G* series console. Other recent contracts in Italy include Metropolis and Psycho Studios in Milan; Suono di Ripetta in Rome; and Capri Digital studio located on a mountain top on the Isle of Capri.

• German broadcaster Suedwestfunk have bought a 2-hour version of the *AudioFile Plus*, for installation at their Mainz facility.

• **EMO** *GEQ* graphic equalisers have been installed at Loughborough Arts Centre, UK, as part of a major refit by Marquee Audio of Shepperton, Middx.

• Recent sales by **Neotek** include two *Elite* consoles bought by Kenneth Copeland Ministries in Texas, USA; and an Elite for Denmark Radio.

• **New England Digital** have announced the purchase of a 32-voice, 32 Mbyte, 8-track *PostPro SD* system by Japan's NTV Video Corporation (NTVV), a wholly owned subsidiary of Nippon Television Network Corp.

• **John Storyk** and studio owner Richie Cannata have announced the completion of Storyk's renovation and expansion of Cove City Sound's Studio B, NY, USA.

• The UK's TVS Television Centre have bought a **DAR** *DASS 100* multifunction digital interface to fully integrate the equipment system in their new digital dubbing suite.

## News from the AES

Our next lecture will be held on Tuesday July 9th, 1991. The subject is **DSP Techniques in Audio** and will be given by Ken Linton, University of Durham. 'It is clear that the professional audio industry is moving rapidly towards a future based almost entirely in the digital domain. In order to satisfy the functional and sonic requirements of audio professionals, large-scale multiprocessor architectures must be realised. With the widespread availability of programmable digital signal processing (DSP) devices, the cost-effective implementation of flexible audio mixing/synthesis is now possible.

'In this talk, the speaker will review novel DSP devices, contrast alternative multiprocessor configurations and discuss the software tools necessary for resource allocation and automatic code generation. It is these techniques which pave the way towards flexible parallel DSP systems for the professional audio environment.'

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

On September 7-9th, 1991, the **10th AES International Conference**, entitled **Images of Audio**, will present delegates with four 'images' of the current state-of-the-art in audio technology and techniques. Two sessions are concerned specifically with sound for pictures, while two further sessions cover exciting current developments in digital audio data compression and signal processing.

The main proceedings are preceded by a one-day **Digital Audio Tutorial** on the principles and technology of digital audio, and this may be attended either separately or in conjunction with the main proceedings.

The Conference will be held at Kensington Town Hall, London.

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

## In brief

• Cambs, UK: **New company out of Spaceward:** A new company called Studio Audio and Video Ltd has been formed by the founders of Spaceward Studios. Designers Mike Kemp and Joe Bull have been joined by Spaceward finance director David Mortimer. At APRS 1991 they launched their new product range based on their *X-S* floating point digital audio processing board for IBM compatible machines.

• Everett, WA, USA: **Rane issued patent**: Rane have been issued a US patent for their new Accelerated Slope equaliser design, which was introduced in the *FMI 14* studio grade mic preamp last year.

• London, UK: **Dreamhire launch new MIDI division**: Dreamhire have set up a new specialist rental division to meet the demands of the expanding MIDI market.

# FUTURE PERFECT

**YAMAHA**
**DIGITAL SYSTEMS**


Few things are certain in the future of an audio production facility. But it does seem likely that the successful ones will be those equipped to serve a broad range of clients – from film and television to the record industry – and certain that they'll be working largely in the digital domain.

With this vision in mind, Yamaha has developed a new, all digital 8:8:2 console, the DMC1000.

With 28 bit mixing and effects, and 32 bit Eq, its sound quality is sensational. Extensive automation includes control of all conventional mix parameters and, being software reconfigurable, the DMC1000 will adapt quickly to a wide range of applications.

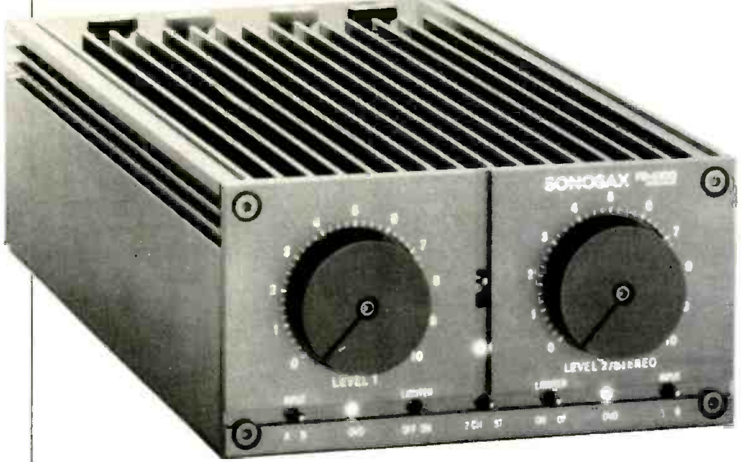
All the major digital formats are handled on-board and analogue auxiliary returns, MIDI, Wordclock and timecode sockets complete an extensive range of interfacing options. In addition, a cascading facility enables expansion using multiple DMC1000s.

And this console of tomorrow, once configured, will be extremely familiar to the engineer of today.

Are we right in our vision of the future? You can find out right now. Call Yamaha Digital Systems to arrange a demo.

**YAMAHA**
**DIGITAL SYSTEMS**
0908 366700

Yamaha-Kemble Music (U.K.) Limited
Professional Music Division




# Sonosax mic preamp, power amp and monitor

Sonosax have introduced three new products—the *FD-M4* microphone preamplifier/splitter, the *FD-A100* compact power amplifier and the *JM-3A* active monitor.

The *FD-M4* is modular in construction and fits into 19 inch Eurocard rack frames. The preamplifier uses fully discrete technology for superior performance and operates from 79 to 0 dB in 1 dB steps for optimum noise performance. Features include excellent transient response, high slew rate and large bandwidth, monitoring section with phase indicator, vu meter and overload LED, four independent balanced outputs and a separate asymmetrical output.

The *FD-A100* provides 2×50 W of power in a 12×5.7×2.85 inch (310×144×72 mm) package and uses discrete technology for optimum sound quality. The amplifier is equipped with a switchable limiter and two independent inputs for each channel, and will operate into loads of 1 Ω.

Specifications include a frequency response of 10 Hz to 40 kHz ±0.02 dB and 0.003% distortion at 1 kHz at full power.

The *JM-3A* compact control monitor is 3-way and equipped with an *FD-A100* amplifier. The active crossover uses slopes of up to 54 dB/octave and separate limiters for each frequency band.

To increase the audio performance, the enclosure has a double-back filled with sand and phase alignment for the drivers.

A passive version of the *JM-3A*, the *JM-3P*, is also available and features LC filters with protection and adjustment circuits for the MF and HF sections.

**Sonosax SA, 1162 St Prex, Switzerland. Tel: 21.806.02.02. Fax: 21.806.02.99.**

# Stage Accompany SA 1310 graphic equaliser

Stage Accompany have released the *SA 1310* constant-Q graphic equaliser for studio and live sound applications. The equaliser features 31 bands on ISO centres between 20 Hz to 20 kHz and a variable 12 dB/octave highpass filter, balanced inputs and outputs on both *XLR*-type and jack connectors (transformers optional), gain trim control and switchable EQ range (6 dB or 12 dB).

The circuitry contains no electrolytic capacitors in the audio path and frequency response is claimed flat to 1 dB between 1 Hz to 100 kHz. The input will accept in excess of 23 dBu with the output drive capability being 22 dBm into 600 Ω.

The *SA 1310* is fitted with a 10-LED bargraph, which monitors the signal post-EQ, with an eleventh LED functioning as a clip indicator.

**Stage Accompany, Anodeweg 4, 1627 LJ Hoorn, The Netherlands. Tel: (0) 2290 12542. Fax: (0) 2290 11192.**
**UK:** Stage Accompany (UK) Ltd, 14-16 Deacons Lane, Ely, Cambs CB7 4PS. Tel: 0353 662278.

# Behringer new products

Behringer have added several new models to their range of processing equipment.

The *Multiband Exciter* employs the Behringer 'Natural Sonic' principle of frequency-dependent phase shifting combined with programme-dependent harmonics processing for the higher frequencies, together with bass excite circuitry for low band processing. The unit also incorporates a surround processor and a de-noiser. The surround processor extracts spatial information from the stereo signal and adds it back to provide an enlarged soundstage.

Though the *Multiband Exciter* is very quiet in itself, the integrated *De-noiser Mk III* is useful in reducing noise inherent in the programme material.

Other features include servo-balanced inputs and outputs, stereo linking and optional output transformers.

The *Suppressor* is a multiband sibilance and feedback processor and finds applications in both studio and stage situations. The unit can be fine-tuned into the frequency bands where feedback or sibilance occur and special circuitry then 'looks' for sinusoidal signals. If these are found, the processor inverts the phase and sums it with the original signal, thus eliminating the sibilance or feedback without affecting the programme.

**Behringer Studio Equipment Ltd, Otto Brennerstrasse 4, 4156 Willich 1, Germany. Tel: 21.54.42.85.21. Fax: 21.54.42.85.23.**
**UK:** Ampsound, 153A Victoria Street, St Albans, Herts AL1 3TA. Tel: 0727 50075.

# Electro-Voice RE27N/D mic

Electro-Voice's new *RE27N/D* for studio and sound reinforcement use is the N/DYM version of the *RE20* studio microphone and provides a crisp high frequency response. Two highpass and one lowpass filters allow various degrees of tailoring the response of the microphone, including a simulation of the *RE20* characteristics.

**Electro-Voice Inc, 600 Cecil Street, Buchanan, MI 49107, USA. Tel: (616) 695-6831.**
**UK:** Shuttlesound Ltd, 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX. Tel: 081-640 9600. Fax: 081-640 0106.

# Monitor Technology Monitor One

The Monitor Technology *Monitor One* loudspeaker has been designed as a wide range close monitor for recording studios or for main monitoring in small control rooms, such as A/V suites or 'workstation' installations.

The system is 3-way and may be powered by a single-channel amplifier or bi-amped, though the internal crossover is still used. Power handling is rated at up to 250 W (provided the amplifier is not clipping).

The specified on-axis frequency response is 55 Hz to 15 kHz ±2 dB, with the −3 dB points at 50 Hz and 20 kHz.

The construction features heavy internal damping and an asymmetrical enclosure to eliminate internal reflections, together with shielding for use in close proximity with video monitors.

A *de luxe* version, the *Monitor One Reference Limited Edition*, has also just been introduced, which features

a new-design woofer and has increased performance in the mid-range.

**Monitor Technology, HC Andersensgade 22, PO Box 1102, 5100 Odense C, Denmark. Tel: 66.14.59.58. Fax: 66.14.91.81.**

# CEDAR scratch module

The Cambridge-based company CEDAR Audio Ltd together with Harmonia Mundi Acustica and Daniel Weiss Engineering of Switzerland, have announced the release of the CEDAR/HMA De-clicker module. Available in two hardware formats the module claims realtime removal of clicks, scratches and ticks however generated. Operating at a sampling rate of 44.1 kHz or 48 kHz, it is available either as part of the HMA *bw102* range of digital processing modules or as an independent 19 inch rackmount unit.

**CEDAR Audio Research Ltd, 5 Glisson Road, Cambridge CB1 2HA, UK. Tel/Fax: 0223 464117.**
**Harmonia Mundi Acustica, In den Sigristmatten 6, D-7800 Freiburg, Germany. Tel: (761) 491506.**
**USA:** Gotham Audio Corp, 1790 Broadway, New York, NY 10019-1412. Tel: (212) 765-3410.

# Heil MTD SR loudspeakers

Heil Acoustics have expanded the *MTD* range of sound reinforcement loudspeakers with the *MTD 112* system. Derived from the *MTD 115*, the system is fully compatible with Heil subwoofer systems for extended bass performance.

The passive 2-way *MTD 112* enclosure is equipped with a coaxial 12 inch driver and provides a coherent time-aligned wavefront. The shape of the cabinet permits a wide variety of uses and configurations, including wedge monitor, flown or stand arrays, fixed installation, etc.

The *MTD 112 LCC* controller is connected in a loop with a suitable power amplifier and provides optimum performance from the speaker enclosure. The controller has two main working modes—Monitor and Front for different working conditions.

**Heil Acoustics, 25 Route de Jouy, Bat B, 91570 Bievres, France. Tel: 1.69.41.80.21. Fax: 1.69.41.39.98.**

# 3M Pro-DAT

A new range of Pro-DAT cassettes has been introduced by 3M. The new tape comes in four lengths—46 to 120 minutes—and is distinguishable from the consumer product by its red door and hubs; a record inhibit switch has also been included. The ultra-fine metal particulate tape has an anti-stat treated back coating, and a binder formulation which according to 3M is capable of handling the effects of the format's 2000 rpm head scanner and high speed searches of 200× normal speed.

**3M Magnetic Products Division, St Paul, MN 55144-1000, USA. Tel: (612) 736-9567.**
**UK:** 3M UK plc, Bracknell, Berks RG12 1JU. Tel: 0344 58551.

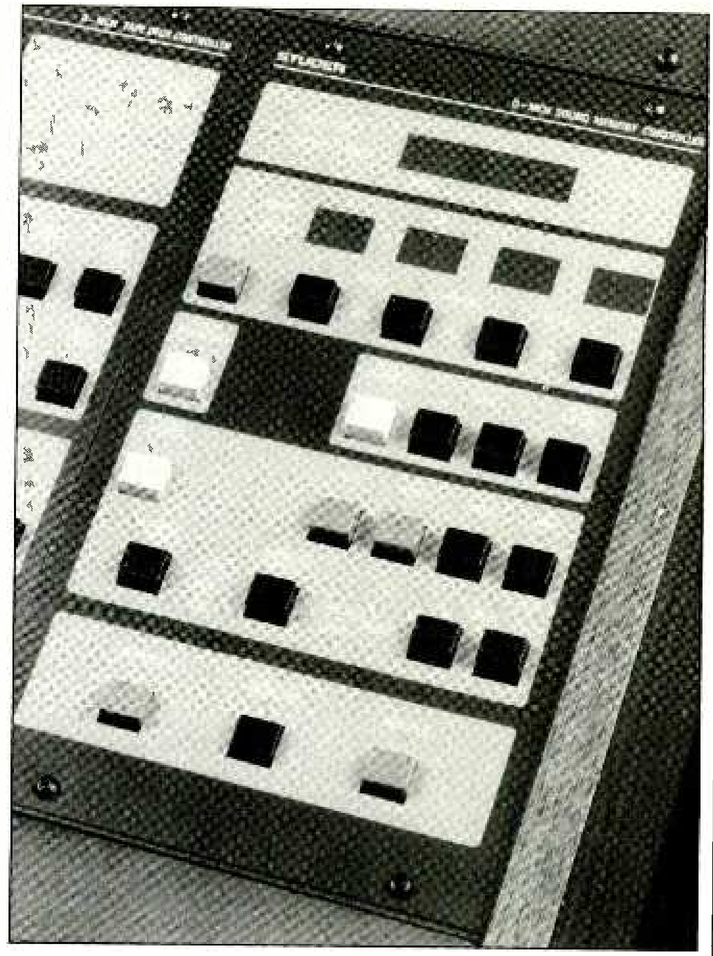
# Circuit Research broadcast processor

A new microprocessor-based on-air signal processor, *The Audio Signature,* has been introduced from Circuit Research Laboratories. The processor is a software controlled, wideband AGC (Automatic Gain Controller) followed by a 4-band compressor-equaliser. Controls include wide-band and multi-band adjustment, variable crossovers and adjustable gain freeze gates. All functions are manual as well as fully programmable (with recall).

Its main application is for on-air FM processing where it may be inserted ahead of an existing FM transmission processor or limiter, storing up to four internal presets (many more can be stored on a PC) which can instantly be accessed to change the sound of the station.

In addition to the digital control system, the unit features realtime analysis output monitoring, input/output and eight position audio diagnostic metering all displayed on the front panel.

**Circuit Research Labs, 2522 W Geneva Drive, Tempe, AZ 85282, USA.**
**UK:** PRECO, 21 Summerstown, London SW17 0BQ. Tel: 081-946 8774. Fax: 081-944 1326.



# Studer D820-48 memory board

The Studer *D820-48* digital multitrack machine can now be retrofitted with a sound memory board. The option allows external or on-tape tracks to be stored as follows—10 secs for 4 tracks, 13 secs for 3 tracks, 20 secs for 2 tracks or 40 secs for 1 track.

Three different methods of storage are provided and user selectable sections can be played back. The memory board also supports 'track slipping', delaying any single channel by up to 40 secs—if all 24 channels are slipped the maximum delay reduces to 1.8 secs. Additionally it is possible to copy up to four tracks simultaneously into four other tracks.

**Studer International AG, Althardstrasse 10, CH-8105 Regensdorf, Switzerland. Tel: (1) 8402960. Fax: (1) 8404737.**
**UK:** FWO Bauch Ltd, 49 Theobald Street, Borehamwood, Herts WD6 4RZ. Tel: 081-953 0091.
**USA:** Studer Revox America Inc, 1425 Elm Hill Pike, Nashville, TN 37210. Tel: (615) 254-5651.

# BBE retrofit for TOA 900

BBE Sound Inc have designed a plug-in module version of their single channel *Sonic Maximiser* to retrofit TOA Electronics *900 series* power amplifiers. The company claim that the *BBE 701* will correct any phase or amplitude distortion inherent in dynamic loudspeakers. Controls located on the front panel include two screwdriver adjustments for definition and low-contour control, and a bypass switch for comparing processed and non processed signals.

**BBE Sound Inc, 550 Bolsa Avenue, Suite 245, Huntington Beach, CA 92649, USA. Tel: (714) 897-6766. Fax: (714) 895-6728.**
**UK:** Stirling Audio Systems Ltd, Kimberley Road, London NW6 7SF. Tel: 071-624 6000.

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

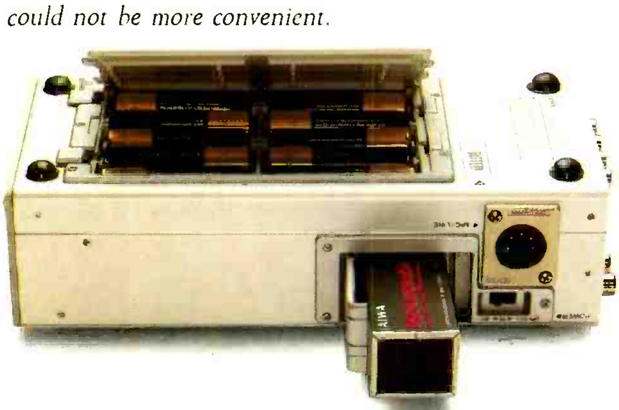

*Every audio professional knows that the DAT format is ideal for portable recording. But at HHB we believe it need not cost the earth.*

*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.*



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

# Mono broadcast of stereo film

Dear Sir, The April edition of *Studio Sound* contained an informative and interesting article on Digital Compact Cassette by Barry Fox. Unfortunately the final section of the article drifted off into Barry's favourite pastime of 'Broadcaster Bashing'. This time we at Thames fell between the sights of his literary *Scud*, ostensibly because we dared to show a stereo feature film in mono.

The film in question was *Beverly Hills Cop II* shown at 8pm on Christmas Day. This film in Cinema Release version is indeed stereo, in fact it is encoded for Dolby *Surround* if my memory serves me right. However, the movie actually transmitted was the TV/Airline version, being the only version capable of transmission at this early hour of the evening.

We deemed Eddie Murphy's tendency to verbal excess unsuitable for a Christmas Day family audience. As you are probably aware, TV/Airline versions are more than just a re-edit of the cinema release. They often contain specially shot scenes and redubbed soundtracks which are created in parallel with the cinema release print. Because they are always destined for video distribution the TV version may only exist in complete form as a video master, usually in NTSC and with mono sound.

So far so good, Barry is correct. However, he intimates that the decision to show this version was made in a 'panic'. This is not so. *TV Times* billed the film as a mono TV version and no on-screen announcement of stereo was made when promoting or introducing the film. The decision to screen the TV version was made four months previously and the correct information supplied to *TV Times* in good time for the copy date of six weeks prior to publication—hardly a 'panic'.

This raises an interesting question of why TV/Airline versions should be in mono only. This really boils down to the fact that, until relatively recently, the US distributors were always asked for mono because that is all that the television companies and airlines could deal with. This is now changing fast and the latest ITV purchased programme guidelines insist upon stereo as a norm for such programming. With the BBC, BskyB and Channel 4 now adding their weight, the distributors are finally changing to stereo working and the incidence of mono TV versions will rapidly decrease over the next few

# Tape life

Dear Sir, The article in *Studio Sound* (December 1990) by Barry Fox on the problems of deteriorating tapes was very welcome.

As the director of a small studio, and not possessing extensive electronic knowledge, it was very reassuring to discover that the problems I had experienced with one of two ¼ inch Reproduction Alignment Test Tapes purchased in April 1982 (one 7½ in/s, one 15 in/s), both recorded on Ampex *456*, were not of my making!

The higher speed tape has, for some 2 years now, been exhibiting exactly the symptoms referred to in Mr Fox's excellent article. All attempts by me to discover the cause of the problem had proved fruitless.

Thank you for helping to solve a perplexing problem.

**Yours faithfully, Jeff Clarke, Emmanuel Recordings, Sherwood Avenue, Newark, Notts, UK.**

months.

It may be of interest to you that Thames are now fully digital stereo, to the AES/EBU standard in our total transmission operation. If we receive programming on the standard *MII* tape format (with PCM) then the audio never leaves the digital domain until it reaches the viewer's loudspeaker. Hence stereo and even Dolby encoded surround sound is preserved perfectly throughout the system. A number of viewers have recently purchased surround sound capable receivers and are enjoying many cinema release films in full surround sound form. We are not up to surround sound for our own TV programming yet, unfortunately!

Finally, Barry criticises the picture quality of the film. I have checked the Transmission log and discovered that the film was graded at between 3 and 4 (out of 5) on a scale of subjective quality for both vision and sound, these gradings having been agreed with the IBA. Such a grade is acceptable and normal for a feature film that has been through a standards conversion process. The film itself was a positive print of good quality but had suffered somewhat during the original tape transfer and grading. This caused highlights to streak visibly in places, particularly on captions and credits, and there was some variation in black level thoughout the film. there was no visible motion smear however, and all things considered, the standards conversion from NTSC to PAL was very good indeed with no visible motion artifacts at all. I do not accept therefore that this film was 'rotten quality' and I doubt that anybody other than Barry Fox thought that they should be listening in stereo, which is probably why no-one else complained.

Thames employs many people dedicated to the pursuit of technical excellence. We do not attempt to 'get away' with anything; the principle being that many people do indeed notice when we do occasionally make mistakes and so we try to achieve the very best that we possibly can for no better reason than we hate criticism of any kind, informed or otherwise.

**Yours sincerely, Graham Stephens, Engineering and Site Operations Director, Thames Television PLC, 306-316 Euston Road, London NW1 3BB, UK.**

# User friendly

Dear Sir, Each issue I enjoy your editorial page; a kind of connection to the larger, international continuum. Regarding 'Broader base saves face', January 1991, much of it tersely confirms the realities our industry confronts.

As to your final 'tangent' paragraph, in which you discuss the validity of the term 'user friendliness', suggesting that many of us might not relate to our equipment as (paraphrasing) 'joined to us in mutual benevolence and itimacy', a thought comes to mind.

On the face of it, this definition, as product manufacturers overuse it, probably seems a bit dramatic, if not emotional. But as a mixer, I find that every time my studio gear brings me successfully through a session or project, I feel *profoundly* joined to it with a full sense of benevolence and intimacy.

**Yours faithfully, Lee Murphy, Brigg's Bakery, 122 West 88th Street, New York, NY 10024, USA.**

# Fantasy Studios

Regarding your article on Sunset Studios (*Studio Sound*, January 1991), you state, 'Sunset is the only studio on the West Coast that still has live echo chambers'.

Well, here at Fantasy we have five large echo chambers, all still very live.

**Neill King, Fantasy Studios, Tenth and Parker, Berkeley, CA 94710, USA.**

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# Ensoniq new releases

A spate of new releases from Ensoniq starts off with the *SD-1* synth that offers 61 programmable velocity and poly-key pressure keys with a 24-track step or realtime sequencer. 3.5 Mbytes of waveform ROM contains all the sounds of the company's *VFX SDII* plus new piano and percussion waves. Integral effects are 24 bit and available in 22 algorithms and new sounds can be added by disk or cartridge to complement the unit's 180 internal sounds and 60 performance presets.

The *SQ-1 Plus* is a development of the *SQ-1* keyboard with 180 sounds, built-in effects, splitting and layering, a 16-track sequencer and 61 velocity sensitive keys. The internal sound circuitry also responds to channel and polyphonic pressure via MIDI.

Aimed at players who can appreciate the difference, the *SQ-2* has an all-new 76-note weighted keyboard with 21 note polyphony, built in effects and waveforms that include the company's 16 bit 'Mega Piano Waves'. The *SQ-2*'s 16-track sequencer can be expanded from its 9,000 note capacity to 58,000 notes.

**Ensoniq Corp, 155 Great Valley Parkway, Malvern, PA 19355, USA. Tel: (215) 647-3930.**
**UK:** Sound Technology plc, Letchworth Point, Letchworth, Herts SG6 1ND. Tel: 0462 480000. Fax: 0462 480800.

# Software

- Fresh produce for the *Mac* from **Steinberg** includes *Cubase 1.8* with MIDI manager mixer maps, interactive phrase synthesis and all that's on the Atari version *2.0* aside from score printing, which will be along soon. *Cubase Audio* for the *Mac* supports Digidesign's *SoundTools*. Amiga version *1.2* of *Pro24* is now also available.

On the hardware side, the *SMP11* rackmount SMPTE device is a stripped down version of the popular *SMP24* while the *VLTCI* is a 1U rackmount video synchroniser.

- **Opcode** have released new modules for their universal librarian *Galaxy* offering support for the Alesis *Midiverb III*, E-mu *Proteus 2*, Ensoniq *SQ-1/SQ-R*, Korg *Wavestation*, Rane *MPE28/47*, Roland *R8M, D70, GP16* and Yamaha *TG77*.


# MIDI pedalboard

The number of outboard effects units and synth expanders now used by musicians makes the task of controlling these devices in a live situation more and more akin to learning a complicated dance routine as the artist concentrates on learning the moves and the steps essential to change presets and alter sounds dynamically. In many instances it all proves to be too much trouble with regression to a minimalist approach often preferred to a minefield of foot controllers, switches and pedals. It's refreshing, therefore, to come across the MIDI *Mitigator RFC-1* pedalboard from US company Lake Butler Sound, which, in MIDI terms at least, promises to be the 'one stomp shop'.

The *Mitigator* allows the creation of 128 named song files each containing a configuration of MIDI functions for the unit's five footswitches. These song files can be combined in four 'sets' that, when selected, exclude song files not required for the performance at hand. What is stunning about the *Mitigator* is its depth of control as each footswitch within a song file can be programmed to execute a string of MIDI commands simultaneously when pressed. These can range from a simple patch change on a solitary guitar multi-effects processor to a complete reconfiguration of the largest of studio systems. Footswitches can be programmed to transmit commands on any MIDI channel including program change, continuous controller data, pitch change, polyphonic aftertouch and channel aftertouch. Add to this, channel altering functions like MIDI mode, reset, all notes off and local control on/off and the ability to generate SystEx commands, systems common commands (song pointer, song select, tune request) and systems realtime commands like start, stop, continue, timing clock, active sensing and systems reset—the potential is staggering.

The low profile unit measures a modest 24×8 inches with a gentle rake that makes the large green 16 character LED display reassuringly legible. The device has a very firm footprint and is unlikely to take flight in response to a kick. It's also extremely rugged—the footswitches are of the type that encourage the player to stand on them with confidence rather than develop the dexterity of Fred Astaire. Bright red LEDs confirm the selection and though the legending looks a little unrefined at first glance it's eminently readable in all lighting conditions.

Song files, which can be scrolled through by increment and decrement footswitches, and their associated footswitch activated MIDI command strings are created and edited via a keypad, dedicated function keys and cursor keys. The process is extremely simple—something the manual never manages to convey—and has the added benefit of appealing to most varieties of user. A MIDI-illiterate guitarist looking to control his outboard will find a well constructed and straightforward device in the *Mitigator* with lots of visual feedback and ample room for any future growth in the guitar rig, while the all-singing all-dancing keyboard player running sequencers, drum machines and a bank of expanders will immediately appreciate that a little pre-gig preparation will make life a lot easier on stage.

Added benefits that may not be blatantly apparent include the ability to generate continuous control change data from a foot pedal that plugs into the back of the unit. Alterations in resistance across the jack socket will be interpreted as realtime MIDI control and this can be programmed to operate across all or selected song files. The ability to generate note on and off information from the footswitches means that the *Mitigator* could also be used as a five-note keyboard (fine for all those Moog *Taurus* drones) or, indeed, to trigger chords. MIDI connections are provided for MIDI out and in, allowing down and up loading of memory as well as external control of the song files by patch changes. Data can also be merged and a useful cable tester program is included.

On the down side, the only things missing are locking connectors for the external power supply and MIDI cables. The membrane-type cursor, keypad and function buttons are a little stiff on the fingers but have the added advantage of being virtually impossible to select accidentally by all but the most concerted effort with a well aimed winklepicker. The system is intrinsically extremely safe.

There is nothing quite like the *Mitigator* and many musicians have been waiting for just such a device. Be it for simple or convoluted MIDI control applications it's about the only pedalboard you'll ever need.

**Lake Butler Sound, 5331 W Lake Butler Road, Windermere, FL 34786, USA. Tel: (407) 656-5515.**
**UK:** (Main retail stockist) Systems Workshop, 24 Church Street, Oswestry, Shrops SY11 2SP. Tel: 0691 655019.

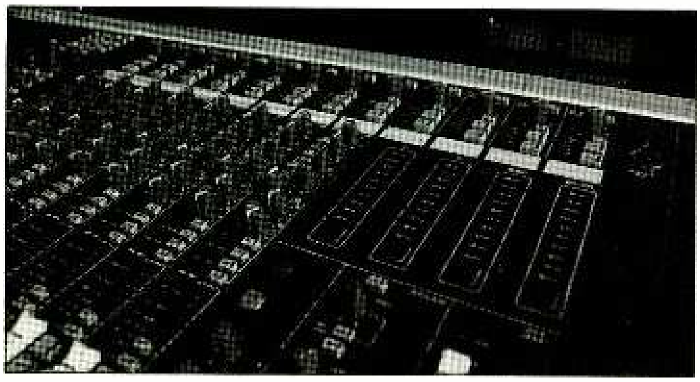
# Sync box

Claiming a frame rate accuracy better than 1 second in 10 hours, the *Syncman Pro* synchroniser from Midiman can read SMPTE from $^1/_{10}$ to 4× normal tape speed. The 1U rackmount with a sizable display supports all SMPTE rates with conversion to MTC, Song Position Pointer and Spot-Lock to video, which syncs once to a piece of longitudinal timecode and thereafter relies on video frame redraws. SMPTE can be generated to internal crystal, house sync, pilot tone or house mains and all the unit's modes incorporate jam sync and freewheeling.

Interestingly most of the *Syncman Pro*'s functions can be controlled by SystEx commands with the creation of tempo maps recorded automatically from a sequencer, entered manually or by tap entry from a MIDI note or external click. The tempo is battery backed and the unit also has the capacity to store 768 MIDI Foley points all of which are editable and can be up or down loaded.

**Midiman, 30 Raymond Avenue, Suite 505, Pasadena, CA 91103, USA. Tel: (818) 449-8838.**

**Studio Sound's Music News is compiled by Zenon Schoepe**



# Soundcraft launch Europa console

Soundcraft are aiming to regain their status in the high-end SR console market with the APRS launch of their all-new VCA-based *Europa* console.

The spec list is headed by switchable input channel noise gates, a sophisticated VCA system, 12 aux sends with global master selection and fully parametric 4-band EQ. Each fully balanced 24-, 32- or 40-channel frame has eight group modules (fed from a routing matrix on each input) plus master module. Dual concentrics are out, while comprehensive metering at every stage is in. Also welcome is a shorter front-to-back reach than the *Series 4*.

*(See 'Live Consoles' for further details.)*

# On tour

• **Sound Hire** have a huge showpiece on July 31st—a free concert by Luciano Pavarotti in Hyde Park, which up to 250,000 people are expected to attend. It has a 'Greco-Roman' stage by Fisher Park and Meyer SR and delay systems, with the FOH vocal and stereo clusters flown from cranes. There's also a live Sky TV broadcast and a recording mobile to keep Richard Leinard's team busy.

• **SSE** fired up Birmingham's Cannon Hill Park fireworks and orchestra spectacular, and the same city's International Conference Centre inauguration ceremony on June 9th. More conventionally, there's UB40 at Finsbury Park; the 'Feile' three-day rock festival in Thurles, Eire; Hothouse Flowers in Phoenix Park and UK tours with Soho, Elvis Costello and Hue & Cry. For their sins, SSE have the seven-week *Monsters of Rock* European tour (including Donington) with AC/DC, Metallica, Motley Crüe, Queensryche and The Black Crows.

• **Two Red Cross** charity shows in May, both broadcast worldwide by the BBC, had all-British productions. London Chamber Orchestra's *Light The Darkness* gig in Geneva followed the Warren-Green brothers' *Power Prom* format with Dimension Audio providing a *TMS-3* SRS plus *PM3000* FOH and Ramsa *S840* monitor desks. At Wembley Arena four days later the *Simple Truth* Kurdish benefit used SSE's *MT-4* rig (already in place for MC Hammer) and, acknowledges Chris Beale, many free contributions from other SR firms too numerous to list.

• **MHA** are supplying Status Quo (supporting Rod Stewart) with a FOH Hill *Concept* 48-channel board and an *M3 S-Series* 32/10 monitor desk plus associated stage monitors.

• **Audiolease,** continue a tour with EMF and their all-new *A-2* SR system. In Europe, they're out with Philip Glass while Julian Cope tours the UK. And they're making full use of their Midas *XL-3* console's 'dual mode' capabilities at London's Shaw Theatre for a musical on the life of Marvin Gaye—it's performing both FOH and monitor duties simultaneously.

• **Autograph Sound Recording** have a large Meyer SR system and a 100-input Cadac console on *Tosca* at Earl's Court from June 23rd. The show is preceded by two weeks of set-up and rehearsal.

• **Britannia Row** are sufficiently busy to be using Meyer *MSL-3* boxes on a couple of tours, namely Lenny Kravitz and James Last. *MSI* systems are out with Gloria Estefan and Jimmy Somerville; *TMS-3* dates include Morrissey, Shakin' Stevens (BRP are covering this for Wigwam), the Irish music festival in Finsbury Park, North London, The Pixies at Crystal Palace Bowl, the Gipsy Kings and two regional festivals—Harlow Pop on July 21st and Cumbrian Rock near Carlisle on July 13th. The new *Flashlight* system is booked for Denmark's Roskilde Festival in late June (with *TMS-3* delays), Joe Jackson's tour and the Pet Shop Boys.

• **Canegreen** suffered a disappointment in May when Deep Purple canned their US tour but are busy nonetheless with Elaine Page, Stranglers, Womack & Womack and Pat Metheney tours in Europe, while Rick Astley enjoys his post-SAW freedom on US dates from mid-July. Nigel Kennedy launched his world tour (strictly classical this time, with a full orchestra) at the Albert Hall in June using 20 *MSL-3*s.

• **Clair Brothers'** Stan Horine says the company is "very busy". European ventures through June and July include Sting, Paul Simon, the reformed all-star Yes (shows in the round), Bob Dylan and a major festival in Italy using Clair's European arm Audio Rent of Switzerland.

• **Concert Sound** have been touring with Harry Connick and Alison Moyet, as well as supplying FOH and control for Joe Cocker's support slots on the Rod Stewart arena shows. They're also providing Level 42 with FOH control in Europe and FOH and monitor boards plus engineers for the forthcoming Dire Straits world tour.

• **Delta Sound** (using Nexo, EAW, E-V, Martin and JBL gear) are into cars in a big way with shows, according to Mark Bonner, for Porsche in Stuttgart, Vauxhall in the UK and Saab in Belgium. There's also a Post Office Counters exhibition and a major HMV conference on their books.

• **Encore** have a summer of rave—at Leeds' Elland Road stadium on June 1st with Happy Mondays, an arena tour with Inspiral Carpets, 808 State at Milton Keynes Bowl, Dee-Lite in Germany and Happy Mondays in Paris. Bucking the trend are a three-day jazz festival in Manchester, the ICA Rock Week and Squeeze in rehearsals.

• **Showco**'s Robin Magruder claims 1991 is "by far our most active summer yet in terms of numbers of European tours". The list is certainly impressive—ZZ Top's stadium dates, Living Color in theatres, The Moody Blues, Robert Palmer, The Bee Gees, Beach Boys and INXS. And a re-formed ELO kicked off their world tour with the Moscow State Symphony Orchestra in late May at the NEC—it's due to wind up in the USSR at Christmas. David Scheirman is overseeing a *Prism* system, which includes a unique SoundLab string mic concept.

**Studio Sound's Live Sound News is compiled by Mike Lethby**

# News round-up

• **Celco**, the live entertainment division of Electrosonic Ltd, has moved. The new address is Hawley Mill, Hawley Road, Dartford, Kent DA2 7SY, UK. Tel: 0322 222211.

• **d&b audiotechnik** took a small slice of the London Dorchester Hotel's mammoth £70 million refurbishment. The refit, which closed the hotel for over a year, included new d&b *E1* systems in the Dorchester Bar and Terrace Restaurant through installer CAV.

• **Nexo**'s Singapore distributors Electro-Systems, apparently busily installing Nexo SR systems in big discos from Penang to Jakarta, have revealed a disturbing new twist to the 'keep music live' campaign. Announcing a prestige contract at Singapore's top nightclub, Fire, Nexo say the venue now includes 14 Karaoke rooms. The world awaits a compilation of inebriated sales reps singing *My Way*—14 different ways.

• **Renkus-Heinz** are aiming their new *C-1A Coaxial Point Source* system at the top of the touring market. This latest version of the *C-1* SRS is, again, a trapezoidal two-box design plus matching subwoofer, with point-source characteristics down to 300 Hz and an improved flying system co-developed by Ampco Holland.

• **Samson Technologies** have announced a new moulded ABS carrying case for their microphone systems. The *SC-1* protects *VLP*, *Stage 11/22* or *MR-1* systems plus cables, accessories and AC adaptor.

• **TAC** have sold an *SR9000* 'superconsole' to the Bremen-based SR hire company Hansa PA through German distributor Mega Audio. The desk is being used on FOH for Bachman Turner Overdrive's European tour. Meanwhile, three *Scorpion II* consoles are going into a new media and arts centre in Walsall, UK via Thatched Cottage, and Belgian SR company Promoscene has ordered two more of the same.

# LIVE CONSOLES

## Mike Lethby gives a summary of live sound consoles and their facilities

It's hard to recall a time since the mid-'70s when there was such fevered activity at the top end of the SR console market. After years of incremental improvements the tide of progress has risen to a flood of new desks aimed at major SR companies, installations and theatres.

The £25-£35,000 sector, for so long the domain of Midas' *PRO-40*, Yamaha's *PM3000* and the venerable Soundcraft series *4*, now sees new 'flagship' boards from Ramsa, Midas, TAC and Soundcraft. They're all fighting for a slice of the action alongside the *PM3000* (which Yamaha will market until its successor arrives, possibly early next year) and TAC's *SR9000* 'superconsole'. Soundcraft say they have not officially discontinued the series *4*, but clearly its days are numbered.

Perhaps the boldest development of all was unveiled at APRS by a company who are new to this side of the market. Clive Green & Company, better known as top theatre console manufacturer Cadac were announcing their remarkable *Concert* FOH desk. This features 'instant recall' of all routings and control settings—the closest any road-going console has come to full automation. The technology isn't cheap but it does promise efficiency savings by way of compensation.

While the other new releases may not offer radical changes they do reflect a welcome new attitude to important matters like ergonomics and customer-led features. But will the traditionally conservative (and, lately, somewhat impoverished) European market be able or willing to support all these new products? And will manufacturers ride out the effects of recession and the Gulf War—neither of which were anticipated when most of these designs first hit the drawing-boards.

## Cadac Concert

The idea of equipping a touring console with something resembling the type of 'scene'-based automation found on theatre sound desks and lighting boards, has been kicked around for years. A few prototypes have appeared at shows but no major manufacturer has ventured into production with a fully working version.

The chief reason, naturally, is cost: £35,000 is getting expensive for a live sound board and if you start talking about doubling that price to add memory automation, your market is going to be fairly small and specialised. This is especially true in Europe, where SR companies' shares of tour budgets are being squeezed relentlessly. So it's no surprise that Cadac see the lucrative US market as the prime target for the *Concert*, a desk that has a very comprehensive automation system and which Green hinted would be priced around the £70,000 to £80,000 mark. It is a bold move. Even though the US represents 50% of worldwide SR console sales, it's still a tough arena. Clair Brothers—as large and resilient a hire company as you can find anywhere—abandoned their own automated console programme late last year when considerable work had already been completed.

Cadac, however, have an advantage in commercial terms. Crucially, they did not have to spend a small fortune conjuring up the concept and writing software from scratch. Green's Luton-based company, formed in 1978, has been building automated theatre boards for years—boards which have well-proven, tried and trusted technology.

Their line-up is headed by the *E* series console, often found in customised form with clients such as *Starlight Express* and the National Theatre. Cadac spent £100,000 in 1989 alone to automate pcb design and production procedures. This R&D groundwork minimised development costs for the *Concert*, which would have made it unviable.

Due for October launch, the *Concert* is designed for touring with an aluminium/steel frame, built to order with as many input modules as the client wants. And since the price you pay for automation is a fixed component in the total cost, larger boards will inevitably offer better cost-per-channel value.

The major feature is an 'instant recall' system, which stores the desk's complete routing status, plus the positions of all input, EQ, aux send and pan rotaries, and control on/off switches' status, in 'snapshot' memories called Cues. While the desk itself can remember basic routing cues, an MS-DOS compatible PC (not supplied) stores and recalls most of the data in realtime. Cadac advise clients to choose a local PC vendor who can provide back-up on demand.

A central control module commands the recall system. Based on a desk supplied to the National Theatre in 1986 and with modifications suggested by top New York theatre sound firm ProMix (servicing *Miss Saigon* among others), the computer reads every control setting on saving a Cue. While channel fader options run from normal analogue types to full-blown motorised VCA faders, recall of levels and settings is not quite in the SSL league. To save the expense of motorising every pot, Cadac opted for LED alerts which on recalling a Cue, light up on any rotary whose setting differs from the previous Cue. To alter any of those positions, you twist the relevant knob until its LED goes out (at which point you've matched the stored value) and then adjust and re-store as you wish.

The standard basic spec includes eight mute groups, 16 discrete aux sends per input, 4-band parametric EQ with low- and highpass filters, routing to 12 subgroups and 12 main groups and an output matrix panel (whose settings are also storable in the Cue memories).

A feature that could swing the economic balance in the console's favour is the provision of two identical inputs on each input module. They offer A, B or mixed A+B and thus effectively double the number of inputs available. On a show with lots of mic lines, for instance with a large live orchestra, one *Concert* desk could potentially take the place of a pair conventional FOH consoles. If the standard capacity proves insufficient, bus connectors allow inter-console linking.

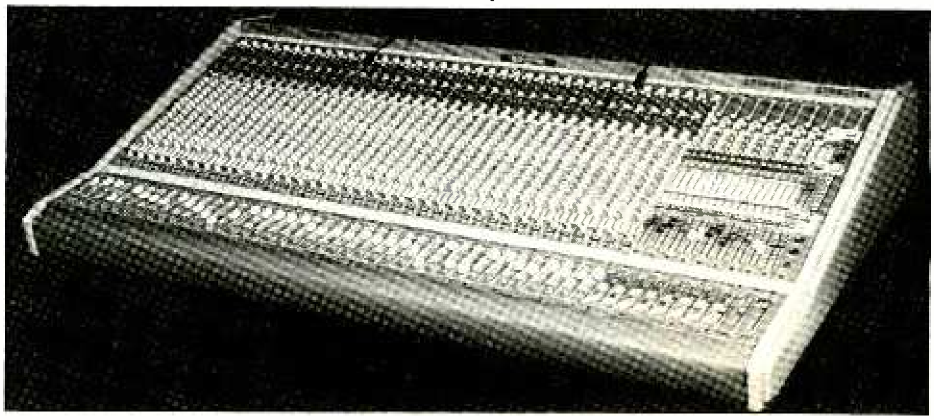
## Midas XL3

The Midas *XL3* is the only console specifically designed for FOH, monitor and theatre duties. The latest in a long (and occasionally patchy) line of Midas boards, it was launched at the end of last summer. Since then, its versatility and sonic characteristics have won it a growing level of respect—not to mention some prestigious sales—out in the field.

Unsurprisingly, initial reactions tended to be sceptical. Chris Rogers, development engineer at Midas, admits that in the console's early days potential clients would arrive with preconceptions, "People often said to us, 'If it's a front-of-house desk and a monitor desk, surely it must be lacking in both areas?' Our reply was simple. 'Take one out and try it for yourself.' Reactions after they'd had a trial run were generally very different."

Those preconceptions reflected the marketing dilemma that Rogers spells out. Admitting the desk was launched without a specific existing demand, he added: "The hardest thing is to discover exactly what people want. On the one hand people will tell you they want a console to have absolutely everything but if you actually provided it, they'd say it was either too complicated or too expensive or both."

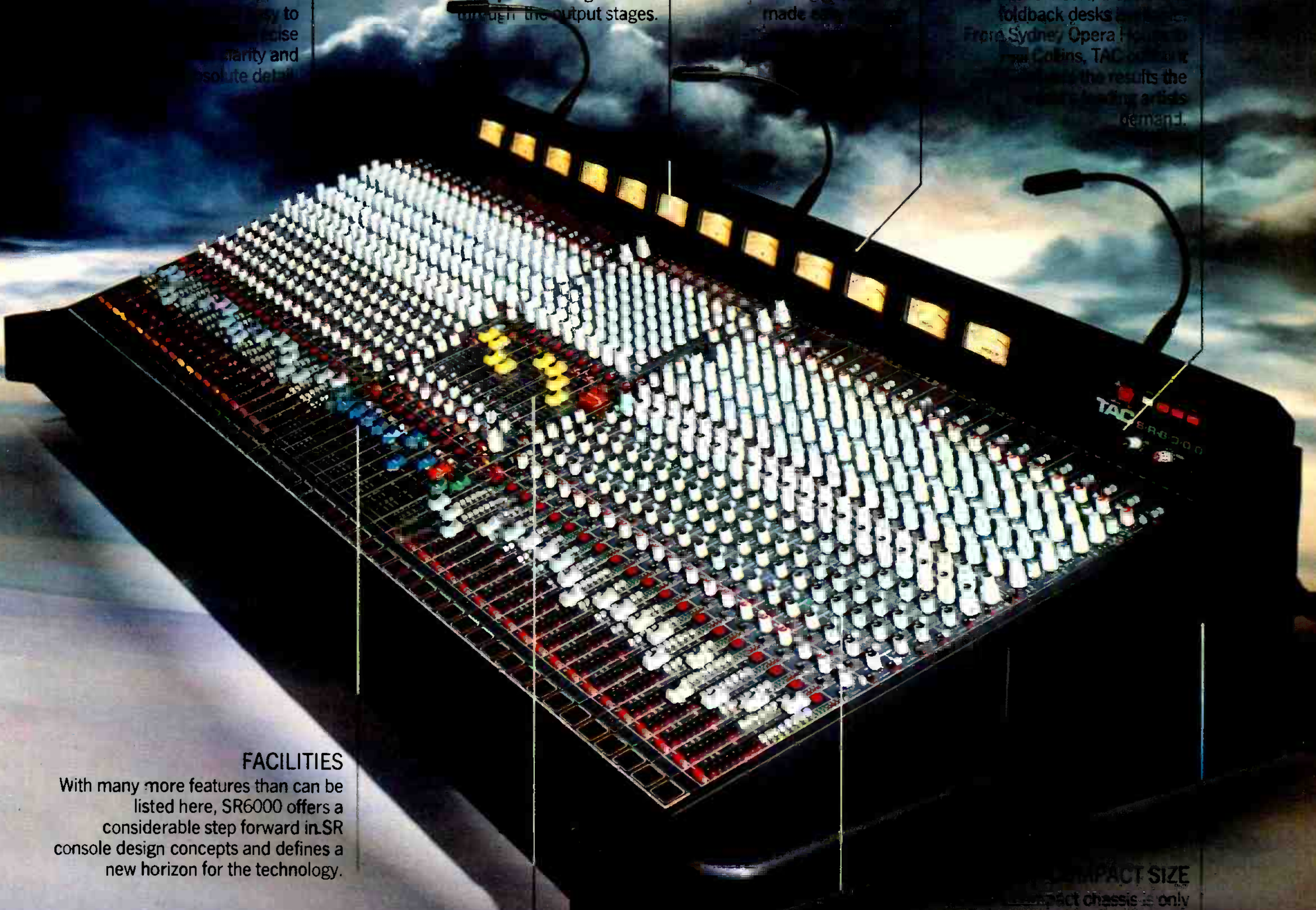
What existing Midas users definitely *did* want, it seems, was the sound of the *PRO-40* (which plenty of top engineers still admire), a usable and quiet equaliser, and flexibility. VCAs, which people love to use but hate to hear, would control the desk's status and Rogers says, "We have an excellent design—VCAs will always add some noise but we've reduced it to the lowest level possible."



**Midas XL3**

# SR6000 from TAC

## THE REVOLUTION IN SOUND REINFORCEMENT



### CLASSIC EQ
...ures a 4-band ...t Eq with parametric ...ids and swept High Pass ... Based on the Classic ... parametric, the ... easy to ... precise ... clarity and ... absolute detail.

### TOTAL METERING
A 7-stage LED meter located next to the mute switch on each channel gives an immediate assessment of levels from signal present to +12dB. 12 VU meters with individual multi-source input selection enable rapid checking of levels through the output stages.

### MASTERS & SLAVES
Linking of multiple SR6000s in master/slave configurations is made easy.

### PEDIGREE
TAC has an acknowledged 12-year track record in the design and manufacture of the best concert, theatre and foldback desks available. From Sydney Opera House to Phil Collins, TAC consoles deliver the results the world's leading artists demand.

### FACILITIES
With many more features than can be listed here, SR6000 offers a considerable step forward in SR console design concepts and defines a new horizon for the technology.

### TOTAL CONTROL
SR6000's output system has been designed to allow maximum flexibility in configuration of output stages. Each input can separately address 8VCA/Mute groups and 8 audio subgroups, all of which are overlapping. The main stereo output and the 10 x 8 output matrix allow multiple speaker arrays to be controlled with ease, while the VCA Master gives overall control of all 10 main outputs.

### SPLIT AUXILIARY SYSTEM
SR6000's revolutionary auxiliary system allows the 8 send busses to be split between left and right sides of the console, giving the engineer 16 auxiliary paths for large, effects-heavy mixes.

### COMPACT SIZE
The compact chassis is only ...mm (66 inches) wide for 40 inputs, 4 stereo effects returns and all output facilities. Robust steel construction coupled with side lift bars and concealed grip space under the armrest allow SR6000 to be manoeuvred and transported easily in the secure knowledge that all circuitry is fully protected from external impact.

**TAC**

Total Audio Concepts
Unit 17, Bar Lane Industrial Park, Bar Lane, Basford,
Nottingham NG6 0HU. Telephone: 0602 783306.
Telex 37329. Fax: 0602 785112.

In the USA: 10815 Burbank Blvd, North Hollywood,
California 91601. Telephone 818/508 9788. Fax: 818/508 8619.

TOTAL AUDIO CONCEPTS is part of AMEK TECHNOLOGY GROUP Plc

And, of course, the desk had to be roadworthy.

The biggest talking point is the 'dual mode' capability. What inspired this design?

Rogers: "This industry is not as rich as it used to be. A PA company will buy a nice FOH desk because it can make money on big tours, whereas the monitor engineer often gets stuck with a desk from the back of the warehouse."

Midas saw that a desk which fulfilled both roles could be an attractive proposition for cash-starved hire firms. At the same time, they had to build a console engineers would want to use—because it's they who decide the success of a console. The design they settled on—which drew heavily on parent company Klark-Teknik's electronic skills—largely bears out Midas' assertion that corners have not been cut in the quest for flexibility.

Channel modules have a new ultra-low noise input stage and a 4-band equaliser with two fully parametric mid bands. A balanced insert/return point and a pre/post fader Direct Out option extend the desk's usefulness in recording, broadcast and theatre work.

To function effectively in monitor and theatre work, lots of mixes and outputs are essential. The *XL3* offers 16 discrete mix sends per input channel, individually switchable pre/post and on/off. These are available as output mixes, subgroups or aux master outputs. All the inputs are also routable to the masters, and there's a 2-way matrix (derived from mix groups and expandable to 8/8). A pair of ancillary record outs completes the *XL3*'s total of 22 outputs.

When it comes to VCAs, every manufacturer has its own idea of the ideal set-up. Midas, says Rogers, simply opted for a configuration, "which we felt was the most usable". They aimed for operational simplicity and a familiar layout, partly out of consideration for engineers whose first exposure to the desk might be in a high-pressure festival situation.

Inputs can be routed to any number of the eight VCA masters, each of which has on/off and Mute. The two Grand Master VCA groups control any combination of the 16 master mix outputs. Muting is similarly comprehensive; all input channels and groups have eight Mute Assign switches, placing each under the control of a Master mute. For extra flexibility, Mix Group mutes can be selected to mute either the complete group output, the group output but not the signal to the main stereo bus, or *vice-versa*.

The *XL3* comes in 24-, 32- or 40-channel frame sizes, with the largest costing around £35,000. As well as the 8/8 matrix option, there is also a 16-channel extender version.

Midas's policy has been to focus on 'ready markets' for the *XL3*, concentrating on Britain, Europe and the Far East and leaving the US for later. In the UK, Encore and AudioLease have bought it for touring. The desk's flexibility and quietness have also found favour with broadcast and installations users.

## Ramsa S840

Ramsa, a division of Panasonic, itself a sibling of the giant Matsushita Corporation, launched the *S840* monitor and house console in 1988. It was designed by Ramsa in the US as a direct response to market demand—namely, for a well-equipped dedicated stage monitor desk at a reasonable price.

Listed at £24,000 (monitor) and £23,000 (house), including two PSUs, both versions share a common 40-input channel frame, master section, master bussing and metering—the essential difference lying in the input modules. While the house model has been a slow seller against much competition, the monitor's price/performance ratio has won it a strong position.

Input channels are headed by a single pad-less 64 dB gain stage, which uses discrete components to minimise noise and distortion.

The equaliser section is a 4-band sweepable design with a variable highpass filter. Steve Spencer, of UK distributors Britannia Row, says, "The EQ is the most subjective area of any console but in my personal opinion this one is extremely musical." While some would no doubt prefer the extra versatility of a fully parametric EQ, the *S840* was, he says, "designed as an affordable console; rather than build in loads of features they made a simple and electronically sound design".

Next are the 18 sends. There are 10 single pots, each with on/off and pre/post switches, and four dual concentrics. The latter, by adjusting DIP switches on the module pcb, can be configured either as discrete mono sends or as stereo sends with level on top and pan below. The variations are: 18 mono; 14 mono; plus two stereo; 10 mono plus four stereo—providing plenty of flexibility in setting up stereo mixes for keyboards, drums, etc.

More DIP switches—nine in all—include direct out pre/post EQ and/or fader; PFL and insert pre/post fade; and aux sends 1-8 pre/post EQ.

The full length fader, designed and built by Matsushita specifically for this desk, is a plastic conductive type to provide a longer life and quieter operation than carbon-track types. Another notable detail is that the rear connector panels form a part of each module; there are no intervening connectors.

The master section houses the 18 outputs, and a 10×8 matrix (standard in both house and monitor versions) controls the eight groups and left/right masters. Here, too, are the first eight mono aux returns; the rest are on a pair of mono FX return modules, each having two returns, full bus routing, pan, level and 2-band EQ. Sited adjacently, these can easily be operated as a stereo pair if required.

The monitor module offers headphone listening by pushbutton to every bus plus PFL and headphone level controls, and machine returns with selection for any input standard including domestic types—a feature that makes the module very flexible at coping with specialised tasks such as broadcast and theatre. A comprehensive line level communications systems is also provided.

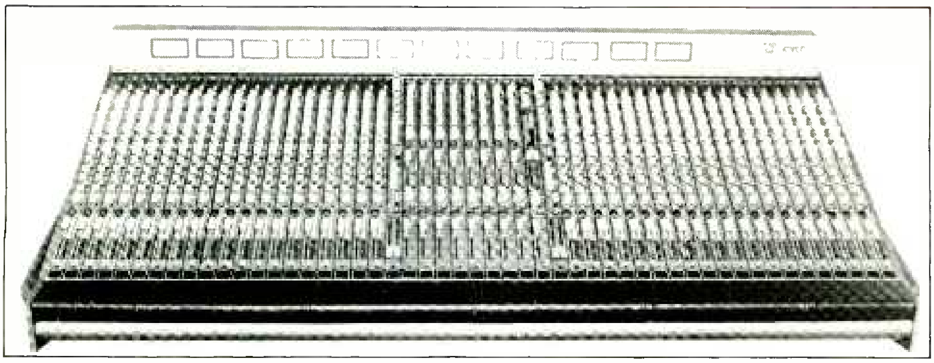
Metering is via an 18-vu meter bridge, monitoring the console's 18 buses. The first eight meters have adjacent pushbuttons selecting the eight matrix outputs; similarly, PFL and oscillator output levels can be selected for metering. Further meter bridge touches include PSU rail warning LEDs and a dimmer control for the three halogen *Littlites* supplied as standard.

European sales to date include Britannia Row Productions, Wigwam, Sound Hire, Canegreen, Capital Entertainments, John Henry, RoadStar, Regiscene, Ampco, Audio Rents and Westfalen Sound. Carlton TV bought a FOH *S840* shortly after its launch.

A final note: BRS are currently working on a prototype (designed in-house) of a VCA grouping and muting option, which should soon be available as a retro-fit for any existing desk or as standard on a new console.

## Soundcraft Europa

Like Midas, Soundcraft's last big live board was



**TAC SR6000**

well into pensionable age before a replacement was mooted. Unlike Midas, that desk—the series *4*—was becoming something of an embarrassment to a company which by 1988/89 had radically reshaped its thinking and its internal organisation.

1991's Soundcraft is a very different beast. Although plenty of good ideas flowed they did not always align with what the market wanted. Quality control was patchy at times, too. In particular, the *4*, successful for many years, had latterly become known more for its noise and uncomfortable size. Although improvements to the grounding improved noise performance on later versions, the Yamaha *PM3000*'s success—itself not an outstandingly quiet desk—showed how far the series *4* had passed its sell-by date.

But by the late '80s, Soundcraft had embraced a new marketing-led attitude. Out went quirky ideas and hasty quality control; in came freshly integrated R&D, marketing and manufacturing philosophies. Japanese-style production line techniques and structured component buying produced a range of very successful budget/mid priced studio and SR consoles. Along the way, too, have come genuine innovations in circuit design: Soundcraft patented the first pad-less mic/line input and an advanced active panpot, to name just two enhancements. All these ideas and techniques can be found across Soundcraft's current range—including the new console, which is intended to restore the company's status in the élite SR market: *Europa*.

First shown in Britain at APRS, *Europa* has been under development for two years. Soundcraft's R&D director John Oakley says Soundcraft users around the world were consulted, "and we've built a no compromise console which answers their acoustic and business needs".

Although he says *Europa* doesn't strictly replace the series *4* (there are no stereo group buses, for example) it will inevitably do just that, and take sales from the high-spec series *8000*.

The console comes with 24, 32 or 40 channels, with eight group modules and a master module as standard. All inputs, inserts, outputs and aux sends are fully balanced and the options list includes dual matrix and advanced stereo modules. Single control pots are used throughout, acknowledging the unpopularity of dual concentrics. Another important nod to users' demands is the combination of a steeper rake and a shortened front-to-back 'reach'.

But the most important features are the *Europa*'s control facilities. These include a sophisticated VCA system, plus a noise gate, 12 aux sends and fully parametric EQ on every input channel. The Input module has Soundcraft's pad-less mic/line gain control and a post-fade routing matrix to eight group buses and left, right or mono mix buses. Equalisation is 4-band fully parametric and the Q controls incorporate end-of-travel switches which select 'shelving' mode.

Evidence that R&D are freely exchanging ideas is provided by the welcome presence of switchable noise gates. Originally developed for the *3200* studio console, the gates have threshold, attack, range, release or hold time controls and a 'shut' LED. For extra control, the high- and lowpass EQ filters can be switched in or out of the gate's side-chain.

Each of the 12 aux send pots has an on/off switch in its cap, a status LED and the facility for time-saving global master selection of overall pre/post switching. Aux 12 also has an override switch that provides local Direct Out control for a multitrack send, leaving other channels' Aux 12 sends free for normal use.

On the fader panel there's a single set of assignment buttons, which, to aid clarity, serves both the eight VCA audio groups and the eight mute groups. To change any group 'live', the engineer presses VCA Write or Mute Write and LEDs indicate the channel's assignments.

The VCA facilities seem well thought-out. Channels can be assigned to either single or multiple VCA group masters, with a full solo facility for each group, while the VCA Grand Master section can assign any VCA group to the Grand Master fader for 'one touch' level control. Other overall operations include Global Cut and Global Mute and further facilities such as Solo-In-Place and Solo Clear (the latter cancelling all PFLs and solos at once).

The Audio group module provides a fader, bargraph meter, cut, PFL and panpot for each of the eight groups, along with matrix routing. Each group has a mono or stereo input/return with width control, 2-band EQ, two pre/post aux sends and bargraph metering. This module also houses the eight VCA masters and eight of the 12 aux masters, with the other four located on the adjacent master module.

With LED ladders provided at every essential location it's a simple matter to check the desk's gain structure throughout any signal path.

Talkback and oscillator routing is comprehensive and protected by numerous 'safety' features. For instance, when talkback is fed to Group, Matrix is cut, preventing crew talkback being fed to the foyer. Similarly, Master Live cuts all talkback, oscillator and solos at a stroke.

The Monitor section provides the control room facilities that theatre users will expect and inter-console linking is provided for with balanced audio I/O and opto-isolated control lines.

Module options include dual matrix modules with eight group level inputs, L, R and mono buses and a mono external input, and stereo input modules with similar facilities to the standard version but just a 3-band sweep EQ.

The alloy frame is stiffer and lighter than the series *4*'s (as well as offering an easier reach to the back), while shorter ribbon bussing and screening plates aim to minimise crosstalk and make the board 'environment-proof'. Connection facilities include space for multiways of the customer's choice.

*Europa* will list at around £30,000 for a 40-channel console.

# TAC SR6000

TAC are among the most recent arrivals at the lower end of this price sector. Their *SR6000* console, launched in May, is pitched specifically at the *PM3000* market at a listed £25,000.

How had the console fared since its introduction? Product manager Carl Reavey: "It's had, generally, a very positive response. Two things seem to stand out more than anything else: one is an incredibly simple item, the provision of a multipin interface panel; the second thing a lot of people comment on is the fact that there are plenty of outputs, which makes it more versatile."

Despite its price and compact dimensions, the *SR6000* offers a quite comprehensive VCA grouping and muting system. Unusually for a desk in this market, it also has a single VCA output group with a master fader. Thus all 10 outputs can be individually assigned to one overall fader, without affecting the relationship between them.

On the input modules, there's an equaliser that's based on the Amek *M2500* design, with two parametric mids and swept HF and LF bands, plus a variable highpass filter control with integral on/off switching and warning LED.

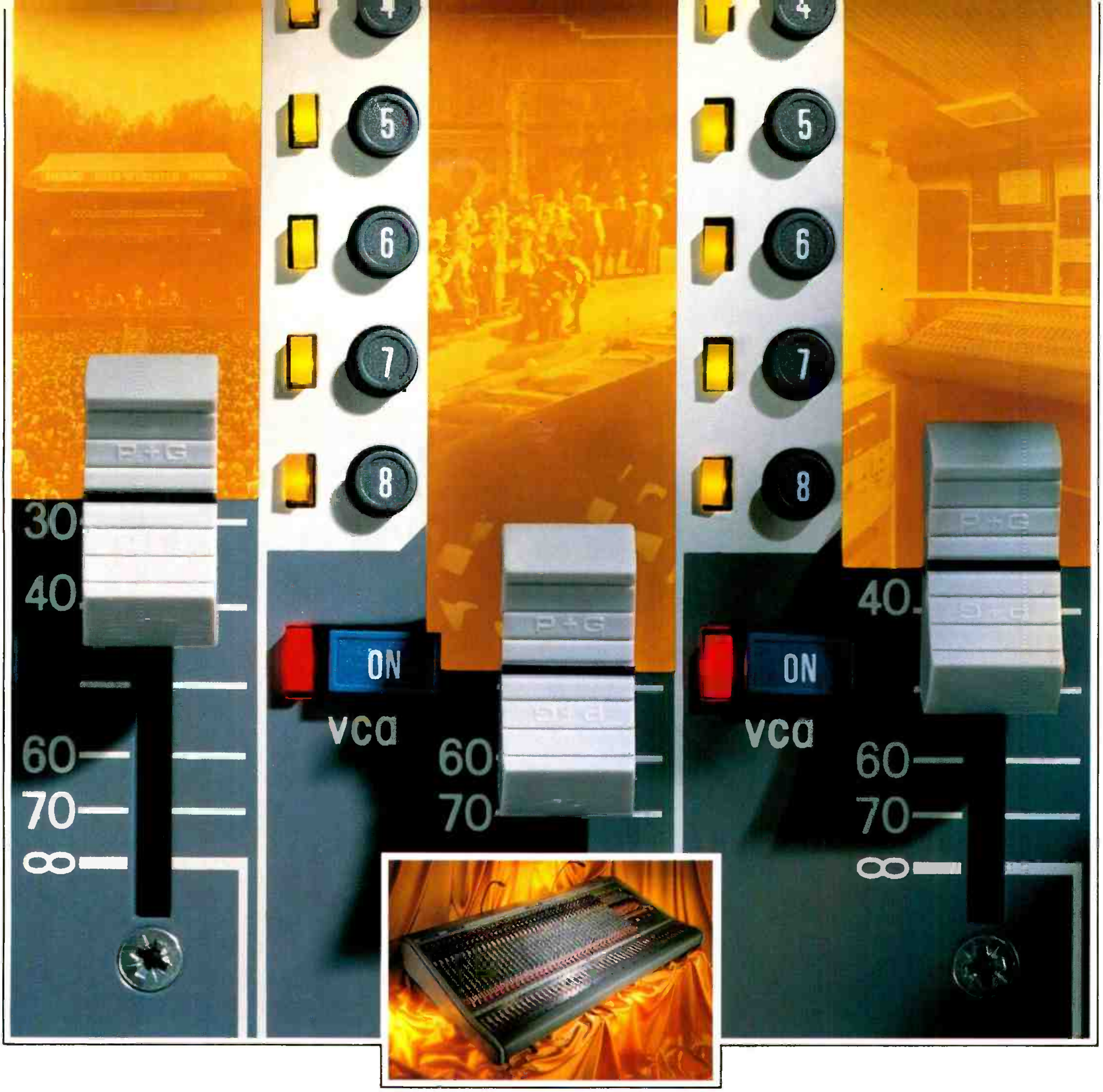
Eight auxiliary sends are provided in the shape of two dual concentrics (configurable internally as stereo pairs) and four single rotaries (switchable pre/post in pairs or individually). Lower down the channel strip are the post-EQ, pre-fade input LED ladder meter, a mute switch, audio routing to the eight audio subgroups and VCA mute grouping to the eight VCA mute groups.

Whatever the frame size, the console's master group section is always located centrally. It accommodates the split auxiliary system and is flanked by identical aux master modules. Each aux master section has its own outputs—a total of 16 in all. These work as standard auxes in 'normal mode', taking the same mix from either side, or can be 'split' to allow the left and right aux masters to control signals from their own sides.

Audio groups are provided as standard on rotaries although engineers who prefer to see their audio groups on linear faders have the option to hit the fader reverse switch on each group module—which swaps the audio group rotary with the matrix output fader. In this section, too, are the matrix outputs with an 8/8 configuration.

It is a compact design, which looks neatly laid out and feels sturdy. Internally, TAC has opted for a steel frame with the rigidity necessary to protect its hard bussing system, designed to reduce signal path length and noise. The matt grey finish is complemented by curved tubular rails at either end which provide grab handles. It's also lighter than some of the company's previous consoles.

To date, the *SR6000* has won sales from NBC in Chicago; Texan jazz club Caravan of Dreams; Hibino Corporation, Japan's top SR company; Norwegian SR firm Nordtek; Herman Scheureman's EML in Belgium; and SSE Hire in the UK. Two more deals were due to be signed at the end of May.□




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.

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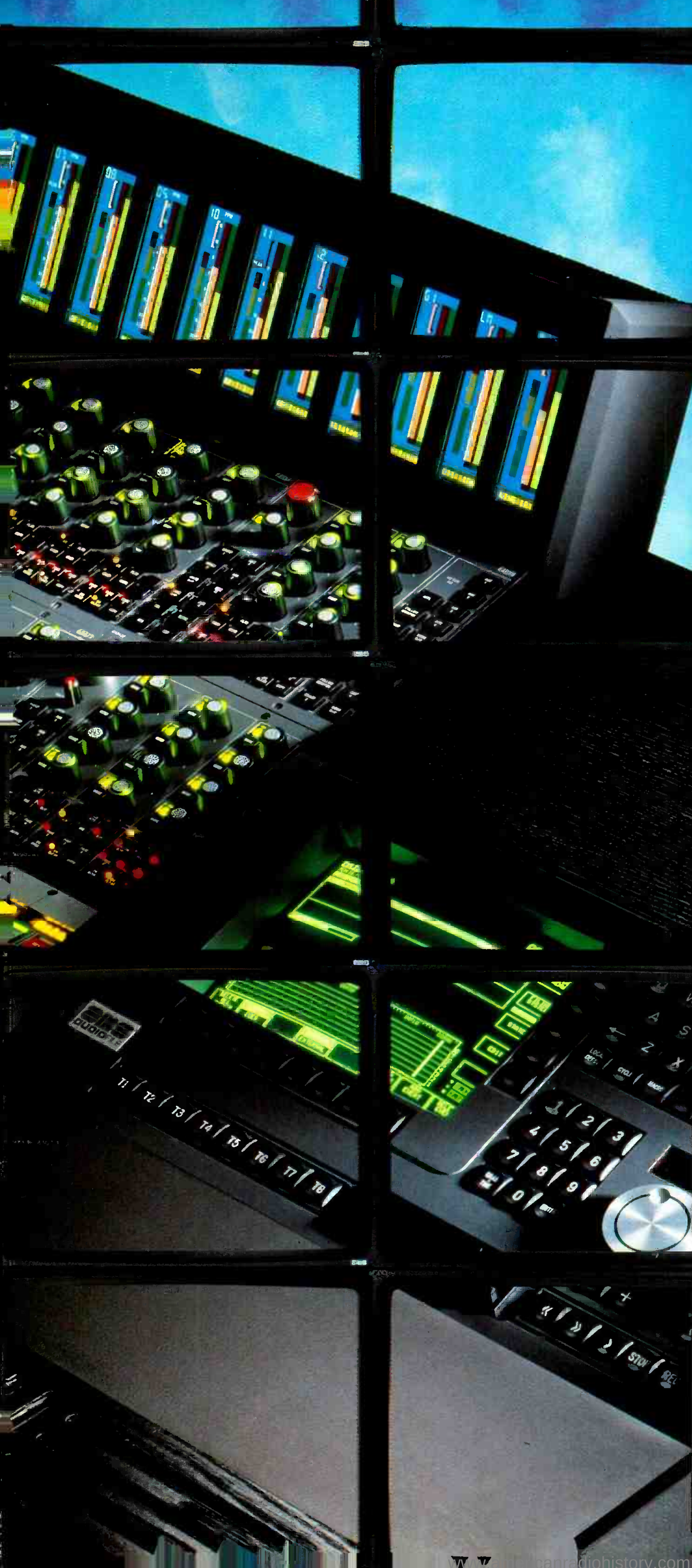
WHY TELEVISION SOUTH WEST SW

# CHED OVER TO THE LOGIC SERIES



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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.

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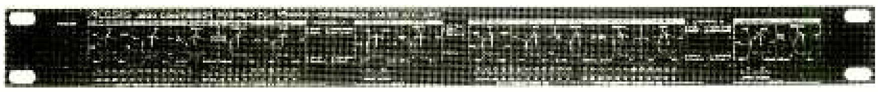
# DYNAMICS

## A list of new compressor, limiter and gate products introduced in the last year, from information available at the time of writing

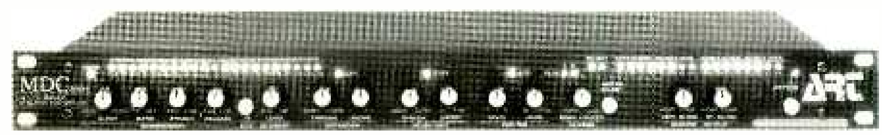
**Alesis:** The *3630* compressor is a dual-channel compressor/limiter featuring switchable RMS/peak compression styles. Each channel has independently built-in noise gates with threshold and delay controls. The *3630* uses the dbx VCA chip for improved low noise compression making it suitable for all studio recording, broadcasting, musical instrument and live sound reinforcement applications.
**Alesis, 3630 Holdrege Avenue, Los Angeles, CA 90016, USA. Tel: (213) 467-8000. Fax: (213) 836-9192.**
**UK:** Sound Technology plc, Letchworth Point, Letchworth, Herts SG6 1ND. Tel: 0462 480000. Fax: 0462 480800.



**Amek:** Amek have introduced a dynamics package that allows each channel of their *Mozart* console to be fitted with a software-controlled compressor, limiter and gate. The hardware is available in groups of eight channels. Several types of gate and compressor can be selected for each channel from a variety of screen displays. All the usual dynamics control parameters are adjustable and are stored with the mix allowing for instant reset when returning to a previously mixed track.
**Amek Systems & Controls Ltd, New Islington Mill, Regent Trading Estate, Salford M5 4SX, UK. Tel: 061-834 6747.**
**USA:** Amek Consoles Inc, 10815 Burbank Boulevard, North Hollywood, CA 91601. Tel: (818) 508-9788. Fax: (818) 508-8619.

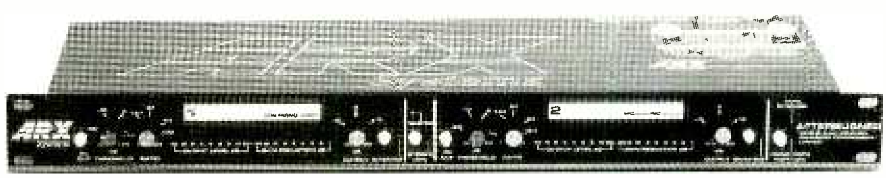
**Aphex:** The *Dominator II* is a stereo multiband limiter that has been designed to cover a wide variety of applications in recording, broadcast, mastering and post-production with the unit being available in two models—the *720* for general use and the *723* for broadcast and transmission applications. The unit is designed for minimum audibility of the limiting actions and therefore greater loudness. Features include a trimmable peak ceiling over a 34 dB range, switchable crossover frequencies and an automatic limit threshold function.

The *320 Compellor* features dual mono operation.
**Aphex Systems, 11068 Randall Street, Sun Valley, CA 91352, USA. Tel: (818) 767-2929.**
**UK:** Sound Technology plc, Letchworth Point, Letchworth, Herts SG6 1ND. Tel: 0462 480000. Fax: 0462 480800.




**ART:** The *MDC2001* stereo compressor/limiter/de-esser/exciter/expander/gate offers two channels of stereo processing. The compressor features independent control of all functions; the expander and gate functions can work separately or be used in tandem. The expander offers a threshold activation level and a slope/ratio control. The gate section has its own threshold and speed/hold controls. The master limiter/clipper serves as an absolute gain threshold that cannot be exceeded.
**ART, 215 Tremont Street, NY 14608, USA. Tel: (716) 436-2720. Fax: (716) 436-3942.**
**UK:** Harman Audio, Mill Street, Slough, Berks SL2 5DD. Tel: 0753 76911. Fax: 0753 35306.

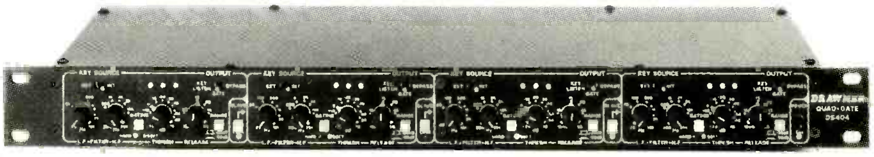
**ARX:** The *Afterburner* is a dual-channel enhanced compressor/limiter that can be configured as a mono dual-band unit, which allows separate dynamics control of low and high frequencies. The Enhance feature allows for frequency correction to preserve the spectral balance of the audio system and to compensate for the sagging low and high frequency response of some loudspeaker systems.
**Audio Research & Technology, 33 Advantage Road, Highett 3190, Victoria, Australia. Tel: (03) 555 7859. Fax: (03) 555 6747.**



**BSS:** The 1U *DPR-404* comprises four channels of subtractive compressor/limiter and HF de-esser facilities. Each channel includes a variable ratio compressor/limiter that features BSS's proprietary 'progressive knee' characteristic. Additionally each channel also includes an advanced HF de-esser previously only available as a dedicated mode on their *DPR-402* dynamics processor.
**BSS Audio, Unit 5, Merlin Centre, Acrewood Way, St Albans, Herts AL4 0JY, UK. Tel: 0727 45242. Fax: 0727 45277.**
**USA:** AKG Inc, 1525 Alvarado Street, San Leandro, CA 94577. Tel: (415) 351-3500. Fax: (415) 351-0500.

**Citronic:** Citronic have added a 2-channel expander/compressor/limiter, the *SPX3-51*, to their range of signal processors. Each channel features separate controls with channels able to operate in stereo mode, discrete 2-channel or as slaves to another unit. Compression includes threshold, variable ratio, attack and release and variable-knee. On the limiting side the threshold is variable and there is a bypass switch.
**Citronic Ltd, Bowerhill, Melksham, Wilts SN12 6UB, UK. Tel: 0225 705600.**

**dbx:** The *363X* dual noise gate offers two channels of noise gating in a compact half-rack dbx *Performer* series package. Applications for the musician and semi-pro recording studio. The *160XT* compressor/limiter now has an update, improvements include the addition of *XLR* connectors; balanced input and output; and true power summing when two units are strapped for stereo.
**dbx Professional Products division of AKG Acoustics, 1525 Alvarado Street, San Leandro, CA 94577, USA. Tel: (415) 351-3500. Fax: (415) 351-0500.**
**UK:** AKG, Vienna Court, Catteshall Wharf,



Catteshall Lane, Godalming, Surrey GU7 1JG. Tel: 0483 425702.

**Drawmer:** The *DL241* dual auto compressor combines a wide range of operational parameters with sidechain automation. Features include expander/gate; soft-knee compression with adjustable threshold and ratio with manual or fully automatic attack and release times; stereo linking can be used to prevent image shifting when processing stereo signals; and a fully balanced hard-wire bypass connects the input directly to the output allowing the signal to pass through the unit with no power applied. The *DS404* quad gate is a 4-channel device offering the choice of hard or soft gating. Chain linking allows all four gates to be linked if required. Each channel is also equipped with variable release, key filters, external key input, choice of 20 dB or 90 dB attenuation and balanced *XLR* inputs and outputs.
**Drawmer Distribution, Charlotte Street Business Centre, Charlotte Street, Wakefield, West Yorks WF1 1UH, UK. Tel: 0924 378669. Fax: 0924 290460.**
**USA:** Quest Marketing, 15 Strathmore Road, Natick, MA 01760. Tel: (508) 650-9476. Fax: (508) 650-9444.

**Electro-Voice:** E-V have introduced the *COL-1* single-channel compressor/limiter, intended primarily for sound reinforcement and paging systems where transient protection and level control is desired, rather than as an audio production effects device.
**Electro-Voice Inc, 600 Cecil Street, Buchanan, MI 49107, USA. Tel: (616) 695-6831. Fax: (616) 695-1304.**
**UK:** Shuttlesound Ltd, 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX. Tel: 081-640 9600. Fax: 081-640 0106.

**Klark-Teknik:** The *DN-504* 4-channel compressor/limiter is a 1U 19 inch rackmount unit offering threshold, ratio, attack, release and output level for each channel. The compressor can be switched between hard/soft knee operation on each channel and there is an auto-mode or full manual operation. The channels can be linked as two stereo pairs. LED metering of gain reduction and output is included on each channel. Noise figures of less than −94 dBm 20 Hz to 20 kHz unweighted are quoted.
**Klark-Teknik Research, Klark Industrial Park, Walter Nash Road, Kidderminster, Worcs DY11 7HJ, UK. Tel: 0562 741515. Fax: 0562 745371.**

**Orban:** the *4000A* transmission limiter has been designed for totally transparent control of peak modulation levels for transmission on analogue or digital microwave or satellite links, telephone lines or on-air broadcast.
**Orban division of AKG Acoustics.** (See dbx entry for address details.)

**Saturn:** Saturn Research have launched the *X24*, a 24-channel variable threshold expander. The *X24* provides 24 noise gates in a single, self powered 19 inch rack unit. The *X24* noise gates work on the expander principle and the product has been developed specifically as a single-ended noise reduction system for routine studio and SRS noise reduction tasks.
**Saturn Research Ltd, Unit 3A, 6-24 Southgate Road, London N1 3JJ, UK. Tel: 071-923 1892. Fax: 071-241 3644.**

**Sony:** *Version 2* software for the *SDP-1000* digital multi-effector includes an improved dynamics processing section that features and additional dual compressor mode and other enhanced facilities including a panpot function; auto fader; event rehearsal and a window help display.
**Sony Corp, PO Box 10, Tokyo AP, Japan 149.**
**UK:** Sony Broadcast & Communications, Jays Close, Viables, Basingstoke, Hants RG22 4SB. Tel: 0256 483366.
**USA:** Sony Corp of America, Professional Audio Division, Sony Drive, Park Ridge, NJ 07656. Tel: (201) 930-1001.

**Studio Magnetics:** The *SG 260* dual gate offers two channels with variable control of threshold, attack, hold and release times, and features a very fast minimum attack time. The unit is housed in a 1U enclosure with internal PSU.


**Studio Magnetics, Unit 4, Radfords Field Industrial Estate, Maesbury Road, Oswestry, Shrops SY10 8HA, UK. Tel: 0691 670193. Fax: 0691 670194.**

**Symetrix:** The *564E* offers four gates in a 1U rack unit—each gate will function either as a gate or downward expander depending on the position of the ratio control. Key listen along with high- and lowpass filters are included, and gain reduction indication is via six LEDs. A fast attack mode causes the gate to open in 50 ms, and specially designed sensors track signals as low as 20 Hz.

The *SX-208* is the newest addition to the *200* series half-rack audio devices. The *SX-208* is the world's first stereo compressor/limiter in a half-rack form. The device has compression ratio and output gain controls and a design, which combines the best of both peak and RMS level detection for accurate response to any level changes.
**Symetrix, 4211 24th Avenue West, Seattle, WA 98199, USA. Tel: (206) 282-2555. Fax: (206) 283-5504.**
**UK:** Sound Technology plc, Letchworth Point, Letchworth, Herts SG6 1ND. Tel: 0462 480000. Fax: 0462 480800.□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# DIGITAL AUDIO PROBLEM SOLVERS

## With the variety of standards prevalent in digital recording chains, Francis Rumsey suggests a few, problem solving devices and their practical uses

Digital audio brings with it the benefits of infinite-generation copying without signal degradation (as long as you use oxygen-free copper cables, of course!). And having invested in all that expensive digital recording and perhaps even mixing equipment it will pay you to keep the signal in the digital domain for as much of the time as possible. This, it must be said, is a less straightforward matter than might at first appear, and from various articles in *Studio Sound* over the past year it is probably clear by now that getting devices to work together digitally may require some intervention by the user—it often being more than a matter of taking a simple cable from one device to the next, especially if the two devices come from different manufacturers.

If discussions on signal synchronisation and the principles of digital interfaces leave you cold, perhaps the following practical suggestions will provide some satisfaction. There is now a collection of digital 'problem solvers' available on the market and, although each offers a selection of common 'fixes', there is considerable variation between the functions of these 'black boxes'.

## The nature of the problem

A host of potential barriers to correct communication exist. It may be that source and destination have different sampling rates, or perhaps even if they have the same nominal rate they are not locked to the same reference and thus drift in relation to each other. Further, it is feasible that the sender will be a consumer device and the receiver professional, having different electrical and data formats. The sender may use more or less bits per sample than the transmitter, so what do you do with the extra bits or how do you make up the missing ones? The sender may use pre-emphasis, or there may be a DC offset in the signal, or perhaps the level of the signal is too low. The recording on the sender may be accompanied by timecode, which is asynchronous.

As Audio Digital Technology's recent advertisement drily notes: 'Standards are wonderful—that's why we have so many of them'.

## Handling different sampling rates

Three, perhaps four, common sampling rates exist in sound recording equipment—32 kHz (for broadcast distribution and transmission), 44.056 kHz (for PCM adapters locked to NTSC video), 44.1 kHz (the CD rate and a common everyday recording rate) and 48 kHz (VTR's, consumer DAT machines and others). Source and destination must be operating at exactly the same rate for signals to be copyable between one machine and another, and even the slightest deviation from any of these standard rates will result in anything from minor signal glitches to complete non-communication.

If two devices with different sampling rates need to be interconnected then a sampling rate converter must be used. This is a little like using a standards converter to convert American NTSC television signals at 29.97 frames per second (f/s) to PAL pictures at 25 f/s for Europe. Sampling rate converters usually have a selection of input and output signal formats and can convert between any of the common rates quoted above, some being continuously variable over the whole frequency range. Examples of the most widely known converters are those from Harmonia Mundi (part of the modular *bw102* system and converting from 44.1 to 48 kHz or *vice versa*), from Sony (any rate from 29 to 52 kHz) and from DAR (as part of the *DASS 100* synchronising system—any rate from 29 to 52 kHz).

Sampling rate converters do their job by calculating new but mathematically accurate samples at the output rate from samples at the input rate, and although this is not a 100% transparent process from the point-of-view of sound quality, the side effects in modern converters are minimal. DAR claim that the noise performance of their sampling rate converter is roughly equivalent to that of an 18 bit linear A/D converter for a full-amplitude 1 kHz signal, but that the noise floor always remains the equivalent of 18 bits below the signal, no matter what the signal level.

You would most probably need to use a sampling rate converter when connecting, say, a CD player to a digital VTR's audio tracks, since the VTR would be operating at 48 kHz and the CD player at 44.1 kHz. The sampling rate converter's output could be locked to the same reference as the VTR (either a video reference or an audio reference in the form of AES/EBU or WCLK). This is shown in **Fig 1**.

A sampling rate converter would also be required if copying between two devices whose sampling rates were nominally the same but which were at extremes of the allowed tolerances for sampling frequency—eg a consumer CD player and a professional tape recorder, as shown in **Fig 2**. The sampling rate of consumer machines can have a looser tolerance than that required of professional systems, thus the nominal 44.1 kHz output of your consumer CD player might be more like 44.098 kHz in practice. Alternatively, a synchroniser might be used in this case and this is discussed below.

In broadcasting, a sampling rate converter could be used between the 32 kHz of the distribution and transmission network and the 48 kHz of audio used in broadcast studio systems.

## 'Wild' signal to reference sync

Some consumer or semi-pro digital equipment runs 'wild'—that is, the sampling clock is internally generated and not lockable to a reference. As in the previous example of the consumer CD player, a nominal 44.1 kHz may in practice turn out to be slightly more or less than this, and any attempt to connect the output of that CD player to a digital system locked to a reference closer to 44.1 kHz will be met with glitches or a lack of communication.

A sampling rate converter could be used in extreme cases (see above) but if the difference between the rates of the two devices is small, a synchroniser could be the solution. DAR's *DASS* system is one of the only commercially-available synchronisers at the present time and it works by reading samples from the input into a large memory buffer (1024 samples) and reading them out again at the rate required by the destination. If the destination's rate is slightly higher than the source's, then the buffer gradually empties (since it is being emptied faster than it is being filled) and thus it must be automatically reset to the mid-point every so often. The same happens if the buffer approaches overflow (which would be the case if the destination ran slower than the source). The buffer reset is achieved by a simple 10 ms crossfade between samples at the current- and mid-points of the memory.

In such a synchroniser, input samples are carried through to the output without modification nearly all of the time, except when a crossfade takes place and thus sound quality is unaffected. Four audio signals within AES

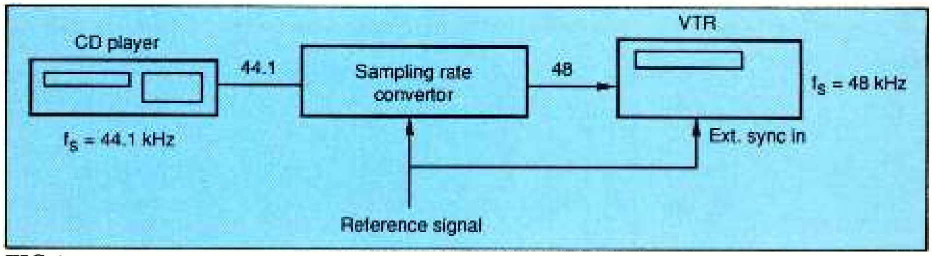


FIG 1

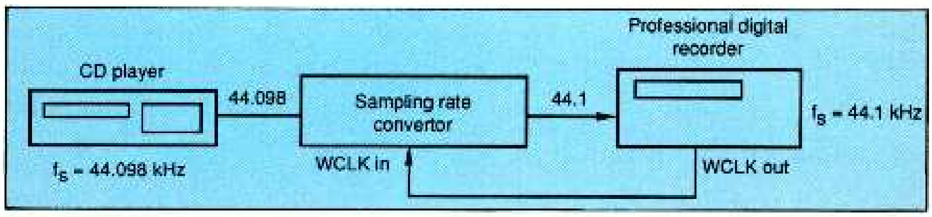


FIG 2

FIG 3: ADT FC-1 digital format converter

tolerances for sampling rate (that is 10 parts per million), a crossfade would only be necessary once every 20 minutes maximum with a buffer of 1024 samples. For signals less far adrift the crossfade would occur less often. A synchroniser would also offer the advantage of stabilising a dirty input signal from a poor quality source, whose clock might be noisy. Since the synchroniser's output is normally referenced to the professional system's clock, any problems with sound quality which might have arisen from the dirty clock affecting the destination's converters are avoided.

One of the other interesting features of the *DASS 100* synchroniser is its ability to lock to a wide range of references—AES/EBU, wordclock, mixed video syncs and even longitudinal timecode (SMPTE/EBU). The timecode reference can be useful when copying tapes whose timecode is asynchronous with the recorded audio. In this mode the timecode is used to reference a new stream of audio samples that is synchronous with the timecode, thus allowing the recording to be copied to a disk-based editor or digital VTR, for example, which expects a synchronous relationship between audio and timecode addresses.

# Handling data format differences

Many of the commercial problem solvers handle conversion between different audio data and electrical formats. ADT's *FC-1*, for example, has inputs and outputs for SDIF-2, SPDIF and AES/EBU, and will convert from any one to any other. It will also allow the user to modify certain of the channel status bits (these are the additional control bits that accompany the audio data) since the destination may require them to be set in a specific pattern for correct communication. The *FC-1*, pictured in **Fig 3**, allows the user to change consumer format data to professional format data, and to manually set the pre-emphasis and copy inhibit flags. This makes it possible to alter incorrectly-set flags in source data, since there have been examples of cases where source data has pre-emphasis but no flag, or has no pre-emphasis but has the flag set. The box will also de-emphasise pre-emphasised source data if required.

Audio Design's collection of *ProBoxes* also come into their own in this application. The *ProBox 1* converts consumer digital sources to professional AES/EBU format, eliminating copy inhibit flags if required—a great boon for users with SCMS DAT machines. The box will accept both coaxial and optical SPDIF inputs. The *ProBox 3* converts between AES/EBU and Sony's SDIF2 in both directions simultaneously, which has uses for interfacing Sony equipment to other manufacturers' systems.

For those with serious interface incompatibilities Prism Systems have developed a PC-based analyser for AES/EBU signals (the *DAS-90*), which displays channel status information on a continuously updating basis, allowing the user to determine the differences between the source's implementation and the destination's. An analogue audio output is also provided to monitor the incoming signal. A similar feature is offered on DAR's *DASS 100*, where a page can show the state of all channel status bits, and allow any of them to be modified until the source and destination talk properly. In addition to these two, RE Broadcast have recently introduced a digital audio analyser, the *d930*, designed to allow comprehensive analysis of AES/EBU data streams, with the possibility for monitoring and modifying control and user data, as well as showing the activity of audio sample data. RE have also introduced another most useful product—the digital headphones *(d940)*—which may be plugged into an AES/EBU line to monitor the audio signal coming down the same!

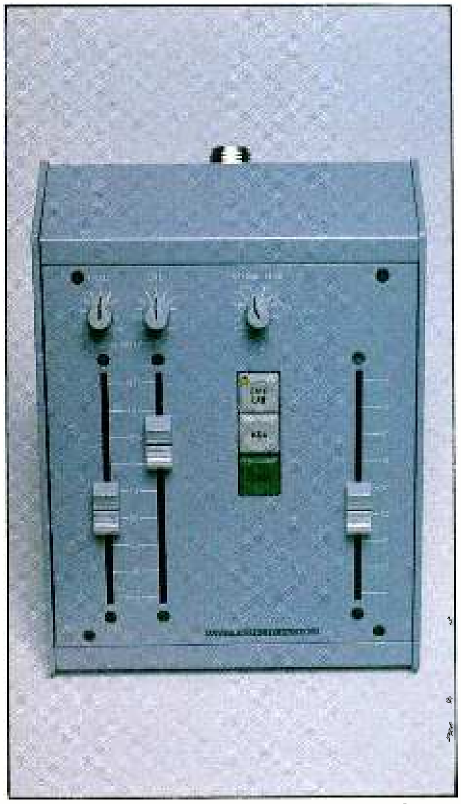

FIG 4: ADT FC-1 remote fader panel

# Signal mods en route

Assuming that communication can be established, it may be necessary to modify the signal to some extent in the process of transferring it between devices. This might, for example, take the form of level correction, DC removal, highpass filtering, channel swap, phase reversal, mixing, or dithering.

Harmonia Mundi, ADT, Audio Design and DAR all offer facilities for gain correction in various forms. Harmonia Mundi's module allows for up to 10 dB of gain and infinite attenuation, as well as phase reversal of either or both channels, while Audio Design's *Level ModeDefier* has two gain scales, either with unity or +12 dB gain at the top, as well as a ±6 dB channel balance control, overload indication and phase and channel reversal.

DAR, Harmonia Mundi and ADT allow for a pair of stereo inputs to be mixed together into a single stereo output. ADT's *FC-1* can be supplied with a remote fader panel (see **Fig 4**), having two faders and two channel balance controls, each of which controls one stereo signal path. An automatic crossfade can be programmed between the two over any duration between 10 ms and 10 secs. 10 dB of gain is available from the *FC-1* and the Harmonia Mundi, and 15 dB from the *DASS 100*.

The ADT *FC-1* and the Harmonia Mundi re-dithering module are capable of re-dithering the output signal to make it suitable for one of a number of destination sample resolutions, and this can be useful when copying, say, a 20 bit master to a 16 bit recording format for CD mastering. If a 20 bit recording is simply truncated to 16 bits for CD mastering it will suffer in sound quality but this can be ameliorated by introducing digital dither into the signal that is correct for 16 bit quantisation. The *FC-1* allows for signals to be re-dithered to 16, 18 or 20 bit resolutions, according to the algorithm devised by Lipshitz.

Another form of signal modification commonly required is pre- and de-emphasis, since some recordings and broadcast signals may be supplied in a pre-emphasised form, or may require emphasising prior to output. There are two main forms of pre-emphasis—the audio recording standard of 50/15 μs, and the broadcast standard of CCITT J17 (used in NICAM 728 transmissions). The *FC-1* is apparently the only device to incorporate J17 de-emphasis.

DC blocking is often useful for removing offsets in converters that often arise in cheaper systems, or misaligned professional systems. This is otherwise known as highpass filtering and simply prevents the passage of DC through the system, allowing everything above 1 Hz to pass. The *FC-1* allows also for HP filters rolling-off at 20 Hz and 100 Hz.

# Conclusion

The foregoing examples don't represent all the possibilities for digital problem solving that may arise in everyday work but they should have aroused some interest in the type of operations that can be carried out without having to convert an audio signal back into the analogue domain. As the all-digital signal chain becomes more of a reality, devices such as these will grow in importance.□

# ROAR RECORDING

## The subject of audience recording is largely undocumented. Simon Croft outlines some practical tips gathered from engineers in fields ranging from TV quiz shows to heavy rock albums

When presented with a star-studded show, it is tempting to regard the audience as a fairly low priority. But the way in which an audience is recorded can be critical to the usability of the material. Audience reaction is the vital ingredient in recreating a live entertainment event for the home listener. It is 'part of the excitement' and contains many of the emotional cues that help generate a feeling of involvement.

Conversely, if the rapport between the performers and the audience is not captured and presented correctly, a negative message will be sent to the listener. If a comedian delivers his best line to virtual silence, the impression is that the joke died. It may be that the audience was simply removed from the mix but effect is not conducive to enjoyment.

Sending out the right message involves more than an accurate transcription of the event. As is the case with a musical balance, the mix often has to exaggerate the basic ingredients in order to give a better impression of 'being there'.

This kind of manipulation becomes difficult if audience coverage has not been well thought out. For instance, undue spill from the sound system can severely limit the room for manoeuvre in post-production. Similarly, very close miking can reduce the apparent size of the audience from a crowd to a discernible number of individuals.

## Measurement

It is perhaps surprising but there is an almost total lack of available documentation on the nature of audience and crowd noise from the perspective of sound recording. The small amount of data available has been compiled in the main to enable the specification of safety and emergency warning systems. For such applications, it is obviously important to ensure that the sound system remains audible above the SPL of the crowd.

Travers Morgan are specialist contractors who installed the SR system at Wembley Stadium, London's largest sports and concert venue. John Staunton, an acoustic consultant at the company says that measurements taken 13 metres away from a Wembley soccer crowd of 80,000 have revealed levels up to 110 dBA (**Fig 1**).

Measurements were taken using a short $L_{eq}$ of 1 second and Staunton points out that peak levels were unlikely to last more than 4 seconds in reality.

Using the same $L_{eq}$ the minimum recorded level was 70 dBA. To give some idea of average level, the stadium's local council, Brent, have made measurements using a 15 minute $L_{eq}$ obtaining readings between 85 dBA and 93 dBA.

Travers Morgan have also taken measurements from within an 80,000 crowd at a rock concert and obtained peak levels of 104 dB.

Typical figures for the other end of the scale are given by Klark-Teknik in their reference work, *Audio System Designer*. Ambient noise in a concert hall or theatre is given as 30 dBA to 35 dBA. For a cinema, 35 dBA to 45 dBA. An individual is said to produce around 75 dBA when shouting.

Data on the spectral content of audience sound is not to be found in the public domain. The nearest rule-of-thumb one can make is to observe the similarity of applause to electronically generated random sound, particularly pink noise.

## Atmosphere

As a music producer at the BBC, Anthony Pugh is interested in the audience's effect on the finished programme.

"You need to introduce more audience as the gig progresses, to build the level of excitement," he explains. If this isn't done, the impression to the listener is anti-climactic. Pugh has known concerts where the audience has made very little noise because it is completely spellbound. This is not a state that communicates across the airwaves, so audience reaction has to be exaggerated to appear realistic.

To illustrate, Pugh plays an SSL mix of a folk concert. The reaction of the audience on recognising the song sounds entirely natural but watching the VCA levels reveals a significant amount of boost to the applause. Playing the same mix with static audience levels simply does not work. A few bars into the song there is noise from the crowd but the significance is no longer clear: it could even be a fight breaking out.

Although it is possible to replace complete sections of applause, Pugh cautions that it is not as straightforward as it appears. Having recorded a jazz festival over several days, he knows how different audiences can sound from night to night. "It was the same venue, the same EQ, the same mics and even the same artists, sometimes, but the sound was completely different."

Humidity, quality of performance and quantity of alcohol are some possible variables.

## Orchestral

As a senior recording engineer with BBC Transcription, Dave Mulkeen has practical experience that ranges from the Stones to Sibelius and frequently covers jazz and folk as well. BBC Transcription differs from other parts of the Corporation, in that it operates on a commercial basis, making radio programmes for worldwide distribution. It offers an SSL-equipped mobile to outside users and it also takes internal contracts to cover events for BBC Radio and Television.

In Mulkeen's experience, orchestral recording is the "complete opposite" of rock recording. The nature of the event means that the orchestra and audience have to be treated as a cohesive whole, rather than miked separately.

"There are so many techniques for recording a symphony orchestra but they all probably involve a stereo pair somewhere up behind the conductor's head. And, to a greater or lesser degree, close mics as well," Mulkeen observes.

In addition, there will be a pair of ambience microphones, much further back in the auditorium. These microphones will also be used to pick up the audience.

Once a concert has started, Mulkeen's general advice is, "Simply leave things alone. A serious music mix is much more static than a rock mix in terms of fader movement. If it's a string quartet you'll probably only use the main fader, having set up. You will use the main fader to take the audience level back, because it is a lot louder than the music." A complete reversal of the rock technique.

The delicate balance between audience and orchestra is amply illustrated by BBC Transcription's innovative techniques for recording in London's Royal Albert Hall. Despite its popularity as a venue for serious music, it is a catalogue of acoustic challenges. The cavernous Victorian dome originally had such a long reverb time it was known as the only place in London where good music could be heard twice, and even the addition of suspended sound absorbing 'mushrooms' in 1969 left an $RT_{60}$ in the region of 3 seconds.

In addition, the audience virtually surrounds the orchestra. The circular seating plan covers about 280°, while six separate tiers extend up to the dome itself.

BBC senior engineer Gareth Watson developed a device called the 'Flying Saucer' in order to gain some control over audience spill. Watson's invention is a clear acrylic disc with two *PZMs* stuck to it which achieves a directional

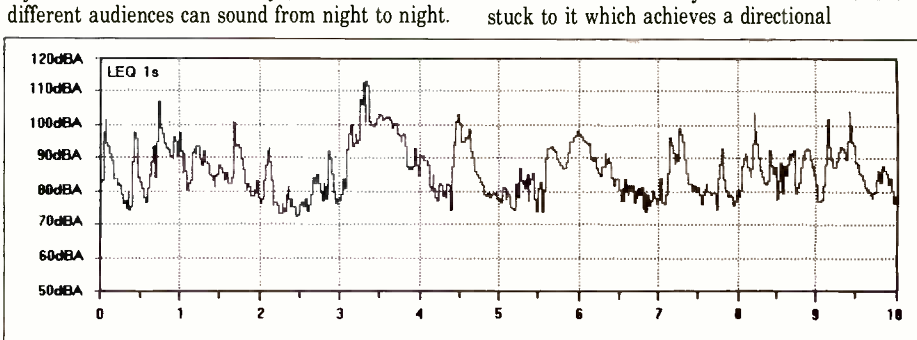


**FIG 1: Level/time plot of major international football match recorded at 13 metres**

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LYREC TR 532 + locator/remote
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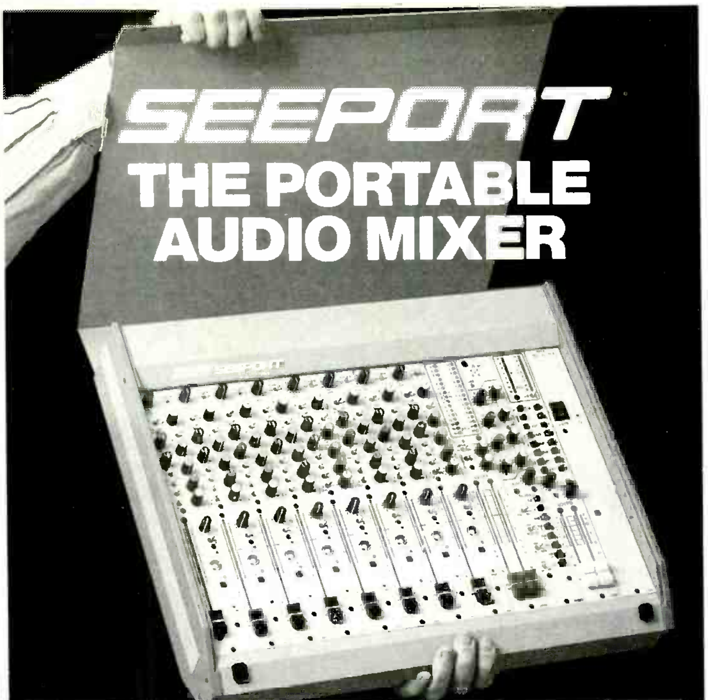


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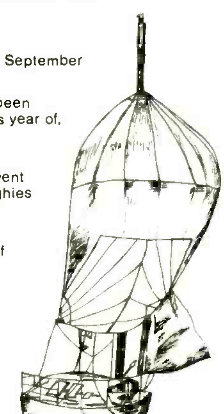

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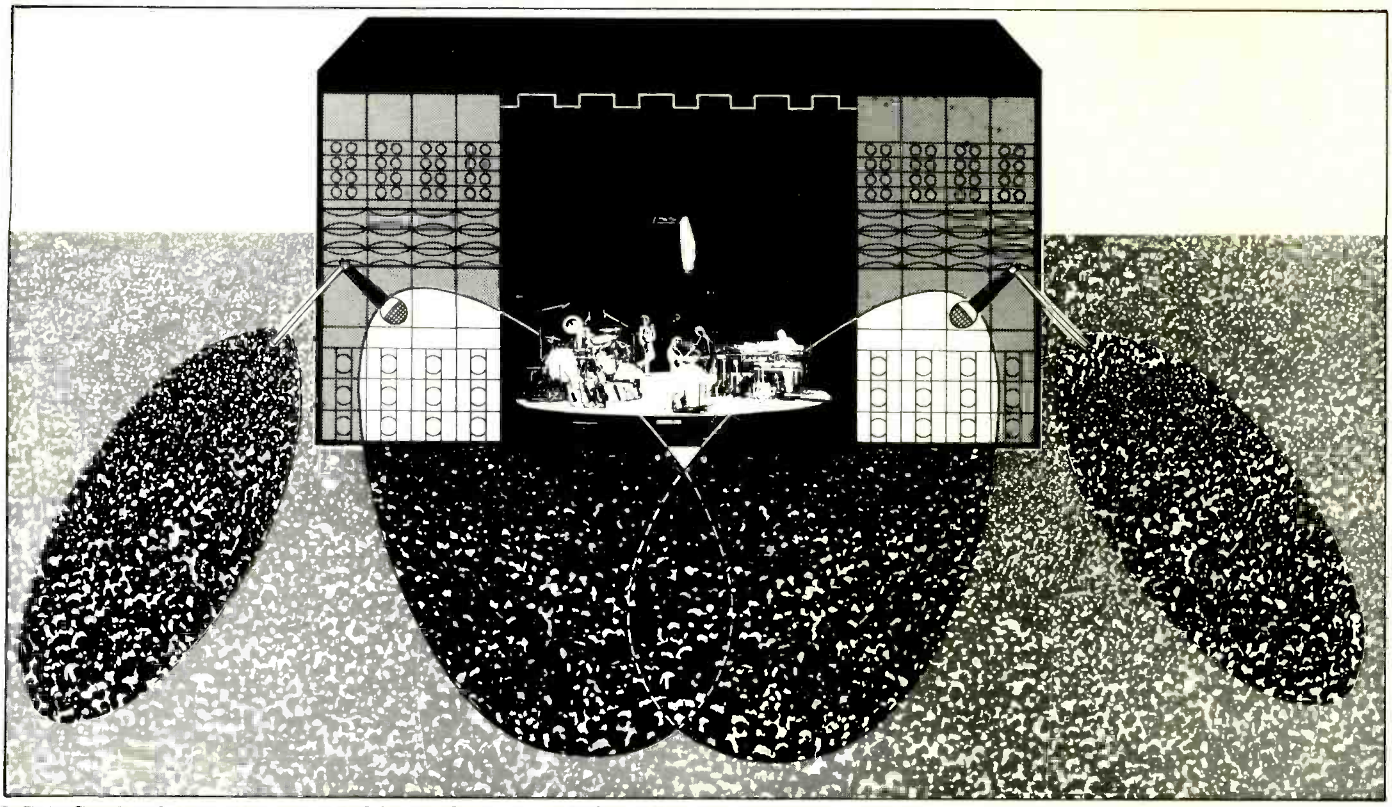

**FIG 3: Cardioid and rifle mic used in tandem on each SR stack for audience pick-up**

characteristic, while what is in effect a second disc provides separation. The device is suspended from the apex of the dome and lowered to just below the level of the 'mushrooms' (**Fig 2**).

Pete Freshney, operations manager, BBC Transcription says the Albert Hall is "a funny one" because the acoustic is "almost the same empty to 80% full but the difference between 80% and 100% is enormous; suddenly, you haven't got an acoustic.

"The audience is everywhere: the reason for the Flying Saucer is that you can angle it in a certain direction and get rid of the audience."

Alternative approaches to ambience/audience miking within the BBC include attaching a pair of *PZMs* stuck to pillars at the rear of the hall. Mulkeen notes, "BBC Radio tend to use a couple of cardioids on a sling between one side of the gallery and the other."

# Rock

Producer Chris Tsangarides recorded one of the all-time classic live albums, *Live and Dangerous* with Thin Lizzy and is currently adding to his long list of credits with a new Iron Maiden album. He has also recorded mainstream vocalists Tom Jones and Billy Ocean in concert.

For audience mics, Tsangarides still tends towards Neumann *KM84*s or Schoeps for crossed pairs, although he has also used *U87*s on the speaker stacks and *PZMs*, depending on the situation.

"Recording the audience in big halls, the normal practice is to stick a couple of mics either side on top of the speaker stacks and then a pair up by the mixing desk or in the balcony. You just wing it actually. Each hall is different.

"Sometimes I've managed to climb into the lighting rig and hung up a stereo pair directly above the stage, centre out. That works well."

Mics at the mixing desk will be about 9 ft up on booms. Each mic will go onto a separate track so there is some leeway during mixdown.

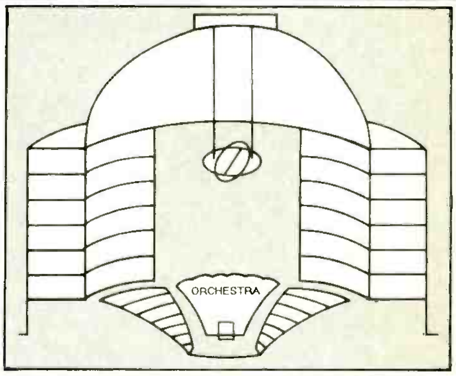


**FIG 2: Position of 'flying saucer' at the Albert Hall**

"I set the level of the audience mics and they stay on through the mix. The audience is pushed up at the end of a song and then back down to its level, so the band sound good. If there are any whoops and yells, you help them along.

"A good little trick is sometimes to throw the phase between the two pairs, the PA microphones and the mixer microphones. That gets rid of the boom."

Due to the distance between the two pairs, there is no danger of creating phase anomalies between the audience recordings.

"One good thing with a rock band is that if they have crowd participation, it's normally a breakdown section, the drums are going and it sounds great anyway because of the whopping great ambience."

But the biggest favour the engineer can do the band is to persuade them to keep the volume at a manageable level.

"The main thing is, if you can get a band to have their monitors in check. But always, they will feed back on the second number. I think it's in the contract somewhere!"

When BBC Transcription senior engineer Dave Mulkeen records rock and pop he uses some similar techniques to Tsangarides but has some different approaches to miking and equalisation.

At the Hammersmith Odeon, which doubles as major rock venue and cinema, he will hang four Neumann *KM84*s over the balcony. The height is adjusted to "a compromise between picking up too much of the PA and, if you get too close, picking up individual claps".

As a practical consideration, Mulkeen also makes sure the mics are "well out of reach" of anyone standing on a seat, so he opts for a mic position somewhere over 10 ft.

Some venues will not allow sound equipment to be placed in the auditorium itself, so Mulkeen has to find other solutions. Like Tsangarides, he sometimes mounts mics on the SR stack.

"You would think they would pick up a lot of the PA but you can place them with the heads behind the wavefront and the only significant spill is through mechanical transmission. And that is bottom end, which you can EQ out."

Using a single rifle mic each side tends to create a 'torchbeam' effect so Mulkeen normally places a rifle mic, plus a cardioid facing inwards, on a stereo bar each side. This helps to fill in the 'hole in the middle' (**Fig 3**).

Mulkeen will generally EQ the audience mics. "Not to enhance the audience but to get rid of PA pickup and the sound of the hall. Halls always tend to honk around 100 Hz to 160 Hz. So I will put a notch in there and then anything below about 80 Hz is irrelevant."

The audience mics are then premixed without further treatment onto two tracks of a 24-track analogue Otari.

"The audience tracks are there for two purposes. One is for audience reaction, like applause between numbers. I also tend to use quite a lot of it to gel the sound. Even when the audience themselves are not making much sound, I like to pick up the hall.

"Having set up the mix, you can move the audience faders up and down and there is a point beyond which it starts screwing up the sound."

Mulkeen is in agreement with Tsangarides over stereo image: "Rock comes from mic splits and DI, so the sound is very clean and the perspective is

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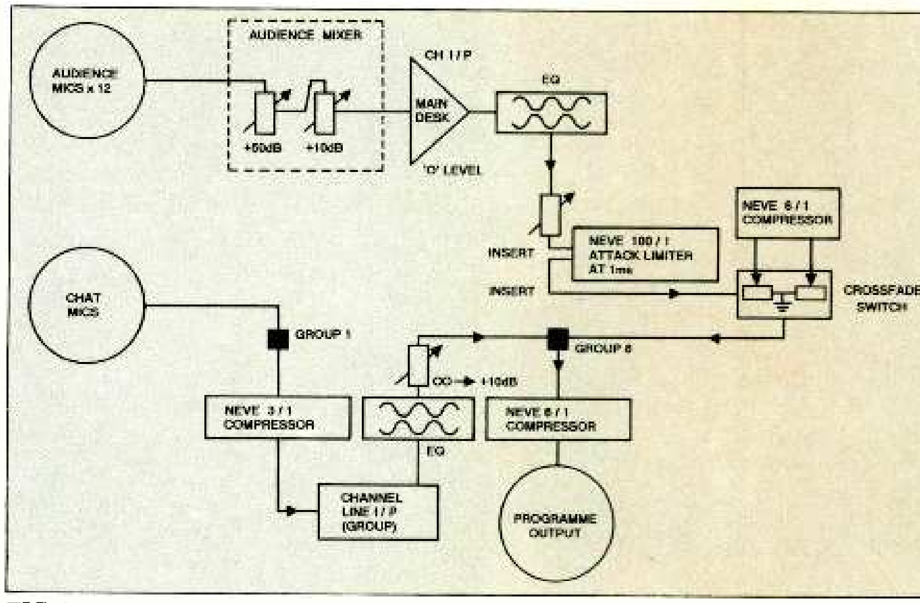


**FIG 4: LWT audience and chat mics signal chain**

artificially devised to sound as if you're looking at the stage. In using audience mics as ambience, you do have to match the two images, so that it does not sound as if you are jumping from one hall to another as you fade the audience up."

Mulkeen also suggests that two sets of audience microphones can be useful where audience participation is likely to be featured. In this scheme, the more distant microphones can be used for applause, while the close mics are brought in for 'sing-alongs'.

# Great outdoors

Outdoor concerts present some unique problems. Apart from the obvious need for adequate wind shielding, the roles and positioning of the audience mics change completely.

"When you bring up the audience, there is no ambience, so you keep it out of the mix until the clapping starts," Mulkeen notes. Paradoxically, the lack of acoustic boundaries means that electronic reverb has to be added to give some sense of space.

Paul Brady, another BBC senior engineer, has experienced the problem at the Cambridge Folk Festival, an annual event in the UK that attracts several thousand people to a rural site, which was never designed for concerts.

"The Folk Festival audience is enthusiastic and noisy but without any reverb, tends to sound smaller than it actually is. Two or three thousand people can sound as if only six bothered to turn up. It is going to sound small unless you have lots of mics, not too close to the audience."

Another problem outdoors is that there is nowhere to hang microphones, so boom stands are used and for practical reasons these tend to be kept to the edges of the crowd, where somewhere nearer the centre would be a more effective pickup point.

Cambridge also offers dramatic changes in acoustic, as Brady explains: "Half the event takes place in a large tent. When it's dry it's like working outside but if it rains that tent starts to work like a metal box."

In the protected environment of a BBC mobile, the first and most obvious sign of rain is howling feedback as the tent's now reflective walls play havoc with the front-of-house system.

# Light entertainment

Game shows, comedy and variety are part of the staple diet of every television channel. London Weekend Television has a successful track record in the genres, with many programmes transmitted on the national network as well as LWT's own area.

Now head of post-production at LWT, Graham Hix cut his teeth mixing live shows, like *Sunday Night At The London Palladium*, and has worked on many and varied programmes where audience reaction is a major component.

Hix carries a wealth of knowledge of hands-on mixing technique and is also able to explain the unique gain control chain LWT has evolved over the years. This has recently been reconfigured to meet the needs of stereo NICAM transmission but for the sake of clarity, Hix describes the original system before covering the adaptations (**Fig 4**).

"The system is not overload proof," Hix cautions. "There is an enormous amount of gain. The channel gains on the audience mixer are typically preset at 50 dB." With another 10 dB added at the fader, there is 60 dB of gain from 12 separate microphones.

In the studio, these are mounted on stands in two rows of six, covering the front and back of the seated audience at a height of about 8 ft.

The output of the audience mixer is fed without equalisation to a line input on the main Neve *V* series desk. (This and other parts of the signal path are of course doubled for stereo.)

The programme then receives "a reasonable amount" of 10 kHz and 3.6 kHz lift and probably 150 Hz highpass filter: "quite vicious bass cut".

Next in the chain is the fader, followed by an insert point for a 100/1 Neve limiter. "It's a peculiar device left over from our old *1066*-type module desks and it's a wonderful limiter. We haven't found anything like it since, so we've reboxed them as the desks become redundant and hung on to them like grim death!"

Given the importance of this device to LWT's system, it is worth noting that the Neve is a limiter only, not the more common limiter/compressor combination. It is normally set for 1 m/sec attack and 100 m/sec recovery, although a faster recovery would probably be chosen if it were available.

The signal is returned from the limiter to, typically, Group 8 on the console. "On Group 8, you will have a Neve compressor, with a 6:1 compression ratio and about +4 dB to +6 dB threshold."

The chain becomes more complicated at this point, as the chat mics that are often on Group 1 are introduced into the scheme. "We break away the Group 1 insertion, so that it is floating, and that goes through a 3:1 compressor. When it comes back from the 3:1 compressor, it is also routed into Group 8.

"So the 6:1 compressor is known as a 'collector', because it collects the audience and the chat mics. It really acts as a safety device for very heavy peaks: the 100/1 is going to hold most of it."

The 6:1 also enables a reciprocal action between the chat mics and the audience mics. If the chat mics are pushed at the end of a laugh, they will tend to knock the audience down in level, enhancing intelligibility.

"It's making constructive use of the classic compressor pumping effect, rather than putting the whole programme through a compressor, which used to happen in the old days." Nowadays, the band or orchestra will have a completely separate dynamic chain.

The set-up described above is the essence of the LWT audience and chat mic system but there are a number of refinements beyond this, as Hix explains.

"There is a crossfade box on a footswitch, which goes between the output of the 100:1 audience limiter and the point where it returns to be assigned to Group 8.

"When you push the footswitch it crossfades another limiter in series with that circuit: when you take your foot off, it crossfades back. That additional limiter is typically a Neve set at 6:1 with minimum threshold, +4 dB. We also use a compressor, set to a threshold of around +2 dB.

"That is used purely for applause. It enables us to crash up the applause and make it more exciting." Because the peak level is under tight control, the dialogue will again cut through, providing close miking has been used.

"If you're on a boom, God help you because there is more applause on the boom than the voice, so you're in trouble."

As a single-channel device, the crossfade box has become a casualty of the changeover to stereo. Although a stereo version may be built, a different solution has been a pair of limiters connected to a simple in/out switch.

"In general, the mono system was refined in a period when heavy audience reaction was in vogue. You would really wham the fader up when you went for laughs."

The move away from that approach is partly the result of the introduction of stereo, Hix believes. On one hand, the separation afforded by stereo means that it is not necessary to push the audience so far up front. On the other, listeners in mono will find the audience a few dB down as a side effect of summing the two channels.

In addition to the main audience mic chain there is an ambience mic, slung perhaps 25 ft above the seating, at fairly low gain. This takes the dryness out of the close mics, which also have a laughter-enhancing midrange lift. The fader for the ambience mic is normally next to the audience group fader.

Stereo has made no difference to the number of mics that are put up for a show. LWT does not use crossed pairs on the audience, for instance.

"The only difference is in the way the panpots are set. We tend to leave a subtle hole in the middle and hopefully push the dialogue up through it."

So far, Hix has been describing a largely static situation. During a programme, a great deal depends on the operator's skill and split second timing. With so much gain in the system, the mixer has constantly to anticipate the audience reaction.

"Hopefully, your sense of humour is the same as the audience's because you can easily go mad on the fader and suddenly you've got 30 dB of horrible-sounding nothing!

To be honest, I think you probably hear the start of the laugh subconsciously and try to beat it. You have to judge how long the laugh is going to last and at the same time watch the screen to see when the artist is going to talk again.

"The trick is to put a plausible tail onto the laughter, without lingering to pick up the house sound system along with it.

"If you leave the fader up with that much gain, the danger is that when the artist speaks, you've got a handful of colouration.

"It isn't so much of a problem in the studio as on location. In the studio our auxiliary feeds to PA tend to be post fader. Again almost subconsciously, when we push the audience mics we tend to duck the artists' mics. So if they were to talk, they wouldn't come out very loud on PA."

While this puts the production mixer in a position of control, Hix admits that this is "bad news" for anyone trying to mix front-of-house "trying to ride levels which are going up and down like a yo-yo".

In a theatre this approach is a non-starter. Given that there is little less likely to provoke mirth than an inaudible comedian, and few things more obtrusive than incessant howlround, the sound reinforcement mixer must have clean feeds.

This calls for a completely different technique from the broadcast mixer.

"We still push the laughs but to try to get rid of the public address spill, because no comedian is going to wait for the laugh to die. We push the performer's mic at the same time as pulling the audience down. It's the exact opposite of studio work."

It is also advisable to use less gain on theatre audiences. This gives less leeway on small laughs but cuts down spill from the SRS and band.

Hix points out that there should be a 'park' position for the main audience fader, which is never completely off. If lower gain settings are used, the console should be adjusted so that the group fader is not sitting at the bottom of its throw where adjustment is very coarse.

Hix believes microphone technique is an important part of the sound LWT achieves in theatres, such as the internationally known London Palladium. A high-slung stereo pair will give a "wishy-washy" sound he says. Tight miking is a better solution.

At the Palladium there is a dress circle and an upper circle, and Hix advocates four mics slung from each, leaving the front stalls out completely.

In contrast, the National Theatre has no convenient hanging points. Here Hix has resorted to Sennheiser *416* shotgun mics, mounted on arms protruding from the wall-mounted ashtrays in the side aisles.

Microphones must be of the same type, Hix warns. "Unless all the mics are the same type, same gain and the same distance from the audience, you can get horrendous pumping because one mic is controlling the others."

Apart from the disruption this causes to the overall audience level, the sound will be overpowered by a small number of people. Hix once made this mistake on a live recording, which went down to multitrack. The problem was compounded because he conscientiously ganged together the limiters on the audience tracks.

"It was the worst of my nightmares. The audience reaction on the multitrack was unusable. I managed to use the laughs but I had to replace every bit of applause! But again, because we tight miked, you tend not to get enormous differences between one venue and another."

For various reasons, audience replacement is not an uncommon requirement at LWT and the department has just put together a 'magic gadget' for the purpose. An Akai sampler is used to store an 'infinite applause' stereo loop but "when you take your finger off the button it crossfades to a proper end, rather than a fade".

But Hix believes that audience replacement is not as predominant as it once was. It tends to be confined to "genuine repair of edits".

Producers realise that if the studio audience does not laugh, "it couldn't have been that funny", says Hix. But if a sequence has to be reshot several times, perhaps to change camera angles, it is unreasonable to expect the audience to find the same gags hysterical.

Often, LWT will help things along by running in applause from an AMS *AudioFile* but Hix has responded to the demand for the device with a new and cost effective solution. Mastered at Tape One, he has just produced an in-house CD of '99 LWT laughs'.

But Hix has replaced clapping with an alternative programme source. Applause is quite similar to white noise, he points out. On occasion, he has simulated rhythmic clapping by simply gating the noise, using the music as the key.□

# MFX IN OPERATION

## Yasmin Hashmi looks at how Fairlight's MFX works

A few years ago, Fairlight introduced 8-channel disk-based recording for their *CMI Series III* sampling/sequencing system. At the time this performed recording only—the intention was to develop editing for the system and call it *MFX*. Unfortunately, shortly thereafter the company ran into financial difficulties. The company was in liquidation but the software development team continued to work together under the name Electric Sound & Picture (ESP). After having joined forces with Australian pro-audio distributors Amber Electronics, they purchased back all the strategic assets of the old company and commenced trading as Fairlight ESP. The company is purported to be leaner and more efficient, farming a fair proportion of its work out to third parties thus allowing it time and resources to continue developing software for the *Series III* and the *MFX*. Rather than simply being an extension of the *Series III*'s 8-channel disk recording, the *MFX* developed into a system in itself, offering to replay and edit up to 16 recordings across 24 tracks. It is now commercially available, and is aimed at audio post-production for video.

## Hardware

Although *MFX* can be purchased independently of the *Series III*, it runs on the same hardware platform. This consists of a unit containing processing, inputs and outputs and hard disk(s). It can support both systems but cannot currently run both simultaneously. There are 16 output sockets as well as a router that allows the output from each of the 24 editing tracks to be connected to a mixing console. The operational hardware for *MFX* consists of a colour monitor and a controller, which is rather like a large alphanumeric keyboard. The system is modular and can support up to 16-channel replay. But the way it does this makes it appear to be a hybrid between a sampler/sequencer such as the *CMI* and a hard disk recorder rather than a straightforward tapeless multitrack.

## System operation

Both the *MFX* and *Series III* require a hard disk and a fair deal of RAM in order to function. From the *Series III* point of view, samples for a particular sequence are loaded from hard disk into RAM whence they are triggered. The number of sounds that can be replayed simultaneously is limited only by the number of outputs the system has but the number of different sounds that can be replayed simultaneously depends on how long the sounds are, ie whether they will fit into the RAM or not. Therefore the more RAM there is, the greater the number of different sounds which can be used and/or the longer they can be. However, once all the sounds for a sequence have been loaded into RAM, they remain there, even if a particular sound is only used once and is not required again. It is, therefore, very important that you have sufficient RAM in order to accommodate all the sounds required for any particular sequence, and this can easily run into tens of Mbytes.

A conventional disk recorder/editor on the other hand, requires little RAM since, as the audio is playing out, the RAM is continually being back-filled from disk. The number of sounds (or channels) that can be replayed simultaneously depends on how many outputs the system has, how many disks are used simultaneously and how fast the disk head can find recordings located on different parts of the disk. In general, the maximum number of channels a disk can continuously back-fill RAM with is eight. However, the number of different sounds that can be used is limited only by how many different recordings will fit on disk, and this is bound to be

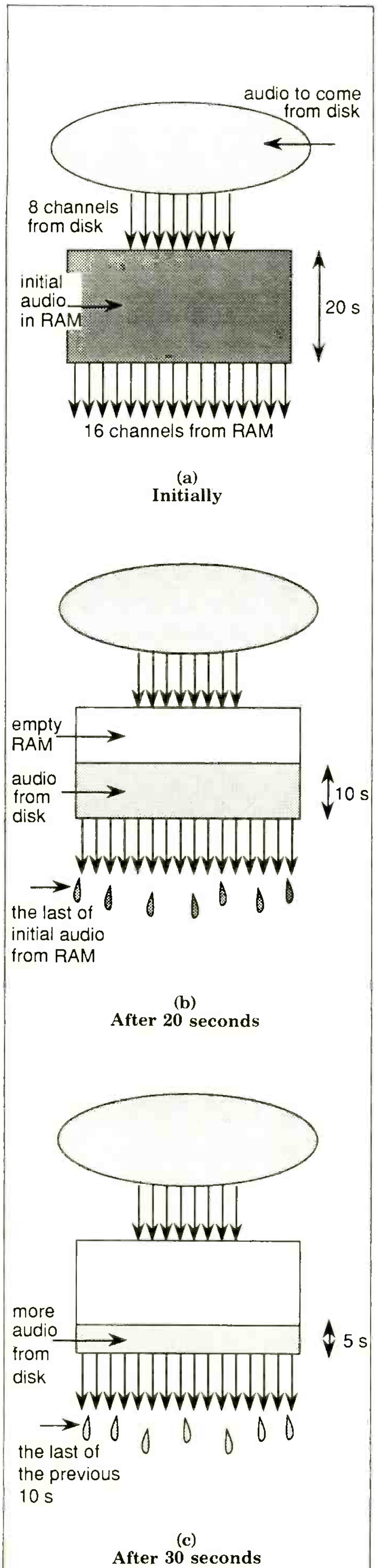


**FIG 1**

significantly more than will fit in RAM.

Like the *Series III* and other sampler/sequencers, *MFX* uses its RAM to provide the polyphony required, in this case 16. But unlike a conventional sampler/sequencer, *MFX* updates (or back-fills) its RAM with new sounds like a conventional disk recorder.

If *MFX* were used as a conventional disk recorder, it would only be able to play up to eight continuous channels simultaneously. But if the recordings are discontinuous (ie successive recordings are not required to play back-to-back and there is sufficient time between successive recordings so that the RAM has time to back-fill) the system can manage to replay up to 16 sounds simultaneously.

The amount of time recordings can simultaneously replay for is dependent on many factors such as how long each recording is, how many recordings are used at any one instant and ultimately on how much RAM there is. For example, if 16 recordings were played for as long as possible, they would only last as long as the length of time the disk could back-fill the amount of RAM available before the RAM emptied completely. If the system had 32 Mbytes of RAM, this holds around 320 seconds of audio at 44.1 kHz. If each sound is said to take a channel, this provides 320/16=20 seconds/channel as shown in **Fig** 1a. While the 16 channels are playing out of RAM, the disk can only back-fill eight channels-worth, ie there is only half as much coming in as there is going out. In the time it takes for 20 seconds to have played out, the disk will have filled the RAM with another 10 seconds as shown in **Fig 1b** and by the time this has played out, another 5 seconds will have been loaded in (**Fig 1c**), and so on. Thus 32 Mbytes can support 16 sounds for around 40 seconds, after which there will effectively be drop out. Fairlight are planning to introduce a card called 'turbo SCSI' which will possibly allow the continuous replay of up to 16 channels from disk.

It is not currently possible to run a *Series III* sequence simultaneously with *MFX* recordings, although this is planned. Once this happens, the user will have to allocate the amount of RAM to be available to the *Series III* and how much to the *MFX*.

# User interface

The colour monitor displays all 24 editing tracks horizontally across the screen. Each track corresponds to an output. The controller provides hard and soft function keys, a jogwheel, alphanumeric keys, two LED displays and dedicated keys for each of the 24 editing tracks. There is no mouse—all system operations are achieved by pressing keys and/or moving the jogwheel. For example, selection of track 12 to be recorded to or edited is achieved by pressing the track key marked 12, which then lights up. Recording or editing functions involve pressing the appropriate key, whereby one of the LED displays will show various options. The option is selected by pressing the appropriate key under the display. The other display is used for providing timing/titling information. For example, the top display can show the names of all the projects on disk. These can be scrolled through using the jogwheel and opened or closed by pressing a key. A project need not be closed while another project is being opened. The system also allows up to 30 macros to be recorded and triggered via 15 double function keys (a macro can remember a series of operations and carry them out when it is activated).

# Recording

If the user elects to begin a new project, a key is pressed, a project title is typed in and 24 empty tracks appear horizontally across the colour screen. Either a mono or a stereo recording can be made to hard disk, a stereo recording is treated as a single data stream, ie left and right channels cannot be separated/edited independently. The system allows recording from an analogue or digital source at 44.1 kHz or 48 kHz and also allows auto-record/punch-in (with a count in if required). A level meter is provided on screen and once a recording has been made the audio appears as a highlighted block in the selected track. Selected (or active) tracks are highlighted in pink. whereas inactive tracks are blue. But there is no indication as to whether a recording is stereo or mono although the user does have the facility for naming tracks and recordings (clips). The name of the track will appear regularly across the screen but the name of the clip, although displayed at the bottom of the screen if the clip is selected, does not currently appear in the block, neither can the names of all clips in the project be listed. Markers can be made on the fly during record and/or playback.

# Editing

Editing takes place on the same screen as recording. There are 24 tracks across which clips can be recorded and assembled and each track can contain up to 99 clips. Like recording, editing is straightforward and consists of selecting a track to be edited, locating in and out points with the jogwheel and cutting, pasting, sliding, copying or overlaying onto the same track or another track. In addition, a group of tracks may be selected and an editing area marked out. Tracks can also be soloed and muted.

The amount of time displayed on screen can be compressed or expanded in steps from six frames to 16 hours. Zooming in and out in this way also affects the jogwheel, which moves across the audio at a rate proportional to how much time is displayed. There are two modes of operation for the jogwheel, either as conventional audio scrubbing or looping the current frame so that normal pitch is maintained. This is useful when in freeze frame to picture and allows frame sized changes in the audio to be detected easily.

As well as conventional editing features, using simple keystrokes allows audio to be deleted or faded from the cursor point to the beginning or end of a clip. Since crossfading one mono clip into another for example, will take up two channels, the system allows crossfaded clips to be bounced internally into one new clip which will then only require one channel. The system will also perform time compression/expansion.

# Synchronisation to video

The system will slave to LTC and can generate timecode at the same time. It can also control an external machine via an RS-422 link using the Sony 9-pin protocol. The external machine can be controlled by the *MFX* transport controls or the jogwheel. Some keys next to the jogwheel allow the quick selection between both audio and video being controlled, video only or audio only. The zoom function for the jogwheel also controls the rate at which the video is scrubbed through, which can be one frame at a time or progressively faster. Control over the video machine is extremely tight. Timecode values can be grabbed and used to place clips or to give an offset.

## Clip organisation

Clips can be borrowed or copied from another project. The difference between borrowing and copying is that borrowing uses the one clip whereas copying actually records another copy of the clip onto disk, so that if the project copied from (and thus its clips) is deleted, the required clips from that project are not suddenly missing from the current one. If *Series III* software is also installed in the system, clips can be transferred to the *Series III* system, edited as samples (ie waveform edited, reversed, etc) and transferred back.

There is no comprehensive library structure for organising clips. The method suggested is that projects can be created that are dedicated to being libraries. For example, a project can be created and named 'Cars'. Each of the 24 tracks could then have up to 99 effects relating to cars. Track 1 could be 99 different ignition sounds, tracks 2 and 3 could consist of 198 different engine sounds, etc. However, there is no facility for recognising clips by name or for searching for a particular clip. In order to find the desired clip, the user must listen through the track until the clip appears. In order to be of any use, the clips in the library would need to reside permanently on disk, thus taking up more and more disk space as the library grows.

Again, if the system, also has *Series III* software, the CMI's sound library structure can be taken advantage of and can be used for storing and cataloguing clips. In addition, *Series III* samples can be transferred as clips to the *MFX* and this may be particularly useful to existing *Series III* owners who have already built up a large sound effects library. Sounds can be auditioned from the *CMI*'s library but the user has to be careful to allocate outputs 1 and 2 to the *Series III* for this purpose.

## Archiving

Once the disk is full, the user may wish to archive its contents. There is, however, the facility for disposing of unwanted audio (such as clips which are not used, tops and tails, etc) in order to create more space for further recording. Archiving of audio and edit information can be done to tape streamer (as is used with current *Series III* systems) or to high speed 8 mm Exabyte tape.

## Costs and configurations

The *MFX* can be purchased on its own or can be an upgrade to any existing Fairlight system. 760 Mbyte or 1.2 Gbyte hard disks are used for recording and these can be daisy-chained to provide more recording time. Continuous recording, however, cannot take place from one disk to another. Channels and outputs come in packages of two, thus for dubbing foreign languages for example, a 4- or 6-channel system would be sufficient. RAM can be purchased in 2 Mbytes with a maximum of 32 Mbytes. The faster 760 Mbyte or 1.2 Gbyte hard disks are recommended since older, smaller disks do not have sufficient capacity or speed of access. A basic system with six channels, 12 Mbytes and 2 hours of storage costs £43,000 and a 16-channel/32 Mbytes/4 hours storage system costs around £68,000. To add *Series III* software costs around £2,000.

## Conclusion

The system is easy and fast to operate and although some functions are not immediately obvious, the manual is very helpful and easy to use. The lack of a comprehensive library structure is rather puzzling, especially since the system is designed to cope with numerous short sounds. The graphics are clear and the control provided by the jogwheel both over internal recordings and external machines is extremely good.

The ability to run the *Series III* from the same hardware platform will certainly be attractive to those interested in both systems, but even when both systems will be able to replay together, there will be limitations since they will have to share outputs and RAM.

Although many of the multitrack tapeless recorder/editors provide eight channels as standard, many of them can provide more. However, for applications where polyphony is important but the duration of recordings need not be too long, the *MFX* provides a unique solution at a comparatively lower price.□

# BRITANNIA ROW

## Ralph Denyer talks to Kate Koumi of Britannia Row about the studio and its history

Britannia Row is a smallish side street in north London. During the '70s Pink Floyd acquired the large building there which currently houses Britannia Row Studios and various companies in the business of supplying lighting and concert sound systems for major bands and venues.

At the time of acquiring the building Pink Floyd were leading European pioneers in the kind of high quality presentation that we all have now come to expect at any major concert. They needed premises in which to store their equipment when they weren't touring. Other major acts were keen to hire the Floyd's sound system and thus Britannia Row Productions was soon a thriving business.

The vast building far exceeded the band's storage needs and before long it housed several companies supplying sound and lighting equipment for major concert promotions. During the mid-'70s the Floyd set up their own recording studio within the building initially for their exclusive use. They recorded their *Animals* album and part of *The Wall* there.

During the early '80s Dave Gilmour and Nick Mason bought the other members of Pink Floyd out of the entire Britannia Row property, the studio and the various ventures it housed. Then subsequently at the beginning of 1984 Mason bought out Gilmour and became sole owner.

The studio had fallen behind in terms of technical specification though even a cursory listen to *Animals* or *The Wall* will reveal that both albums put many recent albums to shame in terms of production values. In order to hire out on a truly competitive commercial basis the studio would need to be completely updated.

Concurrently Kate Koumi was working at Polydor Records as A&R supervisor. Richard Ogden, MD of Polydor at the time, recommended her to Nick Mason and business associate Norman Lawrence. The job on offer was firstly to be solely responsible for the re-organisation of the Britannia Row studio on a commercial basis and to then manage the studio day-to-day. Along with Mason and Lawrence she is now a director of the company.

Koumi had no previous studio experience although she knew the business from a client's point of view through her job at Polydor. Mason wanted someone to set up and operate the studio on a strict commercial basis. Some five years on Koumi says Mason only visits the studio a couple of times a year, leaving her to run the venture virtually autonomously.

She explained, "In May '85 I was employed to build and commission the new studio. I started on the day the old studio shut and literally became site manager which was a bit of a nightmare but at the same time the best possible grounding in acoustics and the technical side."

The existing equipment such as an old MCI console and customised JBL monitors were considered too old to be suitable and only the wide selection of microphones, some vintage compressors and a Mk1 Studer 24-track analogue tape machine were to be retained. The Studer was replaced about 3 years ago.

The studio area of the building was gutted and Westlake was commissioned to take care of the acoustic design only. Koumi decided on a light and airy interior design that, some five years on, still looks as though the studio has just had a complete refit although the general appearance of the place has not altered since the 1985 reconstruction. During the past five years the only significant changes have been in equipment and software.

Britannia Row is a single studio complex with 24-hour on-site maintenance. The studio is designed in such a way as to keep the administrative offices completely separate in order to afford clients' total privacy.

Koumi: "I call it a Single Studio Complex which gives an idea of size and space. All the staff are dedicated to the one artist, producer or whoever is in the studio so we don't whip gates or Lexicons out from one studio to another. You don't have a tape-op wandering in for leads so that when you're quickly trying to patch up, leads aren't there. We give a complete 5-star service as much as costs allow. I think that has been a great selling point."

Plans are already drawn up for a second studio to be located in an entirely separate part of the huge building. But they are being kept on ice for the next year or so. Koumi sees 1991 as crunch time and the year in which only the best run studios are going to survive. Then the pricing war will subside and allow

sensible studio rates to be charged that will allow major investment in new equipment. She is waiting until the market levels back to a sensible number of studios before deciding the pricing level to which the second studio will be equipped. But she has already decided that the two studios will operate separately with clients enjoying the same total privacy as in the existing single studio setup. A conservatory area is being constructed in such a way as to maintain this policy.

Although having 24-hour on-site maintenance is rare and expensive for a single studio setup Koumi is in no doubt about money being well spent in that area: "If someone's a runner you don't take his running shoes off because you think he can run in his bare soles. Maintenance is about as important as that as far as I'm concerned."

About three years ago the studio acquired a Studer *A820* with

remote and Dolby *SR* cards but before long Koumi decided the noise reduction did not fit in with her pricing structure.

"I sold the cards because they cost £15,000 for 24, which was quite an outlay. If people want to record analogue they'll decide there's not that big a difference. And it's so much on the price that if there's not that big a difference in sound, you know, the budgets aren't there to provide an extra £100 a day or whatever for Dolby *SR*."

After the purchase of one major piece of equipment Koumi was horrified to discover that during an initial period the studio, along with several others, was effectively carrying out a major part of the de-bugging of software. Realising that this is to be expected with some manufacturers she opted for an original approach when the SSL *G* series computer upgrade came on the market.

"I resent the fact that one is a guinea-pig for manufacturers and they expect you to let them know what's wrong (with a new product) and then they'll make the new software. So with the *G* series (computer) I thought: I'm going to buy this and I'm going to store it.

"That's exactly what I did. I never fitted the *G* series computer for nine months until all the snags had been found and sorted out. So we never had a single breakdown of the old historical ones of the *G* series computer."

Koumi decided against the *G* series EQ for the studio's 56-channel *4000E* series SSL console. It seems clients were interested mostly in the computer so she opted to go for outboard EQ, buying one Massenberg *8200*, two Focusrite *ISA 110*, two vintage Pultec valve and one Klein+Hummel *UE400* unit, adding "anything else would have been extravagant".

The studio has a wide range of outboard signal processing equipment including an Eventide *H3000 Harmonizer*, AMS *DMX 15-80*s 16 secs, AMS *DMX 15-80* 1.6 secs, AMS *RMX 16* reverb and a Lexicon *224XL* reverb.

Koumi: "I'd say they don't need to hire and therefore their studio costs are what they are. Now from word of mouth they believe it and it has paid off. AMS are still a trial. Their breakdown rate is extremely high and they're very costly. But that's about the only real bugbear we have."

The studio was one of the first to reverse the roles of the main recording room and control room in order to accommodate control room sampling, sequencing, computer and keyboard work.

"I suppose there is a slight compromise in having the equipment rack behind the desk. It's a compromise of the acoustics in terms of the monitoring from the purist's point of view and obviously it doesn't allow a clear wave to surge through to the back of the studio. But that compromise has to come with what flows for the engineer and producer. And rather than have the equipment at the side, which would be better acoustically, it is at the back so they literally just turn behind to operate outboard equipment.

"We have all the EQs in one rack one on top of the other, so that perhaps when someone is sorting out the EQ they would want to be in the centre between the monitors really. So while you're tweaking away you're in the right sort of position. So we go into little points like that, just build up on little points that have a synergistic effect. And in the next year or so, as I say, we won't be spending much more on equipment and just concentrate more on things like that."

The control room is a spacious 21 ft×29 ft with two sizable windows supplying daylight. Directly behind the console is the large and slightly raised outboard rackmounting unit. This actually consists of three units side by side, each of which is two 19 inch rackmounting units wide.

In the past there have been problems with the main monitors at the studio which were rectified by Phil Newell during 1990. Previously they were positioned vertically in such a way that some low frequency material was striking the back of, and passing below, the console. Newell has customised the Westlake *HR1* units, repositioned them horizontally and tilting downwards slightly. Since these modifications clients are reported to be happy with the system and Soul II Soul haven't managed to blow them up once.

Light blue curtaining, curtain folded fabrics, neutral coloured furnishings and carpeting, mirrored surfaces, white light spotlights, Art Nouveau-type wall lights and a ceiling which rises towards its centre all conspire to give a very airy feel to what is already a large control room.

The original Floydian control room is now the overdub room with good visual contact from the console position between the monitors. This is a fairly basic medium size recording room 20 ft×17 ft with mainly brick surfaces providing a medium reflectivity, which has proved most useful for vocals and occasional guitar overdubs.

Sliding glass doors separate the main recording room from the control room and combine with an angled mirror in the room to offer good visual contact, so much so that the video system linking the rooms has yet to be used by studio clients. Again, natural daylight is a feature. When the conservatory is completed it will enclose the exterior area that provides the natural daylight to both the main live recording room and the control room. In order to conserve the natural daylight the design incorporates paved and landscaped courtyards.

The main live recording room is treated so as to have varying acoustic areas. A long curved acoustical structure across part of the ceiling provides a zone that apparently works particularly well for drums. The room has a Steinway grand piano; a Hammond organ can be wheeled between the two recording rooms. The room is often used for recording brass sections and live rhythm sections. There are monitors in both recording rooms with individual power amps.

The machine room is isolated to one side and houses the Studer *A820* and Otari *MTR 90* tape machines which can be used together by means of a *Lynx* synchroniser. There is also an isolated computer room.

A clients' reception room provides a lounge with satellite TV and a cooking area. The conservatory will be equipped with a snooker table, lounge area, kitchen, dining area and showers.

**Art Deco recording area**

For those requiring residential facilities, there is a four-bedroomed house close by with a separate 'producer's flat'. The house and the flat can be opened up to function as one larger domicile. Both house and flat have TV, hi-fi and are serviced daily. Parking facilities are also available.

The studio is not particularly pitched towards either sampling and sequencing or live recording. They cater well for both. It is a very enjoyable place to visit, and as is obvious from a long client list that includes Soul II Soul, Scritti Politti, Neneh Cherry, and a number of top producers, the studio is a pleasing, productive and creative environment to work in. But why?

Accepting that technical industry standards are either met or exceeded, the answer is wholism, the concept that the whole is greater than the parts. In conversation about her aims regarding the studio, Koumi frequently refers to, 'the flow of the work' and employs terms such as, 'synergistic'.

The, 'flow of the work', is paramount. The equipment is selected, set up, modified and maintained with the same thought in mind. Those wishing to survive what looks like being a fiercely competitive decade could do far worse than take note of her example.□

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# CABLE AND SOUND DELIVERY

Towards the end of last year, Philip Newell and Dr Keith Holland made a presentation on the subject of loudspeaker cables to the UK Institute of Acoustics. A résumé of the paper is presented here as a sidebar. Subsequent experimental findings are also detailed by Philip Newell together with some practical conclusions

In between the writing of the discussion document (below) and the conference, a differential amplifier was constructed (following some research funding from Tom Hidley), having some 110 dB of common mode rejection. On initial test runs over 6 ft lengths of cable, the differential unit was realising very considerable differences from cable to cable when driven by a typical domestic amplifier into a 3-way loudspeaker combination.

There were not only level differences, but differences in spectral tilt, and subjective differences in the solidarity of the bass response. Bear in mind that we are not talking about the output from the loudspeaker of the test system, but rather of the output from a monitor connected to the output from the differential amplifier. The differences attributed to the cabling were highly amplified and represented different signals which may well be 90 to 100 dB down on the actual output of the loudspeaker. However, the point was that here was clear, repeatable, measurable evidence of the fact that cables were all performing differently. When these tapes were played to the conference audience, a large proportion of the delegates were undoubtedly surprised by the degree of difference.

## Cables or amps

There has been an enormous amount of material published on loudspeaker cable design criteria. Unfortunately, much of the information has been contradictory. One of the general opinions which seemed to hold at the conference was that this is not surprising as the effects of certain cables are so system-specific that a formula which may be beneficial on one system may be undesirable on another. A further confusing factor here is that if a cable linking a given amplifier and loudspeaker combination were to be changed for another cable, any perceived difference would usually be attributed to the cable. There are instances however where for example, Zobel networks have been dispensed with in amplifier outputs, the manufacturer presuming a certain minimum usual resistance and inductance in the cabling and relying on this for stability. Changing to a low resistance, low inductance, high capacitance cable can have adverse effects upon the operation of such an amplifier so the perceived effect of such a cable change could well be based more on amplifier loading differences than on the cable *per se*. Some people feel that such a problem should be addressed in the amplifier but some amplifier manufacturers would argue that they are producing amplifiers for optimum performance on most typical systems.

Notwithstanding the amplifier manufacturer relying on a given minimum of cable impedance, in which case it could be argued that the cable is part of the amplifier circuit, it would seem hard to argue with the fact that the best cable is *no* cable. Richard Lee of Wharfedale stressed most vociferously that in tests carried out at the company, the only conductors of significant length which could not be detected by a listening panel when inserted between the amplifier and loudspeaker, were two enormous lengths of lightning conductor.

## Oxygen

Cables are in reality, highly complex networks of resistance, inductance, and capacitance made even more complex by the diode-like contacts which can exist between individual crystals or strands where oxygen is present in the form of surface oxides. Indeed, the copper oxide rectifier was the first widely used rectifier in the world of electrical experiments. Removing oxygen from the copper removes one source of variability, but the natural affinity of the copper for oxygen will ensure its eventual re-oxidisation unless steps are taken to use insulators which are as impermeable to oxygen as can be achieved. Unfortunately though, oxygen permeability alone cannot be the criterion for insulating materials as dielectric absorption and charge migration are also important factors in insulator choice. When a capacitor is charged then briefly shorted in order to discharge it, it will sometimes be noticed that after a short period of time, a small voltage will have

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Two thousand pounds or more is no longer an unduly rare amount of money to be spent on a pair of loudspeaker cables, yet many people in professional audio circles are strongly resisting being drawn into any consensus that such cables make convincingly significant audible differences to a system. Many people cite secondary or tertiary effects of the cable installation procedures as the main source of subjective improvement: general care and attention to detail such as clean, tight connectors, careful cable routing and well soldered joints being typical examples put forward for the case against any directly attributable cable benefits. 'Show me conclusive evidence in terms of Ohm's Law' is another often heard demand from the non-believers. The over-hype by the hi-fi magazines has probably done nothing to persuade many of the more conservative professionals that there is any significant substance in the case for such esoteric wonders; yet all along the line, too many people have been convinced of the benefits of certain special cables for me to have ever been inclined to dismiss their benefits out of hand.

Debate continues on the subjects of oxygen-free and linear crystal copper versus conventional copper; co-axial versus twin cables; optimum strand thickness with relation to skin effects; silver solder versus tin/lead solder used in termination; insulated versus uninsulated strands in any one bunch; insulating sheath materials; directionality in terms of one specific end to the amplifiers and the other specifically to the loudspeaker, in other words, non-reversible cables; shielding from external magnetic fields, both LF and RF; general transmission-line properties; and many other controversial areas of discord. Such discussions may be working wonders for the sales figures of the hi-fi magazines, but for the industry in general, it can be doing its credibility no good whatsoever.

## General good practice

Obviously, any reasonable cable is likely to sound better when compared to an excessive length of poorly terminated bell flex. As a general rule, good quality, well terminated, 60 A, multi-strand, conventional copper wire would seem to be adequate for most purposes. At about £1 ($1.90) per metre for a pair of conductors, the price would also seem realistic for most applications. Minimum cable lengths between the amplifier and loudspeaker driver is a virtually self-evident rule of thumb. Conversely, 8 Ω in an absurdly long length of cable will have untold negative repercussions on the performance of a 4 Ω system.

From our findings to date, cable deficiencies are least noticeable on electronically crossed-over, multi-amplified systems, and most noticeable on systems with high level passive crossovers, particularly those displaying tortuous dynamic load impedances. An extreme case of the latter is the Hidley/Kinoshita system whose crossover input impedance drops to around 0.8 Ω at certain frequencies. Given the system power rating of 1000 W, on complex musical drive signals, transient currents in excess of 100 A are not unfamiliar when the systems are driven at high SPL's via their JDF 3200 W amplifiers. On these systems, JDF supply oxygen-free cables, with DC polarised outer screen, directional conductors and overall one specific end to the amplifier directionality. I mention these systems as due to their extreme demands, they have proved to be useful test beds for the highlighting of more general trends.

The amplifier-to-crossover cable would seem to be more critical than the cables from the crossover-to-loudspeakers. The amplifier/crossover cable should be as short as possible as it is in the crossover where the highly complex dynamic loads are realised. If any significant distance exists between the amplifier and the loudspeaker, then the crossover should be brought as near to the amplifier as possible in order to minimise the distance over which the cables are subject to highly reactive loading. It is also over this length

reappeared at the terminals. This is a function of dielectric absorption and to the lesser degree, it is observed in cables; the greatest influence on the degree of absorption being the choice of insulating material. The delayed appearance of a voltage after the short is removed is due to the absorbed charge gradually being released from the insulating material, re-appearing as a charge across the plates.

# Skin effect

The much vaunted skin effect is the subject of great debate. Essentially, higher frequencies tend to travel along the skin of a cable rather than through the core. An early approach to addressing this fact was to use finely stranded cable which would present the greatest surface area to cross sectional area ratio, but it has been suggested that when such strands are bundled together they behave not as strands but as one conductor. Individual strands in the bundle may be near the outside at one point in the cable, suddenly diving into the centre of the bundle in the twisting process then re-appearing elsewhere. The high frequencies thus skip from strand to strand in a way which is both unpredictable and variable as the cable is bent or moved. Obviously, if the cable is made from conventional copper with surface oxides, it can be seen that each inter-strand contact is a potential rectifier, but by-passed by the resistance of the easier route of the individual strand: most electrons will tend to follow whichever is the line of least resistance.

The use of *litz* wire, in which each individual strand is insulated with an enamel, attempts to overcome this contact problem by preventing inter-strand conduction. I have seen loudspeaker cables employing up to 3,000 micro-fine strands per conductor employing the *litz* principle. When wires are drawn so fine, effectively they begin to approximate linear crystals as each of the individual crystals of copper are squashed and stretched. Each individual strand should thus have fewer inter-crystal boundaries and hence less potential for low level rectification. Yet here again, there is controversy over whether to draw or whether to extrude. Extrusion is said to produce long crystals which are in less of a state of internal stress than that produced by the drawing process. Not only are stressed crystals said to be more generally brittle, but they are more prone to fracture due to current stress.

# Magnetic considerations

Current stress is induced by the intermolecular electromagnetic attraction and repulsion which is a function of the magnetic forces resulting from the current flow. Certain materials exhibit the property of magnetostriction; the expansion and contraction in sympathy with the current flow. In multi-strand cables, the individual strands can be moved by interaction of their external fields. Any such movement in cable geometry will inevitably change the inherent inductance and capacitance leaving us with a dynamically varying set of electrical parameters. On a larger scale of movement, the inter-cable relationship was clearly demonstrated in the tests carried out prior to the conference. A pair of FM *Forcelines* multistranded cables were spaced apart while listening to the output from the differential amplifier with signal flowing between the amplifier and loudspeaker. As the cables were brought together, the sound from the differential amplifier output noticeably changed in its spectral balance, clearly suggesting a change in the series inductance of the pair. I note that FM now supply plastic clips to maintain the cables at a constant spacing.

# Beyond power loss

The multistrand cables based on the skin effect philosophy are said by some to have failed to take into account that power loss is not the only parameter to be considered: the factions claiming that larger cables, in the 0.8 mm diameter region, offer a more balanced response. The resistance and linearity changes due to inter-strand leaps, and the frequency dependent pathways through larger cables, together with the attendant impedance differences of those pathways, and magnetic movement of the cables themselves, are claimed to be more audibly noticeable with their attendant phase shifts, than the volt/amp losses due to the skin effect alone. The intricacies of these inter-reactions were clearly demonstrated by Malcolm Hawksford who showed graphs to the conference indicating the behaviour of cables of 5 metres or over above 100 kHz where resonances and transmission line effects could clearly be demonstrated in terms of transfer function errors above 100 kHz and into the MHz region. Here was an area where cables exhibiting good HF transfer could introduce serious levels of RF interference into amplifier circuitry to detrimental effect, whereas a slightly poorer or more lossy cable at HF could be better on a system where RF interference or HF amplifier stability were problematical.

Clearly the potential for a cable to affect performance is established, but the problems seem to be in the precise balance of parameters, and their relationship to the amplifier and loudspeaker characteristics. It seemed to be generally accepted by the conference that the 'correct' specifications for a cable could not be defined as an engineering exercise, as the inter-reaction potential with the rest of the system was too complex. Cables chosen as a part of an overall system were entirely valid in that system, but were not necessarily superior in another system. Many people seemed to feel that in the case of the more expensive varieties, the money could usually be spent elsewhere in the system to greater subjective effect.

# System susceptibility

Obviously, when all else has been done, the cable may be all that is left for improvement, however, system design can help to remove the worst problem of cabling. I have now been designing

of wire that the complex back EMF's from the entire system will impose themselves on the feedback circuits of the drive amplifiers. Whatever impedance or irregularity occurs in this length of cable will form the top half of a potential divider network, the lower half being the output impedance of the amplifier, across which the overall feedback circuits derive their error signal. Any RF or other spurii which may superimpose themselves on the crossover/amplifier cable, including any non-linear conductivity as suggested may be caused by inter-crystal boundaries in the conventional copper cabling, will again modify the feedback signals. I would suggest that some of the benefit attributed by some audiophiles to amplifiers without negative feedback may be partly due to their general immunity from the above effects.

If the benefits of good quality, short cables are to be realised however, the potential for non-linear conduction in this area is greatly exacerbated by the other inter-metal contacts which could include a tinned transistor leg soldered to a copper printed circuit track, in turn soldered (tin/lead) to a copper wire which may be crimped or soldered to a brass eyelet tag, clamped via a steel serrated washer to a brass terminal, in turn connected to a chrome-plated banana plug or spade connector . . . and that is only at the amplifier end. The chrome-plated spring connectors on the loudspeaker chassis are notorious for suffering oxide build-up over a matter of only a few months. In fact, chrome retains its shine by virtue of a thin film of oxide which forms on its surface immediately upon its contact with air. The effect of a short length of adequately current rated, well terminated good quality copper conductor would appear to be small when compared to the other potentially non-linear conductors in the circuit. There are people who say that AB testing of such sensitive nuances via a switch or relay system is invalidated by the introduction of a switch contact into the circuit. I cannot find any justification of this neurosis by experimental measurement, by listening, or by intellectual reasoning when the circuit contains such interconnections as described above.

## Conductor geometry and skin effect

The overall cross-sectional area of the conductors must be adequately capable of passing the highest dynamic currents likely to occur on any given system, without any instantaneous temperature induced resistance rises sufficient to be detected audibly. Cross section will be a function of overall length and inherent resistance per linear unit. The individual number of strands which form that total cross-sectional area are the subject of heated debate. Once again, the problem seems to be aggravated by difficult dynamic loads and minimalised by active crossover/multi-amplifier drive systems. In the latter, in a typical 4-way system it is unlikely that any cable would be handling more than four octaves. In a passively crossed-over system, 11 octaves can make more stringent demands on the transmission line linearity of a pair of loudspeaker cables. Indeed, as an extension of this principle, even in a passive, high level system, it is only the amplifier/crossover cable which carries the full, wide-band programme.

Some manufacturers sing the praises of cable composed of hundreds of hair-like strands in order to maximise the ratio of surface area to cross section—the skin area—while others claim that this approach maximises the potential for surface corrosion and inter-strand non-linear conduction due to the copper oxide rectification principle. It has also been claimed that individual conductors of the size of typical telephone installation wires are the optimum choice for the skin/core balance of current flow. Still other manufacturers insist that such strands must be individually insulated in the bundle, both to reduce the problems of long-term corrosion and to prevent the cable impedance from varying due to the randomised

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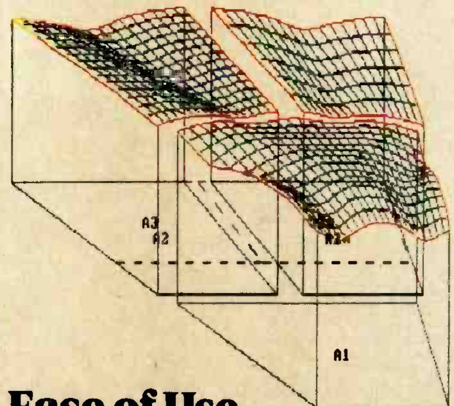

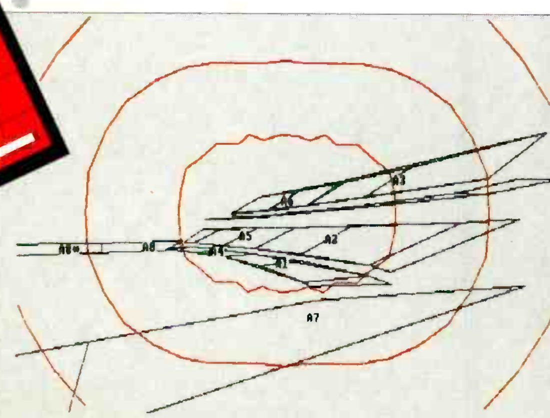

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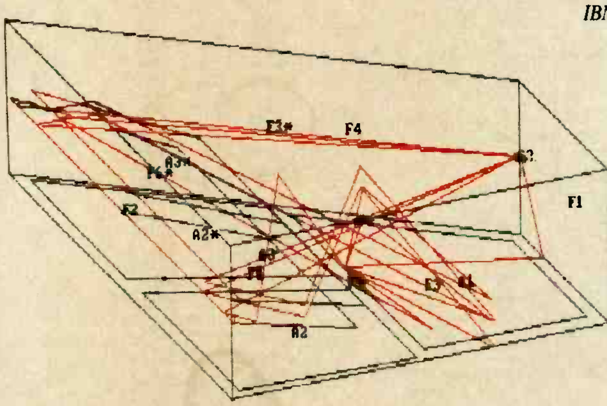

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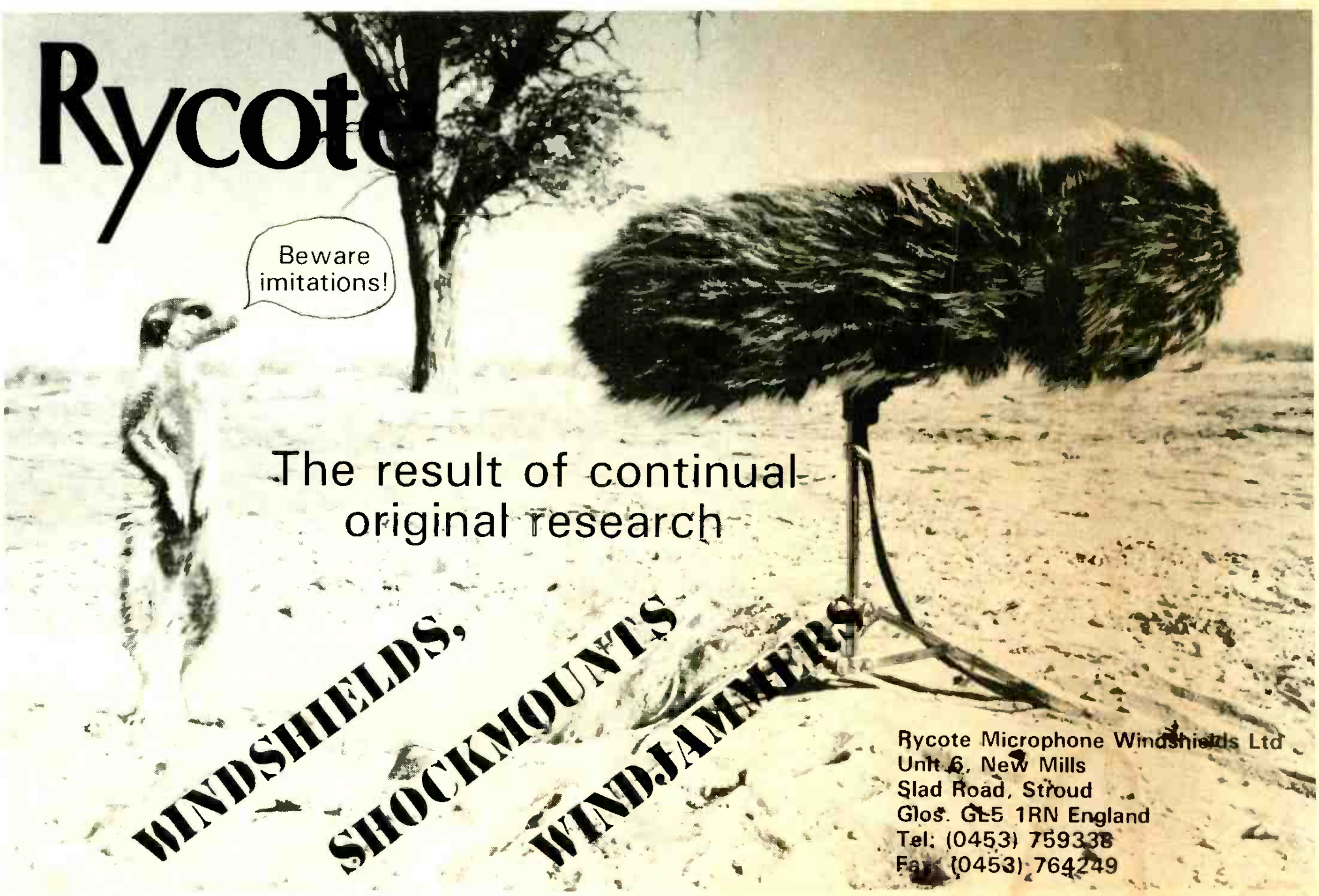

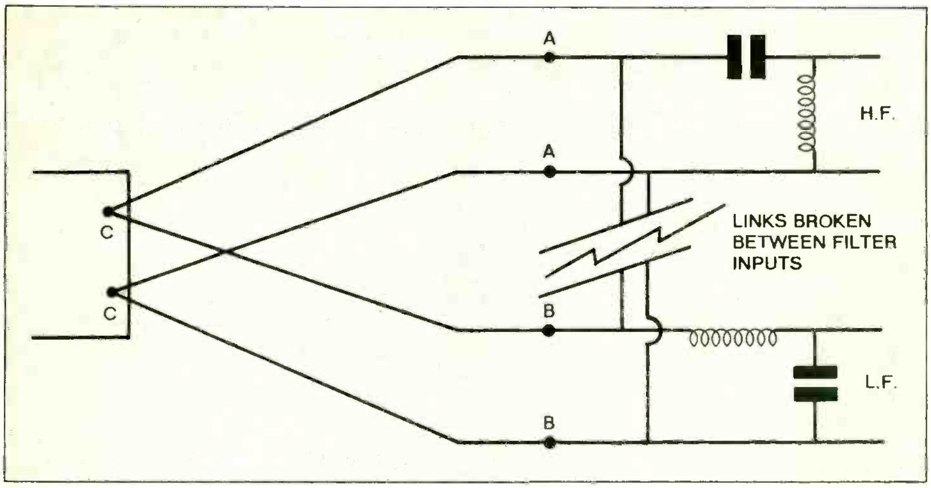


**FIG 1**

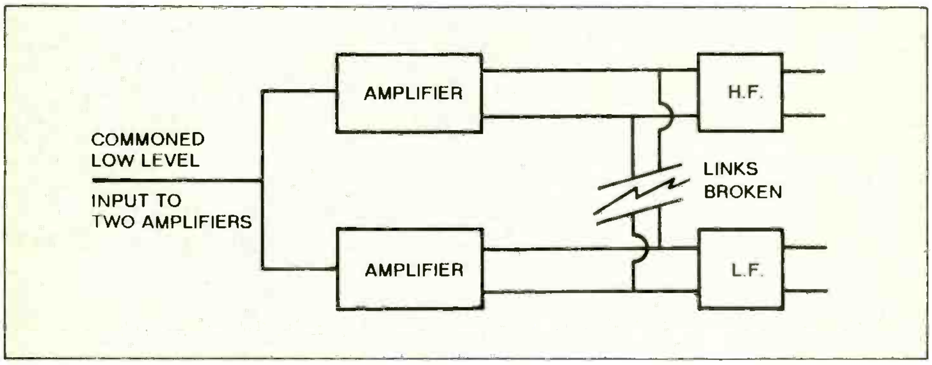


**FIG 2**

and installing studio monitor systems for 20 years, even my first systems were low level crossover/multi-amplifier designs. Adequately sized and well chosen cables has always seemed to render such systems relatively immune from cable problems, whereas at the other end of the scale, the Hidley/Kinoshita, 1 to 4 Ω, 1000 W passive systems seem to be very cable dependent. The explanation appears to be that cabling problems seem to be proportional to length, frequency range and current. A multi-amplifier system with amplifiers located close to the loudspeaker units requires only short cables, and only a narrow frequency band to be carried by each cable, thus reducing two out of the three critical criteria. The third, current, seems relevant only when the other two are large. On a large, wide band system, the low frequency signals can easily be 60 dB above the level of the subtle information carried in the high frequencies. When the signal is carried in one cable the low frequency currents can, by some of the aforementioned processes, modulate, swamp, or mask by distortions the high frequency subtleties which are responsible for clear, open reproduction. Remember, 60 dB expressed in terms of power is 1,000,000 to 1! The override potential is enormous.

The potential for inter-crystal/inter-strand contact distortion is also reduced in multi-amplifier systems. The low frequency drivers cannot properly reproduce the high frequencies which are the distortion products of their high currents. Remember, if the low frequency distortion products are 60 dB down on the signal, they can be on a par with the level of the high frequency information. With the LF distortions restrained within the LF system, they will not be reproduced by the LF drivers which usually cannot reproduce high frequency distortion products and hence will leave the high frequency signals clear but for their own distortion producing potential, 60 dB down on the HF signal and hence causing little problem.

# Bi-wiring

The get-out from this problem for those using high level, passive crossover systems is to avoid any of the cabling in the system having to carry the full range signal. The answer lies in bi-wiring; taking separate cables from the amplifier to the separated high and low frequency filter inputs of the crossover (see **Fig 1**). Although the cables are paralleled at the amplifier output terminals and, unlike the cable of a multi-amplifier system, receive a full range voltage drive, the high frequency cables carry no low frequency drive currents, and hence the magnetic problems which are current related, are not excited. The high frequency signals are unaffected by currents flowing in the low frequency cables. The principle can be extended still further by using two amplifiers as shown in **Fig 2**. In this configuration, the passive high level crossover is still used but the cables are buffered by using two separate amplifiers with the common connection being made ahead of the amplifier. The amplifiers still receive full band *voltage* drive but are only called upon to deliver currents proportionate to the filter sections they are driving. Certainly from my own experiences, bi-wiring probably has

---

inter-strand contact varying as the cable is moved, either by vibration or for relocation.

I have found manufacturers claiming skin effects become significant at around 10 kHz, while others claim that the effect is evident from 1 kHz. I have also met some experienced and learned people who claim that such effects could not be evident until 100 kHz or more.

## Group delays and insulating materials

As with the variability in the claims for skin effects, there are factions who support the concept that transmission line group delays *must* be considered in monitor system design. Again, others contest that such group delays as do exist are usually only in the region of microseconds and are clearly irrelevant to audio applications. Signals reflected back from the non-ideal terminations at either end of the transmission line which the loudspeaker cables constitute, do have finite 'lives' within the lines and can once again, if present to any significant degree, superimpose themselves upon the feedback 'error' signals. The question is, just what degree is significant?

Insulation materials are dielectrics which exhibit charge migration under certain high-level drive conditions, 'bouncing back' again a finite time after the cessation of the drive signal. As with group delays, to what extent are these effects audible and under what circumstances?

Another aspect of insulating material technology is the effect to which the insulation can inhibit or chemically advance the onset of surface corrosion. While this is not a directly audible effect itself at any given point in time, if such effects are evident, over what period of time do these system degradations occur—months, years, tens of years or lifetimes?

## General conclusion

Once a cable has been installed and approved, one very valuable asset for it to possess is consistency. Insidious changes in resistance or linearity of conductivity are alarming properties for any system to have. The suspicion of a system deteriorating with time is unnerving both for the conscientious amateur and the professional alike. If one cannot trust the monitoring to remain relatively constant, then your very judgement is undermined. If skin corrosion is both significant and promoted by certain insulation materials, then cables exhibiting such symptoms are clearly to be avoided. If cables with multiple strands do exhibit perceptible changes due to inter-strand contact irregularity when vibrated or relocated, then these potential variables would also be deemed undesirable, and cables exhibiting such properties should not be specified for any serious system.

While realising that real world applications often make strange demands upon system designers, I doubt that any system could be deemed to be well designed if it had power amplifiers 40 ft away from the loudspeakers, especially if those amplifiers were in turn fed from a plugboard on the end of a similar length of not so heavy flex. May I open the debate, however, by saying that a well designed, well installed system, with amplifiers close coupled to the loudspeakers, will perform optimally using Mk 1 copper cable of adequate cross section. I submit that the more esoteric cables are a means to an end. They may solve the problems of difficult installations or particularly tortuous dynamic impedances but a well selected, conventional copper cable of adequate current rating, short length, and optimum strand configuration for the application will, when well terminated, perform equally well.

a greater effect than any other single aspect of cabling to passively crossed over, full range loudspeaker system.

## Magnetic radiation

While it is impossible to discount aspects of direct human pick-up of the electromagnetic field surrounding loudspeaker cables while being driven, I know of no attempt to qualify or quantify by experimental means any such effects. On the subject of radiation in the reverse direction however, RF interference facilitated by the potential of a cable to act as an aerial, can superimpose itself on the feedback circuits causing grittiness, frequently due to precipitating HF problems in the amplifiers. Such problems are entirely dependent upon environment, so a cable is not necessarily 'better' *per se* because of screening. Again, such a requirement is system specific, depending on the sensitivity of the particular amplifiers to RF interference, and also to the degree of any such interference in the location where the system is to be installed.

## Geometry

Different geometrical layouts of conductors are manifold. All deal in their own particular way with a given designers optimum desired balance of resistance, inductance, capacitance and RF rejection. High capacitance cables were in vogue some years ago. Much of this philosophy was based on line impedance matching, particularly with valve amplifiers where a nominal 8 Ω output could well be realistically fed into a nominal 8 Ω loudspeaker input via a cable of nominal 8 Ω impedance. I say 'nominal' as the output and input impedances are often widely frequency-dependent. High capacitance cables have been blamed for the failure of certain types of transistor amplifiers but conversely there are systems on which they still give pleasing results.

## Insulators

Geometry can also be influenced by insulator design and material. *Teflon* (PTFE) seems to be generally considered a superior material. Insulator thickness can affect dielectric absorption factors and can also vary the inter-strand magnetic attraction/repulsion strengths. Some cables are now being manufactured with strands of insulators within the bunches of individually insulated conductor strands. This is in order to help each strand more closely represent a single core by reducing magnetic inter-reactions and attraction/repulsion forces. Compared to non-litz cables, they are also free from the overall skin effect aspect, reducing the problem of high frequencies skipping from strand to strand as they treat the whole of a tightly packed bundle as a single conductor.

## Conclusions

The variations on all of the different themes are beyond quantification. The complexity of cable behaviour, especially when passing high currents, preclude absolute prediction of how any particular cable will perform with any given system. Although good engineering principles will usually lead to good results, if a cable somewhat against the odds is deemed to improve the subjective quality of a system then that cable should be given due consideration when specifying that system. Actively crossed-over, multi-amplifier systems do appear to be relatively immune from cabling problems as long as good quality, adequately sized cable is used and the amplifiers are kept within 2 or 3 metres of the loudspeakers. The relative immunity is inversely proportional to the width of the frequency band handled by any one cable.

Full range, high power, passively crossed over systems should, where possible, maintain the shortest possible distances between the amplifier and crossover, and crossover and drive units consistent with any magnetic inter-reaction considerations. In almost all such cases, bi-wiring the amplifier to crossover connection will improve results separating the current flow into the two filter sections. This prevents the low frequency currents from modulating and indeed moving the wires carrying the much lower levels of subtle, high frequency information. The benefit is probably further enhanced by the fact that each cable is feeding a much less demanding impedance load, reducing still further the complex current patterns. In many cases, unless everything else has been done to optimise the performance of a system, the money spent on ultra-expensive cables could probably be spent elsewhere in the system to greater effect.

The problem of correlating subjective and objective differences in these areas are largely based on the fact that in response terms we are dealing with the bumps on the bumps on the bumps. The combinations of brain and ear have an awesome ability to resolve fine detail, orders of magnitude beyond our best measuring equipment. The non-linearity of some of the cable phenomena preclude our ability to wholly predict system performance from first principles. There are many 'pet theories' which have no sound basis in known fact; there are also many contradictory viewpoints to be reconciled. The ability of the rest of the system and listening environment needs also to be considered in terms of its ability to resolve some of the cable difference aspects. It is very, very difficult to dictate absolute, hard and fast rules to cover all situations. Considering the complex inter-reaction of performance criteria, both within and between the components of a monitor chain, I do not think that it is too much of an evasion of firm opinion to say that if you can afford it and it works for you, use it! Clearly this latter point may explain one of the reasons for the lack of consensus or undue interest from much of the professional audio world, where rules of thumb and predictability may often count for more than they do in the 'enthusiastic' world of hi-fi. At the end of a professional installation, endless time and resources are rarely available for in-depth studies into the fine detail of subjective system subtleties.

One final aspect of all this is that it may well be that some people are actually far more receptive to cabling effects. It is well known that we all hear differently. I have reports of well controlled circumstances where reputable test subjects have failed to detect differences which others have deemed to be obvious and unmistakable. Should such tests be carried out in groups, the peer pressures and the 'Emperor's New Clothes' syndrome can often carry a consensus when in reality none existed. Having said that, however, there is little doubt that some people do exhibit pattern recognition to a far greater degree than others. Once again the human factors are lurking in the wings ready to muddy the waters.□

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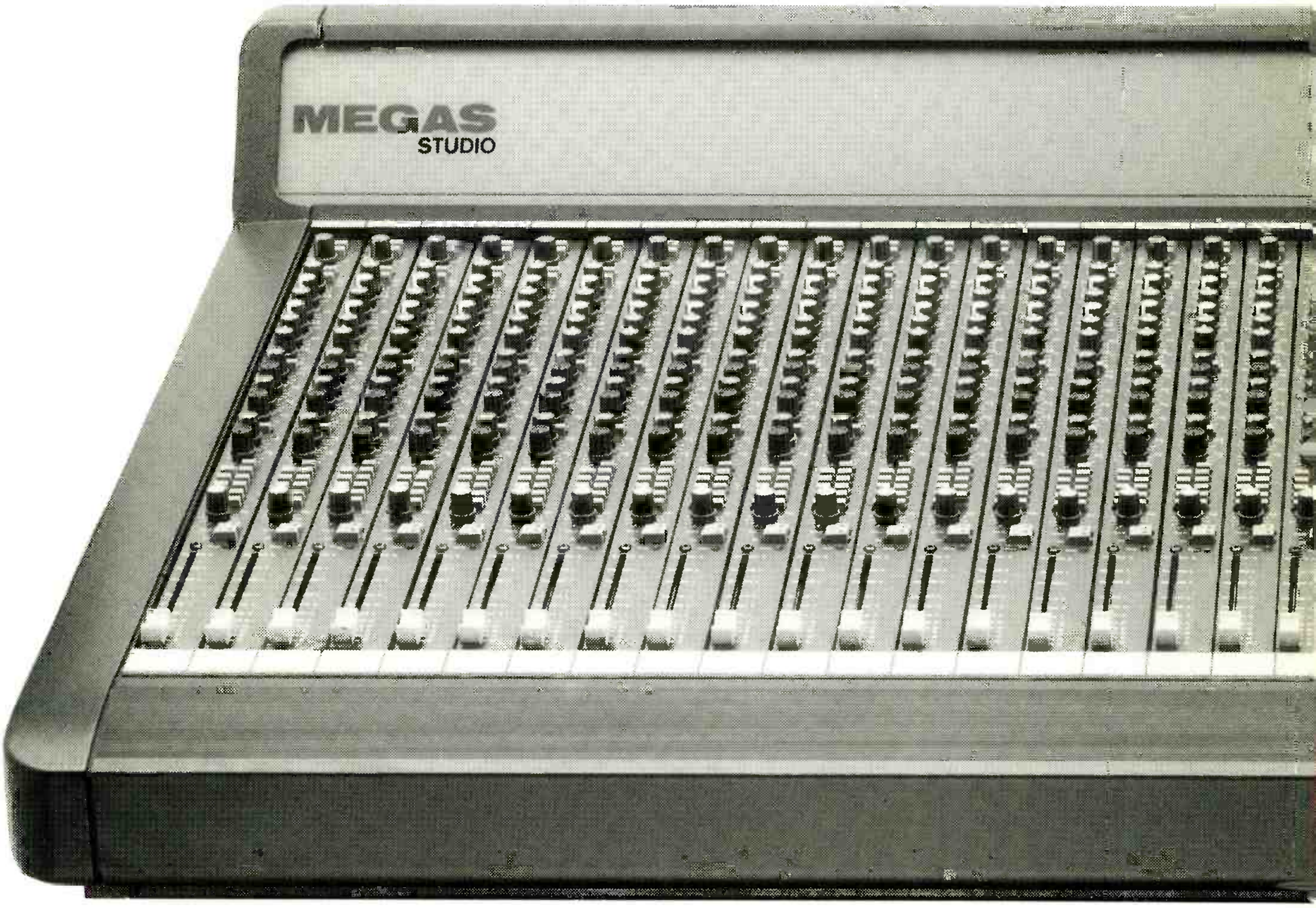


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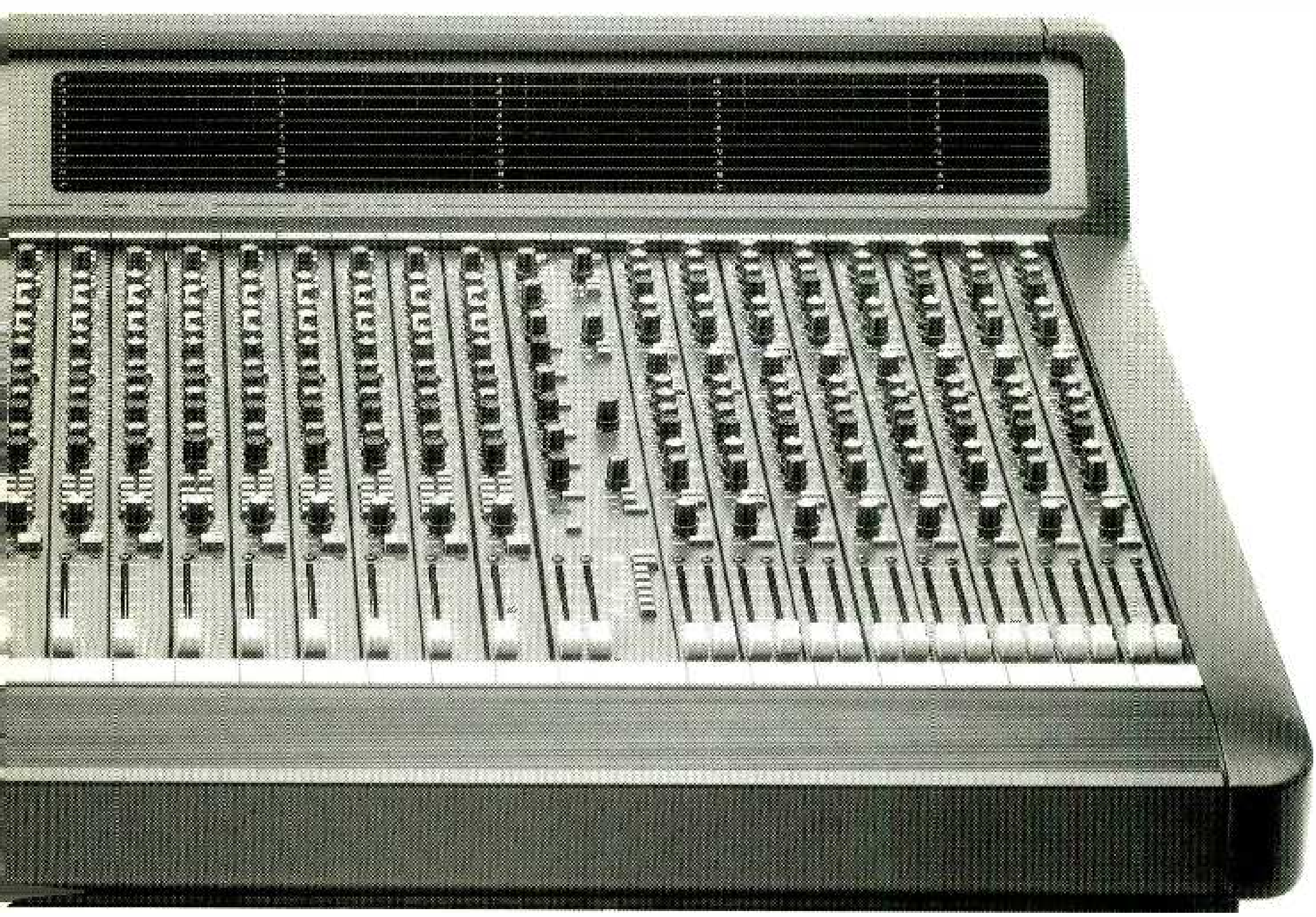


TECHNICAL SPECIFICATIONS: NOISE -127.5db, CROSSTALK >90db, CHANNEL MUTING >90db, INTERCHANNEL >80db.

**Barry Fox**

# Q-Sound reply; TV audiences in 3-D

In mid January, just before the Midem conference in Cannes, I heard on the grapevine that Archer Communications had put out a reassuring statement in the US on the mono compatibility of their *Q-Sound* system. (See also *Studio Sound*, March 1991.)

"As of Jan 1, 1991, all *Q-Systems* (the studio systems used for producing *Q-Sound* recording are equipped with new mono compatibility software programs. . .(which) allow for significantly improved mono compatibility, if required for broadcast purposes," Archer announced. "However, the extent of mono compatibility in a final mix of an audio recording still remains at the discretion of the engineer and creative personnel."

How odd that, after all the years of research supposed to have gone into *Q-Sound*, it is only after broadcast engineers in the UK question mono compatibility that Archer address it.

I asked Archer to explain how mono compatibility can be improved without reducing the spread effect in stereo. On the face of things there will be a straight trade-off between wide stereo with high levels of antiphase and poor mono compatibility, and good mono compatibility with less antiphase and narrowed stereo.

By the time I had left for Midem the only response from Archer was a holding call asking who I was and what firm I worked for.

All this was beginning to feel like a re-run of those unhappy experiences with CBS on *SQ*, *FMS*, *CX* and *Copycode*. In each case the US HQ worked on the principle that there was no need to waste time on what those 'troublesome limeys' were saying. The next step was to announce modifications to the systems in response to what the 'limeys' had said. *SQ*, *CX* and *Copycode* all ended up stone dead. If *FMX* survives and makes headway it will be because the inventors took responsibility for the technology away from CBS.

Danny Lowe of Archer *Q-Sound* later came back with a fuller answer to my question on mono compatibility.

Classical recordings with simple microphone set-ups, he says, often already give poor mono compatibility because of the out-of-phase ambience material they contain. On multitrack mixes, Archer asked engineers and producers what they wanted from *Q-Sound*.

Some felt, "it would be nice if mono would go away altogether". Others felt that a good record should play just as well in mono as in stereo.

"This is ideal," says Lowe, "but not really achievable if one wants a perceivable stereo image."

The majority of engineers polled by Archer felt that the majority of music playback systems are now stereo and that hard panned stereo mix combined with mono in the centre is "an absolute must".

Hence the first generation of *Q-Sound* encoders provided a choice of stereo effect settings that ranged from subtle to extreme. The extreme positions are much less mono compatible.

So the degree of mono compatibility of a *Q-Sound* mix is under the control of the mixing engineer.

Now, clearly as a result of the compatibility issues raised by the broadcasters in Europe, Archer have modified the system, "by reducing the constructive differential in the non-critical bands". This appears to mean that *Q-Sound* now doctors the phase mainly in those frequency bands on which the human ear relies most heavily on image localisation. In other frequency bands, which Archer judges to have less effect on image localisation, there is less phase doctoring.

The theory is that the heavy phase doctoring in the critical bands will create the *Q-Sound* effect in stereo, while the undoctored signals in the non-critical bands will sum their energy to give good mono compatibility.

It would be nice to hear an A/B comparison of the same material, with original and modified encoding. Perhaps Archer can now produce one. After all, according to Danny Lowe:

"*Q-Sound* recognises the need for good mono compatibility and is working very diligently to accommodate monaural playback."

What I would like to know is why, if Archer and *Q-Sound* reckon mono compatibility to be so important, they did not address the question until broadcast engineers sounded the alarm? All we got was hype and puff, obligingly regurgitated by the non-technical music trade press. Why, if the technical issue of mono compatibility was so important, didn't Archer offer some technical background to the technical press? We only found out how the system worked because we found the patent application. And why did Archer simply ignore the first questions on mono compatibility?

Even now, with the modified system, it will apparently still be up to the mixing engineer to decide on a compromise between extreme effects in stereo with poor mono compatibility, and more subtle effects in stereo with better mono compatibility. The radio and TV stations who are worried about mono compatibility (and frankly what radio or TV station can afford not to be?) will still have to listen to every *Q-Sound* track in mono before broadcasting it.

Those with long memories will recall how exactly the same situation arose with the quadraphonic matrix systems of the '70s, CBS's *SQ* and Sansui's *QS*. There was then a trade-off between a strong surround-sound effect with poor mono and stereo compatibility, and reduced surround-sound spread with better mono and stereo compatibility. The designers of the *SQ* and *QS* matrix systems were continually tinkering with them to try to find a better compromise.

The same thing happened with Holophonics, which only worked with headphones. It happened again with stereo TV systems, like Aspex, which were claimed to be compatible, and provide 3D for viewers with special spectacles and perfectly acceptable pictures for viewers without. In that case it was a choice between good 3D with spectacles and horrid colour fringing without, or no visible fringing and no 3D.

How long does it take to get the message across? There is no free lunch.

And now, following hot on the heels of *Q-Sound*, we have *RSS*, Roland's *Space Sound* system.

Although Roland have not yet started pushing the system in the USA, the UK subsidiary have been doing a fine sales pitch. Well aware of the resistance to *Q-Sound*, and indeed any processing system that creates dramatic effects at the expense of mono compatibility, Roland's public stance has been that *RSS* is best used for enhancing ambience and applause, and some special effects.

But Roland's demonstrations to the recording industry have worried some engineers. They fear that once studios have paid £25,000 for a 4-track *RSS* processor, or are hiring it for £1,000 a week, no-one will be able to resist using the new toy on voices and musical instruments. Quite apart from the issue of mono compatibility, some engineers quite simply do not like the effect, even in stereo. And they worry about Roland's suggestion that listeners will 'get used to it'.

Researchers in Belfast, and Michael Gerzon in Oxford, are believed to be working separately on improved systems, which will offer better mono compatibility while giving the ears enough directional clues to make possible true 3-D sound from two loudspeakers.

For, despite the claims that *Q-Sound* and *RSS* are 3-D systems, they only give engineers control over width and height, not depth.

As the first flush of publicity for *Q-Sound* burns out, the BBC have gone overboard on *RSS*. It was featured on Radio 1 and the *Tomorrow's World* TV programme. BBC Records have put out a CD of *Hi-Tech Sound Effects*, which uses *RSS* processing. Roland say that the Rolling Stones are supposedly using *RSS* for the ambience and applause tracks on their next live concert recording. Simple Minds used it for three days, too. And *RSS* spreads the sound for a video game called *Megablast*, developed for Commodore's CDTV interactive video system.

But the independent TV and radio stations are a lot less happy. The Independent Television Commission and Central TV would like it known that, despite what Roland proudly announced, *RSS* was *not* used for *Bob's Your Uncle*, the UK's new game show hosted by Bob Monkhouse that began earlier this summer. Central toyed with the idea of using *RSS* to make listeners with Nicam stereo TV sets think they were surrounded by the studio audience. But Central's engineers realised that if the effects were to be dramatic for the million or so people with stereo sets, the many more viewers with conventional mono sound TV sets would hear Bob Monkhouse greeted with only muted and muffled applause.

For, as with *Q-Sound*, sound that is improved by *RSS* in stereo, is blemished in mono.

Nic Beeby, head of sound and visual operations at Central TV describes Roland's publicity as: "Cheeky. We only borrowed the system for a weekend to try," says Beeby. "We would need to do a lot more work on evaluating its mono compatibility before using it for broadcasts."

Roland have put out several descriptions of *RSS*, of varying technical complexity. None gives anything resembling a clear description of how the process works. But putting the pieces together paints a pretty clear picture.

Bear in mind that in nature humans detect the direction of sound because the human head acts as a baffle. It both delays and reduces the volume of a sound coming from the right of the head on its way to the left ear and *vice-versa*. The brain continually compares the sounds heard by the two ears, and uses the differences in arrival time and volume level to calculate the direction of origin. These phase and amplitude differences between the ears are 'relative' cues.

Conventional stereo reproduction relies on relative amplitude cues and the baffling effect of the head. Both ears hear sound from both speakers but the sound from the left speaker will be less and later at the right ear than at the left ear and *vice versa*. Varying the relative level of the sounds in each channel creates the illusion of a spread between the speakers.

A binaural, dummy head, recording intended for headphone reproduction, sounds wrong through speakers because the left and right channels mix in the room.

Transaural systems, like JVC's *Biphonics*, compensate for this by feeding the binaural signal through a circuit representing the inverse or mirror image effect of the room mixing. But so far transaural systems have obliged the listener to sit rigid at a carefully specified position.

When developing *Q-Sound*, Archer Communications used empirical tests, doctoring the relative phase and amplitude of spot frequencies from a pair of loudspeakers and asking listeners to say where they thought the sound was coming from. Michael Gerzon warned early on that this was a fallacious approach. The results obtained from narrow band testing do not hold good for wide band music.

Roland, it seems, did it differently. Engineers measured the sound field patterns at the ears as sound arrived from points all round the head.

But either way the aim is the same: to doctor the relative amplitude and phase from a stereo pair so the brain is fooled into thinking that sounds coming from a front pair of loudspeakers are actually originating from around the room. And, as with *Q-Sound*, the starting point for *RSS* is a multitrack tape.

Roland have also added extra ingredients. Pyschoacoustic research has shown that when sound reaches the inner ear after reflection from the head, shoulders and ear pinnae, it is distorted by notches, dips and slopes in the frequency response. The reflections, and thus the colourations, depend on the direction of origin of the sound. The brain uses this colouration as an extra 'absolute' cue to the direction of the sound source. *RSS* mimics these ones and also adds transaural processing. It bleeds some of the sound from one channel into the other but in opposite phase so it cancels out when the sound from each speaker mixes in the room.

Although all these ideas are old, such processing is only practical with digital circuitry, which controls phase relationships with extreme accuracy. Analogue circuits are not sufficiently precise and predictable. Roland already makes electronic musical instruments that rely on digital circuitry to generate and manipulate sound in realtime. It was a logical step to develop a realtime directional sound processor.

But the fact remains that all this processing creates phase shifts between channels. This makes music in stereo sound 'phasey', whacks up the level of the difference channel for broadcast or disc cutting, and squashes the top end when, as inevitably happens in any mono TV, radio or cassette recorder, out of phase signals cancel out.

And this is exactly what worried Central TV. As Roland proudly pronounced, the original plan had been to use *RSS* as way of wrapping rapturous applause round listeners with Nicam stereo TV sets. But the massive majority still with mono sets would have heard Bob Monkhouse getting a very dull and lukewarm reception.

Where, in the middle of all this is Hugo Zuccarelli? Whatever *did* happen to Holophonics?□

# Martin Polon

## Our US columnist with warnings for the traveller flying internationally with audio equipment

Alan Leader felt quite pleased with himself. He had first of all scored a major coup in accompanying the University of Northern California (UNC) Choral Society on its near legendary tour of Western Europe; performing at 16 different cities. He held an agreement to record all the concerts and would share the profits with the University from the release of his digital recordings on compact disc. Also, he had put together a remarkable portable recording system. In a beautifully compartmented and lined camera case, he carried his portable DAT recorders, his condenser microphones, his power supplies for the mics, headphones, small monitor speakers and cables. And a large number of spare batteries.

Alan knew there was a war on. But he did not think there would be any problems with his equipment in its beautiful case. He was asked many questions before he was allowed to board his American Airlines flight from the US. His case had been carefully checked after X-ray inspection but the police officer who inspected it was courteous in noting Alan's explanation and passed the equipment. In Europe, everything went exceedingly well. Alan and the chorus travelled all over via the trains with never even a hint of difficulty. The recordings were quite extraordinary, based on the monitored playback. Alan knew that he had a compact disc here; perhaps even two. What seemed to be a daunting challenge had turned into a digital cake walk.

Alan couldn't wait to return home. He had been counting the hours during the last day in Frankfurt and had attempted to reach the airport early. It was just as well that he did since the limo service from the hotel had been disrupted by road maintenance and it took far more time to reach Frankfurt International Airport than it should have. He stood in line for the security check. The officer from the German Border Police was quite concerned by the bag and its contents. Alan was taken into a small office and questioned for some time about the electronics inside his handsome case.

Another man came in and told him that he could not take his bag back with him to the US. It would not be allowed on board any plane without further "testing". Alan was alarmed but asked if he could at least take the DAT tapes back with him. The German officers would have none of that. The airline representative stated that Alan would forfeit his discounted ticket if he did not go onto that flight. Everyone assured him that his bag and its contents would be sent by "the best available way".

As you might not surmise, Alan did see his bag again. It took over a month but it did make it back. The DAT recorders had been disassembled and put back together again improperly. The tapes were badly contaminated with a magnetic field from a large motor that drew them on a rubber belt through an X-ray chamber. Being digital, there was still hope of pulling the performance out from under the garbage. But his beautiful bag was torn, dirty and scuffed. Never the less, Alan was lucky. He had got everything back. Nobody would take any responsibility for the occurrence. The airline blamed the German Government and the Germans blamed the airline.

Needless to say, as always these opening stories are a compilation of various anecdotal episodes and recent personal experiences plus, in this case current news media reports. In fact, the climate for travelling with electronic equipment of any kind—let alone complex audio equipment unfamiliar to airport security personnel—has never been worse. Further, these circumstances are not expected to improve soon, the so-called end of the Gulf War notwithstanding.

First, the current state of frenzy surrounding airport and airline security operations began some time ago. To be precise, it began in the cloudy skies over Lockerbie, Scotland, several years ago. The downing of Pan American flight 103 by a plastique explosive bomb concealed in a Toshiba portable radio, forever changed the view that civil aviation authorities had of passenger-owned electronic equipment carried on board civilian aircraft. The complication of the Gulf War and several terrorist episodes intermediate to the two main events, have caused the most rigorous rule tightening seen through the entire history of air travel.

The following is a relatively accurate synthesis of warning information from the computer reservation systems of several of the world's airlines, designed to help passenger agents advise departing passengers on the carriage of electronics.

1 On all international flights, all electrical or electronic items located in the passenger's baggage, which cannot be identified as completely harmless by the 'local authorities', are excluded from transportation.

2 As above, items such as personal computers, laptops, radios, VCRs and other similar items may be refused. No carriage to be provided either as checked baggage or as carry-on items, if refused.

3 Items such as electric shavers, pocket calculators, small hairdryers, and 35 mm cameras may be accepted as specific checked baggage only on certain international flights. Acceptance of these items is subject to change on a country-by-country basis.

4 On flights from Germany, German Government and German airport authorities are refusing to accept any responsibility or liability for any items denied carriage and left behind.

5 All checked baggage will be X-rayed and passengers may be asked to remove electronic items for inspection or denial of carriage.

6 German-bound passengers are being encouraged to return electronic items to their homes or businesses before they leave the continental US.

7 Express services will be available at US international gateways at the passenger's expense to expedite return of restricted items.

8 Certain countries restrict carry-on or hand baggage to one item only.

9 If there is any doubt, don't bring it.

The pattern of restriction is much greater on international flights between countries but varies significantly from country to country. It is clear from the above regulations that Germany is applying significantly greater pressure against carrying electronics of any kind on-board aircraft than some other nations. Analysts speculate that Germany is responding after the fact to the Lockerbie bombing. Many in the international law enforcement community feel that Germany 'missed the boat' in stopping the radio/bomb from being placed aboard Pan American 103.

Domestically in the United States, Federal Aviation Administration rules are being interpreted by the airlines as requiring anything that uses a battery or batteries to be physically demonstrated by the passenger in question. That means turning the unit on and displaying its function. CD players have to play, radios have to receive, computers have to compute and so on. Items that do not satisfy the security personnel may have to be left behind, though with somewhat more assurance of return by the airlines in a domestic flight situation. The overall airport security schemes are at level four, which is the highest level ever instigated. No one is supposed to be allowed inside the terminals except ticketed passengers.

Now, you may ask what does all this have to do with audio? The answer is that thousands of audio professionals every year use commercial air travel to move themselves and specific pieces of equipment from Point A to Point B. Sales and marketing personnel, freelance recordists, broadcast news personnel, prototype engineers bringing new products to an AES show, etc, all travel by air. The problem is that if security rules and security personnel are inflexible for a known quantity like a video camera or shortwave radio, consider the scrutiny of a *Portastudio* or digital signal processing equaliser. Both will sit mutely while a hulking police officer ponders their fate and yours. "Plug it in and turn it on," has become the watchword of US airport security for

electronic devices.

It might be helpful to list some of the airports where travellers with electronic equipment and audio equipment have reported real difficulties:

- London—Reports of problems with the British Airports Authority (BAA) departure inspection staff. . . especially with electronics. . . especially at Heathrow. . . and especially at Terminal Four. Airline personnel from several carriers including British Airways have commiserated with passengers whose equipment has been disassembled or otherwise negatively impacted. The major problem seems to be the reliance by the BAA on hand inspection techniques to clear most carry-on items. This can thrust the unsuspecting passenger into the tired and overworked hands of the 'Boys in Brown' during peak traffic periods. Carriage of specific electronic items has been denied simply because of the unfamiliarity of inspecting staff with the unit in question.
- Paris—Here the problem is the uncertain nature of the responses to electronic items by the National Police responsible for passenger departure inspections. There has been an erratic pattern of rejection and acceptance in security investigation involving electronic/audio items. In the past, the security staff have been notorious for running clear plastic bags obviously full of 35 mm film cassettes through the X-ray. Now, there are extra thousands of police at Charles De Gaulle and Orly. Bottom line from numerous reports is the unpredictability of the process.
- Frankfurt—It is obvious from the copious regulations enforced, that German Border Guard detachments at the airport are a very tough hurdle to leap. Follow the airline instructions reproduced above, arrive very early to allow for negotiation and plan to rent major items rather than carry them with you.
- New York—Batteries seem to be the focus point of security precautions involving carry-on electronics at La Guardia, Newark and Kennedy International airports. Passengers are requested to remove batteries from all carry-on items to be placed in checked luggage or transported separately by the airline. Shops at the three airports, all operated by the Port of New York/New Jersey Authority, have been instructed to stop selling batteries. The presence of numerous battery powered items and spare batteries has become a virtual guarantee of a detailed inspection.
- Los Angeles—A reputation as a difficult location for carry-on electronics. Also reports of theft from the inspection area, as security staff check an individual with metal detectors while X-rayed carry-on bags continue down a belt to be grabbed by unscrupulous individuals. Seemingly not enough staff at the inspection stations and among the airport police—who nominally oversee them—during peak periods. Relatively inflexible rules generally followed to the letter can cause difficult misunderstandings over unfamiliar electronics/audio items.
- Moscow—In the Soviet Union and for that matter in the whole of the Eastern bloc, reports of the greatest danger with electronics have centred on the expropriation for personal use by guards of passenger-owned and accompanied devices. The usual excuse is the inspection of the item or items at a remote location. The incredible scarcity of these goods in the Eastern bloc is at the root of such occasional problems. To some extent, this footnote to Glasnost has diminished as a problem as the stern and moral eye of the KGB has returned to its business of keeping order, at least in the Soviet Union.

What can the audio professional do to protect audio equipment and/or portable computers while travelling?

- Obey the rules, if you can—If you are told to arrive early, do it! Three hours, four hours, whatever. . . it is the one way you can have some time and latitude to try to arrange a truce for your audio item or items in question. Not everyone will be stopped and not every item will be examined or denied carriage. The one problem with the rules is the relative inability of the 'System' to explicitly define what is expected. Calls to airline reservation centres can be helpful but generally bring the sort of information outlined above from a reservation system. Calls to police agencies are not much more useful since the discretion of the inspecting 'officer' is very much a part of the whole equation. The problems that do occur are most likely to be on the return legs of international flights and no amount of calling will yield substantive answers. To be as sure and safe as possible, be pleasant and polite at all times and enquire directly at the airport in question prior to your departure.

● Carry no batteries—"Leave all your batteries at home and you will eliminate 95% or better of all your electronic equipment problems with airport security everywhere." So states a well-respected airline security expert. Excellent batteries, most usually made by the same multinational megalithic electronics giant you buy them from at home, are available everywhere in the world. That this is contrary to what your mother told you is evident. But children are no longer starving in Europe and batteries are available in Germany... in fact, they invented some of them.

● Cables and wiring in checked baggage—Do just as it says. These items can be replaced easily if lost and will significantly reduce your risk of being stopped and interrogated. Nothing seems to set off security staff more than batteries and wire showing up on X-ray screens.

● Never separate from your electronics—Stay with your equipment. Security people routinely blow up or remotely douse in water upwards of 10 untended briefcases or carry bags per day per major airport. Also watch out for hazards from the non-security minded. Sophisticated thieves 'work' most major airports. They are on the look out for promising bags and carry-on luggage. Watch your bags at airline security inspection stations as the bag goes through and you don't. Put your keys and coins and money clip into the cups provided, before you pass the metal inspection equipment so that you don't become separated from your possessions by a re-check.

● Use airfreight or express services—It's expensive but it is reliable, insurable and dependable. And it sure beats leaving your equipment in a foreign country—in the 'gentle' care of the border guards.

● Carry nothing and rent on arrival—The smoothest way to deal with the current cycle of security hassles. If you can, rent it where you will use it.

The final question to be asked is the most obvious. When will all the hyper-security precautions end and travellers return to a more relaxed mode of carriage? The most realistic appraisal heard yet came from a retired veteran American foreign service officer, speaking without official status: "The facts of life regarding air travel are not what we want to hear; yet it appears that the current high levels of security are probably with us for the near term and possibly beyond. First, the time lag for terrorist groups to operate on a real or imagined act by the West frequently exceeds 18 months. Second, the threat of extremist groups taking action will not fade as long as there is unrest in the Middle East. The Gulf War is over, yet there is no permanent peace as of yet. The Israeli question is another dangling participle. Lastly, terror does not have to come from the Middle East at all. We have recently seen action by those from the Indian subcontinent, from Eastern Europe, from Ireland North and South... from virtually everywhere. Couple that with a strong distaste for a loss of life and aircraft by the airlines and governments involved and you have very little opportunity for change in the short term."

What more can we say!! □

# Opcode Studio Vision

## Mike Collins reviews recording software

Many control rooms have or use some form of synthesisers and samplers together with a hard disk recording system, but how do you work conveniently with both MIDI sequences and hard-disk recordings, especially when you want to re-arrange the structure of a track? One solution could be Opcode's *Studio Vision* software for the *Macintosh*. This is an innovatory 'application' (*Macintosh* terminology for a program), which allows you to record many tracks of audio data (one or two at a time) in conjunction with Digidesign's popular *SoundTools* hard-disk record/edit system, and to select and/or merge these to provide one stereo or two mono tracks for playback. The audio tracks may be replayed from any of the MIDI sequencer tracks in the *Vision* sequencer part of the software. So *Studio Vision* provides an effective combination of hard disk recording/editing and MIDI sequencing. Sections of the audio tracks can be selected along with sections of the MIDI data, and cut, copied, and pasted to re-arrange the music any way you like.

## MIDI sequencer

This is a full-featured professional MIDI sequencer featuring 26 sequences per file with 99 available in each sequence. There is a Graphic Edit Window, which shows the notes as horizontal bars with lengths corresponding to note lengths and heights corresponding to musical pitches. You can just click and drag notes to new start points, pitches and durations.

A List Edit Window is also provided for each track, showing a numerical list of the various events that may be recorded. The bar, beat and clock location of each event is shown, as well as the Note On and Off velocities, and note lengths. The timing resolution provides divisions of 480 ticks for every ¼-note, which is higher than most other popular sequencers, but is considered essential by some for very accurate work. SMPTE sync with MIDI Time Code is available and *Vision* can use the new Apple MIDI Manager software. This lets you run other programs (which also use the MIDI ports) under the *Macintosh*'s MultiFinder pseudo-multitasking software. An example would be Digidesign's new *MacProteus* synthesiser, which comes on a card for the *Mac II*. You could play the sounds in the *MacProteus* from *Vision* with both running on the same *Macintosh*. *Performer* MIDI sequencer for the *Macintosh* actually goes one better by allowing you to have a *Performer* sequence playing the *MacProteus* while writing notes into Microsoft Word wordprocessor. (When I tried this with *Studio Vision*, the sequence stopped playing.)

*Studio Vision* provides a window containing 32 faders, which can alter various types of data according to how they are assigned. This means that the program can provide a certain amount of mix automation before the audio from the MIDI equipment or from the hard disk reaches the mixing desk. You can even use two sets of 16 MIDI channels by using both Printer and Modem ports on the *Macintosh* to connect to the MIDI interface. The program supports loop recording of MIDI data for drum-machine style programming and can also generate new sequences algorithmically.

## Facilities

Opcode suggest several ways in which the software can be used:

- Composers: 'Add "perfect takes" of live instrument solo tracks to your MIDI scores. Record five takes of a guitar or sax solo then cut and paste between them to create the final solo. Quantise rhythm guitar parts. Separate each stroke with the Strip Silence command to create individual rhythm events. Then try non-destructive quantise and track shift commands to get the feel just right. Copy and paste whole harmonic, melodic and rhythmic sections of MIDI and digital audio tracks. If you like a four-bar phrase—in which you've recorded six tracks of MIDI drums, bass, keyboards, strings, horns and digital audio tracks of vocals—copy the complete harmonic and rhythmic phrase. Try using Repeat Paste and insert the phrase four times in a row, or repeat it 10 times at the end of a song for a fade-out. Record and play back the best crash cymbals direct from disk, leaving more memory in your sampler for other instruments. Bounce down unlimited takes of audio with associated automated mixing of volume and pan into a stereo soundfile for playback. Or bounce down several takes of mono guitar to one mono file, leaving a second mono file free for vocals.'
- Songwriters: 'Just as the MIDI sequencer allowed the composer to experiment freely with arrangements, the songwriter can now "fool around" with vocals. Re-order the vocal verses in a song by dragging them to a new start point and instantly hear the result. No more hit and miss flying in of vocals onto your track from a sampler—record them into *Studio Vision* and place them at the bar and beat that you want, and even quantise where they start. Words can be easily stuttered by using the Separate command in the Audio menu, then Option-dragging the word to overlap itself.'
- Post-production: 'Use the digital audio for voiceover and the MIDI for music. Import sound effects digitally at the highest quality via *SoundTools*.'
- Film scoring: 'Add one or two real instruments on digital audio tracks along with your classical MIDI orchestrations.'
- Multimedia production: 'Create MIDI music scores with digitised sound effects and voiceover in a single program. Save the mixed MIDI music as a MIDI File, and premix and capture the digital audio to one audio file for playback. Play back the MIDI using Opcode's *MidiPlay* from *HyperCard*, *SuperCard* or *MacroMind Director*, and play the digital audio at the same time using Digidesign's *Sound Access XCMD*.'

It is easy to see the potential but now, at last, I can record guitar parts and vocals alongside my MIDI sequences and cut and paste them while arranging songs.

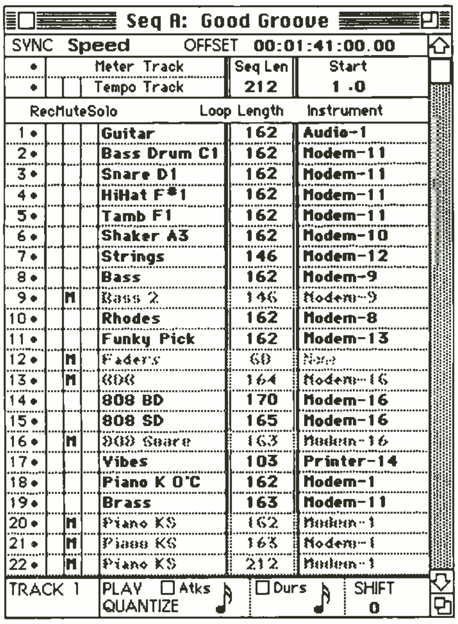
## In action

First you open up a new sequences file and decide whether to record audio or MIDI data first. It is important to decide on your tempo once you have the MIDI data recorded, because you can't alter it once the audio is there without the audio data falling out of sync with the MIDI.

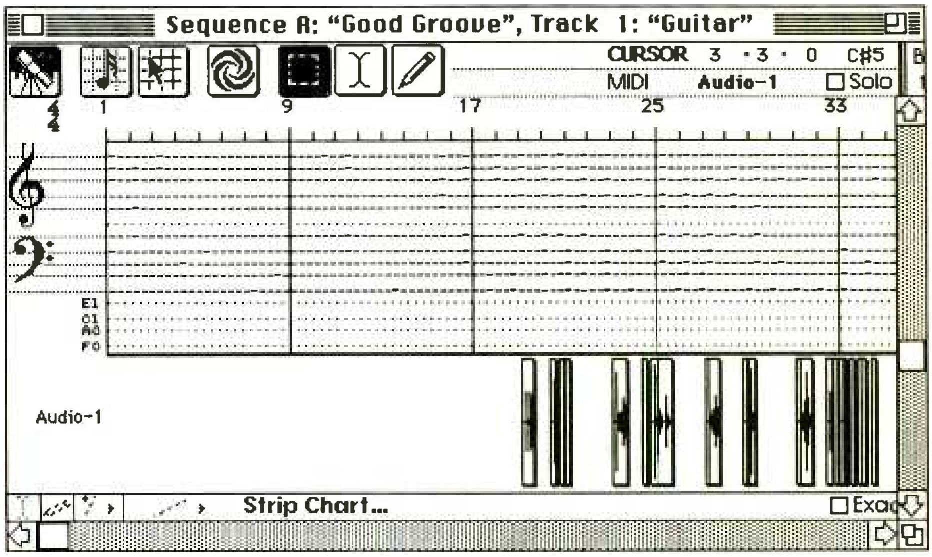
I decided to record several MIDI tracks first, and put down piano, bass, drums and some other parts before recording a guitar track. To record audio, you must hook up the *SoundTools* system on your *Macintosh*. This system includes the AD in analogue to digital converter for the audio inputs, the Sound Accelerator digital signal processing card, which resides in a slot in the back of your *Macintosh*, and *Sound Designer II* editing software. You do not need the *Sound Designer II* software for recording and playback of your *Studio Vision* sound files but you will probably want to use it for certain types of editing of those sound files. You can get a digital in/out unit if you prefer to keep all your audio in the digital domain. A professional digital interface with high technical specifications has also recently been made available.

Once the basic sequences were in place, it was time to select a track to record the guitar into. This was just a matter of typing a new name into an empty track on the **Sequence Window**, and clicking in the record select box. The output of the guitar amp was plugged into the left channel of the AD in unit and then it was time to set the level by referring to the **Record Monitor Window**. This features a meter for each channel with a clip indicator just past the top of its range. The actual level is set using rotary pots on the AD in unit.

Once the guitar track was recorded, the next step was to strip the 'silence' between the notes. The idea here is that you can separate individual notes or groups of notes or sounds into separate audio segments. The playback start points of these sound segments appear in the List Edit Window numerically, and visual representations of the segment waveforms appear in the Graphic Edit Window. These can then be moved around to anywhere you like in the sequence by dragging segments with the mouse in the **Graphic Window**, or by typing in new start points or applying quantisation in the **List Window**. It is




**Sequence Window displaying the audio and the MIDI tracks**




**Graphic Window showing a guitar recording with 'silence' stripped leaving a number of separate sound segments, containing one or more guitar notes**

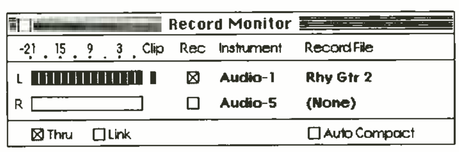
also very easy to shift whole sequences backwards or forwards in time along any track. You can easily try out quantise or track shift values non-destructively, even with the sequences running, and then make these edits permanent when you are happy with them.

# Cut, copy and paste

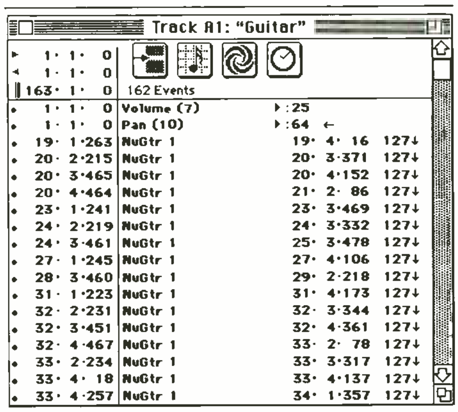
Once the silence was stripped it was possible to compact the file to save space on the disk. With the basic structure of the track in place, including the guitar part playing in the verses, it seemed like a good idea to try extending and re-arranging the piece by copying an extra verse to the end of the sequence and re-arranging the running order of some of the sections. To do this involved selecting the region to be copied, executing the copy command and then choosing a paste point and executing the paste command. Selecting the region to include audio data on the guitar track and *MIDI* data on the relevant MIDI tracks involved shift-clicking on the relevant MIDI tracks in the Sequence Window first, and then dragging the cursor over the ruler area of the guitar track's Graphic Window.

# Editing audio tracks

I deliberately played some of the guitar notes slightly off-time in certain places. I tried quantising to $^1/_{16}$-notes (the correct value for this part) and listened back with anticipation. Unfortunately, worse timing errors resulted than on the unquantised track. So I tried 'spot editing'




**Record Monitor Window checking incoming audio levels. A signal present on the left input channel is causing the clip indicator to light**




**List Window giving the start times of the sound segments in the guitar recording by bar, beat and clock locations**

the dodgy notes or groups of notes using the List Edit window. The clock locations should have read 000, 120, 240 or 360 for $^1/_{16}$-note positions in the sequence. I found one note appearing as a single audio event, which sounded a little late. Its start point had a clock location of 270, so this appeared to indicate that it had been played 30 clocks late. I altered this to 240 and listened. It was still a little late. I moved it 30 clocks earlier to 210 and it sounded exactly in time! I couldn't work out exactly what was happening here. I figured out that the sound segments must have contained small amounts of sound prior to the perceived attack of the guitar notes thus it was necessary for the segments to start playing about 30 clocks earlier for the guitar notes to sound in time. To put this theory to the test I decided to try recording a clave playing ⅛ notes. The clave 'hits' were very clean and percussive, as you would expect, and when I 'stripped the silence' and quantised these, everything worked as you would anticipate—no problems at all this time.

To examine a sound file in greater detail, a menu command is available to transfer you to *Sound Designer II Version 2.0* software, assuming you have this program with your *SoundTools* system. So here you can 'normalise' a recording which you have under-recorded, or apply the range of digital signal processing functions, such as EQ or compression. Finally, it is probably worth mentioning that the current software, *Version 1.22*, features improved playback when linked to SMPTE, and has several small, but useful, program enhancements and bug fixers.

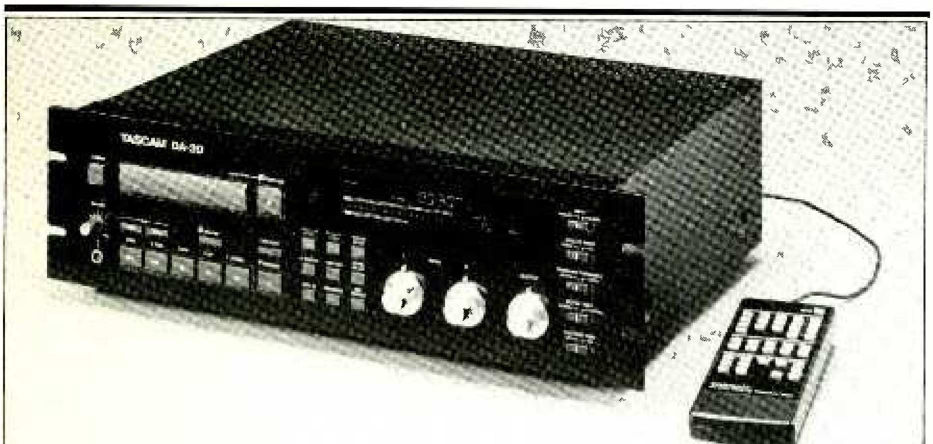
# Conclusions

The MIDI sequencer is first rate, although other *Macintosh* sequencers such as *Performer* or the recently-released *CuBase* provide certain more advanced features preferred by some users. However, the ability to re-arrange audio recordings along with MIDI recordings allows a new dimension of flexibility in the studio. As the manual says 'flying in vocals is now a b-b-b-b-breeze . . .', or words to that effect!

Despite the problems I encountered when editing the audio, *Studio Vision* does let you correct 'live' recordings in a much more sophisticated way than was possible previously. It is a real technological 'breakthrough', and should earn a well-deserved place in the equipment list of any studio wishing to keep abreast of the latest technology.□

# Tascam DA-30

## A technical report by Sam Wise on this studio R-DAT recorder




The *DA-30* is the latest addition to the Tascam range of rackmountable studio equipment, joining the popular *112* and *122* cassette recorders, and *CD-401* CD player. As with these other products, the *DA-30* looks functional rather than ostentatious. It also greatly resembles its family, except for its black front panel, the others being a deep brown.

Most of the legending is white and large enough for easy reading. The meter and tape position readouts are also large enough to read at a distance. Functional layout is good, being better organised than the average domestic machine.

The complete range of input and output connectors is provided, giving both domestic unbalanced and professional balanced connectors and their respective operating levels. All can be left plugged up since they can be selected from the front panel. This allows the machine to be racked and all facilities to be used without any rear access. The remote control has a lead length of 5 metres, and can easily be extended since it requires only three conductors.

The housing is formed from passivated steel in the usual Japanese manner. An internal steel sub-panel supports the front panel components, which are then clad with an extruded aluminium front panel. The rackmounting ears are formed into the aluminium front panel but are backed by a steel angle, increasing the strength for rackmounted applications.

Except for a few largish capacitors, which might be subject to vibration problems, the *DA-30* looks rugged enough to cope with travelling.

The tape transport is simply constructed, being more open and accessible than other units we have tested. It is very easy to remove and should be relatively simple to service. Transport electronics are mounted directly to the mechanism itself, and all connections are miniature plug and socket. Behind this is the power supply. This unfortunately has internally mounted primary and secondary fuses. Why do manufacturers not realise that fuses die of old age more often than equipment failure, and make them externally accessible? Worse, there is an internal fuse on the analogue pcb, which can only be reached by further disassembly or with long-nose pliers.

Two pcbs are mounted on the front panel, one for the switches, the other for the level controls. The digital electronics section is contained on a large pcb on a sub-frame at the top right of the unit, with the analogue section in the shielded area below. This construction should, along with the independent power supply regulator section located on the analogue pcb, provide good isolation from digital noise. The final pcb is used only to interface the serial remote control and parallel external control interfaces. The majority of components are surface mounted, making component level repairs difficult but users rarely do this today anyway. Clear component legends are provided. I was not able to look at a service manual but other Tascam manuals we have seen indicate that it should be of good quality.

## Front panel controls

The left side of the machine houses mainly the front loading transport and transport controls. At the far left is the POWER switch and the PHONES level with headset socket below. The top of this section is taken up by the transport door with OPEN/CLOSE button adjacent. A yellow LED here lights if a DAT cassette is loaded. Beneath the tray are, from left to right: REW, FFWD, STOP, PLAY, PAUSE and RECORD. These perform the expected functions, with PAUSE stopping tape motion in either RECORD or PLAY modes while leaving the tape loaded to the rotary head. This gives a marginally faster start-up than going through STOP. SKIP forward or back controls are sensibly located above the REW and FFWD buttons (being similar functions). They skip in the required direction to the next START ID and then automatically enter PLAY mode. REC MUTE is located above the REC button, and forces the recording of digital zero for 4 seconds.

The last function in this area, STANDBY, is really a mode selector. When STANDBY is active (LED lit), then PROGRAM PLAY, SKIP or DIRECT SEARCH will enter PAUSE rather than PLAY mode on completion.

Pressing RECORD alone places the machine into input monitor mode, allowing levels to be set before recording.

## Remote functions

Further transport functions are available on the wired handheld remote control. For example, a particular selection can be played by pressing its PROGRAM NUMBER on the remote, followed by START. The machine will wind in the required direction, cue, then replay the selection requested.

The REPEAT button, pressed once, causes the current selection to repeat up to 16 times automatically. The function is indicated in the display window. A further press results in repeated play of the entire tape, while a further mode allows repeated playback of a pre-selection of up to 50 items. These functions have proved useful to me when using the DAT as a test tone source, perhaps studios will find a use for them too. Certainly, top quality discos might find them handy while the DJ has a break. Unfortunately

## Manufacturer's specification

**Tape speed:** 8.15 mm/s rec/play (12.225 mm/s play only)
**Record time:** 120 min max (with R-120 cassette)
**Wind speed:** 70 s, full length of R-120 cassette
**Error correction:** 8-fold correction
**Quantisation bits:** 16 bit linear
**Sampling rates:** 48 kHz and 44.1 kHz record and play; 32 kHz play and digital input record only
**Frequency response:** 1 Hz to 22 kHz ±0.5 dB
**Signal-to-noise ratio:** >94 dB
**Dynamic range:** >94 dB
**Total harmonic distortion:** <0.05% (at 1 kHz)
**Channel separation:** Better than 94 dB (at 1 kHz)
**Wow and flutter:** Unmeasurable
**Inputs**
**Analogue:** Nominal input level +4 dBm (*XLR* input), bal; −10 dBV (phono input), unbal
**Digital:** AES/EBU (*XLR-3* input) bal; SPDIF (phono) unbal
**Outputs—analogue**
**Variable:** Nominal output level +4 dBm (*XLR* output), bal; maximum output level +22 dBm
Nominal output level −10 dBV (phono output), unbal; maximum output level +6 dBV
**Fixed:** Nominal output level −10 dBV (phono output), unbal
**Phones:** 100 mW or more
**Output—digital:** AES/EBU, *XLR-3*, bal; SPDIF, phono, unbal
**Remote connector:** 3.5 mm mini-jack
**External control I/O:** Sub-miniature D-type, 15-pin
**Voltage:** 100/120/220/240 V, 50/60 Hz (general export), 240 V only, UK model
**Power consumption:** 34 W
**Dimensions:** 482×150.5×246 mm/19×6 1/16×13 3/8 inches
**Weight:** 9.5 kg/21 lb, exc remote

rewind is not instantaneous, if it were this machine could replace hard disk-based systems in some applications.

The DISPLAY button on the remote causes the display to dim through four brightness levels. This may be a feature everyone wants but even with a blinding headache the standard display level seems fine to me. I applaud Tascam, however, for putting these more obscure features only on the remote and located out of the way at the top of that. The result is a very functional main front panel layout, which is not confused by features only a few will find useful. There are other functions available but no room to describe them here.

A further remote control connector is also provided on the rear panel, having parallel wired logic inputs and pulsed tally outputs for all of the main transport functions. These can easily be interfaced to a custom remote control panel, or controlled from a computer parallel port for special applications. A nice touch.

## Counter controls

The tape counter on the *DA-30* has several useful functions selected by the COUNTER MODE pushbutton. The selected function is displayed in the counter window adjacent to the numeric readout. They are: ABSOLUTE TIME from the beginning of the tape; PROGRAM TIME from the beginning of this item; REMAIN, which gives the time remaining on the tape; and COUNTER, which counts from any arbitary point where it was reset to 0000. COUNTER RESET is located just to the left in the top row of the cluster of nine buttons.

In addition, the counter display can display frames in five-frame increments when a special function is in use to place START or SKIP IDs accurately. To use this mode absolute time must be recorded.

## ID controls

The remaining six switches in this block control the various ID functions. These allow START, SKIP and END IDs to be written or erased and program numbers to be written, erased or renumbered. END SEARCH (which was not working properly on the test machine) locates the end of the last recorded section of tape—that will be useful.

## Level meter/controls

The level meter is peak responding, now the most common metering on digital machines, with a display range labelled from −50 to 0. A bright bar lights if the level exceeds 0, indicating overload. The MARGIN display is useful, indicating the number of dB of clipping headroom remaining due to the actual peak recorded level since MARGIN RESET was last pressed. If margin reads 0 and is blinking, then overload has occurred. If nothing else, this is a good training tool, showing the mixing engineer the difference between his console vu meter reading and peak levels for different types of recorded material. Since it keeps its value until reset, it also provides an indication of peak overloads after they occur.

To the right of the front panel, below the display, are the three level controls: INPUTS L and R and OUTPUT. The left and right input level controls are separate but mechanically ganged. Thus the controls track but can be offset to allow for source level mismatch. As mentioned later, tests reveal excellent electrical tracking accuracy when the input level controls are set to the same level but even small offsets introduced at one control level cannot be maintained when the level controls are altered. This is not a fault, just a fact of life, since audio level controls by design are not electrically linear. Personally, I like the Tascam level control system.

There is a single output level control, since the manufacturers assume the user will match the stereo level correctly on input.

## I/O selection

The provision of multiple inputs and outputs is a handy feature of the machine. As mentioned earlier, since the input source is front panel selected, everything can be plugged in and left. The top switch selects between analogue and digital inputs, while the next selects either the UNBAL (phono socket) or BAL (*XLR*) analogue inputs. These are −10 dBV and +4 dBm levels respectively, though dBm is really a misnomer since the input load is high rather than 600 Ω. The dBu, which is not really an international unit, allows us to assume a voltage level equivalent rather than a power level which the dBm really defines. The fourth switch selects between the coaxial (SPDIF/phono connector, unbalanced) and AES/EBU (*XLR*, balanced) digital inputs.

Sampling frequency automatically tracks the source on playback or when a digital input is in use. For recording analogue sources, the SAMPLING FREQUENCY switch selects 44.1 or 48 kHz. On playback only, or from a digital source, 32 kHz is automatically selected when required.

The last switch is PROGRAM TIME ON/OFF. When off, the *DA-30* will record programme time on the first 9 seconds of an item only. This means that if a tape is loaded which is cued midway through an item, the displayed time will be incorrect. On the other hand, in this mode, START IDs can be altered at any time without a problem. When on, the *DA-30* records the programme time throughout an item, giving instant display of the correct time at any point. This can be very useful in applications like sound effects production. But, if START IDs are altered, the continuity of the programme time is disturbed. Make your choice before you start a tape, according to its expected application. In my acoustic testing applications, it is handy to be able to cue any tape, anywhere, at any time to allow for easy data analysis.

## Rear panel

The only items on the rear panel are the connectors listed in the manufacturer's specifications. Everything plugs in here except headphones. All connectors are of good quality. Gold flashed or plated phono connectors are provided for all unbalanced connections, with *XLR* types (unfortunately wired pin 3 hot) provided for all balanced connections.

# Test results

## Inputs and outputs

The balanced inputs are measured first. Input impedance measures just under 10 kΩ, and output impedance about 10 Ω, compatible with professional operating standards. Input common mode rejection is shown in **Fig** 1, an acceptable performance.

At the nominal input level of +4 dBu, setting the input level control on the front panel to maximum gives a peak recording level of 3 dB below clipping on the meters. In other words, a +7 dBu input level can achieve clipping, a sensible maximum sensitivity. With the output gain also at maximum, this +4 dBu input gives an output level of +19.3 dBu. The input electronics could not be overloaded with the maximum test set output amplitude of +30 dBu.

Assuming a typical professional maximum recording level of +12 dBu, the input level controls must be set to just above position 7 on the marked scale of 0 to 10 to provide the same level 3 dB below clipping. When the output level control is set to give unity gain (+12 dBu output level), it lines up at the same mechanical rotation marking. Very sensible and practical.

Changing to the unbalanced inputs and outputs, and using the manufacturer's nominal level of −10 dBV, the same recording level and unity gain through the system are achieved with the same front panel level settings. The unbalanced inputs overloaded with an input level of +18 dBV, higher than most 'domestic' equipment can produce anyway.

To check crosstalk between balanced and unbalanced inputs, the unselected inputs are set to give a recording level 1 dB below clipping. Then the leakage is measured with the selected input terminated in 50 Ω. The results are virtually identical to the background noise level, ie excellent.

Input gain control law and tracking are shown in **Fig 2**. The law is a good, practical audio taper, and matching is excellent, remaining within 0.4 dB over the range from 0 to −40 dB settings. Remember, however, if the input level controls are offset to correct for source level errors, the matching will not be this good. The output level control is also very good, giving tracking between channels of better than ±0.5 dB across a 50 dB control range.

## Frequency response and crosstalk

All of the following measurements were made on the balanced inputs and outputs unless otherwise stated. **Fig 3** shows the frequency response, which is flat ±0.15 dB across the audio range. There is a genuine small offset between the left and right channels of about 0.35 dB in the balanced mode, which does not occur in the unbalanced mode. This can, of course, be removed using the front panel level controls and is insignificant anyway. The actual response of the left and right channels is superbly matched, with amplitude and phase errors below the measuring system tolerance.

While performing frequency response measurements the absence of an 'Emphasis' recording mode became apparent. However, **Fig 4** shows that when replaying a tape with emphasis, the correct replay roll-off is applied.

The broadband frequency response of **Fig 5** shows the effect of the anti-aliasing filter, which

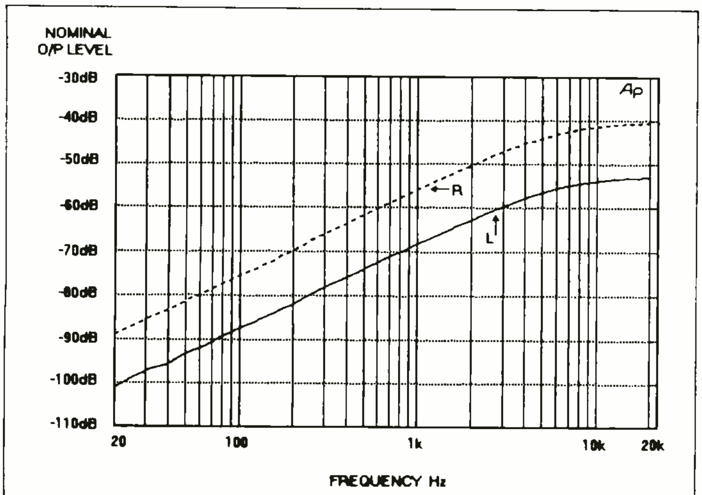



**FIG 1: Input common mode rejection ratio**

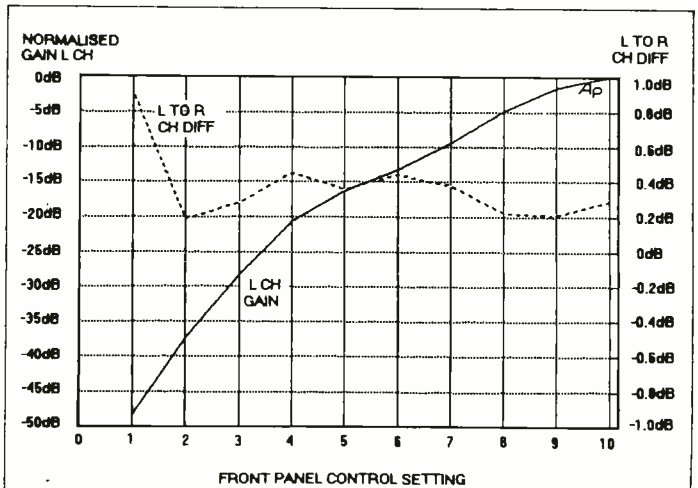



**FIG 2: Law of input gain control**

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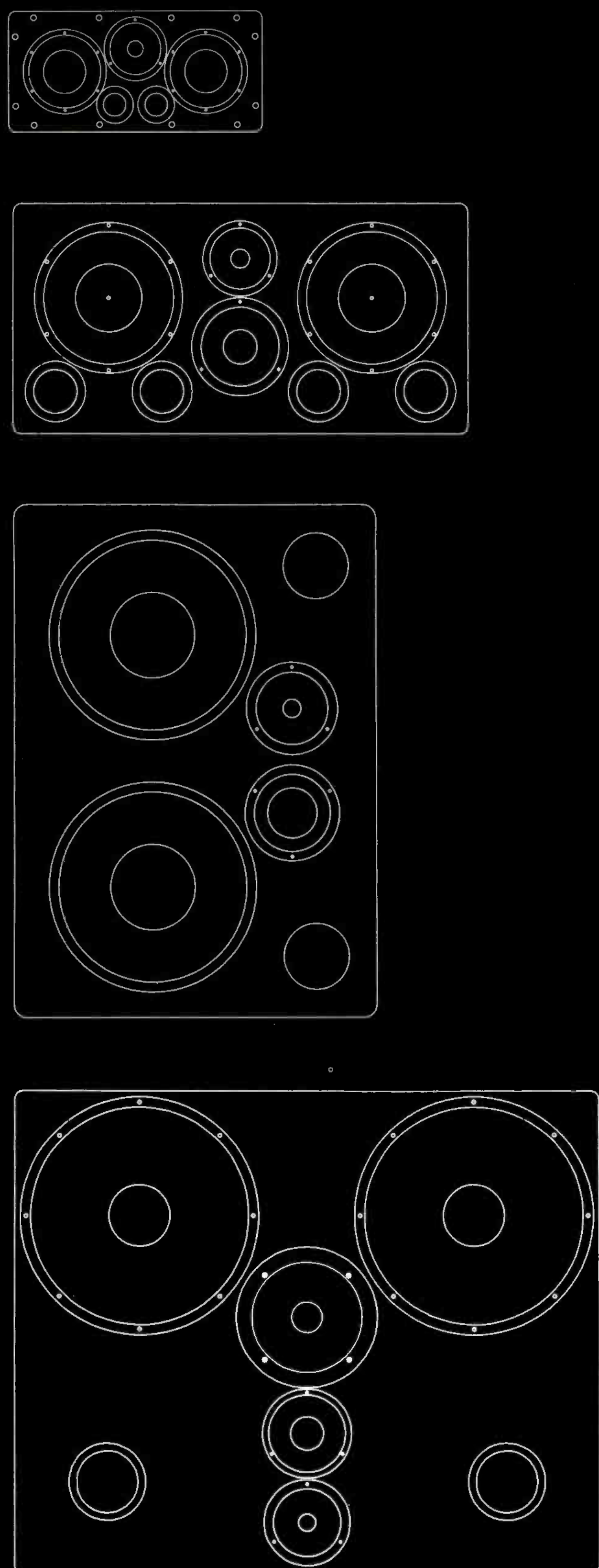

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has no real in-band ripple and a very steep roll-off.

Crosstalk 'off tape' is very good, being below the noise floor. However, when the inputs are connected, left/right separation degrades to about 70 dB, less than the manufacturer specifies. This is the only specification failure we found.

## Noise and distortion

Broadband noise measurements give the results shown in **Table 1**, which are so near the manufacturer's specification as to make no difference.

The ⅓-octave noise spectrum in **Fig 6** is remarkably free from any mains generated noise products and is almost perfectly 'white'. For

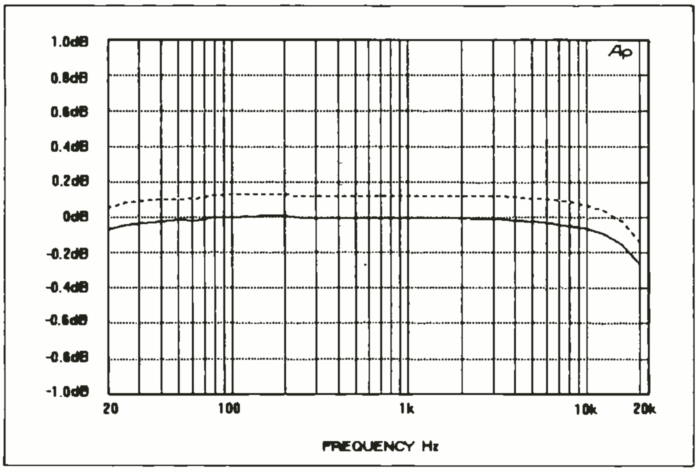


**FIG 3: Frequency response, analogue input to analogue output Output level referenced to 1 kHz on left channel**

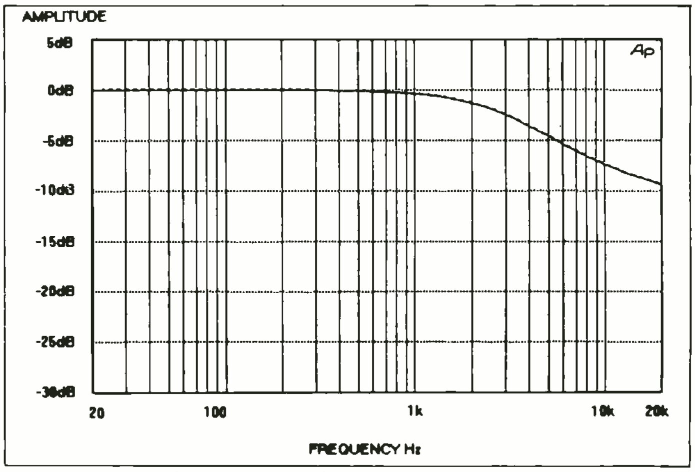


**FIG 4: Replay de-emphasis (dBr) ref LF level Left and right channels are perfectly matched**

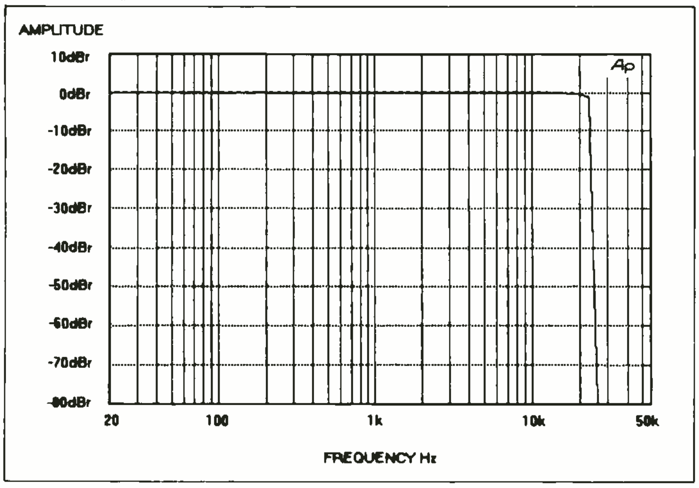


**FIG 5: Wideband response, analogue input to analogue output Output level referenced to 1 kHz of left channel**

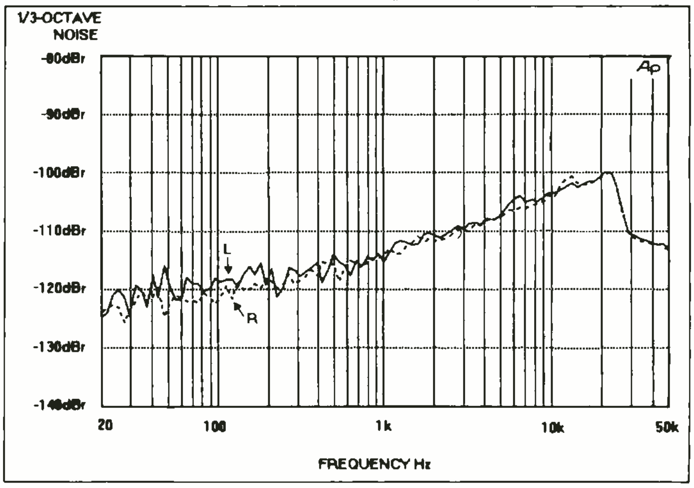


**FIG 6: ⅓-octave noise, analogue input to analogue output**

comparison, an FFT-type noise spectrum was measured using the new digital measurement capability of the Audio Precision Dual Domain test set. The results, shown in **Fig 7**, are flat in frequency response, verifying the 'white' nature of the noise. Some small noise tones are evident but at very low levels.

Modulation noise is also low as **Fig 8** reveals, reaching a maximum of 3 dB at low frequencies but generally less than 1 dB.

Quantisation distortion is shown in **Fig 9** from analogue input to analogue output. The 'glitch' at the right hand end is genuine and was traced to the digital to analogue converter section, which has an error in the most significant bit. Fortunately, this is audibly masked by the high level of the signal itself.

**Fig 10** shows that input/output linearity is excellent, being practically flawless down to levels below 90 dB below clipping level.

Standard audio signal distortion measurements were made, including THD+N, IMD and DIM. All of these gave similar results, being below 0.01% at all frequencies and all levels above −15 dB reference to clipping level. Below this level measurements are noise limited. The D/A converter error is clearly visible as a distortion peak as level is increased through the −6 dB point, but remains below 0.01%.

During the measurement process, signals are routed through all possible pathways: analogue to analogue; analogue to digital; digital to digital; and digital to analogue. This, of course, enables us to determine the cause of some of our observations but also verifies that everything works as expected.

| Method | 22 Hz to 22 kHz RMS | 400 Hz to 22 kHz RMS | 22 Hz to 22 kHz quasi-peak | CCIR weight quasi-peak | A-weight RMS |
|---|---|---|---|---|---|
| A/A | -93.7 | -94.3 | -90.2 | -83.2 | -96.7 |
| A/D | -93.4 | -93.9 | -89.4 | -82.8 | -97.3 |
| D/A (= 0) | -106.8 | -107.0 | -102.7 | -96.1 | -109.4 |
| D/A (dither) | -92.0 | -192.1 | -89.9 | -82.3 | -94.8 |

**TABLE 1**

# Summary

The Tascam *DA-30* R-DAT recorder is an impressive machine, combining versatile features with ease of 'normal' operation. Operating levels are sensible and controls well laid out. The audio performance is also good, being better than most of the machines tested to date. The only omission evident is the capability of recording with emphasis. And, I do not like internally mounted fuses.

We wish it were possible to make some measurement relevant to the durability of R-DAT tape transports and head assemblies. This is beginning to look like a problem with some other manufacturers' machines. Since this is the first DAT offering from Tascam, only time will tell.□

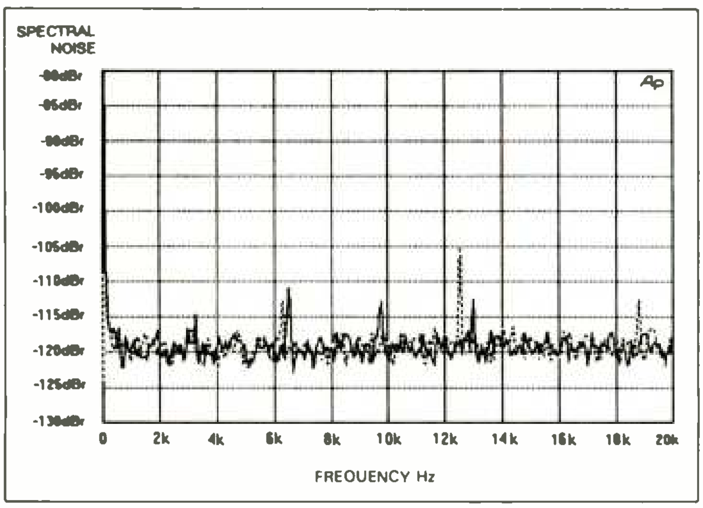


**FIG 7: Narrowband noise using FFT techniques Reference level is maximum output level**

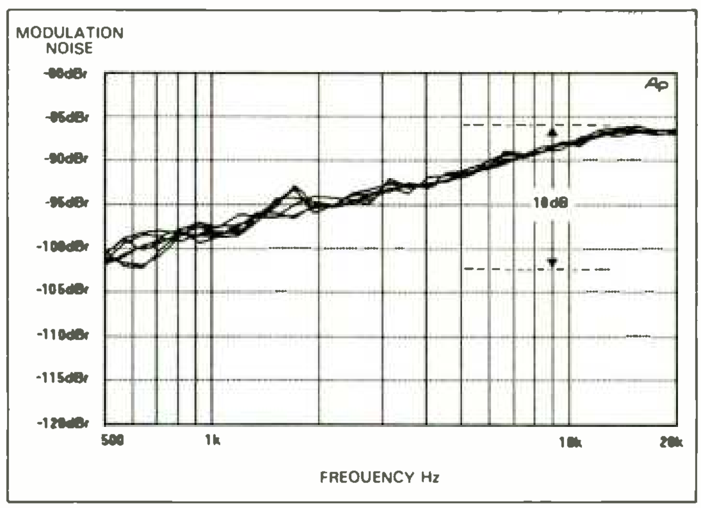


**FIG 8: Modulation noise is difference between max and min curves There is little modulation noise, indicating good linearity**

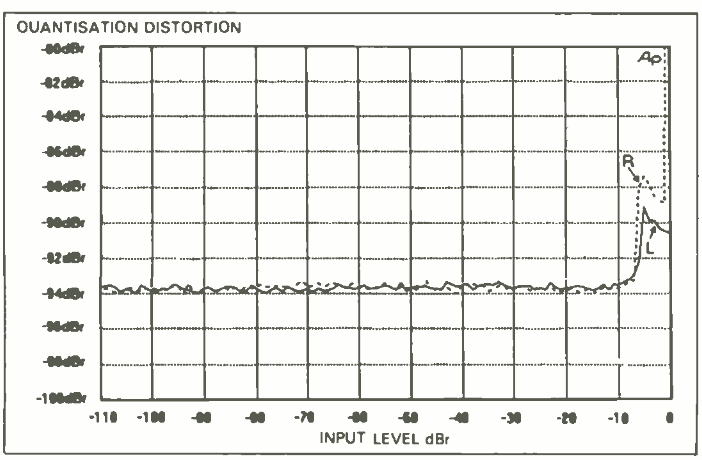


**FIG 9: Quantisation distortion 22 Hz to 22 kHz Analogue input to analogue output Input and distortion levels referenced to max output**

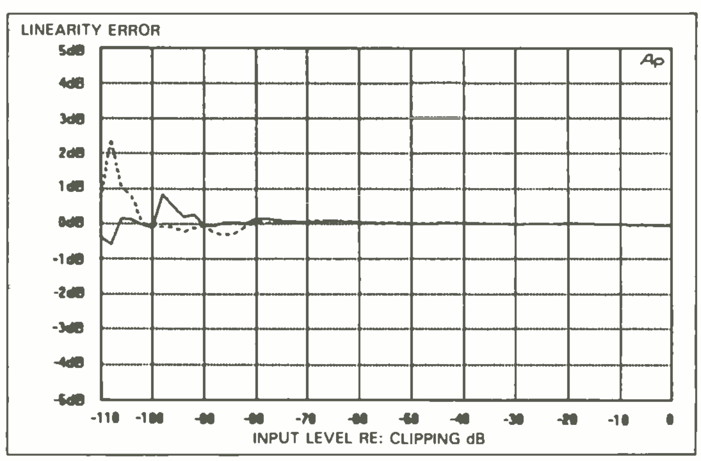


**FIG 10: Input/output linearity at 200 Hz Deviation from a linear I/O relationship. The input level is referenced to the full scale input level. Non-linearity at very low levels is to be expected**

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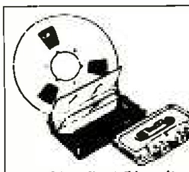
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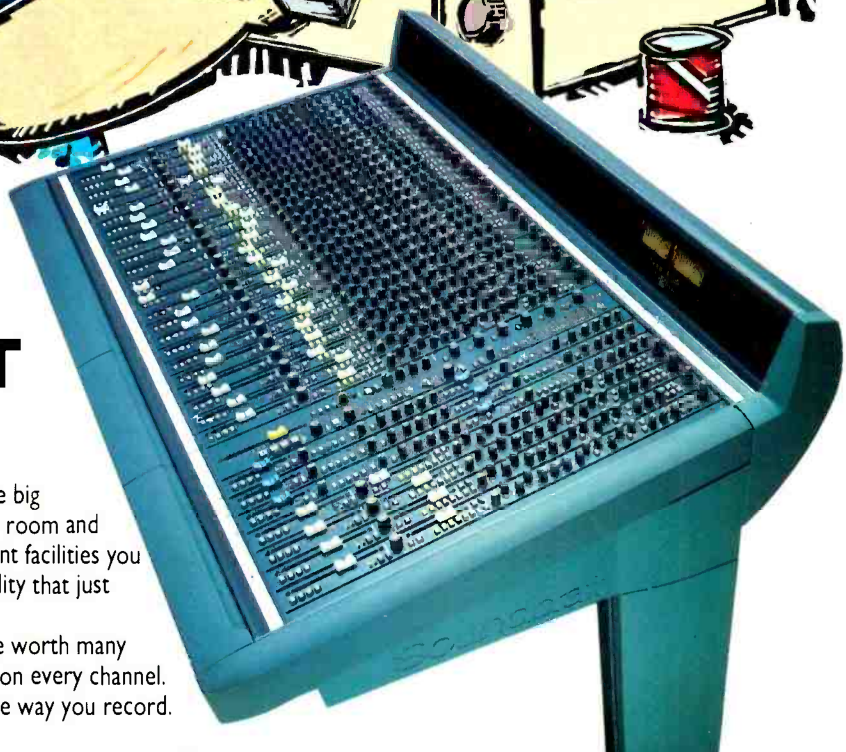

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