STUDIO SOUND

AND BROADCAST ENGINEERING

INTERCONNECTION

Metropolis is one of the most prestigious recording studios ever designed. And within it, you'll find the ultimate in audio equipment.

The new Focusrite studio console.

It embodies Focusrite's committment to the highest standards of sonic purity.

When compared with the approach taken by Metropolis, it becomes apparent that they have shared the same vision.

To create the finest recording environment in the world.

An experience we're sure you'd like to share.

No Compromise

STUDIO SOUND
AND BROADCAST ENGINEERING

Tom Jung at DMP (see page 31)

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TAC Magnum

Broader base saves face

There is little point in denying that times are very hard in the UK recording industry. We hear that there are many other areas of the world where times are also difficult. In London we have seen several influential studios close their doors over the last few months. In some cases this has been brought about through the rising costs of being in Central London and those have either moved out or decided to call it a day. Others have not been able to organise their business in such a way that they attracted work and remained profitable. There are also those who invested heavily but have been hit by rising interest rates, a fall in the property market and the short term attitude and current jumpiness of the banks. All in all, not a very healthy situation.

How long the current climate will last is open for conjecture. At some point, however, we will be back in a positive situation although when that will be, who knows? The only points we can be certain about are that when this happens there will be fewer studios in business; there will be fewer manufacturers and suppliers although much of the attrition in this area will be as the result of mergers and acquisitions; and unfortunately when we do come out the other side many of the previous business problems will still be there—in particular the problem of studio rates. It may even be worse. You could therefore argue that the current situation is something that will change everything and nothing.

Some deep thought is needed to see us through these very difficult times. There are not going to be any heavyweight musical trends that will somehow treble the requirement for studio time and solve our problems. Audio post may well be over subscribed already. Industrial A/V type work looks like it is soon to get swept away in the rush for 'multimedia' where one person has the ability to create both picture and audio without being anything other than average. Creative music making is getting easier and cheaper to completely create within the home environment. And so what is left?

There can be little to be said other than that the studio of the next few years will probably have to become specialists in all types of audio rather than just in one. The experts in a certain aspect of audio will still exist but they may have to be within a broader working environment. Expect to see studios taking on something like being an audio service company where the client will approach with an audio task and the studio will solve it. This will be everything from recording, mixing and production as one would expect but into all other areas of pro-audio as well. There may also even be tie-ups between live sound companies and studios, maybe building manufacturing, servicing, management, publishing, duplication into the equation in some variation. Some studios already have this kind of work base but it may need to be developed even further in the future. To some this may sound remarkably like the way studios used to be; where they were seen as a resource of audio expertise and knowledge rather than just a very specialised recording operation. Should such developments occur then those already with the expertise will be in a far stronger position.

At a tangent, I am finding myself at the receiving end of much information describing software as being 'user-friendly'. Now, my dictionary defines a *friend* as 'one joined to another in mutual benevolence and intimacy' and *friendly* as being 'like a friend'. Now I don't know precisely how you feel about the tools of your trade—both hard- and softwares—but surely this is something of an exaggeration. I choose my friends with care and cannot imagine being joined to a machine 'in mutual benevolence and intimacy'. I think I'd settle for my equipment just being 'user-considerate' but maybe this is where I have been going wrong!

Keith Spencer-Allen

Cover: Photography by Tony Petch

The Avalon Studiosystem.
Some things are just beyond compare.

D&R. Every sound under control.

D'R

New products anger BRPG

Manufacturers who release new technology onto the market without making sure that their products work properly, are evoking the wrath of the British Record Producers Guild. Members of the Guild say they are tired of bearing the cost of further research and development into new recording technology, particularly when it comes down to ironing out bugs in software. They are asking manufacturers to remember that the end user, in other words the producer, wants equipment that works properly, from the outset.

At the last meeting of the BRPG, members said they felt some manufacturers were guilty of trying to reclaim the cost of research and development through sales before the product was ready for general use. Guild members are now warning manufacturers that they are losing patience and that they are not prepared to be unpaid members of the R&D team.

The canopy between the two buildings owned by Imagination in London's West End, creates a very private and spacious piazza. Imagination is a design company with audio facilities and was featured in our September 1990 issue

Exhibitions and conventions

February 19th to 22nd AES 90th Convention, Palais des Congrès, Paris, France.
April 15th to 18th NAB, Las Vegas, USA.
June 5th to 7th APRS, Olympia 2, London, UK.
June 13th to 18th International Television Symposium, Centre des Congres, Montreux, Switzerland.
June 25th to 27th Multimedia 91 conference & exhibition, Olympia 2, London, UK.
July 10th to 12th Pro Audio Asia 91, World Trade Centre, Singapore.
September 8th to 11th Light & Sound Show 91, Olympia 2, London, UK.

Neutrik expands in the UK

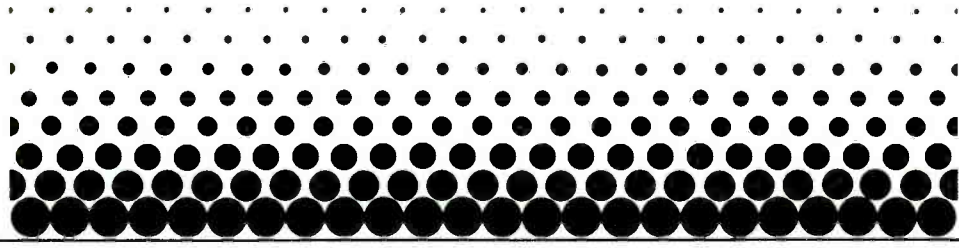Agreement has been reached in principle between Bernhard Weingartner, president and chief executive of Neutrik AG, Leichtenstein and Peter Eardley for the formation of a new joint company, Neutrik Marketing Ltd. The new company will take over from Eardley Electronics, the sole agency in the UK for all Neutrik products, from January 1st, 1991. Peter Eardley will be the managing director of the new company but will also remain chairman of Eardley Electronics.

Pat Walsh has been appointed general manager of Neutrik Marketing Ltd and will be joined by experienced staff from Eardley, as well as Neutrik AG, Isle of Wight.

Neutrik AG UK, the manufacturing off-shoot of the parent company, will move its plant to a new 22,000 ft^2 factory on the Isle of Wight to increase its production capacity of acoustic test instruments as well as the manufacturing of XLR-type and Speakon connectors.

The new address of Neutrik Marketing Ltd will be Eardley House, 4 Uxbridge Street, Farm Place, Kensington, London W8 7SY, UK. Tel: 071-792 8188. Fax: 071-792 8187.

Mark IV buy Klark-Teknik

The boards of Mark IV and Klark-Teknik plc have announced that terms have been agreed for a recommended offer for the whole of the issued and to be issued share capital of Klark-Teknik. Mark IV Industries are the parent company of Altec, Dynacord, Electro-Voice, Gauss and University Sound.

An undertaking to accept the offer has been received from the directors and other shareholders holding 52.4% of the issued share capital. The offer of 69p per share values Klark-Teknik at £11.02 million compared with net assets of £4.29 million at the end of this year to July 31st, 1990.

The present structure of Mark IV is the result of seven major acquisitions in the last 5 years which have taken Mark IV into new markets or have enhanced existing product lines. Mark IV's continuing strategy is to acquire companies that are well managed, are complementary to its core business, and that improve its access to European markets. Mark IV believed Klark-Teknik met these criteria.

Klark-Teknik will continue to be run as a largely independent business within the Mark IV audio division. Mark IV have confirmed that all employment will be safeguarded.

Address changes

● Neutrik's new USA address is 195 Lehigh Avenue, Lakewood, NJ 08701 4527. Tel: (201) 901-9488. Fax: (201) 901-9608.
● Shuttlesound have moved to 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX. Tel: 081-640 9600. Fax: 081-640 0106.
● Benchmark Media Systems have moved to 5925 Court Street Road, Syracuse, NY 13206. Tel: (315) 437-6300.

In-brief

● Glenrothes, UK: Soundtracs head north: Soundtracs plc have recently formed a new Production Division in a 20,000 ft^2 facility at Glenrothes, Scotland. The new factory, which will triple capacity, is to be equipped with automated production lines, complete with robots and will ultimately create new jobs for over 150 people.
● Menlo Park, CA, USA: Dyaxis used to duplicate: Studer Editech, makers of the Dyaxis hard disk editing system, and KABA have reached an agreement where KABA will sell the Dyaxis as an integral part of their 4-track realtime cassette duplication system.
● Stockport, UK: Axis open new facility: Axis Audio Systems have announced the opening of their new hi-tech demonstration facility. Axis felt a need to form a permanent room for demonstrating equipment like DigiDesign's SoundTools and the Akai DD1000 and the S1100 sampler.
● Oxford, UK: SSL's Black Book: Solid State Logic has announced the publication of The Black Book, Volume One featuring whole page black and white photographs of record producers such as Nile Rodgers, Phil Ramone and Hugh Padgham taken in recording studios. The project took a year to complete and features 40 British and American producers with their own short biographies.
● Montreux, Switzerland: NAB/Montreux Radio Symposium: The National Association of Broadcasters and the city of Montreux in Switzerland, have announced a joint venture to stage a NAB/Montreux International Radio Symposium and Exhibition in Montreux in early June 1992. The symposium and exhibition will be the first major European event of its kind, and was conceived in response to the rapidly developing changes and expansion within the radio broadcasting industry.
● London UK: Konk's Studio One has reopened after a fire and features a new control room designed by Recording Architecture.
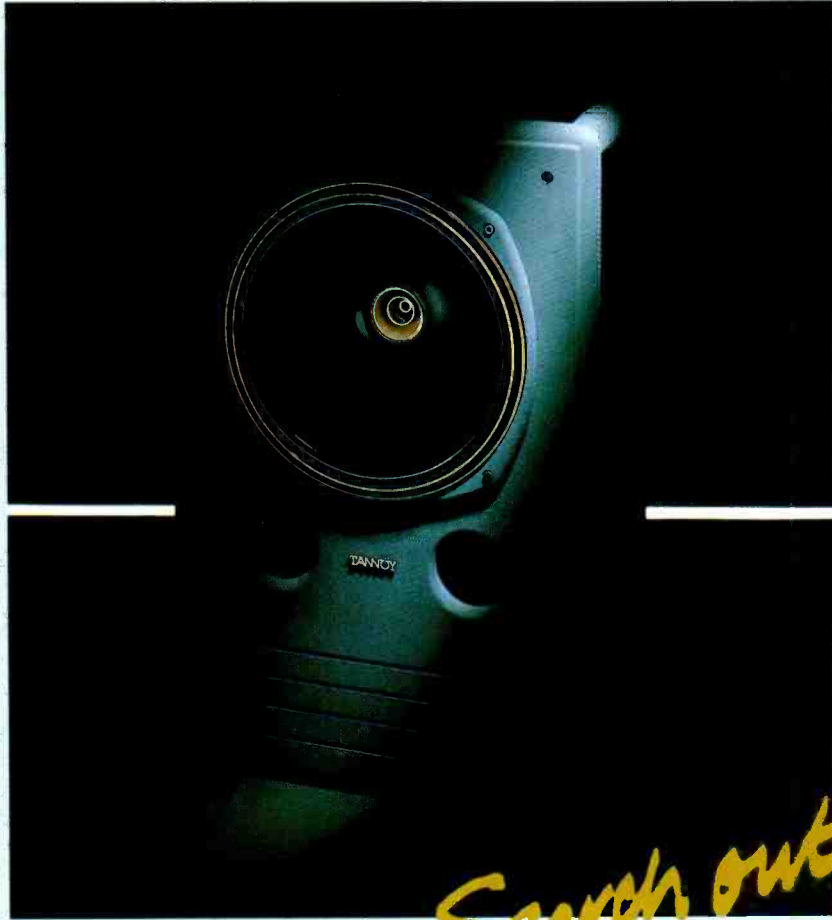
Mozart

"AMEK has achieved a goal which has always eluded me: cost-effective production of mixing consoles of the highest quality.

MOZART, with its combination of advanced signal paths and revolutionary automation, integrated to an unusual depth in the console, impresses me greatly.

The possibilities offered by MOZART are completely new.

During 1989 AMEK invited me to make my own contribution to the MOZART system.

Within the framework of signal flow and automation specified by AMEK, I have designed my own input module which features circuitry built entirely to my specifications. This has resulted in a version of the MOZART which could, as it were, bear my signature.

A new microphone amplifier has been designed which combines the integrity and sonic quality of the highest grade transformer designs with new circuitry which overcomes the fundamental problem of low frequency distortion encountered in even the best of the classic designs.

MR. RUPERT NEVE

Similarly, a very high grade line input stage has been developed which, again, combines the proven qualities of the best transformer with the state-of-the-art performance now achievable. The input device, in fact, behaves much like a transformer and I have called it the T.L.A. (Transformer-Like Amplifier).

Equalization has often been a compromise in the past because it has been expensive in terms of space and component cost to provide the audio quality and completeness of control which might be found in the best outboard equalizers. But this is exactly what I have done in my version of MO-ZART... achieved the best possible audio performance and within a realistically-priced console input channel.

This unique combination of engineering has, for the first time, brought my designs within reach of a greater range of producers, engineers and studios than was ever possible before.

My work with AMEK thus opens up a new era of collaboration in the industry which I hope will help set new standards in console design and introduce a new concept of value."

& I

Agencies

● **ARX Systems,** Australian manufacturers of power amplifiers, processor controlled loudspeaker systems, signal processors, and equalisers have announced the following distribution appointments. For South Africa, Prosound Pty. Tel: 011 337 6556; Philippines, Stagecraft. Tel: 02 834 0539; Taiwan, Prosound Inc. Tel: 04 235 1873; Hong-Kong, Chainford. Tel: 852 601 1293; Italy, Armonia Nova. Tel: 51 765 408; Sweden, Tal & Ton AR. Tel: 31 8036 20; Austria, MTEC. Tel: 724 260860; France, Etelac. Tel: 1 43 002770.

● **Digital Audio Research** have announced the appointment of SLO Viestinta as their distributor in Finland, Markku Haga, SLO Viestinta. Tel: Finland 08382139. Fax: 08382463.

● **HHB Communications** have been appointed UK and Eire distributors of SSL's *Screensound* audio-for-video recording and editing system. The company will also be responsible for UK sales of the *SoundNet* digital audio network. HHB Communications. Tel: 081-960 2144. Fax: 081-960 1160.

● **Audio Concept of Canada** have been appointed distributors for the **Midas** range of live performance consoles throughout Eastern Canada.

● **Clive Green & Co** have signed a general distribution agreement with Siemens AG Osterreich Audio and Videosystems for Germany to distribute the *Cadac* range of consoles. This brings their distribution network to a total of seven worldwide. Gerhard Reiter, Siemens AG Osterreich, Goellnergasse 15, A-1031 Vienna, Austria. Tel: 222 71711 6376.

● **BBE Sound Inc** have appointed various European distributors for their range of signal processing equipment. For Belgium, Titan Audio, rue di Raeymakers 52, B-1030 Brussels. Tel: 322 242 1522; Holland, AEG Netherlands, Aletta Jacobslaan 7, 1066 BP Amsterdam. Tel: 3120 5105 410; Germany, Produktiv Grosshandel, Gilderstrasse 53, B-7530 Ibbenburen 1. Tel: 5451 500180; Switzerland, Embag Musikengros, Militaristrasse 76, Loxhol, 8026 Zurich. Tel: 411 242 9053; Austria, B&T Musikinstrumentenhandel, Marienplatz 43 or 13, 7021 Diassburg. Tel: 431 36 8410; Norway, In Sync Creative Software, Tvetenveien 55, Oslo 6, N-0066. Tel: 472 641430; Denmark, Supersound, Skindergaade 27, 1159 Copenhagen K. Tel: 45 3332 5088; Sweden,

News from the AES

Our next meeting will be held on Tuesday January 8th and will be given by Francis Rumsey on the subject **Does MIDI have a place in Professional Audio?** "MIDI, the Musical Instrument Digital Interface, is widely used as a means of control for electronic musical instruments but is increasingly found on items of professional audio such as mixers, synchronisers and effects units." Francis will explain why professional audio engineers need to understand how MIDI systems may be synchronised with video and audio systems and how MIDI control relates to non-musical devices. He will also discuss the limitations of MIDI as a control system. Francis is well-qualified to talk on this subject as he has written a book entitled *MIDI Systems and Controls*, which is available from the AES.

The meeting will be held at the IBA, 70 Brompton Road, London SW3. The IBA is opposite Harrods and Knightsbridge Underground, between the Nationwide Anglia

Building Society and Boots the chemist. The evening starts with coffee at 6.30pm followed by the lecture at 7.00pm.

To help future planning, the dates, speakers and titles of the following meetings are listed below (more details will be available next month).

**January 8th
Does MIDI have a Place in Professional Audio?**
Francis Rumsey

**February 12th
Loudspeakers, Rooms and Listeners—An Overview**
Floyd E Toole

**March 12th
Digital Audio Broadcasting**
E Meier-Engelen

And now is the time to start planning for the next AES **European Convention,** which will be held at the Palais des Congrès in Paris between February 19th and 22nd, 1991. With a wealth of

papers, workshops and technical visits and the largest exhibition of pro-audio equipment to be seen in Europe in 1991, it is the event not be be missed.

The next event to note is that the AES British Section will be organising a conference with the title **Will You Be Legal?— Implications of 1992 to Audio and Video Engineers** on Tuesday March 19th, 1991. This will cover the effects of legislation to be implemented on January 1st, 1992. This legislation affects manufacturers, designers and installers of professional audio and video equipment and systems. It is intended to reveal the extent and scope of the legislation and will be discussing the strategies for coping with both the legal and engineering consequences of the European performance standards involved. Ignorance of the law is, we are advised, no defence.

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

People

● **Neve Electronics International** have appointed Mike Banks as UK sales manager for the company's range of music recording products.

● Peter Buckland has been appointed general manager of **Nomis Studios,** London. Buckland had been Rod Stewart's manager for live and recorded work for 14 years.

● Dave Pallant has recently joined **Saturn Research R&D.** He previously worked for Soundcraft as a design engineer and more recently at Audio Kinetics as a software design engineer.

● **Quadrant Facilities,** Bristol have recently appointed a new facility manager and editor. Stephen

Soundside Sweden, Asogarten 115, Box 4010, S-102 101, Stockholm. Tel: 468 40 2358.

● **The Rolls Corporation** of Utah, USA, have appointed European distributors. In Germany, Warwick Music Equipment. Tel: 49 9545 4040; Italy, Produx srl, via Calabria 3, 20158 Milano. Tel: 393 11571; Holland, Wils Music Import. Tel: 31 1646 12109; Sweden, Lilliehook Electronics. Tel: 468 506039.

Broadfield is now facilities manager and Andrew Chastney is editor.

● **BBE Sound Inc** of Huntingdon Beach, California, have hired the services of pro-audio and musical instrument marketing consultancy Trevor Cash International to spearhead an integrated European marketing campaign for the company.

● David Bearman has joined **Martin Audio** in the new post of international sales and marketing manager.

● **HHB Communications** have appointed Steve Angel sales manager; Richard Kershaw assumes control of broadcast sales; and Stephen Dunn has joined the company as financial controller.

● Pete Wandless has joined **Hilton**

● **Canford Audio** have secured sole distribution rights in the UK for the Keith Monks range of cable drums. Canford Audio, Washington, Tyne & Wear NE38 0BW. Tel: 091-415 0205.

● **Mark of the Unicorn** have appointed Syco as UK distributors for their entire product range. Syco have also been appointed as exclusive UK distributors for Doremi's *DAWN* (Digital Audio Workstation Nucleus), a disk-based, post-production system

Sound as business development manager. Wandless comes from Focusrite, and before that Harris, Grant Associates.

● **Sarner International,** the audio visual group, have appointed Terry Lockwood as director of sales. Lockwood joins the company from the graphic arts industry. Stuart Marchant, previously director of sales for Sarner AV Equipment has been promoted to divisional director.

● **Studio Magnetics** have recently appointed Martin Knight as sales manager responsible for UK sales of all PA and recording products.

● Bob Ofenstein has joined **tc electronic** of Denmark as their technical director. He will sell and support the entire tc range of products.

and multitrack recorder. Syco, Kimberley Road, London NW6 7SF. Tel: 071-625 6070.

● **Celestion International** have entered into an agreement with Oslo-based CMS Lydsystemer A/S who will now distribute Celestion pro-audio products throughout Norway. CMS Lydsystemer A/S, Sandakerveien 111, 0483 Oslo 4, Norway. Tel: (2) 710645. Fax: (2) 713984.

Contracts

● Likely to be the first installation of its kind, Music Inn Studio in Tokyo have installed a rackmounted **Soundcraft** *200 Delta* console inside the frame of their SSL *4000E* console! Needing extra channels for their 48-track Sony tape machine, the studio chose an 8-channel *200 Delta*, fitted with 12 Dual Line modules, to give them access to an additional 24 inputs. The sale was completed by Soundcraft Japanese distributor SCJ/AKG Ltd.

● London's Pelican Studio has completed the first foreign language dub of a feature film using DAR's *WordFit* ADR and *SoundStation II*. The Danish to English dub of the 90 minute drama *Dance of the Polar Bears* was contracted to Pelican by **Hit Communications**, one of the UK's major international TV and cinema distribution companies.

● Revolver Recording studio, Wolverhampton, UK, have taken delivery of an **Amek** *Mozart LN* desk, redesigned by Rupert Neve. The desk is a 56-channel, automated console.

● Angel recording studio's, London, UK, new Studio One control room has been completely redesigned by Tom Hidley and features a Neve *VR* console, **KEF** *KM1* monitoring and a separate machine room wired for all tape formats.

● Air Mobile in Birmingham, Alabama, recently took delivery of a **Studer** *Dyaxis 2+2* hard disk recording and editing system. The company's services include audio for video, post-production, video sweetening and editing. A *Dyaxis* has also been sold to WBEN-AM and WMJQ-FM in Buffalo, New York.

● Magic Box Incorporated, the new Turkish satellite broadcaster, has invested close to £2 million in Sony broadcast technology. Included in the hardware package supplied by **Sony Broadcast & Communications** Geneva office is a large OB vehicle, featuring six *BVP-70P* cameras. A range of the company's P-Plan post-production suites, including the *DME-9000* digital effects device.

● **QSC Audio Products** have installed 43 model *1200* and *1400* power amplifiers in the Harold Washington Memorial Library, USA. The amplifiers have been specified by Electro Acoustic System of Evanston, IL, to power the paging system in the newly constructed library.

● HHB Communications have installed two racks of five **Neve EQ** modules and five Neve dynamics modules in singer/songwriter Matt Johnson's home studio, which is based around the Akai *DR1200* system.

● SSE Marketing have recently installed two pairs of **Genelec** *1024C* studio monitors in the conference room of Soho Square, London. NHK Technical Service (NTS), a fringe organisation of the Japanese Broadcasting Company NHK, have ordered two pairs of Genelec *1034A*, three units of *S30BNF* and five units of *1019A* for two mobile buses, which are to be used for making high definition TV programmes.

● Austin City Limits, the National Public Television music programme from Texas, have started their 16th season with a newly upgraded sound system including 16 **QSC** power amplifiers. The $150,000 upgrade was designed and installed by Xeno Sound Inc, an Austin-based audio-visual company.

● **Philip Drake Electronics** have taken orders for more than 10 of their *6000* series talkback systems. Various combinations of the system include a 96×96 system for Denmark Radio and in Germany Arri TV has taken a 32×32 system while Premier TV has a 16×16 system.

● Orinoco Studios, London, have placed an order for a **Neve** *VR60* with *Flying Faders*. This is the centrepiece of a studio development scheme that also includes a digital editing facility and a new 24-track programming suite.

● New England Digital have selected **Meyer Sound's** *HD-1* audio monitor as the laboratory reference monitor to be used in the company's further development of their new DSP option mixing and digital signal processing module for the *PostPro* and *PostPro SD* digital audio workstations.

● J C Penney's corporate headquarters in Dallas, Texas, have recently installed a **Studer** *Dyaxis* digital audio hard disk recording and editing system to handle audio post for the company's video production facility. The corporation posts over 100 videos per year for J C Penney and outside clients as well.

● Goldcrest facilities are installing 48- and 64-channel *SL 5000 M* series consoles as part of the company's extensive studio modernisation. The console is configured to accommodate three operators, and features SSL *G* series automation, joystick film panning and *Moving Faders*. It is also fitted with SSL's *Instant Reset* computer system.

Letters

Service contracts

Dear Sir, I read with interest your leader in *Studio Sound* November. I reacted excitedly because you have focused attention on a matter I have campaigned for since 1986.

To get the best out of modern technologies, any owner of complex analogue or digital audio systems must ensure that the equipment is not only satisfactorily installed and commissioned but then maintained at the same levels of excellence.

To expect professional audio products to be sold, interface with other items and remain trouble-free without technical support is totally unrealistic. Any purchaser must realise that there is no 'free lunch' and the cost of trouble-free installation and maintenance is part of the initial capital cost and an essential overhead to be reflected in the charges a facility makes.

'Screwing' the dealer or manufacturer will compound the problem. Without these essential parts of the distribution chain profitably in place there are ultimately no products and no technical backup to ensure they work!

My company pays substantial maintenance charges for service contracts on our CAD and business computer systems as well as the capital equipment used in production. Why? Simply, if any of these complex devices failed we would not be able to function efficiently and ultimately our business would collapse. The cost of this service support is reflected in the selling price of our products through overhead recovery.

Isn't it now time that studios, radio stations, theatres and hire companies, major installations, etc, ran themselves as real businesses with their charges reflecting the real costs of overheads?

Technology has major operational advantages and can improve efficiency no end, however, it has an inherent installation and ongoing support cost. Ask any computer manufacturer!

Yours faithfully, Todd Wells
Managing Director, Soundtracs plc.

Phill Brown

Dear Sir, With reference to the Parkgates article by Bob Stokoe in your October edition of *Studio Sound*, I would like to point out that I was at Parkgates producing an album with The Violet Hour for CBS, and not engineering for Parkgates as the article implied.

Yours faithfully, Phill Brown.

Sennheiser UK

Sennheiser Electronic KG of Wedemark, Germany, have announced the formation of Sennheiser UK Ltd, a wholly owned subsidiary of the German parent company. The UK company is responsible for sales and service of Sennheiser professional products. Sennheiser UK have already appointed RAM Projects Ltd to be distributor responsible for Sennheiser consumer products in the UK and Ireland.

Sennheiser UK Ltd, B2 Knaves Beech Business Centre, Loudwater, High Wycombe, Bucks HP10 9QY, UK. Tel: 0628 850811. Fax: 0628 850958.

Hayden and Nippon Columbia sign agreement

Hayden Labs and Nippon Columbia have recently announced a joint venture between the two companies. Nippon Columbia now has a majority of the equity in Hayden with the balance held by Hayden managing director, Eric Barrett. Barrett commented: "There will be no major changes to either the operation of the company, or its fundamental aim to increase turnover of those products we represent in the UK. I shall continue as managing director. The financial support and strength of Nippon Columbia will ensure that we maximise the potential of our markets and achieve targeted growth in future years."

DIGITAL

ISSUE 5
WINTER 90/91 • A U D I O • T I M E S • ☎ 081 960 2144

SCREENSOUND EXCLUSIVE

ScreenSound has been the subject of a whole series of refinements by SSL. Its new capabilities as a powerful digital audio post-production centre have to be experienced to be believed. Massive off-line storage on optical disc is now complemented by 'Sound Net'. This major system enhancement allows multiple ScreenSound users to share, copy and backup work. As a result, ScreenSound can now offer 56 channel audio playback, and 48 hours of instantly retrievable audio on hard disk. Available exclusively from HHB.
From around £50,000.

NEVE MODULES

You don't have to own a Neve console to access the famous Neve sound. HHB now distributes the Prism series of Dynamics and Equaliser modules and the 33609-12 range of Limiter-Compressors. Prism EQ Module: **£368** Prism Input & Dynamics Module: **£525** 33609-12 Limiter Compressors. **From £1,830.**

AKAI DD1000 TAKES OFF

Now in stock at HHB, the Akai DD1000 is a four channel hard disk recorder providing up to an hour of 16 bit audio. Capable of digital track bouncing, the DD1000 can function as a low-cost multi-track device. It's also ideal for use in any midi-based production environment. Applications include radio programme and jingle production as well as digital editing. The DD1000 can also perform as a highly efficient on-air cart machine or be triggered to play back live backing tracks. **£7,750.**

STEREO MIC FEVER!

What is it with you people? Why have you suddenly gone made for stereo microphones? Could it be all those portable DAT recorders you're buying? Our hottest selling stereo mics are all Sony models: the ECM-979 at **£210** and the remarkable ECM-MS5 at **£695**. We now have the ECM-959 – a superb mic normally packaged with the TCD-D10 portable. Just **£122**.

KENWOOD'S AFFORDABLE CD-R

Now recordable compact disc – to the red book standard – is an affordable reality. For broadcasters and audio-post production specialists, that means instantly accessible jingles and SFX libraries. For recording studios, it allows CDs of different mixes to be supplied to record company clients on demand. HHB is delighted to offer the Kenwood CD-WO (Write Once) system, which is also CD-ROM and CD+Graphics/PQ compatible. The Kenwood CD-WO can function with any IBM PC and blank media costs are surprisingly low. Now available in the UK from HHB, the CD-WO starts at around **£15,000.**

SAMPLING ROLAND'S LATEST

The S770, with its full bandwidth and 24-voice capability, usually comes with 2Mb of RAM. It is supplied by HHB with a further 14Mb of RAM free of charge (normally an extra £1,000). The device has an integral 40Mb hard disk for massive off-line storage of samples and programmes. It's proving as popular with our broadcast customers as it is with recording studios. Comprehensive range of both digital and analogue inputs and outputs as well as SCSI interface to CD ROM and optical disk drives. From HHB, the 16Mb Roland S770 is just **£4,347.**

NEWS FROM EUROPE'S DAT CENTRE

According to Sony's Roger Lagadec, HHB has been responsible for more than 25% of all professional DAT sales worldwide. Be that as it may, the company's principal objective has always been to provide customers with a range of practical, cost-effective solutions each of which offers superb performance. As ever, our DAT range is backed by the most experienced and well-equipped service team in the industry.

SONY DTC-55ES

The first official consumer DAT recorder, the DTC-55ES is also packed with useful features. This includes the ability to automatically record sub-code data, and a long play facility. HHB can confidently recommend this SCMS-equipped model to professional users for a wide range of playback applications. **£477.**

PANASONIC SV-3700

Great value, professional recorder with balanced XLRs, AES/EBU and SPDIF digital I/Os, 4 digit error-rate display, head-hour counter, jog shuttle control and rack kit. **Just £949.**

SONY DTC-1000ES

The Industry Standard. The excellent build quality and no-compromise component specification typical to first generation Sony equipment has helped to make the DTC-1000ES a reliable performer in thousands of professional facilities worldwide. Featuring our legendary 44.1 kHz modification, the DTC-1000ES is available exclusively from HHB at just **£1,130.**

SONY TCD-D3

It had to happen. The DAT Walkman, complete with detachable power supply. SCMS-equipped/SPDIF digital in and out. Useful 'long play' mode. Limited stocks only. **£549.**

AIWA HD-S1

Super-reliable as well as tiny, this model has been upgraded by Aiwa with 1-bit oversampling A/D and D/A convertors. SCMS equipped/SPDIF digital in and out **£520.**

PORTABLES IN THE PIPELINE.

HHB's Portable DAT range is set to grow still further with the introduction of the Aiwa HHB 1 Pro in early 1990, followed later by the launch of the much-anticipated – and highly modular – Stelladat.

APOGEE A/D ARRIVES AT LAST

When it comes to optimising the performance of digital audio systems, be it for CD Mastering or for broadcast uplinks/downlinks, Apogee leads the way. We're delighted to offer the Californian company's latest A/D and D/A conversion technology on an exclusive basis in the UK. The no-compromise A/D 1000 is a stereo, 19 bit, portable processor that incorporates Apogee's remarkable 'PSD' dither technology. Digital outputs to all formats plus word clock at **£1,395**. Another portable device – the 18 bit A/D 500 – offers similar features (minus the dither and some digital outputs) at just **£795**. Apogee's D/A converter, the D/A 1000, is the finest yet devised, incorporating new dual 20 bit, 8x oversampling converters that offer a substantial improvement on previous 1 bit designs. Available from HHB at **£1,195.**

NO-COMPROMISE NEAR-FIELDS

The excitement surrounding ATC monitors is fully justifiable. The superb performance of the SCM 50A and SCM 100A models is already legendary, but the new SCM 20 is following in the footsteps of its bigger brothers. The '20' is quite simply the first near-field design to provide the transparency that professionals require for the monitoring of high quality digital sources. The briefest look at the standard of workmanship and the quality of the components confirms this is no-compromise British speaker design at its best. Listen and you'll understand the enthusiasm. No faddish colouration, just the source as it was recorded. Available exclusively from HHB. **£1,150.**

UPDATE YOUR EVENTIDE

The Eventide H3000 Series is something of a phenomenon. If you already own one of these remarkable 'Ultra Harmonizers' we can add the effects and features from any other model in the H3000 range. Card updates start from as little as **£80**. We can also retrofit the HS322 Sampler Board, which provides 24 seconds of 16 bit audio, just **£995.**

ROLAND SDE-3000A CONTINUES

Excellent value for money, the SDE-3000A offers a delay time of up to 4.5 seconds and eight effect memories. In studio production or live sound, this is one of the most popular effects devices HHB has ever sold. We've also secured fresh stocks. Available exclusively from HHB. **£725.**
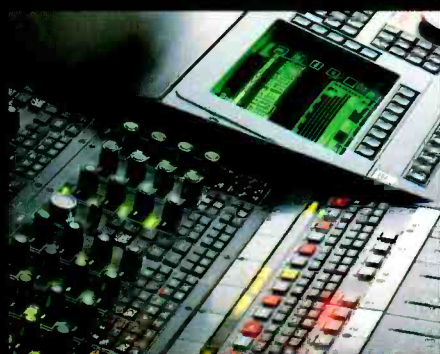
– All prices exclude VAT.

An improvement on the MTR 90 Mk2?
Who on earth could have managed that?

Many an envious glance has been cast in the direction of an MTR-90 Mk.2.

No wonder. With its pinch-rollerless transport, its user-friendly control and its faultless reliability, the MTR-90 Mk.2 has been known as the king of 2-inch multitrack.

But now, it shares its crown with a newcomer. A machine with all its advantages, plus faster wind and rewind, auto alignment on record and repro and frame-accurate mini-locator.

A machine that offers even better audio quality than the MTR-90 Mk.2, but retains all its vital portability.

A machine that even features an optional inboard Dolby* Rack, ready to accept your choice of SR cards or Otari's unique, switchable A/SR combination cards. As well as a host of detail features aimed at helping the skilled engineer to do a better, faster job.

The name of this marvel is the MTR-100A. And if you're still in the dark as to its maker, wonder no more.

Because there's only one machine that can top an Otari. Another Otari.

*Dolby and the DD are trademarks of the Dolby Laboratories Licensing Corporation

OTARI

Neve Flying Faders retrofits

Neve have developed the *Flying Faders* automation system to retrofit a wider range of consoles than the *VR* and *VRP* it was previously only available for. It can now be retrofit to the DDA *DCM232* and *AMR 24*; Harrison *PPM-1*; Focusrite *8924*; Neve *66* series, *8068, 8078, 8108, 8128* and *8232*; Sony *MXP-3000*; Trident *80C*; and Quad/Eight *Westar* and *Superstar*. All the OEM systems use standardised faders with the top plate designed to fit the console required. In most cases the servo-rack can be fitted under the console but in the Focusrite and Neve *8108, 8128* and *8232* models it fits within the console itself. In all consoles the keyboard is free standing with either mouse or trackball except in the case of the Focusrite where the keyboard is integrated into the console with a separate mouse/trackball.
Neve Electronics International Ltd, Cambridge House, Melbourn, Royston, Herts SG8 6AU, UK. Tel: 0763 260776.
USA: Rupert Neve Inc, 7 Park Lawn Drive, Berkshire Industrial Park, Bethel, CT 06801. Tel: (203) 744-6230.
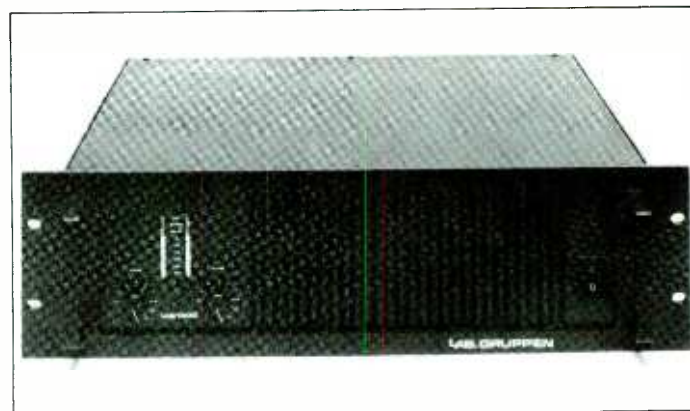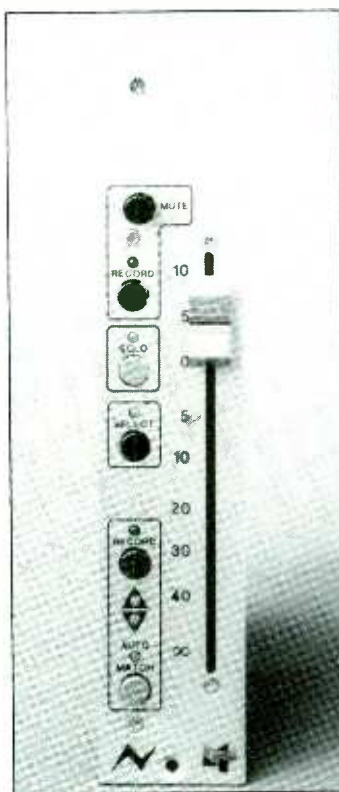
Lab Gruppen LAB 1500

Swedish company Lab Gruppen have introduced a new power amplifier known as the *LAB 1500*. It is rated at 450 W/channel into 8 Ω, 750 W/channel into 4 Ω and 1100 W/channel into 2 Ω at a THD less than 0.1%. The *LAB 1500* will also operate in a bridged mode giving 1500 W into 8 Ω and 2200 W into 4 Ω. Particular attention has been paid to the cooling system based around two proportional speed fans that also cool the power supply and channel air through compact copper heatsinks.

The manufacturers say that the *LAB 1500* can manage extremely low and complex impedances without thermal shutdown. Separate protection circuits are provided for short-circuit, thermal, DC and HF protection, surge reduction and over/under voltage protection. The unit is 3U high and 16 inches deep.
Lab Gruppen AB, Lilla Verkstadsgatan 7, S-434 42 Kungsbacka, Sweden. Tel: 0300 168 23.

Pinc Link A/V workstation

Pinc Link produce powerful work platforms and are ideal for large file servers/network applications, CAD/CAM installations, digital audio and video work stations and can be custom configured. Systems are based around the 33 MHz 80386 board with 128 kbyte write-back cache eight I/O slots, two of which are 32 bit EISA.

An important feature for studios is that the computer system is housed in the Pinc *Full Metal Jacket* 19 inch rackmount with fans and large non-switching power supply. The case is claimed to 'whisper while maintaining complete control'.

As well as providing typical workstation functions, the system is particularly interesting for A/V work as one video monitor can be used to show a variety of screens simultaneously—including the video image that is being fitted with the audio tracks.
Pinc Link, 11684 Ventura Boulevard, Box 910, Studio City, CA 91604, USA. Tel: (818) 760-4539.

Otari Premiere

The Otari Corporation have announced a new console, the *Premiere*, specifically designed for the needs of feature film and TV post-production. The console is available with no maximum number of modules and can be configured to accommodate from one to three engineers. Each module features dual line inputs, 4-band parametric EQ, sweepable high- and lowpass filters, LED input level meters with peak indication, 4-channel panning, eight aux sends with independent level and mute controls. Options are available for mono or stereo 7-band graphic equalisers and 4-pole high/lowpass filter sets.

The *Premiere* has an integrated machine control system that can work with sprocket and tape-based machines as well as a Virtual Monitor/Listen Computer System, which allows the engineers on a multiple operator console to have monitoring access and machine control of any number of tape record tracks. More than one engineer can access the same track simultaneously.

Automation is available in the form of the *Diskmix 3/Film* moving fader system that will provide switch automation on each input module in addition to fader and mute information. The system also provides data entry/read out in feet/frames, scene preset, scene store, scene recall and conforming functions.
UK: Otari (UK) Ltd, Waterside Drive, Langley, Berks SL3 6EP. Tel: 0753 580777.
USA: Otari Corporation, 378 Vintage Park Drive, Foster City, CA 94404, USA. Tel: (415) 341-5900.

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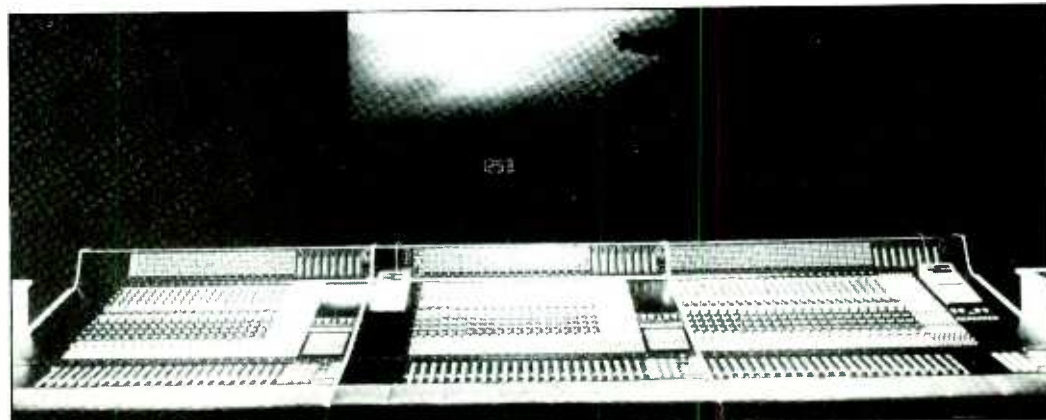
Denon CD cart player

Denon have introduced a new CD player system using the CD cartridge arrangement. In this system the standard CD is placed in an enclosed plastic casing so the user does not have to touch the disc itself and uses it more like an NAB cart. The *DN-970FA* offers operational alternatives to the previous model, the *DN-950FA*, which is still available. The *DN-970FA* has a sloping front panel with setup, servicing and repair access by lowering the panel.

Features available include digital output, ±10% varispeed, serial remote control, external sync capability and versatile search and cue functions by track and index.
UK: Hayden Pro Audio, Chiltern Hill, Chalfont St Peter, Bucks SL9 9UG. Tel: 0753 888447.
USA: Denon Digital Industries Inc, 1380 Monticello Road, Madison, GA 30650. Tel: (404) 342-0637.

Antex SX-8 digital board

Antex Electronics have introduced a stereo audio board for PCs within their series 2 product range. The *SX-8* is a lower cost version of earlier series 2 products allowing recording to a hard disk being a full size card for IBM *AT/286/386*, model *25/30* and compatibles. It offers stereo operation with the ability to simultaneously record and playback two separate signals with 16 bit resolution. Dynamic range is quoted as 80 dB with an S/N of 75 dB. Sampling rates, level, bit resolution, anti-alias filtering and data file are all software controlled. The system supports 16 bit PCM as well as ADPCM for 4:1 data compression. Multiboard driver support is available for operating two or more *SX-8s* for multichannel recording and playback.
Antex Electronics Corporation, 16100 South Figueroa Street, Gardena, CA 90248, USA. Tel: (213) 532-3092.

Spatial Sound SSP-100 sound processor

Spatial Sound have launched the *SSP-100* spatial sound processor for programmable three-dimensional panning effects. It has four audio inputs and eight audio outputs, eight parallel inputs for daisy-chaining *SSP-100* units, plus MIDI, trigger and CV inputs/outputs and FSK (Frequency Shift Keying) Data input and output.

Panning can be accomplished manually via two joysticks or as 32 sequences with up to 99 steps each.

The *SSP-100* features six main operating modes:

Stereo Panning—of four independent signals

Dual Source—allows the simultaneous simulation of two independent sources in stereo or four loudspeakers

Reverb Processing—where an external reverberation unit is used to add proportional reverb to one moving source in stereo or four loudspeakers

Gerzon Processing—uses a special phase technique for realistic spatialisation. One moving sound source can be handled in a two- or three-dimensional array

Three-Dimensional—handles one moving sound source for up to eight loudspeakers in a two- or three-dimensional array for enhanced spatial resolution

Volume Control—bypasses the spatial effect and turns the *SSP-100* into a programmable 4-channel level controller/mixer.

This is only a brief summary of the processor's abilities and a variety of applications are to be found for live sound, video and film post-production, and music recording. Further flexibility is provided by the MIDI and CV ports, which allow external processors to be used with the *SSP-100*.

Spatial Sound Inc, 743 Center Boulevard, Fairfax, CA 94930, USA. Tel: (415) 457-8114. Fax: (415) 457-6250.

Milab LSR-2000

Milab have introduced a new condenser mic designed for live applications. The *LSR-2000* is described as a ruggedised studio mic for touring. Claimed max SPL of 133 dB before distortion is possible and it can be operated on a phantom power supply of 12 to 52 V. The internal circuitry is transformerless. To reduce mechanical noise the capsule is cradled in a silicone rubber mount making the mic suitable for hand held use as well as other instrumental applications.
Milab International AB, PO Box 510, Spinngatan 3, S-260 50 Billesholm, Sweden.
UK: Klark-Teknik Research Ltd, Klark Industrial Park, Walter Nash Road, Kidderminster, Worcs DY11 7HJ. Tel: 0562 741515.
USA: Klark-Teknik Electronics Inc, 200 Sea Lane, Farmingdale, NY 11735. Tel: (516) 249-3660.
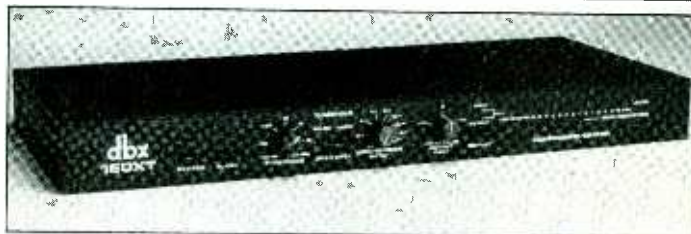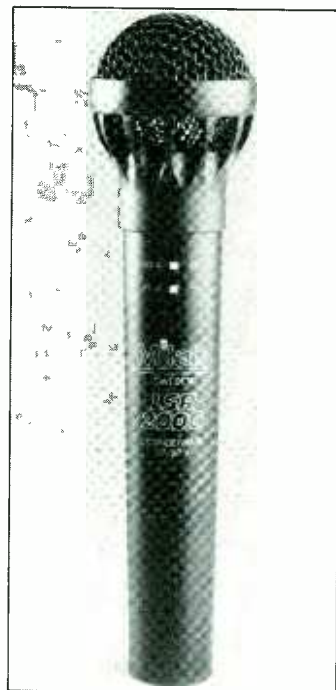
dbx compressors

The dbx *160XT* is the latest version of the model *160* compressor/limiter featuring a continuously variable compression ratio of 1:1 through ∞ :1 to −1:1. Inputs and outputs are electronically balanced with both XLR-type and ¼-inch jack connectors. Other features include selectable *Overeasy* or hard-knee compression independent of ratio; sidechain access, stereo coupling with RMS power summing, and dual meter displays for input and gain reduction.

dbx have also added a new graphic equaliser in the *Performer* series. The 1U equaliser can be switched from mono 31-band operation or dual 15-band and features switchable gain ranges of ±15/7.5 dB for the frequency bands, switchable highpass filter with turnover frequencies of 20, 60 and 120 Hz with EQ bypass switch.
dbx, AKG Acoustics Inc, 1525 Alvarado Street, San Leandro, CA 94577, USA. Tel: (415) 351-3500.
UK: AKG, Vienna Court, Catteshall Wharf, Catteshall Lane, Godalming, Surrey GU7 1JG. Tel: 04868 25702.

BBE for dbx rack

BBE Sound have introduced a new model of the *Sonic Maximizer* designed to fit the standard dbx *900* series modular mainframe. The BBE model *702* is a 2-channel module with front panel controls for process, low EQ, process and clip indicators and a hard wire bypass switch. The performance specifications are the equivalent of the *822A* top of the range model.
BBE Sound Inc, 5500 Bolsa Avenue, Suite 245, Huntingdon Beach, CA 92649, USA. Tel: (714) 558-3963.
UK: Stirling Audio Systems, Kimberley Road, London NW6 7SF. Tel: 071-624 6000.

PERFORMANCE YOU CAN DEPEND ON.
PRECISION YOU CAN SEE.

However large or small an installation, three factors are of overriding importance.

Reliability, accuracy and sonic performance.

All of which are an intrinsic part of every Turbosound enclosure – and every Turbosound array.

PRECISE ACOUSTIC DATA IN DETAIL.

Our range of compact enclosures is designed for maximum flexibility and ease of installation in any type of venue.

So for every enclosure there's a detailed Engineering Information sheet which covers all the angles.

Literally.

Each includes a set of spherical co-ordinate isobars, derived from rigorous independent tests, conducted by Mark Engebretson of Summit Audio Laboratories in California.

Guaranteeing the accurate information on directivity prediction that's essential in specifying a system.

And confirming too, exactly why our dispersion theory is a major step ahead of current thinking.
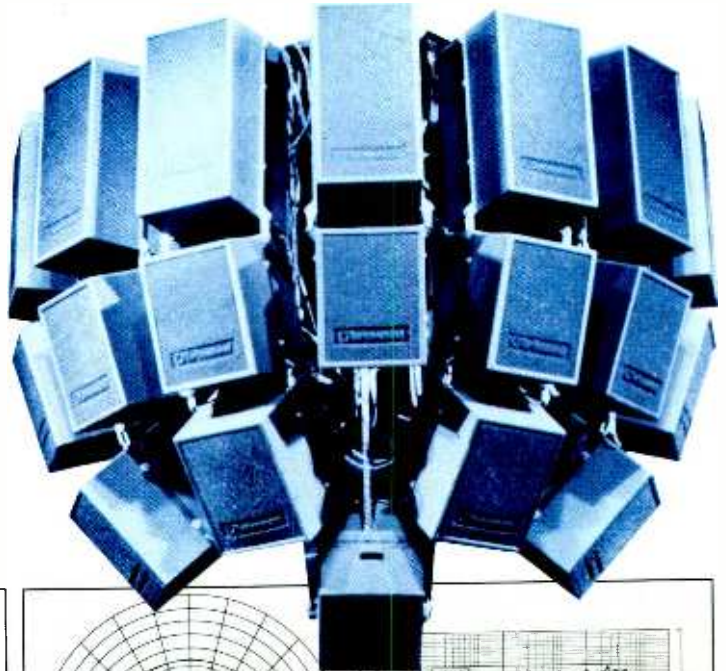
Grand Old Opry ®, Nashville
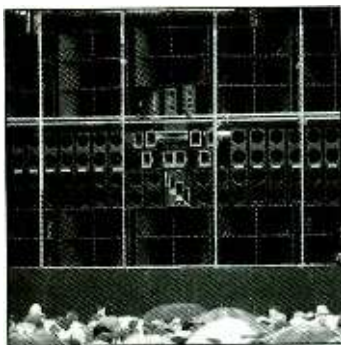
Point source array – 1979
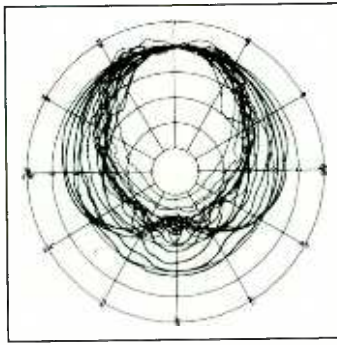
Composite Polar Set
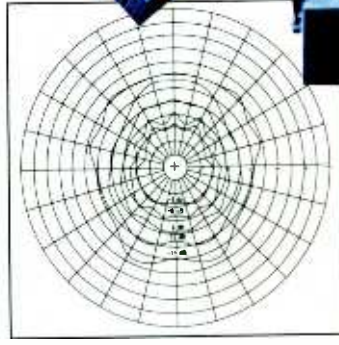
Spherical Isobar
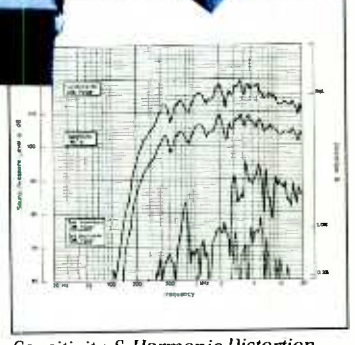
Sensitivity & Harmonic Distortion

PROPORTIONAL DIRECTIVITY™:
A HIGHER-PERFORMANCE ARRAY.

Contrary to other claims, Turbosound developed the world's first curved spherical array, using ultra high Q enclosures, back in the late 1970s.

And ever since, we've been questioning the fundamental wisdom that constant directivity is the key to perfect dispersion.

Because in exploring its benefits, we also exposed the flaw that what was good with a single unit didn't hold true for arrayed multiple enclosures.

The results of our thinking are depicted in the directivity sections of our Engineering Information sheets.

You'll notice the pattern becomes progressivley narrower with rising frequency.

We've called it Proportional Directivity.

Unlike conventional theory, we knew that narrow dispersion was only part of the answer. But Proportional Directivity provided the ideal route to a perfect array.

Our patented TurboMid™ device allows the directivity factor (Q) to increase smoothly through the frequency range.

Trapezoidal Array Problem
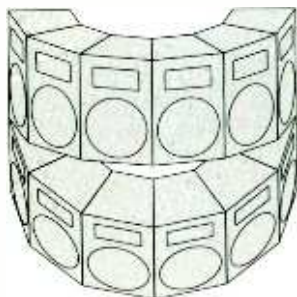
This minimises the problem of multi-source interference (combing) and creates a virtually seamless transition between adjacent enclosures – over a substantial range of angular increments.

And since both horizontal and vertical dispersion accuracy are a major factor, we've evolved a range of compact, rectangular section enclosures rather than enter the current vogue for trapezoidal designs which fixes only one position. Rectangular section enclosures give considerable freedom in the design of multi-tier arrays and will produce an even spectrum response in the most complex situations.

As can clearly be seen from the illustration, a multi-tier array made up from trapezoidal-style enclosures can produce severe dispersion anomalies.

The technology is complemented by a unique, flexible and complete flying system, designed to allow the creation of 3-D 'true point source' spherical arrays – fully safety-certified by a UK Government authorised testing station.

To get the full picture, call your authorised dealer for a Engineering Information Pack and a demonstration.

◯ Turbosound ✚
ONCE HEARD NEVER FORGOTTEN

TURBOSOUND LIMITED, STAR ROAD, PARTRIDGE GREEN, WEST SUSSEX RH13 5EZ TEL (0403) 711447
TELEX 878723 TURBO G FAX (0403) 710155 IMC TURBO-UK

Meyer VX1 stereo programme equaliser

Designed principally for frequency response shaping of stereo programme material the *VX1* can be considered as a virtual crossover with each channel divided into three bands. Each band has its own level control (±12 dB) with the crossover points being variable from 60 to 1000 Hz and 1 kHz to 16 kHz in 31 detented steps.

Other controls include master output gain (also detented). EQ bypass switch (which does not affect gain settings) and mono/stereo switch for independent 2-channel use.

The bandpass response and 6 dB/octave curves at the crossover points allow a wide variety of response curves to be created and first order minimum phase networks keep the overall sound very natural.

As well as finding applications in mastering and shaping of stereo programme material, the *VX1* is also useful for simulating curves, eg cinema or PA house curves, when doing mixdowns or transfers.

Other features include the Meyer ISO balanced inputs together with balanced outputs. RCA connectors for −10 dBV signals are also provided. **Meyer Sound Laboratories Inc, 2832 San Pablo Avenue, Berkeley, CA 94702, USA. Tel: (415) 486-1166. Fax: (415) 486-8356.** UK: Autograph Sales, 102 Grafton Road, London NW5 4BA. Tel: 071-485 3749.

Casio DAT

Casio have launched a new portable DAT machine based on the earlier *DA-1* and *DA-2* machines. The *DA-7* has both analogue I/O at 48 kHz and digital I/O at 44.1 kHz with the digital I/Os being SPDIF standard. The unit is equipped with mic amps. Casio have implemented the Serial Copy Management System (SCMS) so the unit's application in the professional environment is rather restricted. The unit comes complete with mains power supply, carrying case and rechargeable Ni-Cad battery pack.
UK: Casio Electronics Co Ltd, Unit 6, 1000 North Circular Road, London NW2 7JD. Tel: 081-450 9131. USA: Casio Inc, 15 Gardener Road, Fairfield, NJ 07006. Tel: (201) 575-7400.

In brief

● The TAC *Bullet* has been upgraded with a new mono input module fitted to all consoles now as standard. The *B1018* has six discrete aux sends arranged as four mono sends from two dual concentric pots, plus a stereo send.
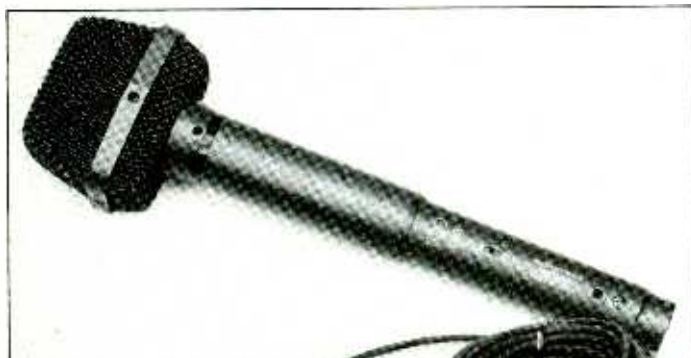
Audio-Technica microphones

Audio-Technica have introduced five new mics of widely differing types. The *ATM35* is a small electret condenser for use with musical instruments. It is equipped with the *AT8418* mic clamp with a flexible gooseneck adapter enabling the mic to be attached to musical instruments such as the rims of drums and brass instruments, etc. Manufacturer's specification quotes a frequency response of 30 Hz to 20 kHz, a max SPL of 145 dB and a phantom power requirement of 11 to 52 VDC.

The *AT825* is a stereo electret condenser suitable for handheld use. The capsules are positioned at 110° to each other in an XY arrangement. The handle contains a low-cut filter of 6 dB/octave from 150 Hz. Powering is by battery or phantom power with a figure of 300 hours continuous use quoted for battery life. Specifications include a max SPL of 126 dB, 30 Hz to 20 kHz on-axis frequency response and S/N ratio of greater than 67 dB A-weighted.

The *AT804* is a robust omnidirectional dynamic mic designed for speech applications such as ENG and interviewing. Frequency response is given as 70 Hz to 10 kHz at a weight of 220 grammes.

The *ATM41HE* and the *ATM61HE* mics are both dynamics with Neodymium magnets and a hypercardioid polar pattern. Both have a 'floating diaphragm' and 'double shock mount' for reduced handling noise, heavy duty mesh guards and a non-reflective black finish. Specifications are similar but the *ATM61HE* has a further 5 dB improvement in HF response over the *ATM41HE* giving an overall specification of 50 Hz to 18 kHz.

UK: Audio Technica Ltd, Technica House, 11 Lockwood Close, Leeds LS11 5UU. Tel: 0532 771441.
USA: Audio-Technica US Inc, 1221 Commerce Drive, Stow, OH 44224. Tel: (216) 686-2600.

AT825 and ATM35

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you've got the effect you've been looking for. As soon as you try any Ultra-Harmonizer, you'll notice that it doesn't sound like any other effects processor. The effects are inspiring, different, creative.

Or try the H3000S. The sounds all feature the same sparkling audio quality as the H3000SE. 48 of them are taken up by the remarkable Steve Vai Preset Collection, perhaps the best starting point any musician could imagine.

Many broadcasters choose the H3000B. It provides an impressive palette of ready-to-use treatments and sound effects to dramatically enhance any on-air presentation.

While we're on the subject of broadcast applications, take a look at the BD-980 stereo profanity delay – a unit that takes the stress out of running phone-ins and live radio shows.

HHB can quickly transform your H3000, giving it the features of any other model in the range.

We could go on to talk about the H3000 series' versatile MIDI implementation, the informative LCD display and the ease of programme editing – but that might be rubbing salt into other people's wounds. So call HHB for a demonstration, and you'll understand why there are a few tears being shed in the East.

ART Power Plant overdrive preamp

Incorporating the saturation curves of 12AX7 into 6L6 tubes ART's *Power Plant* overdrive preamp has clean and distortion channels with 3-band EQ in each. Outputs are provided with independent EQ curves for connection to guitar amps or rigs and balanced lines for mixing consoles.

The distortion circuitry incorporates an input drive control, a pre-emphasis control (labelled 'Harmonic Superdrive') and a master volume.

It's heartening to find scales of 1-13 on the first two of the aforementioned rotary controls, every little bit helps after all.

If the *Power Plant* shares any common distortion characteristics with the ART *SGE* multi-effects processor then it surely deserves a listen.

Continuing with the attention that the company seems to be paying the guitarist, ART could be answering the very real need for a cheap but proficient MIDI foot controller. The *X-II* allows foot operated selection of 128 patches via MIDI and comes with an LED display, steel reinforced switches and an all-steel chassis.
ART, 215 Tremont Street, Rochester, NY 14608, USA. Tel: (716) 436-2720.
UK: Harman UK, Mill Street, Slough, Berks SL2 5DD. Tel: 0753 76911.

Focus on MIDI/CV

The palette of extraordinary contemporary sounds afforded by today's digital synths has not tainted the affection held for some of the more traditional and memorable pre-MIDI beasts. While some companies are enjoying the popularity they are experiencing with repackaged modular MIDI versions of the old titans, interfacing the CV-based systems with the 5-pin DIN has never been easier. Users will also be surprised at the level of sophistication on offer. Time to dig out the old Oberheims.
● PAIA's MIDI/CV Interface (MCVI) works in conjunction with stackable expansion MUX cards to give MIDI control of CV synths, lighting and other voltage-controlled devices. For applications where only one CV input and output is needed, only the MCVI is required. More complicated arrangements require MUX multiplexer/demultiplexer cards, which sequentially sample a number of inputs and refresh a number of outputs while reading or generating

MIDI.

Functions controlled include pitch, velocity, pitch wheel, modulation wheel, pressure, foot pedal, volume and data slider. Modes supported by the system include mono input and output, eight synth MIDI Mode 4, controller/switch input and output, note to switch conversion and Poly Mode to eight CV outputs, with two different types of output assignment.

Systems are available in kits or pre-assembled and tested.
PAIA Electronics Inc, 3200 Teakwood Lane, Edmond, OK 73013, USA. Tel: (405) 340-6300.
● Digisound's *80-25* is capable of controlling eight CV synths from MIDI and can operate in a 4-voice mono multitimbral mode. Features include an 8-note unison mode, aftertouch, legato gate retriggering, intelligent note allocation, internal or external pitch bend and Mono and Poly velocity can be assigned to any outputs.

Modified units are available for driving Moog, ARP and EMS synths.

Digisound, 16 Lauriston Road, London SW19 4TQ, UK. Tel: 081-946 0467.
● Groove Electronics offers two MIDI/CV devices. The *M4CV* operates on four independent MIDI channels running four CV and filter outputs receiving velocity, pitch bend, modulation wheel and aftertouch. Twelve trigger outputs are included.

The *M2CV* runs 1 V/octave mono synths with assignable receive channel and trigger type. It receives pitchbend, velocity and aftertouch. Options include an ARP arpeggiator trigger driven from MIDI clock.
Groove Electronics, Unit 22, Barnack Industrial Centre, Kingsway Trading Estate, Wilton, Wilts SP2 0AW, UK. Tel: 0722 743712.
● Companies specialising in independent internal MIDI retrofits to some of the more established analogue machines include Groove Electronics and Kenton Electronics, 137-165 Hook Road, Surbiton, Surrey KT6 5AR, UK. Tel: 081-974 2475.

JL Cooper CS1 control station

Aimed primarily at the Mac/Digidesign *SoundTools* user, JL Cooper's *CS1* control station is a dedicated control interface with a jogwheel and 10 buttons that can interact with sequencing software. Templates stored in the *CS1*'s memory support Opcode *Vision* and Mark of the Unicorn *Performer* and a

MIDI version is available to work on Atari. Function keys and a footswitch can be programmed to execute any task and the unit can be regarded as the *SoundTools* user's equivalent to the company's *Fadermaster* MIDI command controller mouse substitute. Any reservations held about the mouse-driven inaccessibility of the

Digidesign hard disk system would now seem unfounded.
JL Cooper Electronics, 13478 Beach Avenue, Marina del Rey, CA 90292. Tel: (213) 306-4131.
UK: Sound Technology plc, 6 Letchworth Business Centre, Avenue One, Letchworth, Herts SG6 2HR. Tel: 0462 480000.

Software

● Opcode's *Studio Vision* combines the company's *Vision* sequencer with Digidesign's *SoundTools* on one Mac II or SE/30.

The system offers SMPTE sync and fully automated mixing of the two dynamically allocated playback channels with cut and paste editing features executed in one domain applicable to the other. *Studio Vision* can be upgraded too from *Vision*, will be able to absorb any developments in *SoundTools* hard disk playback abilities and integrates with Opcode's *Galaxy* librarian.

Opcode has also released Version 3.0 of its *Cue* film music management system for the *Mac*. Features include cue sheet, tempo calculation, custom score paper layout, production summaries for creating master cue lists, MIDI triggered sound effects, clicktrack, tempo tap and MIDI file playback.
Opcode Systems Inc, 3641 Haven Drive, Suite A, Menlo Park, CA 94025-1010, USA. Tel: (415) 369-8131.

Samples

● New for the Roland *S770* is TSC's library available on optical disc or 45 Mbyte cart. Included are collections of sound effects, drum samples and orchestral samples donated by an ex-*Synclavier* owner and now *S770* convertee.

TSC are continuing their policy of not charging clients for the use of their vast sound library.
The Synthesizer Company, 9 Hatton Street, London NW8 8PR, UK. Tel: 071-258 3454.
● Greytsounds have released disk sample libraries for the Akai-Linn *MPC60*, Akai *S1000* and *Emax II*, compiled on Mac *Alchemy* and *Sound Designer*. Drums covered for the *MPC60* include *SDX*, Linn, *TR808*, *DMX*, Alesis *HR16* and *16B*, Roland *R8*, EMU *SP12* and *1200*, the almighty *TR606* and *303*.

The *Emax* and *S1000* are served by similar collections of drum and orchestral samples along with the *Prophet 10* and *VS*, PPG *Wave 1* and *2*, ARP *2600*, *Taurus* pedals and *Mellotron*.
Greytsounds, CA, USA. Tel: (818) 993-4546.

Studio Sound's Music News is compiled by Zenon Schoepe
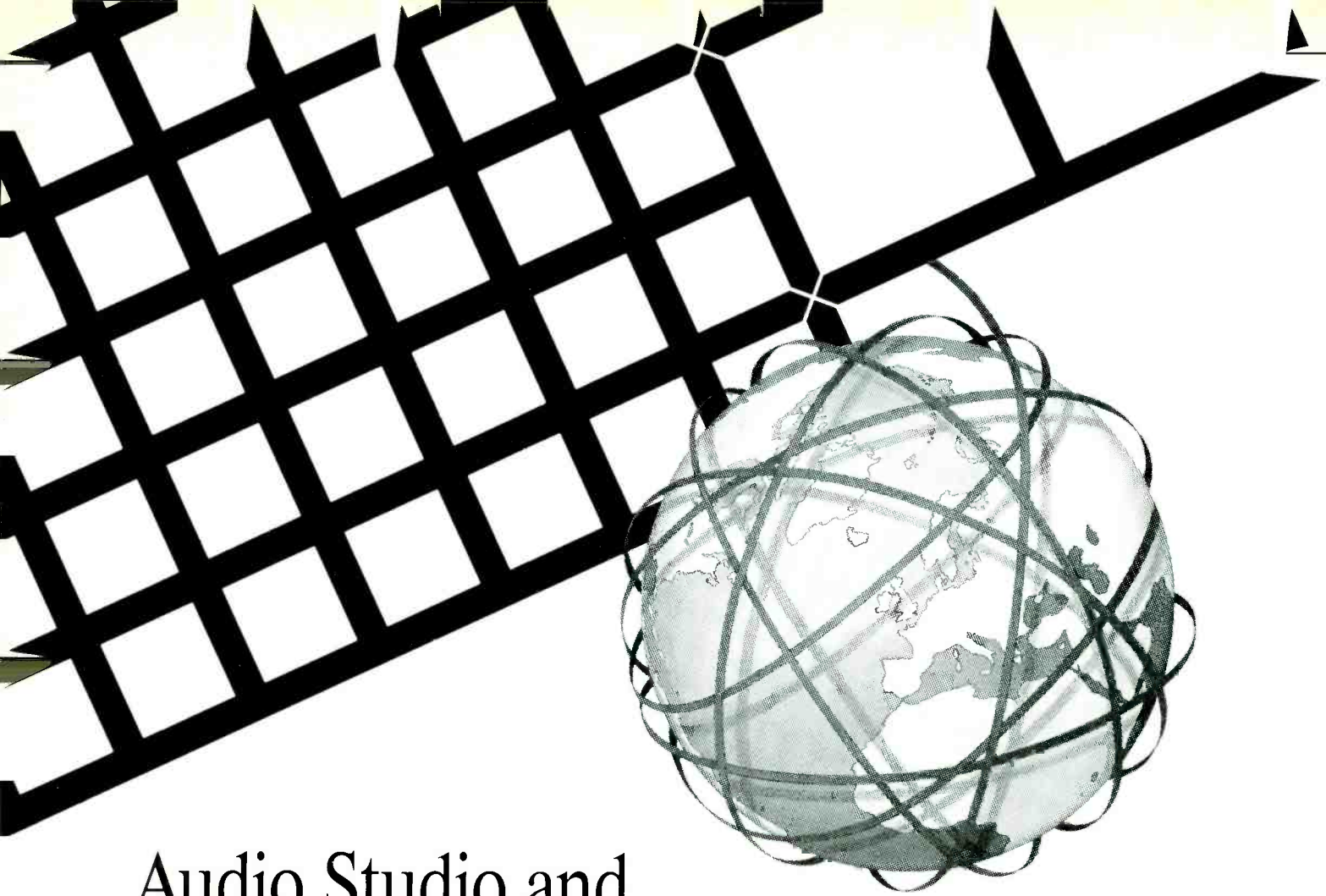
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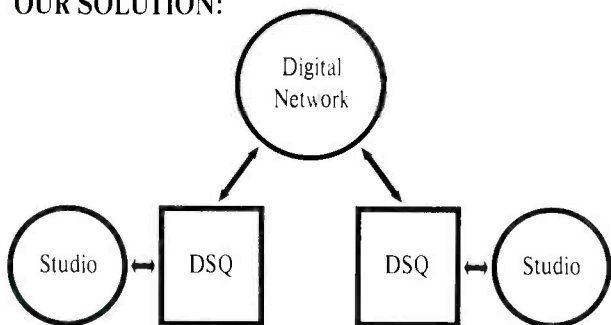
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Knebworth 1990: stage courtesy of Upfront Productions

Upfront goes Outback

The production arm of London-based staging and production outfit Upfront is reborn this month as Outback following a buy-out by Upfront co-owner Roger Chopin, along with Martin Crick, Roger Heighton, Simon Lowry and Shane McCarthy. Outback will continue their predecessor's production services while Upfront EGS Ltd maintains the existing hire stock, the bulk of which will immediately move to Upfront Inc's US production operation, based in Austin, Texas, and headed by Spike Falana. Chopin's partner Roger McCue has sold most of his stake in Upfront to concentrate on his new artiste management project Mix Centre.

Clair's Euro HQ move

Clair Brothers were set to undertake a major reorganisation of their operations in December—relocating their European headquarters from London to Basle to form a new partnership with Swiss rentals company Audio Rent.

London boss Stan Horine says the move has two aims: to cut overheads and to facilitate Clair's desire to exploit their live PA expertise in tackling the UK and European installations market. Installations are increasingly featured in the company's US work and the London office recently finished their first contract at the Lewisham Theatre in South London.

Clair's US R&D team, meanwhile, have rolled out their latest PA processor/limiter package (to replace the hybrid system currently used in Clair PA racks) and its automated sound reinforcement console project continues apace. In Europe, current major tours include Janet Jackson and The Waterboys. The new telephone number in Basle is +41 61 701 5515.

Samuelson's merger

Samuelson Group plc have brought together their three lighting and sound companies under the guise of Samuelson Lighting & Sound, in a new purpose-built centre at Greenford, West London. The restructured division—with former Kodak Copiers sales chief Roger Dix at the helm—incorporates Vari-Lite Europe Ltd (headed by Brian Croft); Samuelson Concert Productions Ltd; and the Lighting and Sound & Vision operations of Theatre Projects Services Ltd. Alan Thomson heads up both Samuelson's and TP's lighting operations. The new base has a large training and rehearsal studio and a demonstration showroom; Dix says the new division "is headed for exciting growth in the industry".

SCP has also formed an alliance with leading US trade show and theatrical lighting company Vanco.

News round-up

● JBL have launched a complete pre-wired, packaged sound reinforcement system, a simple off-the-shelf concept designed, say Harman UK, for the smaller disco and PA market. The 2 kW JBL *Control Package* includes two *Control 12SR* 2-way enclosures plus stands and one *SB15* sub-bass enclosure, and is designed to be drivable by one amplifier. Larger applications are catered for by two *Sound Power Systems*, both complemented by a dedicated rack, crossover and cabling, and JBL-recommended amplifiers.

● DDA have notched up a successful run in the past few months with sales of their new top-of-the-line *Arena* PA console to leading Canadian PA rental firm Bruit Bleu (for the latter's New York *Aida* shows) and a contract through Audix for seven *D* series and eight *S* series consoles in Birmingham's new International Convention Centre. Its centrepiece is the Symphony Hall, new home of the Birmingham Symphony Orchestra.

● Nexo have a lengthy list of new activities in Europe—including recent installation and hire systems through Stockholm-based distributor LEAB and contracts through CAMCO of Kreuztal, Germany.

● Electro-Voice note that David Bowie's world tour has featured *N/D757A* (vocal) and *N/D408A* (drum) microphones—at the behest of FOH engineer Buford Jones. Jones, named as *Mix* magazine's 1989 Sound Reinforcement Engineer of the Year (why not 'producer'—is there an element of the poor relation here?) notes in typically dry Texan off-hand manner: "I've used *N/DYM* microphones for some time...the *N/D408A* is particularly great for drums."

● MHA Audio's new Hill *M6* PA system, first used by George Michael and Cliff Richard in the past year, has had its US debut on the B-52s/Ziggy Marley 'arena and sheds' tour with supplementary control hardware provided by The Toy Specialists of New York.

● Outline of Italy have two interesting controller-assisted speaker systems: the *TopFly* 2-box 3-way system and the *FlyOne* 3-way full-range system. Both, say Outline, have been developed with the latest CAD technology to combine high sensitivity and power handling in medium-sized trapezoidal enclosures. *TopFly* employs custom designed ferro-fluid cooled drivers, hybrid bass loading and a *Quadraplug* loaded MF horn, which achieves 113 dB/1 W/1 m. It's designed for portability and easy assembly in arrays and, claim Outline, eliminates comb filtering problems.

● Canegreen have a packed schedule embracing the Grateful Dead's Wembley shows (using an unprecedented total of 88 Meyer *MSL-3s* plus subs) and UK/European tours by Van Morrison and Suzanne Vega (the latter also venturing into former Eastern Bloc territory) as well as Aztec Camera's US dates.

● SSE are touring with The Pogues, World Party, Del Amitri and a couple of (legal) House Raves with their *MT-4* and E-V speakers and TAC *SR9000*, *Scorpion 40/12* and Soundtracs *SPA* consoles. The company's Irish subsidiary, meanwhile, has Paul Young and a charity night hosted by former Beirut hostage Brian Keenan at The Point in Dublin.

● Sound Hire, Richard Leinard's Meyer rentals company, maintains its impeccable cultural connections with productions for Pavarotti, the Bolshoi Opera and Shirley MacLaine. Sound Hire has also invested in three RAMSA consoles (two FOH, one monitor), which Leinard says represent "a quantum leap in quality and reliability", and a brace of Meyer *MSL-10s*. The latter, 'big brothers' to the standard *MSL-3* enclosure, weigh 440 kg apiece and, Leinard states, "are very powerful, very high Q, ideal for large events and the next logical step in our upgrading".

● Gazza: We have received a press release announcing Lindisfarne's 25-date UK Christmas tour, the highlight of which was a reference to the rap by 'Gazza' (Spurs and England World Cup soccer star Paul Gascoigne) on the band's reworked classic *Fog On The Tyne (Gazza Rap)*. The release, however, coyly omits to mention whether the tearful folk hero himself might tread the boards...

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

GHL TRUCK

GHL run a remote truck in the American Mid-West. Designed, built and run by Gary Hedden, the truck reflects his specific approach to mobile recording which he explains to Keith Spencer-Allen

There are many reasons for choosing Nashville as a location for a recording operation. The reputation as the home of country music is, of course, one attraction. For a remote truck there is also the fact that it is very central with a major part of the USA being within a day's drive. And finally it is a very pleasant area with a far better balance between the quality of life and the recording business than found in most other major music cities. It is for a combination of these reasons that GHL are based here.

The GHL truck is a 40 ft Great Dane air ride freight trailer made specifically for this purpose. Its base is one of the many Nashville business premises that appears to be a house but actually contains offices, a workshop and a small studio for overdubbing. The truck spends quite a lot of time working at base and a fixed umbilical between the truck and studio makes linking up very easy.

The operation started in Columbus, Ohio nearly 7 years ago. Gary Hedden designed and, for the most part, built the truck over a period of 14 months—8,000 man/hours. He himself has a long history of involvement in all areas of pro-audio in the Mid-West since the age of 14 including being a partner in a Chicago-based studio with two 24-track rooms and disc mastering as well as several Grammy and Clio nominations to his credit. Over 20 years' experience was therefore put into the truck with the design being very much for Hedden's own way of working. Having begun work out of Columbus, the operation was moved to Nashville 4 years ago.

Designing a mobile has to be the art of obtaining the most workable compromise. Building within the restrictions of a truck means that there are not the possibilities available to the designer of a bricks and mortar studio. Or to quote Hedden: "Basically you are trying to create a control room in a mobile

tunnel—which is very difficult."

Early in the design it was realised that the truck would probably be doing a lot of remote recording as well as location work. It was therefore necessary to create a type of control room environment that could function as both a mobile studio operation where creative work could proceed for days, and on the other hand be used for the efficient one-man operation, live recording situation. Hedden tackled this with a series of crucial design decisions: the interior of the truck is divided into three quite separate areas; the acoustic treatment has been carried out to preserve as much of the 102 inches maximum legal width of the truck as possible; and by placing the Harrison console across the width of the truck.

The control room is 22 ft long and faces towards the rear of the rig and takes up the centre section of the trailer. The console is a 36-input Harrison MR4 with an extended patchbay to fit the available space. For live work they have a 'sidecar' PR07 with an additional 12 inputs that sits at a right angle to the console giving the "48 inputs that appears to be the magic minimum these days". The MR4 has been a popular choice and has proven reliable on the road. The locality of the manufacturer is also useful although this was not the case when the choice was made. The MR4 is fitted with ARMS fader automation, which is a tape-based system, and although Hedden concedes that a disk-based system would be quicker, the ARMS has been very reliable and with the on-board three-machine lock-up system spare tracks for data are no problem.

The main monitors are modified Fostex LS-3s, which are 3-way systems with a 15 inch LF unit and are powered by Hafler ProSeries amplifiers. The mid range horn has been removed and mounted on top of the cabinet with the high dispersion tweeter (which is external to the LS-3 cabinet anyway) within a Sonex lined soffit. This monitor layout was developed by Hedden and is a design that he has used in several previous rooms. He finds that the width of the stereo image tends to be controlled by the dimensions of the soffit itself.

Hedden: "I was trying for the widest stereo image possible within this truck. I actually had much of the equipment in my hands when we were in rough construction. We were able to play with the distances and placement until it sounded right and said that this was the way that it was going to be. The truck was then literally built around this system. We have had up to 25 people in here for a playback and it works."

Situated between the monitor speakers are a pair of video monitors for programme feed and a CCTV monitor and two racks of processing equipment. In front of the console and not visable in the photographs is a sizable couch that keeps non-operational people out of the way but still gives them a reasonable monitor sound.

The compactness of the console gives access around one side of it, the other end joining with the built-in racks and work surfaces. Immediately to the right of the operator are remote panels for the pair of Otari MTR90/II 24-tracks carried as standard. For synchronisation the truck uses a BTX Softouch system with Cypher and two Shadow IIs capable of locking-up three machines. The other tape machines carried as standard are Otari MTR12 (running ½ inch 2- and 4-track) and two MX5050B ¼ inch 2-tracks.

The table surface immediately behind the console operator becomes the producer's area with a talkback mic that can interface to the Clearcom communications system positioned in the rack below it. When the truck is functioning as a fixed unit then this space becomes a place for the keyboards.

Within easy reach of the engineer are the dynamics control racks containing an Aphex Compellor, a dbx 900 rack, an Aphex rack with six CX-1 compressor/expanders and four EQF-2 EQs, six Valley DynaMites and a couple of LA3s. When needed the 900 rack can be fitted with compressors but also contains two 902 de-essers and a 906 flanger. There has been much effort to keep the equipment in the immediate vicinity of the operator out of the audio path but still in easy reach for one-man operation and it seems to have achieved this.

Other processing equipment carried in other rack space includes two Lexicon 200s, two Lexicon PCM-42s, Eventide 910 Harmonizer, Marshall Time Modulator, Quantec QRS and Alesis MidiVerb II. The choice of equipment reflects Hedden's approach. "At the time many of these equipment choices were made, 6 or 7 years ago, most trucks only had one kind of

reverberator and spring units were quite common still. The *224* was a popular choice at that time but I elected to go for a couple of *200*s and the delays which I felt suited the remote business better. Processing added since that time is not fitted within the truck but carried when needed. The collection of gear still works well as a system to me."

Other control room mounted equipment includes Dolby *A* and *SR* for the *MTR12*, two Nakamichi *MR1-B* cassette machines, Technics *SL-XP7* CD player and a JVC *CR-6650* U-matic machine. The latter can be selected as one of the machines under the *Softouch* synchronisation system for mixing to picture or as a recorder for the dbx *700* digital processor also carried in racks above the multitracks. The *700* was used very heavily but DAT has since overtaken it. Hedden: "As a recorder it sounds really wonderful. It was certainly inconvenient to edit on having to go through analogue to get product out. We have even recorded the Space Shuttle launch on that system without distortion."

The control room has a spacious feel to it and unusually for a mobile has differing floor heights for separate areas. The decor finish is largely blue cord material with plenty of light wood. Isolation from the outside world seemed good despite the design aims of maximising internal dimensions within the trailer shell.

The Otari multitracks are at the far end of the control room with plenty of access space around them. They are mounted on special tuned vibration isolators. All the equipment on this rear wall actually backs into the space behind the control room—'The Shop'. In this space access to all the rear connectors are available so the machines don.'t have to be moved—and they never have. There are also hour meters installed for the multitracks.

For recording the machines are generally run at 30 in/s without noise reduction and with rental systems to meet any specific requests. Digital multitracks are hired in as needed but the space available allows a pair of Mitsubishi machines to be carried as well as the standard equipment complement.

The workshop doubles up as so many different types of spaces. There is a bench across the back of the area and everything is

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INCH AFTER INCH,

SESSION AFTER SESSION,

THRILL AFTER THRILL.
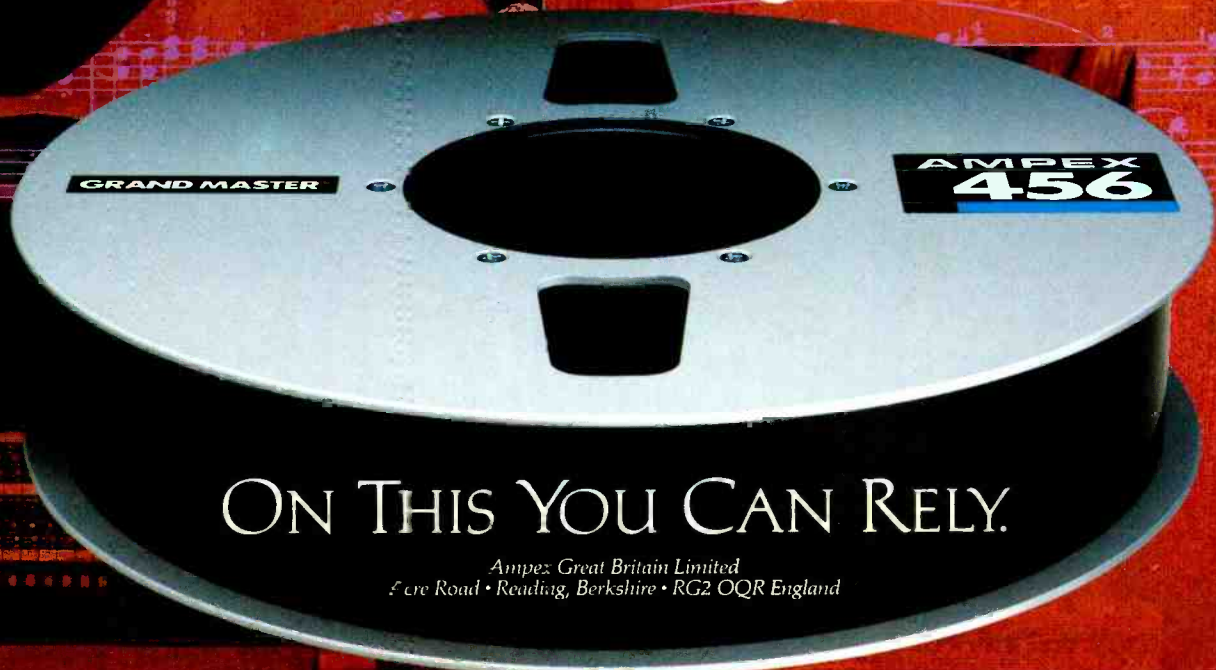
GRAND MASTER

AMPEX
456

ON THIS YOU CAN RELY.

covered in carpet including the resident fridge and microwave. Also carried is a photocopier, which Hedden considers the most popular piece of equipment that they take to a remote show.

The workshop carries a full set of spares, back-ups and test equipment on board at all times. It has been used for modifying guitars and amps, making adapters and so on. Having had several experiences of mending cables under stages in the dark Hedden decided that he would resist any suggestion that the area be made into a lounge. The space has also doubled as an announcer's booth in an emergency, a video assist area on a camera shoot as well as having an acoustic function being a 'resonant cavity' taken into consideration during the acoustic design.

On the test equipment side there is an oscilloscope that is also used for monitoring timecode and sync as needed. In conjunction with an Amber distortion meter it is also used for machine alignment. Hedden: "We put the scope output onto the tape machine operator's Clearcom headset so he can put on headphones and hear whatever he is looking at and still be in touch with the stage. Because of the monitoring and test signal input on the 90s we can do a complete alignment with distortion analysis during a soundcheck without tying up the console. There is never enough time and this is one extra way to get another slot in there."

The mobile generally carries a mic complement of about 50 and normally includes Neumann *U89* and *KM84*s, AKG *C414* and *C460*s and Sennheiser *MD 421*s. There is also a smaller number of mics from E-V, Sony, Shure, Crown and Beyer as well as a dozen DI boxes.

Outside and under the trailer is a vast amount of storage space. With a tractor/trailer rig you do not have a drive shaft and frame to deal with and so it can have other uses. Access is from either side and is the whole area under the trailer in front of the rear wheels. The space is used for carrying stage equipment so that the control room can travel ready set up without the need to clear it before working.

The rear section of the truck is accessed through the back doors and contains all the snakes, air conditioning and power supplies and conditioning systems. The 54 mic lines are arranged on two 27-pair snakes of 100 metres each on motorised reels. There are also motorised auxiliary snakes, a CCTV camera snake that also carries power and the GHL dedicated video truck interface that provides six balanced audio and three video lines.

The air conditioning is a microprocessor controlled system for maintaining constant control room temperature and additional cooling for power supplies, etc, allows remote control of fresh air mix and auto switches to a back-up system in the event of a failure. The console is also provided with a positive air supply to reduce dirt accumulation. The system has successfully maintained working conditions over a temperature variation of −20°F to 90°+ with 100% humidity in southern Florida.

The truck has a power requirement of single phase 170 to 260 V at 80 A. Power is conditioned with a Sola Ferro Resonant transformer so that it sits at a constant 120 VRMS. This system has the effect of making equipment lamps last longer as well as helping equipment running with unregulated power supplies.

All this rear utility section of the truck can be sealed against adverse climatic conditions even when operating. The exterior of the truck is 12 ft 9 inches high excluding the antennae on the roof for cellular phone and radio activated alarm system. Tractor units are hired as needed. With the truck often working back at base or on location for long periods of time it is far more economic to work in this way.

The influence of Hedden is seen in another area of the design: "I am known as something of a fanatic on grounding." The interconnection of the equipment has been carefully worked out with a large copper star ground system implemented within the Harrison MR4. The console's own grounding design was also modified to meet Hedden's requirements. "All components within the truck are connected to the star with *AWG 1* welding cable. We monitor the fault current from the star constantly and there is a warning light if it exceeds 50 mA. All the audio cables are terminated at the console end and fitted with RF bypass capacitor at the peripheral end on some 600 lines. If there is anything wrong the light comes on in the console long before you hear the hum."

Eighteen months ago GHL added a DAR *SoundStation*—the

Looking towards the front of the control room over the Harrison MR4 to the customised Fostex LS-3 monitors

Towards the rear of the control room with dual Otari multitracks

first such system in North America. At the moment most of its work is CD premastering back at the base but Hedden is experimenting with its musical applications and multitrack vocal 'comping'. The system capacity is eight tracks, 4 hours. Hedden: "For me, personally, the interest is in regaining a leading edge viewpoint and moving into what I feel is a valid new direction for the industry rather than reel-to-reel multitracks. I felt that this system had the greatest potential of those I was looking at."
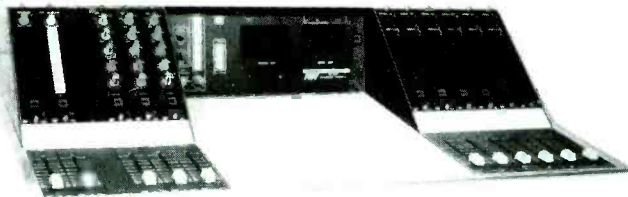
The *SoundStation* is installed in flight cases and Hedden often takes it out to work in musicians' homes to do mixing or solo recordings. The plus here is that it is portable and fits in his pick-up truck rather than having to take the mobile. And apparently it also travels well.

The mobile finds itself over a large area of the Mid-West from Illinois to Florida although Hedden describes most of their work as being in the north central region. The design of the truck has meant that the type of work is varied from live satellite uplinks through audio/video work to custom gospel records. Country music may be the largest single type of work undertaken but it is far from the only type. And unlike many mobiles GHL does a lot of fixed location work functioning as a mobile control room.

Hedden: "I tried to design this truck to be as comfortable as many studios to work in. It still is my favourite place to mix."
GHL, 2807 Azalea Place, Nashville, TN 37204, USA. Tel: (615) 269-5183.

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Tom Jung should need little introduction to lovers of well-recorded jazz and classical performance on both sides of the Atlantic. An early involvement with digital production during the late '70s began at his own facilities, Sound 80 Studios and Road 80 Mobile, based in Minneapolis, using the early 3M *Digital Mastering System*. Tom's production credits include the first digital multitrack remote recording for Billy Joel's *Songs In The Attic* album, using the 3M 32-track *DMS*; his numerous film score recordings have included such films as *Dressed to Kill, Blow Out, Star 80, Cotton Club, The Fan* and *Jagged Edge*.

Tom Jung's focus during the past 8 years, however, has been directed towards his own record label, Digital Music Products, which specialises in all-digital jazz releases.

"We set up DMP in 1982," he recalls, "to achieve two goals: first, to extend the limits of digital technology; and second, to return to the use of more fundamental and sonically pure recording techniques.

"I find that today's emphasis on multitrack production techniques, close-miking, extensive overdubbing and all of the other 'control-room trickery' can harm the music. With the setting up of DMP, I wanted to return to the days of good, basic techniques and attempt to capture valid musical performances with the least amount of intrusion from the recording hardware.

"The single most important consideration for me is the music; that's the bottom line. While it goes without saying that I need to capture a sonically accurate 'picture' of what's happening on the studio floor—and that is where nearly 30 years of hands-on experience in this business begins to pay off—I never forget that the music is of greater importance. You can have the greatest players and most proficient production team in the world, and be using the most advanced equipment but if there's no music to play then we are all stroking ourselves!"

Since the early '80s, DMP has issued nearly 40 compact discs, all of which were recorded digitally, and feature such artists as Warren Bernhardt, Flim & the BB's, Bob's Diner, Chuck Loeb, Manfredo Fest, Dial & Oatts, Bob Mintzer Big Band and Thom Rotella. The majority of DMP's earlier releases were made direct to 2-track digital—usually a modified Mitsubishi *X-80* using a custom-designed Cello analogue console with discrete analogue Class-A electronics. Built by Mark Levinson, the Cello console features four stereo mic/line inputs, two mono mic/line inputs (with around 20 dB of gain trim) and five stereo-pair line inputs, plus one stereo and two mono effects sends—and no EQ. More recently, Tom Jung has been using Yamaha *DMP7D* 8-input digital consoles.

"Be they analogue or digital based," says Jung, "each and every component in my recording systems were chosen for their consistent 'musical accuracy'; they need to be able to accommodate and faithfully reproduce complex musical wavefronts and steep transients."

Beyond digital 2-track

From his first direct-to-digital sessions, during recent years Jung has beeen exploring various digital mixing and overdubbing techniques— 'creative multitrack' as he refers to the process. "For a long time I always stayed with direct-to-digital 2-track, because I just couldn't put up with the sonic degradation caused by the multiple conversions from analogue to digital and back again during the conventional overdubbing process (using digital multitracks and an analogue console). Then, having discovered the Yamaha *DMP7D*, which allowed me to do all of my mixing and processing entirely within the digital domain, I began to seriously consider a 'modest' amount of overdubbing but only when absolutely necessary.

CONVERSATION WITH TOM JUNG

Producer Tom Jung is president of the Digital Music Products record label which specialises in direct-to-digital jazz recording for CD release. Interview by James Douglas
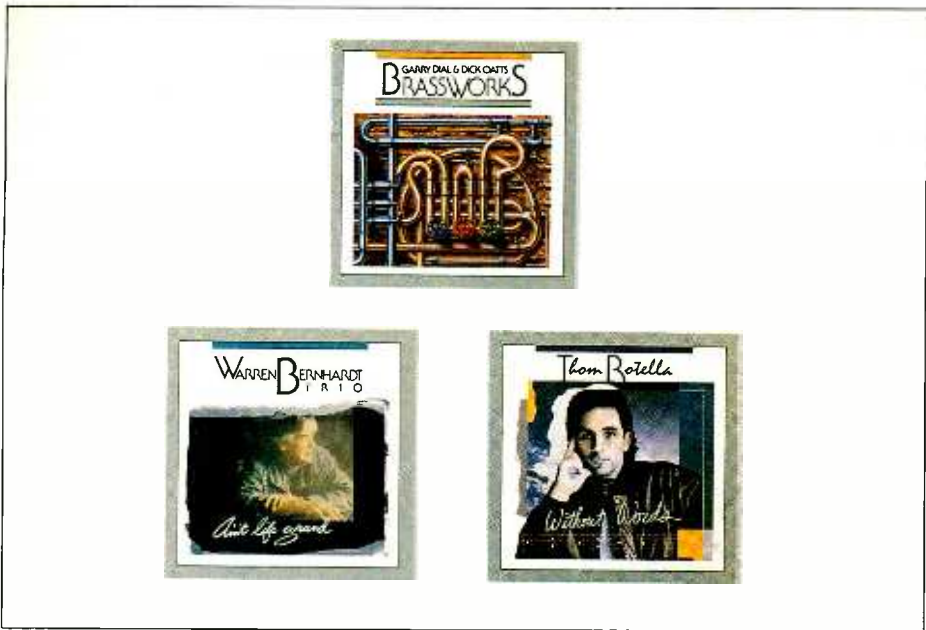
Three DMP CDs whose production details are covered in text

take a few takes to get the structure tight and everybody playing together. We'll go back and touch up but the first solos are almost always the best ones."

What factors are involved in selecting an appropriate recording environment? "It is always a mutual decision between the artists and myself. From a musician's point of view I look for a room that is easy to play in; that sounds good; and one that has a natural acoustic. They also need a good cue system. Generally, I prefer to work in a big room, because I like the length of early reflexions that can enhance and add body to the sound. In New York, I like Clinton Studio A, and in LA I've worked at Oceanway and Westlake Studio D."

Mic selection

Jung considers that he has a 'minimalist' approach to recording: "For me 'less is more'— although I don't necessarily believe in the concept of just using spaced omnis to pick up an entire ensemble, which is often the case with classical music recordings. Instead, I try to reproduce the band as it might sound in a good acoustic space, with everything recorded in stereo—all piano, drums and percussion—and with a three-dimensional character.

"I use two mics on piano—usually a pair of modified B&K model 4003 spaced omni mics, with a custom-designed high voltage power supply and discrete Class-A preamplifier."

Mic signals are run through either PSE (Professional Systems Engineering) and Studio Technologies preamps via high-performance 'audiofile' cable to Yamaha AD8X (19 bit 64× oversampling, sigma-delta) A/D converter units and hence to the DMP7D.

"Normally, I use a Speiden SF-12 stereo ribbon—which, by the way, I first heard about from Studio Sound—as a solo overhead mic on

"I made the decision primarily for logistical reasons. Sometimes it isn't possible, either because of the size of some of the ensembles I wanted to record, space considerations, or people's availability, to have all of the musicians gathered together in one place at the same time. During a recent Dial & Oatts project, for example, I chose to record the basic tracks of a jazz quartet, edit together the best solos and then lay over a string section as a live-to-two overpass.

"For those sorts of sessions, multitrack wasn't really a production tool—in the sense of providing punch-ins, etc—but more of an assembly device. I could take the various elements of a jazz quartet and string section and have the ability to overlap sections but still keep the entire production

within the digital domain. In other words, the person who purchases the compact disc is hearing only one analogue-to-digital conversion. But I didn't want to turn DMP into another 'multitrack' label!

"Jazz is a very creative process when played live. Musicians play off one another and improvise; a lot of ideas develop during a performance. So jazz lends itself to my trying to capture an overall performance. Because of a tune's complexity, however, and other factors, it is sometimes too difficult to play it through in its entirety. Most of my editing results from the fact that during live-to-stereo dates, the earlier solos are almost always the best but the rest of the tune may not be fully worked out, so that it may

Microphone and digital mixing configuration used during basic tracking sessions at Westlake Studio D for Thom Rotella Band's 'Home Again' CD release on Tom Jung's DMP label
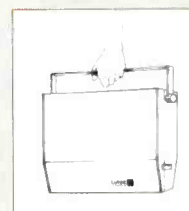
the cymbals or toms. I use
! condenser boundary,
-V *ND308* dynamic mic for

...er favourites include Sennheiser *MKH20* spaced omnis on percussion; B&K *4007* or Shure *SM80* omnis on acoustic guitar; Beyer ribbon *MC130*, *160* and *260*s on saxophone and brass; and a Shure *SM80* for string bass, mounted on foam between the end of the bridge and fingerboard. Orchestral miking might comprise a pair of spaced B&K *4007* omnis over the high strings (viola and violin) and another spaced pair of *4007*s over the low strings (cello and bass).

"Since I've been working with digital," the producer confesses, "I have rediscovered the beauty of ribbon mics. Ribbons can be a little unwieldy—because of their low signal output, they are not friendly to the concept of wide dynamic range! But, by using ultra-quiet preamps—located within 12 to 16 ft of the mics—I've had some excellent results with ribbons. I love the way they

sound on cymbals—condensers that are commonly used as overheads on drums can end up sounding 'edgy' and 'bright' through some A/D converters.

"I use omnis almost exclusively, because they don't produce the low frequency proximity build-up exhibited by cardioids. Leakage from other instruments needs to be picked up as uncoloured as possible—rather than use a separate room mic, for example, I normally set up an instrument close to the drums with an omni mic and let it serve as my room mic. I prefer that technique of double service, rather than using an extra, dedicated room mic, just so that I don't need to bring yet another source into the recording. Less is more!"

Session breakdown

The primary production stages during a more complex, multitrack-based session, Jung explains, might run as follows:

1—Mix the basic tracks via one or more Yamaha *DMP7D* 8-channel digital console at a sampling frequency of 48 kHz and record the stereo balance on to Sony *DTC1000*, *PCM2500* or Panasonic *SV3500* DAT machine

2—Transfer the 2-channel DAT tracks digitally to a Mitsubishi *X-80* PD format or Sony *PCM3402* DASH format deck and perform razor-blade editing of the best takes

3—Transfer the edited basic tracks digitally to a Mitsubishi *X-400/X-850* PD format or Sony *PCM3324* DASH format multitrack

4—Add the various orchestral tracks, additional solos, etc, to as many as six tracks of the digital multitrack overpass masters

5—Remix the multitrack masters digitally through a Yamaha *DMP7D* console to 2-track DAT

6—Perform a second digital transfer from DAT to *X80/PCM3402* for final editing and sequencing of stereo masters

7—Transfer the edited masters digitally back to DAT for dispatch to the compact disc mastering plant (usually Sony's Digital Audio Disc Corporation, Terre Haute, IN) at which point the tapes will also undergo the 1× sampling frequency conversion from 48 to 44.1 kHz and onto Sony *PCM1630* format for CD manufacture

How does the producer decide when a multitrack overpass process such as this might be necessary, we queried? "It depends upon the type of session I'm planning," the producer considers. "One song on a recent Thom Rotella tune, for example, required three mandolin passes, which were recorded against the live to 2-track basics. My previous technique of a live overpass against a 2-track DAT worked fine for one pass via the *DMP7D* console—the digital equivalent, if you will, of 'sound-on-sound' or, more realistically, a live 'overpass'—but for more complex sessions I might use a digital multitrack.

"The *DMP7D*'s on-board equalisation offers a great deal of control. The bandwidth Q is very flexible—you can dial up extremely broad curves and shape the EQ response any way you might need. I firmly believe in digital EQ; it does far less harm to the music by processing in the digital domain. I can also interface the *DMP7D* directly to the *SPX1000* digital processor and delay unit, for ambience and reverb. The DDL and chorus internal to *DMP7D* also work very well, I also prefer the palette of sound processing 'colours' available from the *1000*."

On one particular session, Jung linked together three *DMP7D* consoles. "That configuration of synchronised mixers gave me a total of 24 live inputs via three Yamaha *AD-808X* 8-channel analogue-to-digital converters. One *DMP7D* handled DI electric bass, drums and percussion; the second acoustic guitar and piano; while the third covered electric keyboards and additional instruments."

Digital vs analogue mixing

How would he summarise the sonic advantages of mixing signal sources within the digital domain, rather than mixing analogue and then converting the 2-channel balance? "Analogue processing, no matter how good the devices have been designed, can often degrade signal purity. For something complicated like a piano waveform, the complex harmonic content is much more well preserved by getting the signal as quickly as possible into the

digital domain, and then doing all the processing there—including mixing.

"Individual waveforms do not overwhelm, if you will, an A/D converter in the same way as a complex, mixed programme signal might. Drums and percussion, for example, end up sounding much cleaner and 'crisp' when I can mix them in the digital domain, rather than mixing analogue and tracking the composite to DAT—important sonic detailing between individual instrument is preserved in a way that I find more musically relevant. The *DMP7D* also features a 24 bit digital bus; the future protocol for all high quality all mixing and signal processing in the digital domain should be to a high resolution of at least 24 bits.

"The A/D converter is the weakest section in any digital recording system. Analogue op-amps used in the front end of A/D converters tend to 'smear' a lot of the subtle details, and compromise harmonic accuracy. With digital consoles I can retain the individual elements from discrete sound images; the end result is a cleaner, clearer signal to tape."

Digital mastering

Jung has been using DAT transports as data-storage devices—a term he prefers over 'recorders'—for many years now. "When I first heard about DAT I got hold of a couple of the Sony *DTC1000ES* consumer machines; the *PCM2500* is a professional version with AES/EBU digital I/O and error indicators, which are very important for me. I have also used Panasonic *SV3500* Pro-DAT machines, which are very good. I master my final remixes to four DATs

simultaneously, using Sony brand tapes, which I have used throughout my DAT recording experience with excellent results.

"For razor-blade editing of masters I prefer the Mitsubishi *X-80*. Of all of them, I still prefer my older *X-80*, which is ultra-reliable, and a proverbial 'digital workhorse'—it quite literally has never let me down. The *X-80* reminds me of the old tube Ampex *300*. Back when I first started in this business in 1963, the *300* had a very solid transport with very low wow & flutter performance; the *X-80* also is a very solid, reliable digital machine.

"The various *X-400/X-850* and *PCM3324*s really serve as the multitrack/overpass masters and are just digital data storage devices—I simply go into and come out of them in the digital domain, with no editing. I suppose that I could have used a hard-disk system to store the data, although I prefer the speed, familiarity and user convenience of reel-to-reel transports."

Jung always uses a 48 kHz sampling frequency throughout his projects. "The conversion to 44.1 for CD mastering is done at the DADC mastering plant through a Harmonia-Mundi *bw102* system, which I feel is sonically superior to any of the

other sampling frequency converters."

Auditioning the results of various D/D transfers between DAT and reel-to-reel machines, Jung says he finds "very little difference between them. I can pass back and forwards all day long in the digital domain, and not hear any sonic anomalies. I have two very good reference D/A converters that I use all the time for monitoring what's happening within the digital domain: a Krell *SBP64X* 64× oversampling DAC with Class-A outputs, and the Wadia *1000* with 64× oversampling. The Wadia retrieves more ambience information than the Krell, but the Krell is harmonically more accurate, and relates more closely to the kinds of D/A converters used in consumer units. So if I used the Wadia exclusively to judge, for example, the amount of reverb within a master mix it would sound 'dry'

when played back through most home systems, because they do not decode the nuances of ambience patterns. As a result I use both units, and pass back and forth between the two, because they have a totally different sonic character."

For interconnecting the various elements within his all-digital production system, Jung prefers to use short, high quality cables. "If I use long cables between SPDIF digital I/Os—more than a meter or so—there might be no sonic difference but the data transfer is less reliable and I can hear the occasional glitch. The AES/EBU I/O is more robust: it is balanced and can handle longer cable runs. Fibre optics make more sense for very long cable runs—for a classical session, maybe, where the mic preamps and A/Ds can be located on stage, and one could run light to the console's digital inputs."

Monitoring loudspeakers

For accurate monitoring on location in the studio, the producer uses Westlake *BBSM-6s*, "powered by whatever the studio has that's suitable—a Bryston, maybe, or I'll bring my own PSE or Krell amps. I'll always take a reference source to set up the monitor chain and the room acoustics, plus the console reflectivity and general acoustic disturbance in the nearfield. Control room monitoring has always been a problem for me—I have never heard a good set of 'commercial' monitors that sound really accurate.

"In my DMP mastering environment in Connecticut, I am dead-nuts on down to 20 Hz. In terms of accuracy, that's an octave and half I don't hear in the average control room! I worked on the design with acoustician Peter D'Antonio of RPG Systems, and we've used his diffusers after a full *TEF* analysis. Here I use a permanent monitoring system, based on a listening room concept: a small Harmonia-Mundi *bw102* control surface on a small table for digital transfers between DAT and reel-to-reel 2-track, equalisation and editing. After my DATs leave here there's nothing more that needs to be done to them except a 48/44.1 kHz sampling frequency conversion.

"My monitors are currently Australian Duntechs, a line-source monitor that is extremely accurate and has very extended bass. My other current favourites are Merlins, which are made in upper New York state, with five drivers arranged 3-way: a pair of 8 inch LF drivers, a pair of 4½ inch mid range drivers and a single metal dome tweeter—all transducers are laid out in a line, with the tweeter in the centre. I use a Class-A Krell *KSA250* amplifier to drive the Merlins or Duntechs."

The bottom line

Summing up his successful career as producer-engineer on what have come to be regarded as some of the finest jazz releases in recent decades, Jung is candid about his creativity and production strengths. "My role is to provide a unique combination of musical sensitivity and technical expertise—one without the other is a disaster. I feel that I can gather together the creative forces in the studio and capture a musical event on a medium that lets that 'magic' of the occasion reach the listener.

"It is all too easy to lose sight of what it is we are trying to do; to lose track of the music. I like to think that I can translate the musician's ideas into a viable product and share in the creative process. Anything less would be a disservice to the great musicians and artists with whom I have been privileged to work."□

Further information

The little known companies mentioned in the text can be found at these locations:
PSE (Professional Systems Engineering, Inc), 7401 Lyndale Avenue South, Minneapolis, MN 55423, USA. Tel: (612) 866-4984.
Duntech Audio, 2401 South 1560W, Woodscross, UT 84087, USA. Tel: (801) 299-0400.
Merlin International, 2705 South Main Street, Hemlock, NY 14468, USA. Tel: (716) 367-2390.

OUR MONITORS

Some classics

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Producer Al Schmitt at work in Studio Two

SUNSET SOUND

Sunset Sound, California, was one of the first independently owned recording studios in the USA. Julian Mitchell reports on why and how it still thrives

"Someone from Prince's people phoned up on the morning before he was due in to record. They said that he needed a bed to be set up in Studio Three and could I arrange it. Of course, I did. I bought a very good double bed from the local store and a bunch of purple sheets and linen, then set about making Prince's bed in the recording area of Studio Three. Apparently he needed the bed for lyric writing."

Just another day at Sunset Sound on Los Angeles' famous Sunset Strip for studio manager Craig Hubler, bed-maker to the stars. But when you're a studio that has recorded such classics as *You Send Me* by Sam Cooke; *Riders On The Storm* by The Doors; *Sympathy For The Devil* by the Rolling Stones; *I'll Be There* by the Jackson 5; *Fire And Rain* by James Taylor; *Stairway To Heaven* by Led Zeppelin; and *Beat It* by Michael Jackson, you start getting used to client idiosyncracies.

Sunset Sound is coming up for 30 years as a recording studio. People love working there and its own legend draws more people in. All this from a company who, when asked, couldn't provide a glossy studio brochure or any promotional photography. But perhaps when you're that famous any more promotion is just icing on the cake.

Bibbidi-Bobbidi-Boo

Sunset's founder Salvador 'Tutti' Camarata came into full time recording through Walt Disney. Camarata had spent a lot of time setting up London Records with Decca's Sir Edward Lewis after the war and had become head of their US division. After some years expanding the London Records catalogue Camarata gave up his executive status with the company and became involved only on specific projects. It was on his return to the States that Camarata was approached by Disney with a view to starting up the Disney record label. Fourteen years and 300 children's albums later Camarata approached Roy Disney about finding a place to exclusively arrange and record the Disney music. He was told that they would rather be Camarata's customer than the owner of the studio. Soon a property had been found on Sunset Boulevard, and by 1959 Sunset was up and running with one studio and recordings of family favourites like the music for *Jungle Book* and *Cinderella* were well on the way. Through the next few years record producers started showing interest in the place and the mixture of that and the death of Walt Disney moved the recording emphasis towards rock 'n' roll.
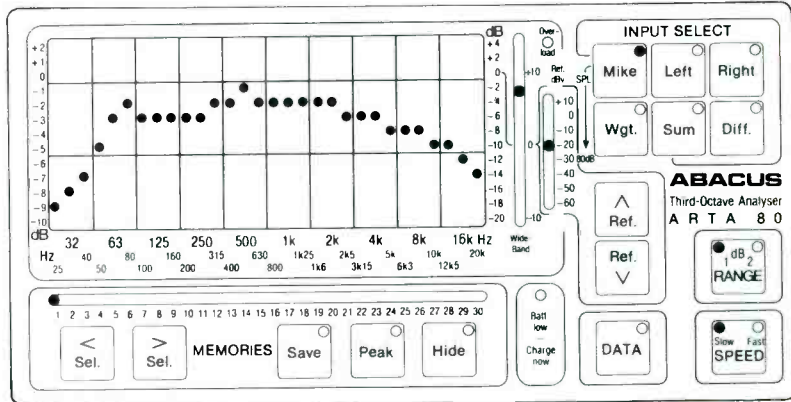
Craig Hubler, "A couple of record producers started booking time and all of a sudden Janis Joplin was in here and The Doors, pretty soon we turned into a rock 'n' roll studio; the Disney connection just started fading away. The only other independently owned studios were Radio Recorders, Goldstar and United Western. The rest of them were record label owned. We're the only surviving studio out of that list that's still under the same ownership. Tutti is kind of semi-retired now and his son Paul runs the place as CEO."

The next few Sunset years seemed to have been dominated by

Herb Alpert who apart from his own international success was producing artists like The Sandpipers and Sergio Mendes with Brasil '66. Around this time an old hair salon area was bought and turned into what was to be Studio Two. Hubler: "Sunset started in mono, single-track, ¼ inch and we have just progressed through all the formats throughout the years. The studio complex used to be a little shopping plaza. Over the course of the years we started occupying up all the other properties in this complex and building studios in them, until by the mid '70s we had taken over the Studio Three building."

The mid '60s was boom time for rock music in California and Sunset Sound were right in the heart of it. A young engineer at the time, Bruce Botnick, recalls the excitement of those days, "Between '66 and '68 it was like a train out of control. Everybody who was anybody was recording at Sunset Sound. I think we were the hottest studio in the world at that time. I would come in at 8am and work on commercials, in the afternoon we'd work on Disney records, and then at night I'd do rock 'n' roll until 4am." These anybodies included Buffalo Springfield, The Turtles, Righteous Brothers, The Monkees, Lovin' Spoonful, Frank Zappa, Captain Beefheart and Deep Purple.

Sunset claim that through these years recording innovations were coming thick and fast, like creating isolation booths for vocals and for strings. A trend that was quickly followed throughout the world. Sunset were also the first studio in Los Angeles to have an 8-track machine, on which the second Doors album was recorded.

The Sunset sound

Tutti Camarata had worked for many years in Britain and had worked on many, if not all, the desks around at the time. When Sunset was being set up he realised that none of these desks would do for his studio, Hubler, "He never really liked any console that he worked on. It just didn't do what he wanted it to do, so when he started this place and he was in the market for consoles, there still wasn't any console commercially available that satisfied his requirements in sound quality, flexibility, whatever. So he just said, 'Let's build one ourselves or we'll hire someone to build one for us to our specifications.' So over the course of 25 years we have assembled teams of people specifically to build our consoles for us. We consulted engineers,

Studio Two

guest engineers, maintenance people as to what they wanted to see and Tutti funded the project for the desks to be built and installed."

An idea to commercially produce the Sunset Studio One desk was mooted around that time and a tentative order from the Department of Defense for Armed Forces Radio gave some credance to that idea. Unfortunately, the order fell through due to production delays. The desk was, however, already causing some interest in the industry. Hubler: "A lot of people had the impression that it was an API console, well, it kind of has the qualities of the API, but the API's guts are not anything near what this thing is in its sophistication. At the time we built digital logic assignment systems from scratch, we developed some proprietary circuitry, like there was a negative feedback distortion reduction system to compensate for the transformer 'population' in the console. We were able to drive distortion down to half the level of what even transformerless boards are capable of with this system without colouring the sound. I believe the dynamic range of this console is almost 150 dB, so this thing was digitally ready a long time ago, anything we feed into it, it can handle without any kind of problem."

Sunset desks also found their way into the other two studios but in different configurations. All studios have the same monitoring, 3-way bi-amped systems designed by George Augspurger. Hubler, "We were George Augspurger's first clients. The monitors all have JBL components in them, dual woofers, mid-range driver and JBL super tweets. Studio One is slightly different from the other two in that it has TAD woofers only because the TADs for that room are very punchy. We found those were better for that room's acoustic. George Augspurger designed all the rooms. The performance areas we haven't touched, they have the original tile floors and the original ceiling tiles."

Those rooms

Studio One was the first room at Sunset but the control room has changed quite a lot through the years. The first desk there was very small and mono, by the time the second custom desk was installed the room had gone back 8 ft and the ceiling had been raised in sympathy. The room is now the main draw for the studio but not just for tracking and straight music mixing. As more film soundtracks are rock or pop orientated producers are taking them to music studios like Sunset to mix. The studio is fully Dolby *Surround* equipped and offers a cheaper alternative to the big mixing stages up the road in Hollywood. Hubler, "We do what we like to call premixing. We get everything down to maybe a 1 inch 8-channel and that goes to the re-recording stage and they just fly it in during re-recording for the final movie mix with the dialogue and effects."

Studio Two has another unique desk, unique to Los Angeles that is, in the Amek *APC 1000*. It was a major departure for Sunset to buy someone else's desk. But a decision had to be made either to carry on making a Sunset desk, which was the ideal scenario or picking a desk that could replace a custom desk but still attract the clients. The decision seemed to have been made for them with the prohibitive cost of running an R&D department in a studio. They couldn't keep up with technology and run a studio at the same time. Hubler, "There was a lot of consideration, first of all the monitoring in this room is considered by just about everybody, including the room tuning people, as the reference standard so we did not want to change anything in the dimensions of this room. We wanted to find a console that would fit in the prescribed space as the previous one, which was only a 32-input board. We also wanted a console that could handle double machines so it had to have at least 48 inputs and then, of course, digital and having the PD format here, 64. So we shopped around, the length of the console was a very big consideration. we also wanted GML automation, a recall and reset system.

"Basically the SSLs and the Neves in those configurations would be too big for this room, however, audibly they weren't our choices anyway. The Amek is the closest thing on the market sounding like our custom boards. We had double blind shootouts with guest engineers, we had Neve modules, SSLs, the Amek, our custom boards, Calrec, we ran these tests over several days with drums, vocals and everything set up in Studio

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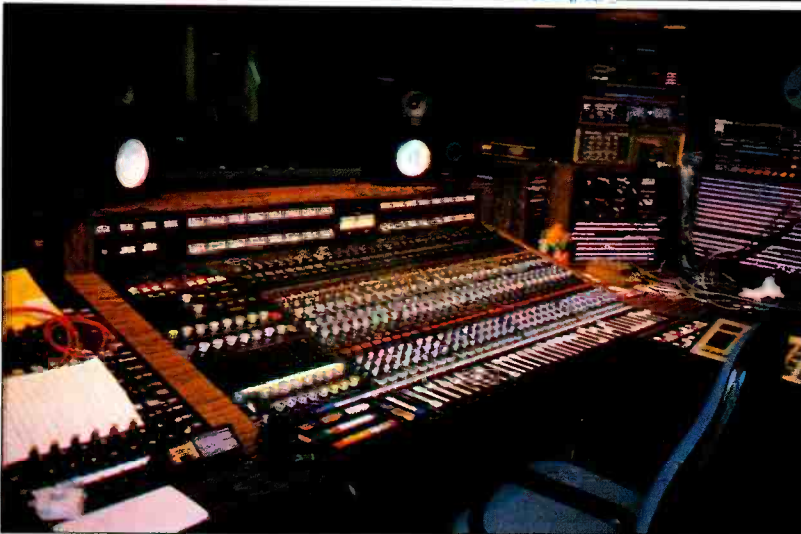
Above—Studio One; below—Studio Three

Three. We did this test and in virtually every category the Amek or the custom board came out number one."

The GML automation was built and installed under licence by Amek in Manchester, UK. This situation was borne out of George Massenberg's admiration for Graham Langley, Amek's main console designer. Hubler, "Graham Langley is probably the only console designer that George Massenberg completely respects. They very much wanted a working relationship."

Tutti and Paul Camarata had visited Amek 8 months before the console was actually installed at Sunset in 1988. Graham Langley recalls the meeting, "They would've liked to have made their own (desk) but they couldn't for whatever reason, I think they thought it would take too long and they realised they weren't console manufacturers. I don't know how long it took them to make their Studio One desk, I thought George Binder

was working on it but apparently he wasn't. But there was essentially no other console on the market that could actually do what their console could do. It was, I think, based very much on API modules anyway, but there was nothing around that had switch reset, motorised faders and everything else.

"Part of the reason they chose the *APC* was that we do still customise things, probably more than most other people, and they really wanted to do what they had done before, use the component parts and make their own console. For instance half the mic amps were designed and built by them and put in racks underneath the console, even the jackfield was a weird configuration, it was 32 jacks per strip, which nobody had ever heard of so they had to source a lot of the parts. But basically it operated in a similar manner to their console, it had the right size and we customised some of the software for them.

"A lot of the LA studios don't want the console that the studio down the road has, it's got to be personalised to them, like 'The Sunset console'. Although the Amek *APC* was the same size as the old Sunset desk they still had a size problem. When you build your own console on the premises there's no problem in the installation, you're just moving components from one room to another. Amek is in Manchester, so to get the console into Studio Two they had to knock down the control room wall, which backs onto the street."

But Sunset had taken all this into consideration when they bought the desk, and more. Hubler: "We had the whole desk wired in Mogami instead of their standard whatever they were using. We had them custom build a TRS patchbay because we can't stand TT bays, they're too difficult to maintain. We made some modifications, we added some API *550*s for our diehard clients, 24 of our own custom mic-pres, which can bypass the Amek mic-pres that sound very good. Just recently we modified their dynamics modules a bit for a much faster attack time on the gate, some of our clients weren't happy with it. Most of the problems we had with the board were with power supplies, so Amek provided replacements from an outside vendor, which have settled down the console.

"There are three different computers in this console, all this

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digital logic, recall and reset. The whole console can be reset-up for a 64-channel mix in about 5 minutes and multiple people can work on the board resetting it at the same time, whereas with SSL you have to do one module at a time, address it and reset it. With this one, five people can be setting up individual modules, which just facilitates things very quickly."

All Sunset's studios are on the ground floor of the complex with the maintenance room and some offices. Coming out of Studio Two you walk across a courtyard, with obligatory basketball net, over to Studio Three, Sunset's smallest room. Studio Three's performance area used to house Audio Rents back in the time when Tutti Camarata, who had started the company with Alan Myers, had an interest in it. Hubler, "This is really a bread and butter room, mostly tracking and overdubbing, people don't mix too much in there except Prince."

Studio Three's desk is the smallest of the Sunset custom line but is always in use, Prince used to work here 8 months of the year, "It sort of looks like a desk that Mattel would make, it's got these funky little knobs and the thumbwheel bus assignments. But it's the 'ballsyist' little console you have ever heard.

"Studio One's desk is a refined and improved design over the one in Studio Three with new types of op-amps and so forth. But Three works 365 days a year, we could have probably got a new board in here by now if we could stop the clients coming in. We do little mods as we can as it is rather limited in what we can do but we've done 32-track live digital tracking dates in this room, double machines, you know, we make it work and the clients put up with it because they like the sound of the room and console. But it's pretty much not changed since 1977."

The rise of Sunset

The impression of Sunset Sound is of a very well run studio on all levels, not least financial. Through the years Sunset have ploughed most of the profits back into the business, even at the expense of equipment that some other studios would deem vital like Studer tape machines. Until around 1985 Sunset got by with Ampex MM1200s, now they have Studers and an Otari digital multitrack. When the question of expansion did start arising it wasn't Sunset that was on the receiving end. They, in fact, bought another studio, The Sound Factory, a block away in Selma Avenue, which was up for sale in a bankruptcy case, again astute business from the Camaratas.

They bought the studio and the land where The Sound Factory

stood (as they had done with Sunset Sound) and renamed the studio Sunset Sound Factory. The new studio was bought in 1980 and has become Sunset's 'budget' facility. The studio gave Sunset more room for marketing packages for both studios. Hubler: "Often we'd do these packages where someone would track at Sunset, overdub at The Sound Factory and come back to mix, so we have budget flexibility with the record companies, they didn't have to pay the higher rates for the whole project.

"But over the year it's developed where The Sound Factory has its own client base and we have ours and only from time to time do the clients go back and forth. Sound Factory has all separate management and staff and we operate independently of each other operationally and financially. However, the digital recorders that we own do float between the two sites."

Maintenance staff are shared between the two studio sites. There are three full time maintenance members and one of them is available for Sunset Sound Factory. There is a high premium put on maintenance at Sunset, which seems to be in contradiction with the European situation where maintenance is like insurance, the last thing you take into account and the first thing that is looked at when times are hard. Craig Hubler says that if times were that hard he would go before a maintenance man.

Engineers are not in short supply in California as it's there that people naturally steer for. To have Sunset Sound on your *curriculum vitae* is virtually priceless in the recording industry but usually people don't want to move on. You can understand the lure of such a place, Sunset is the only studio on the US West Coast that still has live echo chambers—three large ones. Hubler likes to take digital-reverb fed people to see them.

"There are three very large live echo chambers, four EMT *140* plates and all this stuff sits on the third floor of this building. We have, like, what we call 'Echo Central' which you go and you can patch the chambers to any room and you can have them come up on the patchbays here. The newer crowd's only exposure to reverb has been digital and some people say, 'What's a live echo chamber?' I've had to take them up to the third floor and open the door and say, 'This is a live echo chamber,' and of course it sounds like your bathroom times 50, and they go 'Wow, that's fascinating'."

You can't help being impressed by Sunset Sound, even before you heard the story and read the guest list. Their attitude is a mix of conservatism and unleashed creativity, perhaps the perfect balance for running a recording studio.□
Sunset Sound Studio, 6650 Sunset Boulevard, Hollywood, CA 90028, USA. Tel: (213) 469-1186.

Equipment list

Recording machines
Studer *A-800 Mk III* or *A-820* 24-tracks
Ampex *ATR-100* 2-tracks
Aiwa *AD-F780* cassette decks (+4 dBm)
Otari *DTR-900* 32-track digital (extra charge)
Ampex *ATR-100* 4-tracks
Aiwa *AD-F660* multideck cassette dubbing cart (+4 dBm)
Monitoring
JBL/TAD custom designed monitor arrays
Yamaha *NS-10M* and Auratone *T-6* nearfields
Headphones from AKG, Fostex, Sennheiser, Sony, Culver
Mixing consoles
Studio One: 56/56/16 Sunset Sound custom console with GML moving fader automation and API *550-A* EQ section (all modules)
Studio Two: 64/64/48 Amek *APC-1000* customised console with recall (switches) and reset (knobs), GML moving fader automation, 48 dynamics channels and 24 auxiliary Sunset Sound custom mic preamps
Studio Three: 32/24/8 Sunset Sound custom console with 24-channel monitor section and API *550-A* EQ section (all modules)
Outboard
Comp/limiters: Universal Audio *LA-2A* (5), *LA-3A* (2), *LA-4* (1); dbx *903* (8), *160* (2), *165-A* stereo (2), *160-X* (4); UREI *1176-LN* (12), *176* (4); Inovonics *201* (3); API *525* (2); Drawmer *1960* stereo tube (1); Neve *33609* stereo comp (1)
EQs and filters: API *550-A* (4); dbx *905* (4); Orban *621B* (2) and *622B* (2) parametrics; GML *8200* (1); Pultec *EQP-1A* (6), *EQP-1A3* (6), *EQH-2* (2), *MEQ-5* (1); UREI *556* bandpass filter (1)
Digital delays: AMS *DMX 15-80S* 9.7/6.4 (3); Roland *SDE-3000* (5); Yamaha *SPX-90-II* (3), *SPX-1000* (1); Lexicon *Prime Time II* (3), *Prime Time I* (1)
Reverb/echo: AMS *RMX 16* (3); Quantec *QRS* (1); EMT *250* (1);

Lexicon *PCM-70* (3); Yamaha *REV5* (3); EMT *140* plates (4); Live Chambers (3); *Echoplate* (1); AKG *BX-20* (1)
Pitch shifters: Eventide *H-3000-SE* (3) and *H949* (2) *Harmonizers*
Kepex & gates: *Kepex I* racks (3) and *II* racks 16 cards (4); Drawmer *DS-201* gates (3); Aphex *612* stereo gates (3)
De-essers: dbx *902* (8), API *560-A* (4); Orban *516-EC* (2)
Synchronisers: TimeLine *Lynx* (6)
Other: Publison *Infernal-90* audio computer (2), Dolby *SR cat-280* cards (25); Kenwood *DP-7010* compact disc players
DAT
Panasonic *SV-3500* PRO-DAT recorder (1); Sony *DTC-300-ES* DAT recorders (2)
Video
Sony *BVU-850-SP* U-matic editor/recorder (1); video sync generator—House Sync (1)
Microphones
Neumann: *U-87* (7), *U-67* (6), *U-47* (6), *U-47* FET (4), *KM-84* (4), *M-49* (2), *U-64* (1)
Electro-Voice: *635-A* (4), *RE-20* (1), *RE-15* (1)
Altec: *633-A* (2), *21-D* (2)
RCA: *22-D* (2), *44-BX* (2)
Beyer: *MB-301* (3)
Sennheiser: *421* (10), *441* (6), *MKH-405* (1)
Sony: *C-37A* (9), *ECM-22P* (3)
AKG: *C414-EB* (7), *C451-E* (6), *The Tube* (1) *D-12E* (1), *C-12A* (1)
Shure: *SM-57* (11), *SM-546* (3), *SM-56* (1), *SM-59* (1), *SM-60* (1), *SM-7* (1)
Crown: *PZM-150* (2), *PZM-130* (2)
Telefunken/ELAM: *251* (3)
Studio Dimensions
Studio One: Control room—19×20 ft; performance area—22×36 ft with two iso rooms
Studio Two: Control room—16×23 ft; performance area —30×40×15 ft with two iso rooms
Studio Three: Control room—18×21 ft; performance area —20×50 ft with three iso rooms

INTERCONNECTION PRIMER

Until the equipment is interconnected a studio remains a collection of bits. Correct procedures and methods for putting the project together need an overall understanding of the subject for optimum results. David Dearden takes a look at the basic requirements

The design of a studio does not stop when the acoustician leaves and the last piece of timber is knocked into place. You now have to look at the equipment and make that side work—and work so you can build a reputation for quality recordings. Care has to be taken in the planning and execution of the installation and this can be looked at in eight defined areas.

Equipment location

The location of the large number of ancillary pieces of equipment with which every studio is blessed (or cursed) requires considerable thought. This equipment can be divided into two categories: adjustable and preset.

Adjustable equipment means limiters, compressors, equalisers, effects devices, etc, and these need to be located within easy reach of the operating position so adjustments can be heard in the correct acoustic perspective as they are made. Remote controls for tape machines should be easily accessible. On some of today's large consoles it can seem quicker to walk straight to the machine rather than reach for the remotes situated at the other end of the console.

Preset equipment includes power supplies, power amplifiers, monitor equalisers (if any), crossovers, etc. Power supplies should normally be mounted remote from the console to obviate hum induction into the console transformers and sensitive low-level sections, but beware of mounting the supplies too far away as problems may then be encountered due to voltage drop in the supply cables. If remote sensing is employed, cable inductance may cause instability in the supply unless properly compensated. In general, use the largest diameter cable practical and limit the length to a maximum of 10 metres. Take care that the supplies are not mounted too close to any other hum sensitive equipment such as tape machines.

Monitor amplifiers should be mounted as close as possible to their respective monitor speakers. In general large diameter cables should be used to connect the amplifier to the speaker particularly when high power is being used although there is a variety of proprietary cables that may be considered. Any appreciable resistance in the

speaker cables will reduce the amplifier damping factor causing 'woolly' bass due to the amplifier not keeping the speaker cone under tight control at all times. Ideally the speaker cables should be screened to prevent crosstalk into other cables sharing the same trunking although large diameter screened cable is not always easily available and so a good solution would be to keep those cables on their own. It is, however, preferable not to run speaker cables anywhere near other signal cables, especially its own amplifier input. Most power amplifiers are critical in terms of phase so any coupling between output and input may induce effectively positive feedback giving rise to instability on signal peaks.

Electronic crossover units are best situated close to the monitor amplifiers remembering, of course, that the monitor amplifiers produce a fair quantity of heat so it is preferable to place the crossover below the amplifiers.

Cable routing and trunking

Having decided the location of all pieces of equipment, the next point to consider is how to get the cables there. Floor ducts and trunkings are the normal method but the common problem is that the floor is usually carpeted thereby inhibiting easy access. One solution is to use carpet tiles, which are easily removed. Another way is to construct the floor ducts with removable covers that are carpet covered. With a little care the duct covers can be made completely unobtrusive yet easily removable. Wall trunkings if designed to fit flush with the wall treatments can be similarly disguised although close liaison with the acoustician/interior designer is essential. If it is impossible to keep all the trunking accessible ensure that sufficient spare cable capacity is available for all future expansion. The small extra expense is well worthwhile when the amount of down time is considered if half of the control room has to be destroyed just to add a few more cables.

All cable trunk casings should be earth bound at every join and finally returned at one place only to the main studio ground point. This could be either the electricity supply company's

incoming earth or the studio technical earth if one has been installed (and in many cases this is the only way to provide a satisfactory earth). Never rely on the trunking to provide the audio common to the equipment. Separate trunkings should be provided for high- and low-level cables; high level in this case means AC wiring and possibly loudspeaker cabling. Some organisations run all high level trunking at ceiling height and low level at floor height. Avoid running low level cables especially microphone cables near high current equipment such as air conditioning plants.

AC mains wiring should be divided into two distinct circuits wired separately back to the mains distribution board. One circuit, designated 'clean', should supply all the audio equipment, while the other designated 'dirty' should supply everything else, eg fridges, typewriters, lighting, etc. This will help to ensure freedom from annoying clicks and pops. In persistent cases, or when the studio operates on existing wiring, troublesome equipment can often be modified to make use of zero crossing switches. This will ensure that when switched on, the actual switch-on is delayed until the next zero crossing of the mains waveform instead of being switched at some random point of the waveform. As the switch-on is at a zero-crossing, ie the voltage is zero, the current is therefore zero, and a click is not produced, whereas if switched at some random point in the waveform the high instantaneous energy drawn will cause the short duration spike superimposed on the mains to appear on all susceptible equipment connected to the same circuit. Any contact that occurs will also be transmitted to other equipment. A similar problem occurs when using thyristor and triac type light dimmers. These dimmers work by switching on at various parts of the mains waveform to provide varying amounts of energy to the lamp.

In the worst case condition, when switch-on occurs midway through the cycle, the extremely fast rise time of the voltage generates high-energy signals into the megahertz region, creating interference to audio and radio equipment.

Generally, in a control room the audio equipment is well screened and at a reasonably low impedance so problems will not be caused, however, in the studio it is a different matter.

Guitar amplifiers, etc, are usually high impedance devices, open to all forms of interference, and problems will usually be encountered. Dimming of studio lighting should be provided by Variacs, which are variable rotary transformers. These vary the amplitude of the mains voltage, without distorting the shape of the waveform.

Motor-driven types are available for remote control but unfortunately, Variacs are much more expensive than thyristor dimmers of an equivalent current rating, however, they will avoid any problems.

Equipment ventilation

All equipment produces heat, some more than others, and some more than they should. A large percentage of all studio faults are caused by equipment overheating, especially faults of the intermittent variety. Careful placement of equipment and ventilation frills plus a small fan or two will minimise these problems. When installing equipment in a rack always try to place the main heat producers near the top. Placed near the bottom the heat will have to travel upwards over every other piece of equipment. Always leave a ventilation space or panel at the bottom to allow cool air to enter. In extreme cases a fan mounted on the bottom panel blowing air into the rack will considerably improve through-flow ventilation. Low-noise fans are available, or two normal fans may be wired in series to give a lower rotational speed while preserving the volume of air being forced into the rack.

If it is not practical to place heat-producing equipment near the top, intermediate ventilation panels should be provided. Where the manufacturer has punched ventilation holes in the top and bottom panels of his equipment it is reasonable to assume that he didn't intend them to be blocked by another piece of equipment sitting above or below it. If fans are provided it is beneficial to fit air filters to them to reduce the ingress of dust, but it is then essential that it becomes part of the regular maintenance programme to clean the filters.

Heat may not always be self-generated. A spotlight shining on a piece of equipment may look quite pretty but after a few hours, the amount of heating caused by the lamp may make the equipment front panel too hot to touch, especially if it is black (a colour preferred by a large number of manufacturers). In the worst case these temperature variations can cause failure of the device or annoying drift of its characteristics.

Wiring considerations

To take full advantage of the potential audio performance of consoles it is essential that the installation is carried out with care and attention. All audio signals are referenced to the system earth, which must be clean and noise-free, and essentially equipotential. In addition, the earth system integrity is absolutely necessary for safety. Do not disconnect the mains earth from each piece of equipment as this could create a hazardous situation. If in doubt consult a competent engineer and your local electricity supply company to ensure safety regulations are not infringed or negated.

You should decide on a central point for the main earth system and star feed to all mains outlets and equipment racks from this point. Common electrical wiring practice is to daisy-chain wires from outlet to outlet but this is not recommended for audio installations. The location of the earth system star point should be in a convenient, easily accessible position such as the main equipment rack. The star point must then be connected to the incoming mains earth or in extreme cases to a totally separate technical earth (if local regulations permit). This is usually formed by sinking three or four earth rods into the ground preferably at least 10 ft apart. The number of rods and the depth will depend upon the soil type and dampness. All rods are connected in parallel to reduce the earth impedance, using at least 6 mm² cable and taken to the star point. Ensure conformance with local regulations and have the system checked and approved by a competent engineer.

It may be necessary to install an isolating transformer for the clean supply to ensure adequate isolation from mains-borne interference. The isolating transformer must be of adequate current capability and should incorporate a Faraday Shield connected to the incoming mains earth.

Audio wiring

Having provided all equipment with power and earthing connections, consideration must be given to the method of providing audio interconnection, and adequate screening of those interconnections. This must be done in a logical sequence to avoid problems and assist in the localisation of problem equipment:

(a) Connect control room monitor system to the console, and check for any hum, buzz or radio frequency interference. Only when you are satisfied with the quietness of the console and monitor system should you proceed to the next step

(o) Connect multitrack tape recorder, via noise reduction system if applicable, and again check that the system is still clean

(c) Connect stereo tape recorders, studio monitors, echo and foldback sends, one at a time, checking and isolating any connection which degrades performance

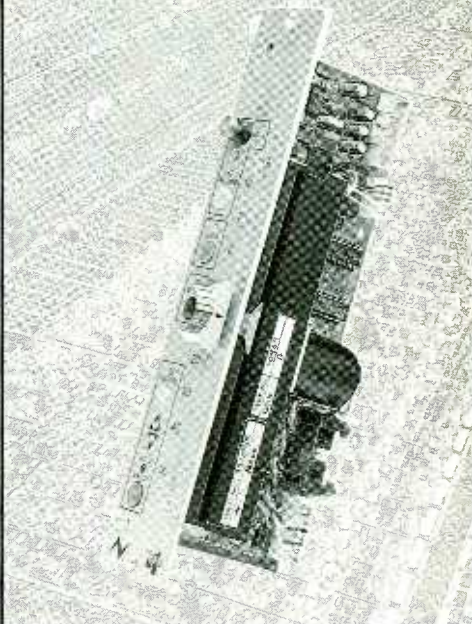(d) Connect all peripheral effects devices

(e) Connect all microphone lines

By following this sequence much time and future trouble will be saved and the result will be a quiet, stable system.

Shielding

Audio equipment is supplied with a variety of input and output configurations, which must be taken into consideration when deciding where the screen connections should be made. There are three sources of unwanted signal being impressed on the screen as follows:
extraneous electrostatic or electromagnetic fields;
noise and interference on the earth line;
capacitive coupling between the screen and the signal wires.

To minimise the adverse effects of the unwanted coupling to the signal wires, it is important that the screen is connected at one end only, ie, the screen must not carry any signal current. Any signal on the wires within the screen will be capacitively coupled to the screen, and this current will ultimately be returned to the source

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of the signal, either directly, if the screen is connected at the signal source end, or indirectly via the entire earthing system, if the screen is connected at the signal destination end. The indirect connection will cause an increase in high frequency crosstalk and should be avoided wherever possible. In general, therefore, always connect the shield only at the signal source end. In high RF areas, the screen can also be connected to earth via a 0.01 µF capacitor. This will present a short circuit at radio frequencies, thus lowering the effective shield impedance to ground, however, at low audio frequencies the reactance of the capacitor will be sufficiently high to not cause an earth loop problem.

Combinations of unbalanced, balanced, and electronically balanced (differential) systems mean that there are nine interconnection permutations. The optimum connection of the screen in each case is shown in **Table 1**.

It is important to remember that all equipment connected to the mains is a potential source of hum and interference and may radiate electrostatic or electromagnetic radiation. In addition, the mains will also act as a carrier for many forms of RF interference, generated by electric motors, air-conditioning units, Thyristor light dimmers, etc. Unless the earth system is clean, all attempts to improve hum and noise levels will be futile. In extreme cases there will be no alternative but to provide a completely separate and independent 'technical earth' to replace the incoming 'noisy' earth, however, always consult your local electricity supply authority to make sure that safety regulations are not being infringed.

Equipment interfacing

Although the problems of interfacing equipment have been drastically reduced in recent years, difficulties still exist in certain areas. Dual standards in the method of wiring connectors, odd input and output impedances, strange levels and headroom inadequacies all combine to add difficulty to a supposedly simple task.

Microphone impedance—most microphones commonly found in Europe have nominal rated impedance of 200 Ω but this does not mean that the input impedance of the console microphone input should also be 200 Ω. This is a nominal mid frequency impedance and at high frequencies, could be much higher. An attempt to match impedances between microphone and console will usually result in increased high frequency distortion due to the microphone being unable to supply sufficient current into a low input impedance. These microphones are designed to operate correctly into an impedance of at least fives times their rated impedance, ie into an input impedance of at least 1,000 Ω. This is voltage matching, where the input impedance does not appreciably load the microphone, which is then effectively working into an open-circuit so generating its highest possible output voltage.

On the other hand, older microphones were usually designed to be able to operate into an

TABLE 1

Output	Input	Screen
1 Unbalanced	Unbalanced	Source
2 Unbalanced	Balanced	Source
3 Unbalanced	Differential	Source
4 Balanced[1]	Unbalanced	Destination
5 Balanced	Balanced	Source
6 Balanced[2]	Differential	Destination
7 Differential[3]	Unbalanced	Source
8 Differential	Balanced	Source
9 Differential	Differential	Source

1 The shield is connected to the destination earth point, which is the opposite of normal practice, because the signal wires being shielded are referenced to the input earth, not the output earth
2 If the output transformer is centre tapped to earth, the screen should be connected at the source
3 When an active differential output is operated in unbalanced mode, it is very important that the output current returns to earth via the shortest, least reactive route. Check for instability at the output

NB
A In all cases, use good quality twin screened audio cable. Do not use single screened cable
B Always connect both conductors at both ends and ensure that the screen is only connected at one end
C Do not disconnect the mains earth from each piece of equipment. This is needed to provide both safety and screen returns to the system star point
D Equipment which has unbalanced inputs and outputs may need to be electrically isolated from the equipment rack and/or other equipment to avoid earth loops

input impedance equivalent to their rated impedance without causing an increase in distortion. They are designed for power matching, ie for maximum transfer of power from microphone output to amplifier input.

Impedance and cable length—most ancillary studio equipment, such as limiters, equalisers, etc, has also been designed for voltage matching. The input impedance is usually greater than 5,000 Ω, thus causing an insignificant load on the previous stage. Output impedance is usually less than 75 Ω, allowing the unit to drive into loads of 600 Ω or greater with negligible effect on the output level. 600 Ω input and output impedances are relatively uncommon these days in general studio practice, however, 600 Ω still seems to have some magical significance when it comes to specifications.

Consider though what happens when an amplifier drives a length of cable. The cable has capacitance and in conjunction with the circuit impedance will form a lowpass filter. At what frequency will the high frequency loss start to occur? For instance, an amplifier has an output impedance of 600 Ω and is connected by a cable 50 metres in length, to another amplifier with an input impedance of 10,000 Ω. If the cable capacitance is 250 pF/metre the total capacitance will be $50 \times 250 = 12,500$ pF or 0.0125 µF. The formula to calculate the −3 dB frequency is:

$$F = \frac{1}{2\pi RC}$$

where R = circuit impedance in MΩ (source impedance) and C = circuit capacitance in µF

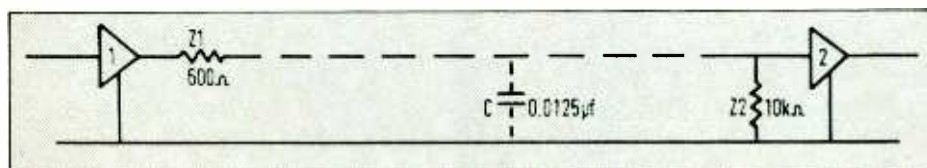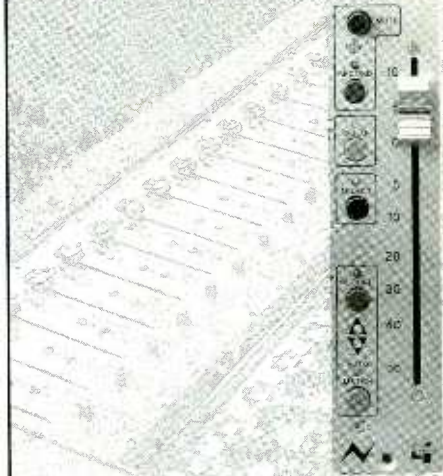$$= \frac{1}{2 \times 3.1416 \times 0.6 \times 0.0125} = 21.2 \text{ kHz}$$

Fig 1

The response will now fall at a rate approaching 6 dB/octave above this frequency, which is obviously an unsatisfactory situation but what can be done about it? A lower-capacitance cable could be used, or the circuit impedances changed. For instance Z, could be reduced to say 75 Ω, meaning that the -3 dB frequency would be 169 kHz, which would be much better.

In the real world of course, things are never quite as simple. Source impedance may vary with frequency, and a length of cable is not a pure capcitive load, however, for a first approximation, this simple calculation should suffice.

This illustrates the importance of a low output impedance when driving cables of any appreciable length, a point that must be borne in mind when installing equipment and choosing cable types. A common example is the cable used to connect an electric guitar to its amplifier or DI box. Here, circuit impedances are normally very high and 6 or 7 ft of cable can cause problems. Another common problem source is cable length between disc pickup cartridge and amplifier—the frequency response aberrations can become very complex when there is also an inductive component present. This, combined with the cable capacitance, will resonate at a certain frequency, which may be within the audio spectrum—the simple piece of wire to connect one piece of equipment to another is not really so simple after all.

Amplifiers, etc, should generally have as low an output impedance as practical, and should connect into a reasonably high input impedance of at least 5,000 Ω, a bridging load. The classic 600 Ω to 600 Ω matching situation has no real application in normal recording studios but only when dealing with extremely long lines, such as telephone lines, where the line itself has a characteristic impedance determined by its capacitance and inductances per unit.

Cable connectors—the studio business is fraught with dual standards and confusions even down to the simple job of wiring an *XLR* connector. Much American practice has been to use pin 1 as earth, pin 2 as signal low and pin 3 as signal high. European practice reverses pins 2 and 3. When all equipment contains input and output transformers, this does not usually cause any problems, provided all cables are connected the same way. However, when the equipment is unbalanced, there will be a problem because in a system wired to the European standard, pin 3 will be signal low, ie earth with an unbalanced system. If this is now connected to an American standard unbalanced piece of equipment with pin 3 being signal high, a short circuit will be present, across input and/or output. This is a very common cause of new equipment being returned to the dealer, with the message 'it doesn't work' although the dealer is subsequently unable to locate any fault. Each studio should decide upon

which standard to conform to and wire all cables that way. If unbalanced equipment is purchased and wired to the other standard, the equipment internal wiring should be altered to suit.

There is yet another dual standard regarding the *XLR* connector. This is the choice of sex for inputs and outputs. Most commonly the male connector is the output while the female connector is the input, however, certain broadcasting organisations have reversed matters and manufacturers of equipment destined for broadcast applications supply equipment, wired this way. When this equipment finds its way into studios, again it is best to change the equipment's internal wiring.

Cable checking and phasing

Much wasted time and uncertainty is caused in any studio by suspect cables and patch cords, which may exhibit any or all of three faults.

Incorrect connection at one connector causing phase reversal in balanced circuits and shorted input/output in unbalanced circuits. A very simple cable check box is shown in **Fig 2**. This is well worth the small amount of effort needed to construct and will indicate any of the above fault conditions. LEDs are used as indicators as these will respond rapidly enough to enable the user to see intermittent faults, which only occur when the cable is moved or stressed.

The checker box, although shown with only one pair of connectors, may be made with as many parallel connectors of different types as required to encompass the studio's needs. When a cable is plugged in, all LEDs will light, assuming the cable has electrical continuity. By depressing each pushbutton in turn, the corresponding LED should go out verifying the phasing, unless the connectors have been cross-wired, when it will indicate reversed connections. If a button is depressed that does not extinguish any LED, a short circuit exists between connections.

The final connection

Following the points made in this article will not make you an expert in studio wiring and interconnection but it will point you in the right direction and hopefully avoid those basic and time wasting (and possibly very expensive) errors that can be so difficult to correct. A successful installation can make a significant difference to the performance of the installation and help realise the full potential of your chosen equipment.□
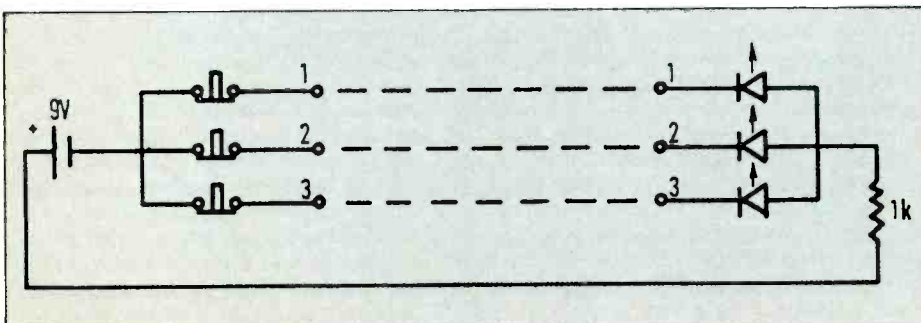
Fig 2: Cable checker

SYMETRIX DPR 44

David Miles Huber looks at the new digital audio workstation from Symetrix

One of the latest multichannel digital audio workstations was recently unveiled at the 89th AES Convention in Los Angeles. The *DPR44* (Digital Processing Recorder) from Symetrix, is a random access recorder, editor, mixer and multifunction signal processing system for use with music editing/mastering, commercial, soundtrack and audio-for-video production. Initially, this system will soon be released in a basic 4-'track' configuration that will be capable of simultaneous 18/24 bit recording, processing and playback at industry standard rates of 32, 44.1, 44.056 and 48 kHz. Due to its modular design, however, the *DPR44* may be expanded in the future to support a greater number of channels (with a possible maximum capacity of 40 tracks).

The *DPR44* is comprised of a central audio processing rack, which is designed to accept plug-in signal processing, memory and digital I/O cards in a modular fashion. Control and graphics interface is provided through the use of an Apple *Macintosh II* series computer with a 13 inch high-resolution RGB colour monitor. Communications between the processing rack and computer is accomplished via a standardised Local Area Network (EtherNet), making it possible for more than one computer terminal to share a single processing rack, or for multiple racks to be connected within a single production system.

Control over many of the *DPR44*'s transport, selection and control functions is carried out through the use of a hand-held Graphics Control Tablet and Stylus. By placing a colour template overlay onto this lightweight tablet, control over transport, on-screen selections and editing functions is accomplished by surface movements using its pen-like cursor stylus.

Virtual control

Although the *DPR44* makes use of a standard *Mac II*-type computer to perform basic control and screen drawing functions, it does not make

Edit screen

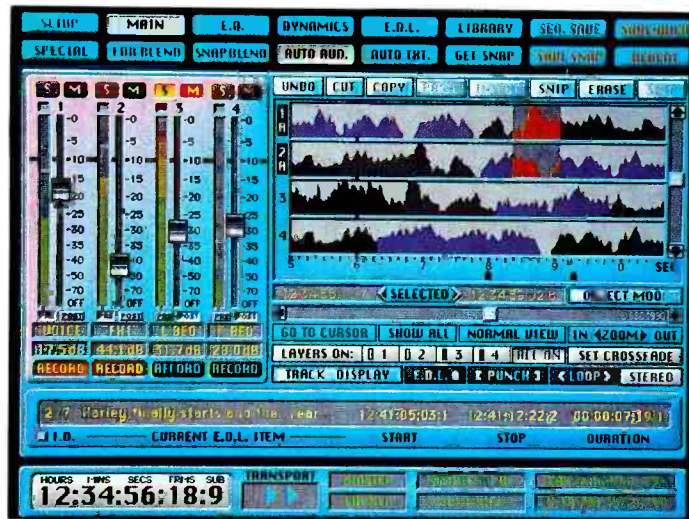**EQ screen**

Dynamics screen

use of the standard window-based graphics system that is commonly associated with the *Mac*. Instead, Symetrix has consciously opted to use a set of screens (known as Virtual Control Panels), which individually appear as a visually appealing, hi-res colour screen. Each control panel may be directly accessed from any other panel, through the use of SPEED buttons that are located at the top of each screen. By selecting the SPEED button that relates to a particular processing task, it is possible to obtain instant access to any editing, processing or other task-related function. Control over these panels is accomplished through the movement of on-screen data faders that allow a range of task-related parameters to be adjusted in realtime; keyboard entry, for entry of specific values; and on-screen Data View Windows, which allows task-related data to be graphically displayed and controlled from the graphics tablet.

Edit control

The Edit Virtual Control Panel serves as the system's main screen. It contains a set of four active track edit windows for displaying four tracks of waveform data; four full-length channel faders providing automated level control over reproduction from hard or optical disk; and a number of related track and audio processing controls. Indicators are supplied for displaying channel level (pre/post fader), while individual track legends display user-defined track names, numeric fader settings (in dB) and record/ready/safe status.
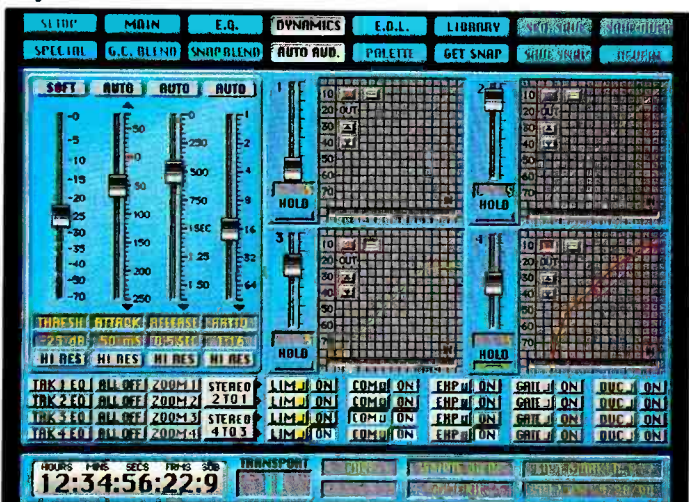
Once audio has been recorded into this panel, the *DPR44* is capable of auditioning and editing audio in two different ways: through the more conventional style of editing, scrolling and marking a continuous stream of digital audio data or through a novel method known as object oriented editing. Symetrix defines an audio 'object' as a predefined block of sound that contains a hard-boundary start and end point, along with timecode designations and duration. Through the use of object oriented editing, defined

segments of audio can be assembled, edited and auditioned by using an 'audio building block' approach to random access editing.

Any soundfile that has been recorded into the *DPR44*'s edit window will automatically be placed into the system's library as an object under a user-defined label. Conversely, all objects that are selected from this library will be re-entered into the edit window with its original sample boundaries and timecode information left intact. When operating in this mode, entire segments of predefined audio can be slipped, edited, cut or copied by selecting an object region and performing the desired function.

Another unique feature of the *DPR44*, is its ability to 'layer' up to four audio objects on a single track. This does not mean that these objects will be mixed together in realtime (a feature that could be useful) but rather that they are conceptually stacked on top of each other, allowing the front-most (or selected) layer to be reproduced, while the background layers will remain masked and inaudible. By selecting between alternate, colour-coded layers within the edit panel, the user can easily choose between a number of possible takes or audition various forms of sound effects or dialogue against any other track. In addition, any object layer can be slipped in time relative to other layers and tracks, and should any foreground layer be deleted or moved, any exposed layer revealed behind it shall become active and will be reproduced.

Equalisation control

A dedicated Equalisation Control Panel allows up to five, fully-programmable digital equalisers to be assigned to each of the *DPR44*'s four output tracks. These five EQ sections offer realtime signal processing options of lowpass and highpass filtering, low and high shelving, bandpass, parametric and notch equalisation.

Upon being selected and assigned, a frequency curve for each digital filter will be graphically represented within the effected Data View Window as a colour-coded Rubber Band. In addition to these individual EQ curves, a red rubber band graph is used to detail the overall response curve that results from summing these EQ sections together onto a single track.

Equalisation parameters may be adjusted in realtime by moving any of the three on-screen control faders that provide control over centre frequency (20 Hz to 20 kHz), Boost/Cut (±12 dB) and Bandwidth (0.5 to 2.0 octaves). These faders may be adjusted directly, or a desired numeric value can be entered from the control tablet or keyboard. Additionally, any of the above EQ parameters can be graphically altered from the control tablet and graphics pen. This is possible as

it allows the user to grab any rubber band curve and redraw its response in realtime. Using this method, side-to-side movements of the graphics pen will increase/decrease the band's turnover or centre frequency, while up/down motions will directly affect the bandwidth Q.

The *DPR44* is capable or performing non-destructive, realtime equalisation upon a track in either of two ways: by assigning a series of EQ settings to a dedicated output track (thus processing all track-related program material), or by assigning pre-defined EQ settings to individual audio objects within the edit window (allowing various EQ profiles to be applied to segments of program audio in realtime).

Dynamics control

The *DPR44*'s Dynamics Control Panel provides each track with simultaneous access to compression, limiting and to expander/gate/ducker functions. By using the panel's fader controls, it is possible to obtain realtime dynamic control over threshold (continuously adjustable from −70 dB to +20 dB), attack time (10 μs to 2.5 s), release time (10 ms to 5.0 s) and compression ratio (1:1 to 1000:1 'limit').

Graphic display over dynamics processing is accomplished in much the same way as within the equalisation panel and will allow up to five simultaneous dynamic sections to be actively displayed within each of the four data view windows as a series of colour-coded rubber bands. These individual bands are used to represent the dynamically-processed input/output curves. Control over their parameters can also be directly varied from on-screen dynamics control faders by direct entry of a value or grabbing and re-drawing a rubber band to a desired dynamics curve.

In addition to manual parameter adjustments, each dynamic section can be placed into an Automatic Program-Dependent Mode. This method allows the contents of each soundfile to be analysed (over a user-specified period) and then to apply computer-selected parameters, which are optimum for the signal content.

As with the equalisation panel, realtime dynamics processing can either be dedicated to a particular output track, or applied to audio objects within the edit window (allowing non-destructive profiles to be assigned to segments of programme).

Soundfile assembly control

Soundfiles and objects that have been saved to hard or optical disk can be retrieved through the use of a soundfile Assembly Control Panel. This

control panel allows for soundfiles to be loaded into the appropriate track edit windows, through the use of two object selection regions. An upper region is used to display a listing of the individual soundfile objects that are stored on computer disk. Each listing provides the user with an identifying label, timecode information, duration, in addition to sampling rate and mono/stereo designations.

Any soundfile from the upper region can be entered into the *DPR44*'s edit window through the use of a lower soundfile assembly list. This is accomplished by using the control tablet and pen to select and drag a soundfile object into the lower assembly list. This will allow any defined object to be placed into a consecutive, end-to-end playlist and assigned to a specific output track (or pair of tracks for a stereo source).

Once the objects within this list have been entered into their appropriate edit windows, relative start times, layer designations and crossfade parameters may then be modified by the user.

Library functions

The Library Control Panel enables a wide range of control panel settings to be stored to or recalled from hard disk at any time during a session. A dedicated Snap (snapshot) Library allows any number of equalisation, dynamics or mixing screen settings to be stored to disk by selecting the SAVE SNAP button that is located in the SPEED button section of each control panel.
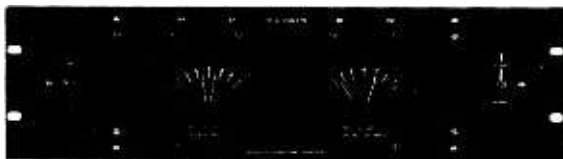
Up to eight setup snapshots can be selected from the library and placed within the main edit control panel. In this way signal processing 'palettes' can be chosen quickly and assigned to any number of audio objects, in a realtime, non-destructive processing power. In addition, the library includes a special Fader Blend and Snap Blend function, that can be used to compute automatically a crossfade between two snapshot settings to reduce the audible effects in level, EQ or dynamics when moving from one setting to another.

Inputs and outputs

Analogue signals are input into the *DPR44* via Delta-Sigma 18 bit, 64× oversampled A/D converters and are output using 8× oversampled stereo D/A converters, while digital transfer can be accomplished from any of the following ports: AES/EBU, SPDIF-2, IEC 958/II (S/P DIF and IEA-J CP-340). Control over digital and analogue I/Os can be managed from the *DPR44*'s Setup Control Panel and allows the user to select sampling rates, stereo/mono grouping modes, analogue/digital sources or outputs and other system parameters (such as machine control over external audio and video transports).

In conclusion, the Symetrix *DPR44* has the potential for offering a great deal of production and processing power, with a control interface that is both graphically appealing and intuitive. In particular, I am 'wowed' by the system's extremely fast processing speed, allowing for realtime digital signal processing and potential for transferring audio data to/from another device, while allowing the user to continue production work on a currently loaded project. Symetrix are also rolling out the welcome mat for third-party developers, whose job in the years to come will be to add new processing features and applications to this multichannel digital engine.□

MIDI PATCHBAYS

Soft or hard? Vic Lennard looks at the case for the MIDI patchbay

Before considering software vs hardware MIDI patchbay technology, perhaps it is worth looking at why there is a need for the MIDI patchbay at all. After all, most MIDI devices have a MIDI Thru, which can be used to create a 'daisy-chain' of units. There are two reasons why this is not to be recommended.

The first is down to the opto-isolator on the input side of the MIDI interface; a poor quality component will cause distortion of the MIDI signal squarewave due to the slow speed of switching. While this will probably not lead to corruption of the MIDI signal immediately, problems will occur after two or three further connections.

The second factor is the limitation of the 'daisy-chain' arrangement which can only be used for transmitting MIDI data from a central source to various modules. Changing the master device would require repatching while merging, and multiple routing would be impossible.

Different types

MIDI patchbays can be placed in four categories. The first has front panel switches through which the MIDI signal passes and so physically switch this signal. These are passive and include units from Philip Rees and the now defunct Quark. The disadvantage of this technique is that the switches can only be changed when no signal is passing through otherwise MIDI messages will be corrupted, however, they have the distinct advantage of simplicity.

The second is the type with a processor. This is used for sensing the input and only allowing the routing to be changed during a gap in the MIDI signal when it is safe to do so. Sycologic's *M8* and *M16* are of this sort and having a processor on board allows for response to MIDI program changes, a liquid crystal display and a pushbutton interface. Expansion of such a system is likely to be very difficult unless this is considered at the design stage.

The third type is software-based. This means that the functions of the patchbay are controlled by a program held in ROM and uses a processor along with multiple Universal Asynchronous Receiver Transmitters (UARTs), which deal with the serial to parallel conversion of data. Immediate advantages are the ability to perform complicated tasks but this is offset by delays that increase with the signal load and lead to the inevitable overloading of the system at some point. Consequently, the number of inputs and outputs are limited. There are many examples of this type including units by Akai, Yamaha, Roland and Audio Architecture.

Finally, there is the pure hardware system, which routes MIDI signals through a complex matrix via switches. Delays are minimal and so size of the matrix is almost unrestricted. Hinton Instruments' *MIDIX* is one such example.

Software-based patchbays

Each input or output will have a UART. When a MIDI byte is received at an input, the UART will hold the byte and inform the processor of this situation. This will be carried out under interrupt as the processor has to give its entire attention to this input for a small time interval. The processor then has to store this byte before another arrives and overwrites the first; the byte is placed in a memory location in RAM. The processor 'speaks' to the software in ROM and memory in RAM via two buses: Address and Data. An enquiry is made of a location via the Address bus to obtain an instruction from ROM or a byte from RAM which is then returned via the Data bus. In a simplified form, depending on the selected functions the processor then has to carry out, the byte will be taken from RAM, altered as required and queued up in RAM ready to be sent out from the assigned output (**Fig 1**).

As an example, a Z80 processor within a patchbay can typically handle around 600 programming instructions between the receipt of MIDI bytes and has to deal with the storing, processing, queuing and outputting of data as well as the general 'housekeeping' such as display refreshes and button scanning. Consider the following scenario. A MIDI Note On arrives at an input. The three bytes have to be stored in order so what happens if the Note On status byte arrives but without the Note number and Velocity data bytes? How long does the processor wait until that message is aborted? The longer the waiting period, the worse the delay while too short a waiting time will lead to a lost MIDI message. Current UARTs have a small two or three byte buffer so that an entire MIDI message can be received before there is the risk of overwriting data but these bytes have to be saved in the specific order they are received. What this point is leading to is the realisation that a highly efficient program is necessary for a software MIDI patchbay to function effectively and that there is an inherent minimum delay due to the movement of bytes to and from RAM.

Various optimisation techniques can be followed and these are likely to be proprietary. To give an
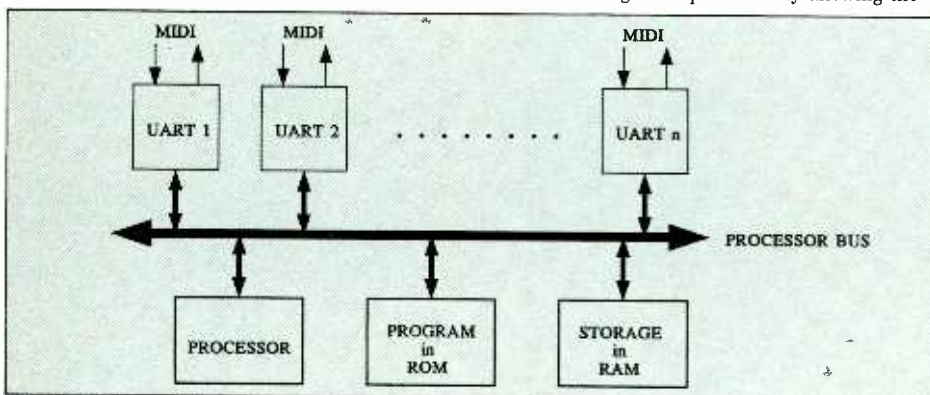

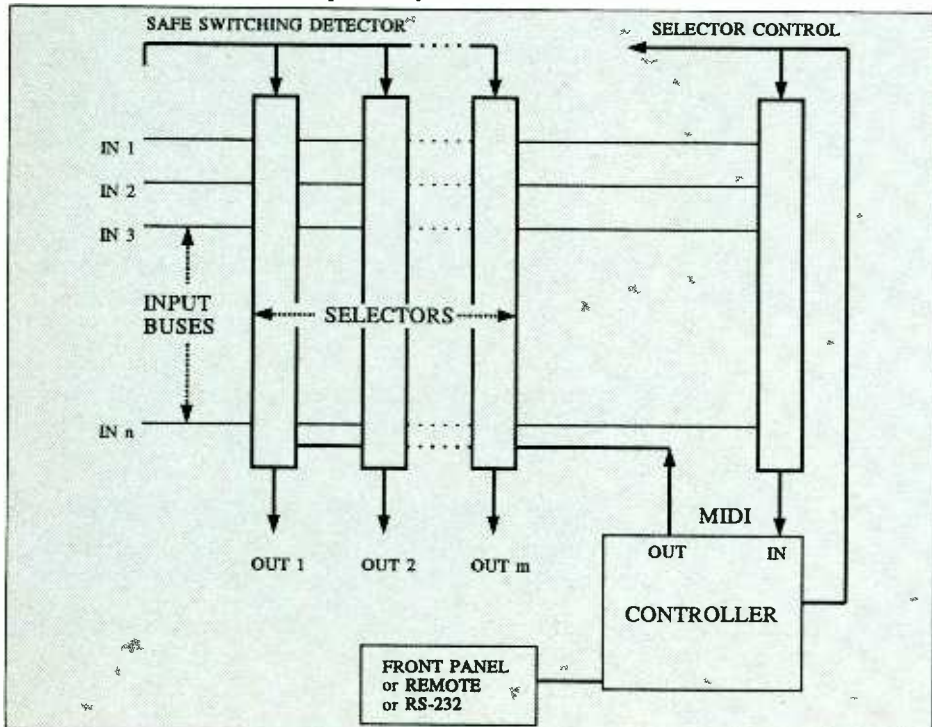FIG 1: Software-based MIDI patchbay

FIG 2: MIDIX hardware-based MIDI patchbay

example, if an input is routed to just one output there is no need to store the incoming bytes. The data can be received and relayed directly to the output with minimal delay. One input going to several outputs entails the duplication of bytes, which again means that buffering is unnecessary.

The real problems occur when merging MIDI data and handling System Exclusive. MIDI commands cannot be split up and so the processor has to ensure that the correct data bytes follow the status byte. Certain MIDI commands will need to be given priority, such as System Realtime which deals with the synchronisation of devices. Also, if MIDI data is being received in running status—the status byte is assumed to have remained the same as the last one unless otherwise changed—the status byte being held in the running status buffer has to be placed with the relevant data bytes. These procedures take time and so increase the workload on the processor.

The flow of System Exclusive varies from manufacturer to manufacturer. It is quite possible to have SysEx flowing at full MIDI bandwidth, which may leave insufficient time to multiplex it to several outputs. This will depend on the complexity of the other functions being carried out.

Consequent to the above, it should be realised that the performance of such a patchbay is limited due to the bottleneck that will occur on the processor buses and the quality of software programming. It will certainly not give guaranteed performance in all situations. However, given clever programming, software MIDI patchbays can do a lot more than just route data. Multiple layers of velocity cross-switching, zoning according to note value or MIDI channel and the remapping of MIDI controllers are just a few of the complex tasks that can be undertaken.

Hardware-based patchbays

While MIDI data in serial form has to be converted to parallel data via a UART in a software-based patchbay, a professional hardware equivalent does not operate in this manner. MIDI bytes received at an input will be sent straight to the relevant output by passing through a wired matrix with switches. The MIDI signals pass through this matrix under the control of a processor and each switch will be controlled by a safe switching detector that ensures data corruption does not occur.

A professional system should be modular. This is imperative to a professional studio because should a MIDI patchbay fail so will the studio. The patchbay should be viewed in a similar light to a mixing desk without which the routing of audio signals would be impossible. A professional mixing desk uses high quality componentry including metal *XLR* connectors and socketed ICs. So should a MIDI patchbay of similar standing. In the software example, merging data compromised the efficiency of the routing. The answer is to have the merger as a separate module that can be patched in as necessary so avoiding that compromise.

The Hinton Instruments *MIDIX* system runs along the lines of the above (see **Fig 2**). Merging is via the 3-input MIDIY module and the processor can be controlled by the front panel, remotely or via an external computer using the RS-232 interface. No special software is needed, only a comms package with a VT100-compatible

display. Illegal patches to prevent MIDI loops can be defined as can groups for soloing and MIDI event filters. MIDI events can also be set for each output to be sent in an emergency and System Exclusive can be requested and captured from any device attached to the matrix and saved to computer via the RS-232 connection. MIDI control of audio signals is also possible with the MIDIC module, which uses high quality dbx VCAs, balanced audio inputs and outputs and logic controlled FET switches.

Delays between input and output are in the region of 200 ns and are not affected by the input load. Consequently, the maximum number of inputs is determined purely by necessity.

Bottom line

If a hardware MIDI patchbay is so superior then why do the software counterparts exist? The

answer is simple—cost. A 16×16 software version with complex functions and multiple merging costs around £700. The MIDIX system of the same size with a 3-way merge module will cost four times this. More to the point, it is not practical to have a software patchbay beyond 16×16 without using fast, and expensive, processors while *MIDIX* can be expanded to 96×96.

The problem confronting people using MIDI is the lack of a middle ground. One possible future solution lies in the use of template ICs, which can be blown like PROMs and can be custom programmed to handle MIDI data. As each chip will have its own processor and a small buffer, one of these on each input and output would allow independent control of each connector. These chips would then be able to 'talk' to each other. The only fly in the ointment is that such a project would be expensive unless a fair volume of units could be sold. Is the market big enough for a manufacturer to take the plunge?□

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MONITOR SYSTEMS
INTERFACING PITFALLS
PART 9

Phil Newell assesses feedback on the series and begins to draw conclusions

Sufficient time has now elapsed to enable some analysis of the many letters and communications received in response to Parts One to Eight of this series. The one significant criticism seems to be on the question of whether the design of such systems is an art or a science. A monitor system, here includes the entire chain from the monitor outputs of the mixing console to the ears of the listeners. A monitor system includes amplifiers, crossovers, cables, connectors, loudspeaker drive units, loudspeaker cabinets and the listening room itself.

A great degree of science is, of course, applicable to the design of each individual link in the chain. There are also strong scientific bases to the interconnection of the individual links that form the chain but unfortunately, as no single link is 'perfect', compromises must be made. Those compromises are highly individual, as they follow the subjective preferences of different human beings and few agree on where the lines should be drawn. Where such individual compromise differences cannot be quantified or qualified, which is in most cases, a more artistic interpretation is inevitable.

This by no means advocates an artistic final approval, it is merely that the required science is just not developed to the necessary degree. Reproduction via a monitor system must be considered as a painting of the original, not a photograph. It is theoretically impossible to reproduce an original sound field, so any reproduction must be less than perfect. The lack of attainable perfection pervades the entire system: crossovers that do not sum in terms of power and voltage; cables that are not perfect conductors; amplifiers that distort and do not have infinite bandwidth; loudspeakers suffering similar defects as amplifiers; rooms with finite, non-uniform reverberation times; and the non-linearities of the air itself. The variables in the above imperfections are infinite. Quite clearly, there will be combinations that sum in an undesirable manner, while other combinations either cancel or are benign in their manifestation. For this reason I have long advocated an integrated approach to the entire system, as I firmly believe that the individual building blocks cannot be specified adequately in isolation.

Room and drive unit imperfections

It is all but impossible to build two truly identical rooms. Introducing the listener will disturb its performance further.

In Part Eight (March 1990) I discussed at some length the electro-mechanical and human perceptual differences in the mid range driver tests carried out with Keith Holland at the Institute of Sound and Vibration Research in late
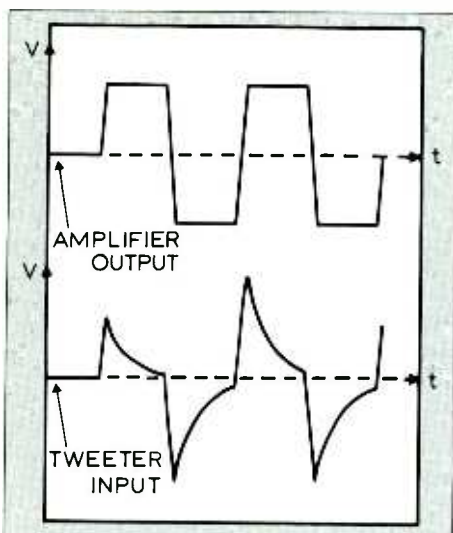

Fig 1: An example of the effect of a passive crossover on voltage

1989. No two loudspeaker drivers were deemed sonically similar over our entire range of test signals; all were imperfect. So, two of the five links in our monitor chain—the rooms and the loudspeaker drive units—have already fallen by the wayside in the attempt to achieve any degree of perfection.

Crossover anomalies

The third link, crossovers, were discussed at some length in Parts Five and Six in December 1989 and January 1990. Since those parts were written, more definitive work has been carried out in an attempt to clarify some of the pros and cons of the different approaches. Tests were constructed in which electronic, low level, active crossovers were tailored to match, to a very high degree of accuracy, the transfer functions of some passive, high level units. No matter how accurately the overall system performance was matched when measured on steady state or noise signals, the systems showed differing impulse responses and most certainly did not sound the same. There is little wonder at this outcome as the electrical loading and interaction characteristics of the two approaches are entirely different. While the active crossovers are usually terminated by a relatively constant impedance, their high level passive counterparts are rarely so fortunate, as a loudspeaker driver rarely presents anything as

simplistic as a constant impedance load. In the active unit, the driver and crossover are buffered very effectively by the intervening power amplifier, which provides an almost ideal termination to the crossover filter and an excellent drive source for the loudspeaker. In the passive, high level units, close coupling of the filters to the drive units not only preclude anything approaching an ideal filter termination but also have the unfortunate effect of decoupling loudspeaker drive units from the low impedance signal source provided by the amplifier.

Although swept sine and noise signals can be shown to perform with an admirable degree of comparability between the two approaches, transient signals react in totally different ways and when approaching overload, the performances of the two systems are radically different. When the time constants of the resonant filter circuits are close coupled to the electro-mechanical problems of a loudspeaker driver, some very unpredictable events may take place.

If a cone is pushed forwards under the influence of the signal from say a bass drum, then when the transient has passed, the loudspeaker cone/coil assembly will attempt to return to its rest position under the influence of its suspension elasticity. The resultant back EMF generated by the coil will be modified by the filter circuits *en route* to arrival at the amplifier's output terminals where damping will be effected by the low output impedance of the amplifier. The back EMFs must pass backwards through the filters in order to reach the feedback circuits of the amplifier. Tighter control is exercised when the system is actively crossed over and the amplifiers and loudspeakers are directly connected to each other. Some most peculiar currents can be generated when further transient and steady state combination signals arrive via the amplifier at the crossover terminals of a high level, passive system while the back EMFs are being fed in from the other end by a mechanical restoring force from the loudspeaker suspensions. The combination possibilities of potential musical drive signals is so enormous that trying to predict the behaviour of such a system is beyond practicability. It is like trying to fight a controlled naval battle in rough seas with a loose cannon thrashing around on the gundeck. These dynamic parameters are effectively impossible to model in an active version of a passive crossover.

What sparked this comparison work was the noticeably higher failure rate of some mid range drivers in systems with passive, high level crossovers, when compared with almost identical systems driven via active crossovers. From some of his computer models, dependent upon the drive

RELIABLE PLAYERS

signal, Holland suggested that if the system briefly overloaded on a low frequency signal, the resultant distortion products fed into the mid range units could be much higher than 'rules óf thumb' would lead one to expect. Effectively, there are situations where the time constants of the passive crossover circuits can discharge before the clipping of the low frequency half cycle has ceased. As the signal then swings from the full positive to the full negative rail of the amplifier, a product of short duration having rail-to-rail peak amplitude can be fed to the mid range driver. Discharging once again, the crossover is then primed to deliver a rail-to-rail negative to positive product to the mid driver. The crossover is acting as a voltage doubler circuit and hence, in bad cases, a power quadrupler. A typical result is shown in **Fig 1**. Such effects are signal dependent and of very short duration, and while there is no risk of thermal overload of the coil, the mechanical overload risk does exist; fatiguing the drivers prematurely when amplifier overload levels are reached.

In order to overcome these overload problems, the initially obvious conclusion would be to build a low level active crossover to drive the system in a bi-amped configuration, while adhering as closely as possible to the parameters of the passive unit; thereby removing all low frequency interference potential from the mid/high driver. On an experimental unit, the slopes were matched on the plot to within the thickness of a pen line, and the systems were measured on noise such that they were deemed to be, to all intents and purposes, identical.

Unfortunately, they did not sound the same. Their dynamic characteristics were very different: their impulse fingerprints were different. Which was 'better' or 'worse' is not relevant in these circumstances as the object was to reduce the driver fatigue while maintaining the well-liked, well known, and well appreciated sound of the original system. It is the Butterfly effect again in what are very complex, interactive and not totally linear systems. The musical drive signal has a great bearing on the nature of these perceived performance differences and distortion products, as **Fig 2** shows (from Schroeder's *Models of Hearing*). The two plots are composed of the same 31 harmonics of equal amplitude: only the phase relationship of the harmonics is different. At high levels, (a) would clearly be more stressful to the driver than (b). From this it can be seen that subtle differences in the way complex musical signals interact can lead to benign or highly awkward envelopes, even though the original signal components are identical. From these examples, it will be seen that the crossover format must be taken into account at the outset of the design of the entire system: deciding on the crossover afterwards is not realistic; it is not an add-on, it is a lynch-pin.

Amplifiers

In the discussion on crossovers in Parts Five and Six, I stated that I tended to prefer a type of amplifier for use with my own systems because I had found them to be the ones generally considered most suited to my systems. Since that was written, a most startling series of tests have taken place on the premises of a well known manufacturer of hi-fi and monitor loudspeakers.

A new range of monitor loudspeakers had been developed and tests were carried out on four different sizes of systems to choose an amplifier to be recommended for use with the range. Of the four loudspeakers, two were of the same generic type with similar crossover topology and general format. The largest loudspeaker in the range was assessed in listening tests while being driven in turn by each of four amplifiers, which we shall call A, B, C and D. When it had been generally agreed that amplifier A was preferable, seeming to give the overall character with which the greatest number of listeners were content, the test was repeated on the next size down in the range of loudspeakers. To a certain degree of surprise, amplifier B was chosen to give the best performance on that loudspeaker system. It had clearly not been the first choice amplifier with the previous loudspeaker. When the tests were repeated on the two other models, amplifier C was chosen. Only amplifier D failed to be nominated first choice on any of the systems.

When the tests were repeated with Keith Spencer-Allen (editor, *Studio Sound*) in the panel, the results corresponded with the original tests. The differences were not subtle, or only discernible by 'golden ears'; furthermore, the tests were also eminently repeatable. Not one of the test amplifiers could therefore be deemed to be *generally* superior to the others. On the other hand, reports and articles have been published
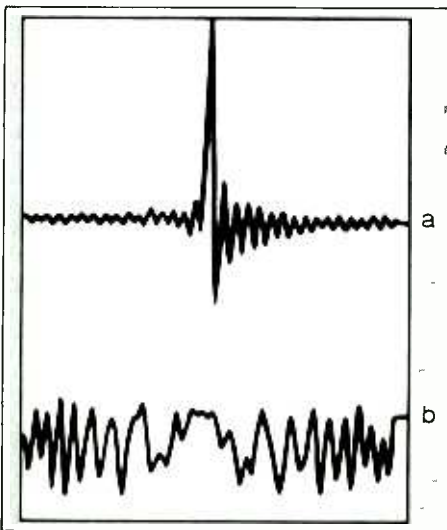

Fig 2: Photo showing effect of differing phase relationships

claiming that all high quality amplifiers of better than a given performance are indistinguishable under normal operating circumstances. How come these widely disparate conclusions? If it were possible to produce perfect amplifiers of zero distortion and infinite, uniform bandwidth, then by definition they would all sound the same; they would all be perfect. Choices between makes and models would be solely on output power, physical appearance, weight, reliability and cost. One would not sonically be any better or worse than another. Unfortunately, amplifiers are imperfect and the discrepancies in the results of listening tests are surely due to drive signal and dynamic loading conditions, which play to the strengths and weaknesses of each design.

Given loudspeakers with a generally benign dynamic impedance characteristic, then on one type of music, it is feasible that the performance of a whole range of high quality amplifiers could be said to be very close indeed. Changing the drive signal from a mainly smooth to a mainly transient nature or *vice-versa*, may then produce more distinct differences in the perceived performances. On the other hand, loudspeakers with 'obstacle course' dynamic impedance properties may well highlight some alarmingly noticeable performance differences in the amplifiers under test: instantaneous current delivery capability possibly then becoming very significant between amplifiers of nominally similar, steady state output powers. In fact I would go so far as to say that all other things being within reasonable 'hi-fi' limits, instantaneous current capability and slew rate are probably the prime factors governing the sonic performance of amplifiers.

The response from some manufacturers in the past has been to attempt to design passive high level crossover circuits that produce relatively uniform load impedance/frequency ratios to the amplifiers. Some very clever and very complex crossover circuits have been designed to achieve this goal but, as we discussed a few paragraphs ago, changing the crossover changes the dynamic loading to the loudspeaker drivers as well as to the amplifiers, and hence almost certainly will change the sonic performance. The drive EMFs go through the crossover one way, the back EMFs travel in a different way and the dynamic signal coupling between amplifier and loudspeaker drivers can become very complicated indeed.

After reading some of my articles, studio designer Tom Hidley showed interest in some of the lines of thought. He has since funded some of the research work on amplifier and cabling differences. In all his current installations, he uses monitor loudspeaker systems designed by Shozo Kinoshita. The system uses a high level passive crossover said to be capable of handling 3,000 W peaks before saturation. The impedance drops to around 0.8 Ω at a couple of frequencies and the system is rated at around 1 kW of music programme. Kinoshita had intimate knowledge of the drive units as I believe he was largely responsible for their design at Pioneer/TAD in Japan. These loudspeaker systems were always very demanding on amplifiers, the current systems being installed with JDF amplifiers costing around £8,000 per mono unit. On a more kindly load, the JDF is probably not readily discernible from another high quality amplifier, yet the options to tame the electrical impedance gymnastics of the Kinoshita crossover, or to electronically cross over and multi-amp the existing Kinoshita systems, are not available. Hidley's clients seem to like the systems sounding the way they do. Either of the above changes in the drive system would change the sonic performance. Their only option has so far been to leave this difficult crossover as is, and develop an amplifier capable of driving it, the JDF, delivering 3,200 W into 0.5 Ω. The extra cost does not make the systems any easier to sell but if that particular sonic performance is what the clients want, then the system as it stands is the only option available, amplifier choice being a critical factor.

Loudspeaker cables

What is more, the cable harness for the above system is now in the order of £2,500, they are very elaborate affairs indeed. One of my own monitor systems is comparable in size, frequency range, output SPL and most other general parameters to the Kinoshitas. The main difference lies in the fact that my systems are 4-way active.

Special esoteric cables have been shown to have only marginal effects on the performance of these systems, while on the Hidley/Kinoshita systems, the effect of esoteric cabling is readily noticeable. Once again, the effects of complex impedances seem to play a significant part in the necessity or

otherwise for using expensive cabling. Complex impedances make more demands of amplifiers and are more likely to produce premature overload and consequent driver fatigue. However, if a passively crossed over system works sonically, together with its complex amplifiers and cabling systems, then it must be considered as a system. All our work to date shows that a system can be modified radically in terms of many of its components and structures, in such a way that parameters measured against standard test procedures show remarkably little discernible differences. At the same time, another minor change to a system can cause seemingly disproportionately large differences in perceived sonic performance.

The cable issue is still somewhat contentious. We are currently investigating the possible correlations between measurable aspects of cable performance and perceived audible differences. Despite much hard work, the correlations are still very tenuous. Implications currently appear to relate to effects on back EMFs and feedback rather than forward current transfer. In terms of resistance, even allowing for skin effects, which

some Japanese engineers have claimed to be noticeable above 10 kHz, we still seem to be dealing with differences in the order of $\frac{x}{1000}$ Ω on a typical loudspeaker run. By means of a differential amplifier with over 110 dB of common mode rejection, we have so far failed to detect any noticeable differences between good quality Mk I copper cable and the more esoteric offerings, however, differences with cheap cables of inadequate size are unmistakable. I do not say that there is no difference, as for example, the general improvement on the Kinoshitas but in many cases, it may well be in some other area rather than purely in the cable itself: terminations for example.

Connector reliability

Many of the claims made for the linear crystal and oxygen-free copper cables relate to the electrical imperfections of the inter-crystal boundaries of conventional copper cable. Again, in

tests, if these effects manifest themselves at all, it seems to be in the form of very low level signal disturbances. The two immediate points to cast doubt on this very low level non-linearity being the root of the problem are firstly that our tests on audible similarity of loudspeakers showed little if any correlation between non-linear distortion products below −40 or −50 dB affecting their sound. But caution here, while some large discrepancies are seemingly innocuous, other very minor disturbances can be noticeable. Secondly even the inter-crystal boundaries of conventional copper are superb conductors when compared with the inter-metal contacts of the connectors. In other words, when a copper cable is soldered via a tin/lead alloy (solder) to a brass (copper/zinc) terminal, in turn screwed to a chrome plated copper spade terminal, soldered to a copper film on a printed circuit board, which leads to a transistor socket, which contacts a tinned leg of a transistor, and so forth. These things are all thermocouples producing their own spurious voltages in response to any temperature change. How can the wire alone make that degree of difference? Again, dissimilar metal corrosion between all these contacts will insidiously reduce the contacts' effectiveness over a period of time causing them to behave in a truly unpredictable manner. There is an insidious risk of performance deterioration with time, which may be very hard to detect without a strip-down and rebuild.

Holland is strongly of the opinion that a person paying hundreds of pounds per metre for cable will make very sure that good quality connectors are used and that all soldered joints are of good standard. Connections will probably be made very carefully indeed and attention to detail will be rigorously applied. I know of one classical engineer in particular who resolders his leads every 2 or 3 years to try to ensure good contacts but is that practicable for a mixing console or tape machine?

Many loudspeaker drive unit terminals have chromium plated push connectors, which most certainly deteriorate over a period of time as the surface oxides build up. Except under highly controlled and very repeatable conditions, it can be difficult to isolate some of the above inter-metal contact problems from the effects of the cable itself.

Variability of air

We have established variables in terms of performance shortfalls in the loudspeaker drive units, the crossovers, the amplifiers and the interconnecting cables. The fifth link in the monitor chain is the air in the room, which is itself an interconnect between the loudspeaker, the room boundaries and the ear. To the uninitiated, air is air but a brief digression into the world of aeronautics may help to emphasise just how variable this medium can be.

Say an aircraft has a service ceiling of around 10,000 ft, at which altitude the rate of climb at full power is virtually zero. In a suitably hot, high and humid climate, the same aeroplane would be unable to leave the ground, no matter what length of runway it had available.

When air is saturated with water vapour, it is much less dense than when dry. That is because water, H_2O, has a molecular weight of $18(2+16)$ whereas each molecule of oxygen, O_2, has a molecular weight of $32(2\times16)$ or molecule of nitrogen, N_2 a molecular weight of $28(2\times14)$. The three gases vie for space in any given volume and displace each other in such a way as to maintain

the number of molecules per m^3 for any given gas pressure. Water, oxygen and nitrogen molecules exist as H_2O, O_2 and N_2 respectively. Given that the atomic weights of hydrogen, oxygen and nitrogen are 1, 16 and 14 respectively, then the above molecular weights can clearly be seen. What is more, the air density reduces by around 1 mbar for every 27 ft increase in altitude. Denver, CO, USA, at over 5,000 ft above sea level would have air some $5000 \div 27$ mbars less pressure than Oxford, UK, for the same sea level conditions. If the barometric pressure on a typical day at Oxford were, say, 1,000 mbars, then Denver would be some 185 mbars ($5000 \div 27$) less. Add some heat and water vapour to the air in Denver, which both reduce density, and you could well be experiencing pressures of around 750 mbars—enough to prevent the aircraft, which would have happily risen to 10,000 ft over Oxford, from being able to take off. As the pressure falls, the speed of sound also falls, so frequency to wavelength ratios in a studio in Oxford would be significantly different from the wavelengths in a hot humid studio in Denver. Remember, the difference is not subtle! Build yourself a studio in Mexico City at 8,000 ft and the differences will be even greater. Climate and weather can be significant factors in system performance and human perception.

Variability of perception

The variables and non-linearities in all the links of the monitor chain can interact in such a way that system prediction from individual component parameters is not feasible. If we cannot predict, then we can only listen but organised listening tests have regularly failed to live up to expectations or to produce conclusive results. Evidence is now emerging that when we listen in a relaxed manner or are concentrating on something else, the main brain activity in response to the music is in the right hand hemisphere. As soon as we begin to concentrate on that music or to listen for differences in sounds such as in A/B testing, the more logical left hand hemisphere of the brain will dominate the response. When this shift of predominant activity from one half of the brain to the other occurs, we are entering a very different mode of perception. Little wonder then that some of the more subtly perceived differences can be so elusive under more controlled conditions. Begin to look, and you change the conditions of the object of observation.

That the two halves of the brain are different in terms of perception is easily proven by the classic experiment of giving a person a pair of oscillators, one fixed and one variable, one being fed to each ear via headphones. When asked to tune the variable oscillator to the fixed one, there will frequently be a discrepancy in the tuning when checked electronically. When the headphones are reversed, the discrepancy will be in the other direction, showing that one half of the brain, is perceiving a different 'absolute' pitch to the other. The difference can easily be in the order of 3 or 4 Hz at 1 kHz. Under normal listening conditions, no such pitch distortions are evident.

Subtlety relevance

Whereas audiologists may frequently deem that very small differences detected under controlled circumstances are generally too small to be of

consequence, the artistic environment of the working studio can experience a human response to much more subtle changes. The claim of the audiologists that if a monitor system difference is so subtle, then it must be specific to a system and will have no bearing on the mix when played elsewhere, is harder to argue against. However, the studio environment is sometimes such a delicate factor in the creative process that it is hard to argue that there is anything too small to have an effect on that creative process. The chaotic interaction with the human factors will almost certainly ensure that such things cannot be dealt with in a purely scientific context.
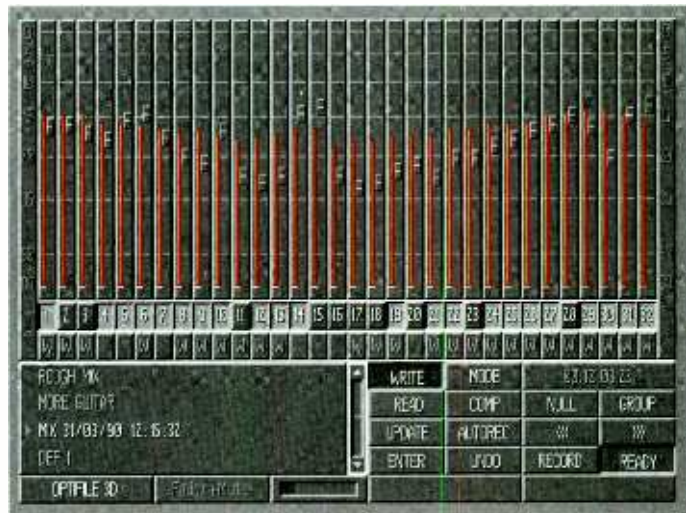
If a system works, and a progressive series of modifications or developments is deemed to improve that system, then those improvements are positive steps in the performance of that system. Modifications that prove beneficial to that system will by no means automatically prove beneficial to other systems. While it is obvious that many of the major pre-requisites must be met before a system could really be taken

seriously, the more subtle and esoteric concepts frequently discussed in the hi-fi press are not always so accommodating in stating their cases for inclusion. Until much more work is done in some of these areas, cost effective solutions to the rat's nest of cross-correlations are unlikely to be forthcoming. I know of a designer with some large cheque-book clients, who operates on the concept 'If some people think it sounds better with X while others are not so sure, then as long as nobody thinks that X makes matters worse, we will incorporate X.'

That approach does seem to work but on a process of diminishing returns in the financial area. To be able to define absolutely when X is beneficial or when it is a waste of time is the main thrust of much of the current work with which I am involved. By that means, to know when to use X to benefit rather than as a generalisation, will free the money to also apply Y and Z in their most beneficial roles. Comments on any of these points will be greatly appreciated.□

Kodak are planning to change the face of the cinema industry. The company have been working for 2 years with the Optical Radiation Corp of Azusa, CA, on a system that records six channels of hi-fi sound as digital code in an optical soundtrack down the side of a cinema film.

It has previously been thought impossible for film to resolve, and film projectors to read, the many millions of tiny black and white dots per second needed for digital hi-fi sound. Kodak have developed new film with better resolution and ORC have built a solid state sensor, which compensates for the sideways 'weave' of film as it runs through a projector.

So far the *Cinema Digital Sound* system works only with 70 mm film, as used only by major city cinemas, but the inventors claim they will later make it work with 35 mm film. Kodak and ORC have equipped a dozen cinemas in the US, most in the Los Angeles area, and two in Manhattan. They were supplied with digital versions of *Dick Tracy*. The first demonstration in Europe was given at the Kinepolis cinema complex near the Heysel football stadium in Brussels, a modern complex housing 25 cinemas under one roof, all already equipped with Dolby stereo and *THX* sound systems.

The sound of *CDS*, as heard at Kinepolis, is impressive. But Kodak's hard-sell on *CDS* as 'the ultimate sound experience... (which) can make every local cinema sound better than today's first-run theatres playing 70 mm Dolby surround sound' may prove dangerously over-optimistic.

In fact both Kodak and ORC are selling the concept of *CDS* more like double glazing than a new film system. This is exactly the opposite approach to the softly-softly line used by Dolby Labs to break down half a century of hide-bound inertia in the film industry.

Richard Wood, president of ORC, claims, "In the 35 mm format cinema digital sound will be superior to current 70 mm Dolby stereo sound."

A publicity film produced by ORC, with clips from many features including *Top Gun*, *Apocalypse Now* and *The Wizard of Oz*, has impressive sound but is clumsy in the way it almost completely ignores even the existence of Dolby Labs, let alone the company's achievements in cinema sound.

'As is their usual fashion, Dolby engineers are just sitting quietly, watching, obviously believing that both Kodak and ORC have gone in over their heads with claims to robustness that may prove hard to justify, especially for *CDS* in the all-important 35 mm format.

The conservative film industry took over a decade to embrace the Dolby analogue system and will not readily spend the $20,000 Kodak and ORC will now ask for a digital reader to convert an analogue stereo projector to digital. Film studios will not make *CDS* prints unless cinemas can use them.

Dolby Laboratories started equipping cinemas with stereo systems in 1976, when *Star Wars* was released with an optical stereo soundtrack. Although most new films are now made in Dolby stereo, there are still only 15,675 cinemas in the world equipped to play Dolby stereo. Kodak

Barry Fox

Six-channel digital soundtrack to enhance film sound; storage for 100 photos on CD by 1992; the case for CD 'gold'

estimates that there are 90,000 cinema screens worldwide, which leaves nearly 75,000 cinemas still content to use low-fi mono sound systems.

There is not room on a film for both analogue and digital soundtracks, so to support *CDS* the film studios must produce two versions of each film.

CDS records five channels of sound to be spread across the screen and round the cinema. A sixth channel carries deep bass for music and special effects from all directions. Two extra channels convey code for synchronising other equipment, such as screen curtains, laser light shows or even electronic musical instruments. *CDS* sound is coded to the same standard as domestic compact disc (16 bit words streaming at 44.1 kHz) and all eight code streams are interleaved to create a single digital stream running at 5.5 Mbit/s.

For recording onto film, this single stream is sub-divided into 180 parallel streams, which are recorded in the soundtrack area as 180 very narrow parallel tracks. Each contains tiny black or white dots representing digital ones and zeros. Under a microscope the soundtrack area looks like a chess board.

The projector is fitted with a CCD sensor, with 512 individual light sensitive spots in a line lying across the film. The CCD remains stationary but however much the film weaves sideways while running through the projector, a block of 180 sensors can always align accurately with the 180 film tracks.

Digital error correction, similar to that used for compact disc, compensates for dust or scratches on the film soundtrack area. Kodak and ORC claim that their system is so robust that a film print can be run 1,000 times and still sound perfect. Cinema engineers, who know how badly cinemas treat films, remain suspicious of this claim.

Sceptics also await proof of ORC's claim that *CDS* will work with 35 mm film. The wider 70 mm film runs through a projector at 22.5 in/s (57 cm/s) compared to 18 in/s (46 cm/s) for 35 mm film. *CDS* uses the same soundtrack width for both film sizes, and ORC promises the same 5.5 Mbit/s data rate. So the 35 mm system must rely on a 25% reduction in the already tiny bit size.

The digital sound reader can be added to any existing projector, even clamped to the wall of the

projection booth. This is possible because the digital information comes off the film ahead of the matching picture images and is stored in a buffer memory with a control that lets the projectionist adjust the delay to compensate exactly for the position of the sound reader with respect to the lens beaming pictures onto the cinema screen.

Because most cinemas still use only 35 mm film, *CDS* can only succeed if it works with 35 mm prints. ORC were promising a short demonstration to prove the feasibility of 35 mm *CDS* in Hollywood in October. But the *real* test will be when cinema projectionists have mishandled 35 mm *CDS* prints.

For those paying £20 or £30 each for blank write-once optical disks, note this. Kodak (and Philips) promise Photo CD for 1992. A write-once CD disk will be used to store 100 high quality photos as digital code. The disks being used by Kodak are either made by Taiyo Yuden in Japan, or by Kodak in Rochester, NY, using Taiyo's dye polymer technology. Kodak has been working secretly on a disk that sandwiches a photofilm between the layers of a CD.

A photo lab will pay £60,000 for a Photo CD transfer station, including scanner, Sun workstation, encoder, picture processor and disk writer.

Kodak promises that a member of the public will be able to get 24 pictures transferred from photo film negatives to Photo CD, with zoom and colour correction, etc, for each frame, and blank disk thrown in—all for £9.

Bearing in mind the capital investment and labour costs, that means the price of a blank must be less than £1.

Mobile's latest publicity material adds more weight to the argument that gold-coated CDs are likely to last longer than the conventional aluminium discs.

Mobile acknowledge, "Under normal conditions, with reasonably careful handling and storage, the aluminium disc should perform well for a very long time," but go on to give further information on the severe life-acceleration tests conducted in conjunction with Tonen Sekiyukagaku Petro-Chemical of Japan, and the Naval Air Development Center of the Department of the US Navy. Tests were conducted on three types of aluminium compact disc from various manufacturers, and two of gold.

All discs, aluminium and gold, were subjected to the same conditions of 80°C and 90% humidity. The block error rate, BER, measurement indicated a degradation of all the aluminium CDs. After 275 hours, all aluminium samples were unplayable, whereas the gold samples remained well within the Red Book specifications on BER. They also remained playable.

Further tests by Ultech in Japan put 10 aluminium disc samples, and six gold samples, through cycles of extreme temperature and humidity. Most of the aluminium discs were unplayable after 30 cycles while the gold discs still played on.□

Kevin Stone was tired. He had put in another 18 hour day yesterday at Artesian Recorders and felt it. But he felt better now that he had his lunch in front of him. A rich and satisfying bacon pudding awaited his approval. Accompanying it was a nice order of those steak cut double-fried chips he so fancied. But he couldn't start eating just yet. His stomach hurt and he popped another remedy. He was bothered by the current problem at the studio caused by Jack Bartholomew. Jack was his number two and a minority partner and his 'bloody' cousin. He guffawed at the sound of 'minority partner'. It made Jack sound like an East Asian. No, Jack was a problem. He always had to futz. He couldn't leave well enough alone. Now he was trying to connect the timecode DATs to create a 4-track DAT system. Whether it could or could not be done was not the point. They had a post-production suite with 24-track recording capability, SR and timecode. It was much more than good enough for the film trade. Everyone who used it told him so. But all his competitors had such capability too, or so it seemed, and demand was light for post services. They had had to let their maintenance lad go and Jack was supposed to take up the slack. Instead he was still futzing around.

Kevin pondered to consider that the Anchor and Crown pub was indeed his home away from home. He most often consumed his lunch and dinner there, if he ate away from the studio. He especially liked their savoury puddings, their bubble and squeak (Brussels sprouts and mashed potatoes fried together), the fish and chips and most of all, their famous steak and kidney pie. Not all at once, of course. As he greedily consumed his lunch, the thought of Jack's previous escapade continued to haunt him. Jack had tried to automate their small Neve console. He had bought an automation kit from a small advert in a trade rag. He had never cared for that magazine and its back-of-the-book ads were of questionable value. Nevertheless, Jack had forked out over a thousand quid for a box of parts and cable and so-called automatic faders. The net result had been a ruined console that not even the service people who worked on the older Neve units could restore to total function. The console and its room had been down for nearly 3 months.

He and Jack used to meet for breakfast at the neighbourhood McDonald's restaurant. He had loved the food but he had gradually reached the point where arguing with Jack was driving him crazy first thing in the morning. The whole point of meeting early was to air the daily dirty linen between themselves and not in front of the other employees. But that hadn't stopped Jack from engaging in bouts of noisy confrontation all day at the studio, so it seemed senseless to continue to meet in the morning. Jack continued to eat at Micky D's, so he had no choice but the necessity of the little Greek-run greasy spoon on his corner. They served up a barely passable breakfast of fried eggs, bangers, fried potatoes, tomatoes and fried bread. Not that anyone should assume he ate this way all the time. He frequently had some nice cereal at home in the morning.

The day had started like many others. He and

Martin Polon

You gotta have heart. Comment from our US columnist

Jack had engaged in a running argument over the new 'multi-DAT' room that Jack was sure would bring in oodles of new business. Kevin had gone out to lunch to escape Jack, rather than consume some nice Cheshire cheese with several croissants that the phone girl would bring him back when she went to lunch. Last night, he had stayed late to go over the books. Many was the night he would work late at his desk, accompanied only by the two Wimpy burgers without fries that the night recording staff at the studio would bring him. He noticed how his office chair upholstery had become frayed and worn. He pondered how the state of the chair had changed...probably reflecting how much time he really spent in it.

At 3.30pm, Jack swung into Kevin's office doorway. Jack hollered, "Haloo, I need your help with the turret for the 'Double-DAT' post system. It's still packed on the truck."

Kevin began to steam. Jack knew he wasn't supposed to order the imposing metal console without Kevin's go ahead. Far worse, he had to rub it in Kevin's face by asking him to help unload it from the lorry. There were several technicians available who could have helped but it was obviously more fun to goad Kevin. Kevin became angrier and angrier as he helped muscle the huge monstrosity over to the lift gate on the lorry. He pushed the behemoth metal shell the last few inches when it happened.

He doubled over with the most intense pain he had ever felt. At first, he thought that somehow the 'thing' had fallen over onto his chest. He then realised that he was short of breath and sweating profusely. He began to disappear into the pain as any thought or movement seemed so impossible to contemplate...let alone to do. His last conscious memory of the experience was Jack standing over him and yelling, "Kevin, come on and get up—it really isn't very funny. It's going to rain and we have to get this inside so it doesn't rust."

Kevin's abysmally bad luck began to change at that exact moment. In fact, he was about to have the best luck of his life. Michael Moore, a local PC (police constable), had been walking his beat and had entered the alley/mews behind the studio. He had been watching with studied disinterest as the two men had been struggling with the large metal enclosure. PC Moore heard the laboured gasping of the smaller man as he fell to the bed of the lorry. He had audible evidence, rather than visual, of the first external signs of Kevin's heart attack. Having been well trained to be alert to

such circumstances, the PC decided that Kevin's fall grasping his chest demanded immediate action. It had taken him just a few seconds to touch the microphone/speaker buttoned to his shoulder loops and contact the dispatcher at the area communications centre for that region of London. He then leapt up to the bed of the truck to assist Kevin.

As fortune would have it, an ambulance had just dropped off an elderly patient several blocks away, who had been accompanied—more's the luck—by her physician. The emergency call was taken by that van and in a matter of 180 seconds, Kevin was in the hands of trained medical practitioners. A short trip through London traffic to the nearby Middlesex Hospital followed, with help from one of the best cardiac departments in the country being rendered. Kevin's luck had indeed changed for the best. He survived his massive heart attack and recovered most of his powers after a long hospital stay and equally long recuperative period.

Traditionally for this column, the opening piece of illustrative writing is made generic by being fictionalised. More unusual is the fact that rather than being an amalgamation of half a dozen or a dozen different actual happenings, this one represents the combined experiences of ten's of thousands of individuals. I was very nearly one of them. The fact is that the majority of studio owners, operators, managers and experienced mixers and technicians in the major studios are over 30 years of age, with many over the age of 40 years. The same can be said for most other branches of audio engineering, installation, mixing, repair, sales and management. But, there is no conspiracy today to keep younger people out of the audio business. There is rather the result of historical imperative, with the tens of thousands of post-World War II 'babyboomers' having been young people at the heart of the growth and development of the audio industry. Not to mention that the self-same babyboomers are, in addition, the demographic group that purchased records in the greatest numbers during the '70s and the '80s, pushing the record business to stratospheric levels. What has happened is that the world audio industry has entered its maturity and along with it comes the aging babyboomers.

It is one thing to age and it is another to age well and with good health. For many of us, as it was for me, the pressures of busy and stress-filled lives had changed who we were, how we lived and—most important—how we ate and how we exercised...if we did at all. For those of us over 40 other medical conditions involving thyroid functions, metabolic slowdown, adrenal and hypothalamus activity, etc, can further affect how we live and how we metabolise what we eat. For many of us, our jobs change. We start out as young, eager, aggressive individuals. We tote that barge and haul that bale. We give credence to the classic definition of soundman (and soundwoman).

'Look...up there in the sky...is it a bird or a plane or a forklift? No, it's soundman...able to schlep huge sound systems at a single bound.' But all that early activity kept us healthy and

protected us from the cockamamy diet we all eat with the stress and unpredictable hours of the audio industry.

As we get older our jobs change. We become mixers instead of schleppers. We become managers instead of technophiles. We become leaders instead of followers, owners instead of employees. And most important, as in my case, we become 'computer potatoes' instead of spry active individuals. The net result is that a lot of us have become 'stocky' or 'heavy', developed 'mature waistlines', refer to ourselves as having a 'big frame' or 'large bones'. In fact, we have become FAT. How much time did many of us spend at the last AES Convention standing around the bar (drinking expensive 'name' water with a twist), discussing our latest diets, the inexorable medical or personal reason we were putting on weight 'beyond our control', how our closets stored at least two or sometimes three wardrobes to cope with our varying weight and what was the latest 'fad' food to avoid.

Of course, weight gain is not the only measure of our decline. The combination of poor diet, lack of exercise, stress, bizarre and excessive working hours, smoking and substance abuse sounds like an accurate description of life in the fast lane (or is that the FAT lane?), for the audio business. In fact, the net result of such abuse of our bodies is measured in obesity, high blood pressure, high blood sugar, high triglycerides, high serum cholesterol, diabetes, heart disease and stroke. What is equally curious is that it is possible to reverse the whole process. It happened to me. A recent trip to the Pritikin Longevity Center in Miami, Florida, has restored me to good health and helped me to shed in excess of 30 lb. The key to the programme is education: a high fibre, fat-free, sugar-free and salt-free therapeutic diet; intense medical supervision; a personally customised exercise plan based on walking; complete abstinence from tobacco, alcohol, caffeine and other abusable substances. The results were just plain miraculous for me.

Not everyone in the audio community can avail themselves of a programme such as the Pritikin system but there is a great deal each one of us can do to add years to our lives. It is also of note that younger individuals are equally benefited by a healthy life style and can avoid the path of self-destruction that much sooner. Let us examine some areas of potential change:

●**Smoking** Smoking tobacco has been linked to coronary heart disease and stroke by literally dozens of studies. Add to that the proven risk of emphysema and cancer in the lungs, larynx, lips and mouth and you have a partial listing of the results of inhaling the devil's brew of tars, nicotine and other chemical byproducts. The answer is simple. Stop smoking. A ban on all smoking in the audio workplace is also essential since smoking creates polluting deposits that can affect audio equipment.

●**Caffeine** A mixed bag of research studies indicates that excessive consumption of caffeine contributes little that is positive and much that can possibly contribute to heart disease, hypertension and stroke. Better to switch from coffee to coffee substitutes and from tea to herbal teas. Especially if high levels of stress mark the life style of the caffeine user. Many people become addicted to caffeine without realising it. If a conscious effort is to be made to lessen caffeine usage, a gradual reduction is the recommended format. The other problem with caffeine is that many carbonated soda drinkers are totally unaware of the caffeine hit they get from colas and some other soda beverages. Check the label before you drink.

●**Substance abuse** In this category we find alcohol and legally controlled substances. Any abuse in this area usually has the long-term effect of destroying the user's health and diminishing function to the point where one cannot be considered capable of performing even the most rudimentary tasks while under the influence. To stop substance abuse, one almost always needs to get professional medical help and/or organisational help.

●**Exercise** The best exercise is also the simplest and the most satisfying—walking, biking and swimming. One of these three, done four to six times per week for at least ½ hour sessions, will help improve the quality of life and the health of the exerciser. A doctor should always be consulted

All these efforts can make life a much more enjoyable experience and a much longer one at that

before starting an exercise programme and a professional exercise physiologist can be a great help in designing a programme for an individual.

The other side of the exercise coin is to add exercise to your daily schedule by climbing stairs instead of using elevators and escalators, walking to the bank instead of driving two blocks, walking around a facility to hold a conversation rather than using the intercom or an internal telephone, etc. The more energy you expend during the day, the more calories you will burn.

●**Stress reduction** Reduce the negative impact of your daily work life upon yourself. At the same time, admit to yourself those character traits you have that put more stress upon yourself. If time in general and your own late arrival at meetings in particular is your problem, consider controlling your time environment. Such control can take many forms. Set your watch 5 or 10 minutes ahead so even if you run late, you will end up on time. If you know you are always playing meeting times, right down to the wire, write the time in your calendar with a 15 minute pad. For example, log a 2.30pm meeting in as being at 2.15pm. The worst that will happen is that you will arrive early. Work to reduce the impact of daily stress on your body. Exercise is one of the best stress reducers known. If pressure makes you feel like you will explode, go take a walk. Do a work out

with whatever form of exercise you enjoy most. You can see a mental health professional to learn relaxation techniques and/or you can buy a book on the subject.

●**Sodium/salt consumption** Sodium, in the form of table salt or as Monosodium Glutamate (MSG) will inevitably hold water in the body and cause the blood pressure to elevate in susceptible individuals. There is almost nothing you can say that is good about sodium, except that it does enhance flavour for those whose palates have been so dulled by excessive salt consumption that only more salt will do the job. Decrease your salt consumption by reading labels. Try to keep your salt consumption well below the 1500 ml mark per day. Replace salt with lemon, herbs, spices and other salt-free or sodium-reduced seasonings.

●**Dietary fat consumption** The British and American national diet finds *per capita* consumption of dietary fats in excess of 30% and frequently closer to 40%. Fats become deposits in arterial passageways and slowly block the flow of blood. When the blockage is total or impedes a blood clot: in the heart we have a heart attack; in the brain we have a stroke. All in all a most undesirable turn of events. Again, read the labels on food products you buy. Swap fatty items in the diet for less fatty or non-fat items. Replace butter with margarine or diet margarine. Replace beef with chicken and chicken with fish. Use low fat yoghurt instead of sour cream and other cream-based products.

●**Dining out** It seems obvious that few of us will ever see old age on a steady diet of fast food cheeseburgers, fried potatoes and milkshakes. The alternative is also much better in reducing stress. Patronise Italian restaurants and order a salad topped with vinegar plus an order of pasta with plain tomato marinara sauce. Simple and satisfying. The same can be said for ordering steamed chicken, vegetables and noodles in a Chinese restaurant. A little soy sauce and chinese mustard can be used to flavour the entrée.

The conclusion that one cannot avoid reaching after an experience like this, is that all these efforts can make life a much more enjoyable experience and a much longer one at that. What is so gratifying is that one begins to see and feel a difference relatively early on in such a life improvement programme. Also, though not everyone can change every category indicated above overnight, the rewards of even a modest programme of self-improvement will become evident in a short time. What is so puzzling, is that in a profession where we pride ourselves on how we take care of our equipment, we take so little care of ourselves. Many studio owners 'baby' their equipment, sometimes to the level of fetishistic behaviour. Yet these very same individuals smoke, drink and consume a daily diet of steaks and prime rib. It is obvious that they are thinking about how long their precious equipment might last but to them it is probably not an issue that their equipment is likely to significantly outlive its owners.□

Fostex 4020

An operational report on the Fostex 4020 event controller by Dave Foister
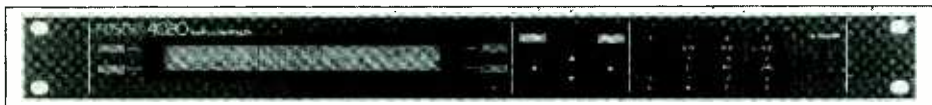
With the excitement surrounding the D-20 editor, visitors to the Fostex stand at this year's APRS exhibition in London may have overlooked another product tucked away in a far corner of the display. The Fostex 4020 event controller may not have the appeal of the new editor but it provides some thoroughly useful facilities and deserves a closer look.

The 4020 allows several types of event to be controlled by a cue list triggered at appropriate SMPTE times. It incorporates its own reader and generator, handling all timecode formats, and its programmable outputs comprise a MIDI port, two serial data ports, a tape transport controller and a set of eight general purpose switch contacts. The tape machine controller is a parallel interface that is immediately compatible with Fostex transports, and its output pulse format can be configured for other machines. The two serial ports are RS232, with one switchable to RS422, while the general purpose outputs consist of eight single-pole changeover switches, all connected via a 25-way D connector.

The operation of the unit is reasonably straightforward: it is controlled via a menu hierarchy on a large well-lit LCD window. The options on the screen are not always self-explanatory, necessitating frequent dips into the manual until one is familiar with the unit. Unfortunately, the manual is a particularly fine example of how not to present a manual. It is difficult to follow, poorly translated and lacks detail on many of the unit's functions. In particular, it completely fails to explain how to use the MIDI and RS232 outputs, although I eventually worked it out for myself.

MIDI output events are defined purely as the numeric value of the codes to be sent. A string of two-digit hex numbers is entered to form an event and, since this raw data includes the MIDI channel number information, it is not necessary to define a single MIDI channel for the unit. In fact a single event can address several devices on different channels, sending different messages to each. While this system offers powerful flexibility, its obvious drawback is that few of us carry in our heads a full list of MIDI codes, and the manual does not explain any of the above nor does it include a MIDI message table. I would suggest that this could be a worthwhile future addition.

If a sequence of MIDI codes is to be used frequently, it can be defined as a named command, like a macro, and then called by name

when required. Such a command can include any combination of types of event and can incorporate time offsets of up to 10 seconds between events.

The timecode points at which the programmed events are to take place can be entered manually via the keypad or captured on-the-fly while reading code. An events list is easily assembled, with the 4020 automatically numbering events in time order. Below the current timecode value as read off tape it will display the time and nature of the next programmed event.

While programming of the 4020 is straightforward enough, it can be frustratingly time-consuming because of the way the menus and selection buttons work. For instance, assigning an output event to a selected timecode point involves first finding the correct menu and then stepping through all the available options—the button only steps forward and only moves one step at a time so must be pressed repeatedly. To make matters worse, the option I imagine would be most often used—the user-defined commands—is the last item in the list.

Despite these grumbles—most of which I imagine could be addressed with a software upgrade—the 4020 is an extremely useful piece of kit, forming the missing link between all sorts of equipment. My own main use is simple but illustrates the potential. I use a D-20 for making cassette copies of live concert recordings. Live recordings are a nuisance to copy because of the problems of making a clean end at the point the cassette needs to be turned over. If the point is missed it is time consuming to find the correct point on the cassettes, erase the excess on the end and find the right place on the master to start side 2 and it causes excessive wear and tear on the machines. The 4020 allows me to find the turnover point in advance, at which point it will initiate a MIDI-triggered fade on a Drawmer M500 and 10 seconds later stop the D-20, leaving the cassettes to record silence to the end of the side ready to be turned over and restarted. The time and aggravation this saves can only be appreciated by those who have had to suffer this kind of work.

The control possibilities of the 4020 are enormous. I understand, for instance, that they were used on the recent Wall epic in Berlin to douse the projectors among other things. I would like to see more thought given to the user interface, and particularly the manual, but even as it stands I can see the 4020 finding its way into all sorts of facilities, solving problems from the mundane to the bizarre. ☐

www.americanradiohistory.com

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R & W's Third Edition of the 'Industry Bible'

The third edition of Raper & Wayman's Professional Audio Catalogue, which has become an audio industry standard reference work, is now available. People will notice a difference — the cover is now in glorious colour — but its logical, signal path format and comprehensive information on a wide range of audio products have been retained, and it has been expanded to 88 pages to include two new sections covering Audio Furniture and Problem Solvers. The Raper & Wayman price list is now updated on a quarterly basis and sent to all recipients of the catalogue.

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convention news

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What singles out the Venue as the best PA desk in its class?
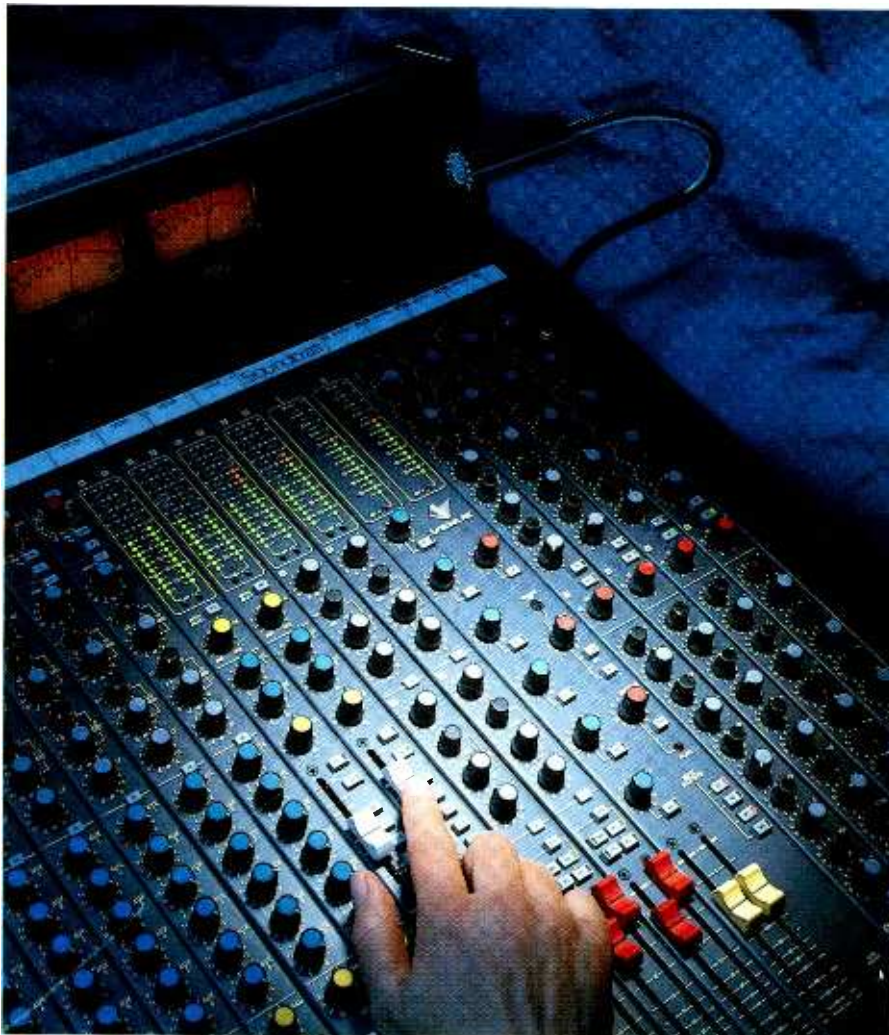
The Soundcraft Venue is certainly full of surprises.

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And together with the option of a powereful matrix module Venue's output flexibility is enhanced in even the most demanding situations.

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