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The picturesque setting of the Berlin AES Convention—see page 39

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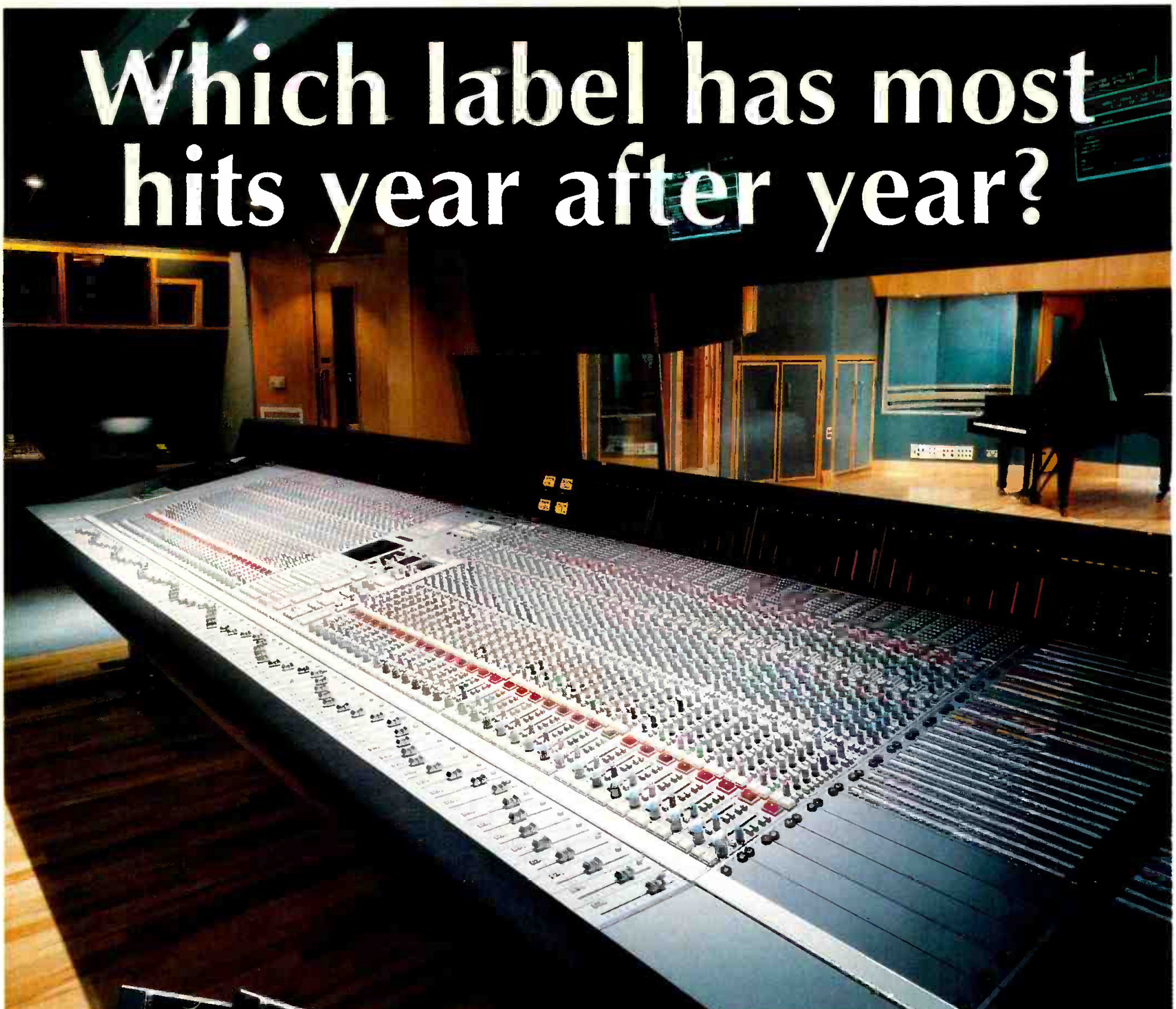

Photographed at Abbey Road Studios, London

Billboard

1992 STUDIO ACTION

PRODUCTION CREDITS FOR BILLBOARD'S No. 1 SINGLES

CATEGORY	Produced on SSL consoles*	Produced on ALL other consoles
DANCE	94%	6%
R&B	79%	21%
MODERN ROCK	79%	21%
ADULT CONTEMPORARY	77%	23%
HOT 100	71%	29%
RAP	59%	41%
ALBUM ROCK	59%	41%
COUNTRY	56%	44%

*Recorded and/or mixed on SSL consoles

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Off the Record

It was not until I had raised this topic with a variety of industry figures that I realised how diverse opinions on the subject actually are. One particular figure proclaimed—with due apologies for the predatory nature of business—that business carries but one obligation: that of making money. And it is with this opening comment that I would like to discuss the responsibilities of the major record companies.

Paying full price for a CD—in Japan or the UK at least—hurts. But whether you are buying the first uncut recording of Prokofiev's *Cantata for the 20th Anniversary of the October Revolution* on a prestige label like Chandos or The Prodigy's *Experience*, a full price CD is now a purchase most of us have to think twice about. And sometimes it is not even the first time we have paid for the music...

Since the popular acceptance of the compact disc format, the major record companies have been capitalising on their artists' back catalogues. The new format is effectively good enough to have replaced the vinyl LP—with a little help from the majors' marketing strategists of course. As a result, many music lovers are 'retiring' their vinyl collections in favour of CD. I am not about to suggest that it is unreasonable of the record companies to expect us to pay a second time for music we have already paid for once, but I would suggest that, with the profits that have come from the music biz over recent years, there comes a moral responsibility to reinvest in the artists and the industry in less prosperous times.

Drawing material from its back catalogue relieves a record company of the recording overheads necessarily incurred in making a new recording—and starves music recording studios of their business. Should there not be some acknowledgement of this economy in a CD's purchase price? Why is it necessary for the record companies to keep the price of rereleased recordings in line with the price of new recordings? Especially as those same companies—in certain territories—are finding it increasingly difficult to justify such high prices for new material? And on top of the present situation, we are about to be presented with the opportunity to buy the same material all over again in two new formats at top-line prices.

Certainly there is an overhead involved in introducing a new format, but whose expense should it be? Who is going to reap the financial benefits of the success of DCC or MiniDisc? Could it be the same parties who are presently making massive profits from the success of compact disc?

Looking at the state of the professional recording industry, it is fair to say that studios are suffering because of the current dearth of recording projects. Had the record majors sought to invigorate the record industry by using the profits generated by their back catalogues to subsidise new artists stimulating studio business along the way, their pricing would be thoroughly defensible. Alternatively, those lower overheads could have been instrumental in making CD a far more affordable medium as a means of stimulating interest in record sales. Surely it is in everyone's interest to restore the 'speculative' buying that used to be so commonplace in the heyday of the vinyl LP. The majors are contributing to a depression of record sales and a promotion of one of their greatest fears—piracy. Is it actually home taping that is killing music, or is it the circumstances that make it so attractive?

My fatalistic colleague would have me direct my arguments towards increased musical education. While I will happily acknowledge the value of education, I would still argue that a moral responsibility rests with the record companies—after all, if they stifle the music that facilitates their sales of software and hardware, where are they going to make tomorrow's massive profits? ■

Tim Goodyer

Cover: Amek & Doremi's *Dawn II* at Molinare, London. Photography: Nik Milner

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This is not an extravagant claim on our part, but a consensus of opinion from thousands of delighted owners and members of the press worldwide.

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"In terms of future reliability, ADAT's solid construction and attention to detail make a bold statement about the company's commitment to the pro market." *Mix Magazine*

"We did what any engineer worth his or her salt would do to a shiny new toy: we tried to break it – the transport never flinched."

"We can't find any reason not to love ADAT."

"ADAT works great, sounds amazing, and is priced right. That's good enough for us. We're buying in." *Electronic Musician*

"Affordable digital multitracking that doesn't require a mind-bending learning curve to access."

"This is a product that, when held up against virtually any musico-techno achievement of our era should – and undoubtedly will – hold its own, if not set a few new standards. We're sold, we're jazzed, and we're getting one (or two or three) for ourselves." *Keyboard Magazine*

"My review unit has passed every test I could throw at it."

"Tell you what. If you buy one and don't like it, give it to me, and I'll add it to the stack. I'll even give you album credit and pay the freight." *EQ*

"It is usually bad practice to bounce tracks and record the mix to an adjacent track – yet such problems do not concern ADAT at all."

"Playback seemed identical to the CD with no tape noise, no hiss, no hum . . . nothing but the pure sound." *Home & Studio Recording*

"In fact ADAT is probably easier to use than the average VCR."

"ADAT is good enough to be used in pro studios whilst being affordable enough to be found in the better-off home studio too."

"If you had any reservations about the recording medium, the quality of circuitry that Alesis would be providing for the price, or anything else, put them aside. This is no-compromise digital audio."

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TEC 1992 WINNER RECORDING PRODUCT OF THE YEAR

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AUTO 2 ▸ 1
PITCH ▼
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LOCATE 0
AUTO PLAY
FAST FWD
STOP
PLAY
RECORD
EJECT

In-Brief

● Test equipment speaks: Mike McMillan, an audio technician at the University of Reading, has very limited sight and is registered blind. He could not, in fact, see the display on his audio test equipment and was in danger of losing his job because of it. Paul Skirrow of Lindos Electronics has enabled Mr McMillan to keep his job and communicate with his system by enabling his Lindos *LA100* to talk to him. Des Fisher of NaTCH Engineering wrote the software that interfaces between the serial output of the LA100 and a PC; the program then controls voice system (Dolphin Systems' Apollo Voice Unit working with the Hal Screen Reading program). Lindos Electronics. Tel: +44 394 380307. NaTCH Engineering. Tel: +44 786 833541

● Frankfurt, Germany: In what is believed to be the first digital audio network of its kind in Europe, Studio Besser in Frankfurt has initiated a franchised network of studios to transmit audio voice-overs via an ISDN digital telephone network. Current studios who have joined the network includes facilities in Hamburg, Dusseldorf, Munich and Biarritz — with more to follow. Studio Besser selected apt-X from Belfast based Audio Processing Technology for the audio data compression system used.

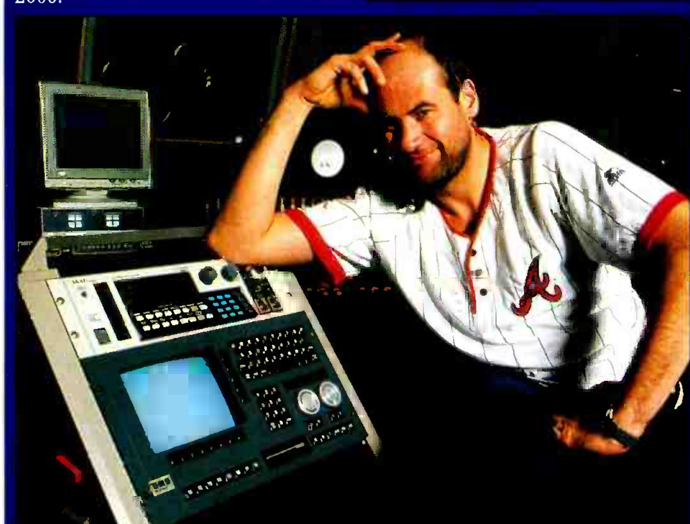
● Middlesex, UK: The Acoustics Laboratory of the Services Electrical Standards Centre (SESC) at Bromley in Kent has opened a new facility at its 'Aquila' site in which it aims to offer outside customers a fast, efficient and cost-effective service in sound level meter and microphone free-field calibration. A new anechoic chamber designed and built by IAC of Staines is to be used for the service. SESC Tel: 081 467 2600.

● Berkshire, UK: Producer Robin Millar has been selected Chairman of the British Record Producers. New Chairman Millar commented, 'My wish for the Record Producers Guild is that it will strengthen the bond and the understanding between those of us involved in the creative supervision of music making and those who have the responsibility of passing that music on to the public. The closer that bond, the healthier our industry will remain.

● European digital TV includes HDTV: The partners in VADIS, a pan-European project on digital TV, have agreed to expand the goals of the project to include digital HDTV. Having produced good results using the standard definition pictures, the project is now developing a multi-layer picture compression scheme which is matched to European requirements for a digital television system offering both standard and HDTV resolution pictures. For example this system could give the consumer the option of using a small portable receiver to decode standard definition TV or a larger fixed receiver to decode HDTV from the same broadcast. ■

ERRATA

A couple of mistakes currently being made by many—including our own Patrick Stapley in last month's 'Air Movements' — concern Lyndhurst Hall and DynaudioAcoustics' Andy Munro. Specifically, the monitoring system being installed in Air's new Studio 1 is a joint project between Air and DynaudioAcoustics not Air and Andy Munro. Also, the drivers loaded into to the system are manufactured by DynaudioAcoustics and not Dynaudio. Apologies for any misunderstanding. ■



Fish in his Funny Farm Studios with an AMS *Audiofile Plus* for compiling mixes for his latest album *Songs from the Mirror*

Artist Jamie Reid let loose again at Strongroom

Strongroom opens new studio

In its ongoing moves to corner the market in production suites, Strongroom Studios has built another three rooms which have again been snapped up by permanent clients.

In late 1991 three programming suites were built at Strongroom's East London premises for Rhythm King production team, The Beatmasters; John Coxon of Warner Music and Go! Discs. Studio owner Richard Boote spoke then of his plans to build more suites in the vacant warehouse next door and a year later that scheme has come to fruition.

The three new suites, like the others, are KFA's *The Box* systems, prefabricated modules built from 600mm wide panels. It did not take long to find interested clients. 'Because of the feedback from the other three rooms, we got people straightaway who wanted to take them,' says Studio Manager Siobhan Paine.

Two of the rooms have been leased by UK band The Outfield and producer Gareth Jones, currently working with Ugly, Glen Coulson of Heaven 17's new project. The third room which is self contained and includes its own office has been taken by production team Ian Curnow and Phil Harding, currently working with Espirity and Dannii Minogue. Their suite has been decorated by Jamie Reid who is responsible for the renowned vibrant graphics in Strongroom's studios 1 and 2.

All three new suites are larger than the first three which were an identical 4.8m x 3m x 2.3m high. Gareth Jones has the largest room which he has filled with his analogue keyboards and samplers. His room is also alone in having no vocal booth.

Curnow & Harding have vacated Strongroom's own programming suite, previously leased by The Beatmasters for two years before their own suite was built. However, before Strongroom could start advertising its now-vacant production facility it was rented through into early 1993 by Orbital. Go! Discs is now about to vacate its suite after a year, but Paine does not think it will be vacant for long.

Boote's plan to build a budget studio in the warehouse has been shelved until the market has settled down but he aims to go ahead with the building of a bar and restaurant facility. 'There are so many people working here that we need somewhere for them to relax and get together,' he says. He is also planning to build some offices in the warehouse because 'quite a few management and production people are interested in being based here.'

Another plan is to install an SSL console into Studio 2 replacing the Amek *Mozart*. 'A lot of people prefer working on SSL and it will complement the Neve in Studio 1,' says Boote. He is currently looking around for a second-hand SSL which he hopes to install by March.

Strongroom Studios.
Tel: 071 729 6165.
Fax: 071 739 1973.

Caroline Moss

Soundtracs spend on Spendor

Soundtracs plc, makers of mid market audio consoles to the studio and live market, have bought Spendor Audio Systems. Spendor Audio was formed in the late 1960s to manufacture studio monitor speakers for broadcast and hi-fi. The company will remain at their Hailsham base and will continue to operate independently of Soundtracs. However following the aquisition Mrs Dorothy Hughes, the founders wife, will be retiring as Director/Company Secretary and Todd Wells, Managing Director of Soundtracs plc, will be appointed as Chairman and Chief Executive of Spendor.

Soundtracs plc, 91 Ewell Road, Surbiton, Surrey. KT6 6AH.

Super Studio boom!

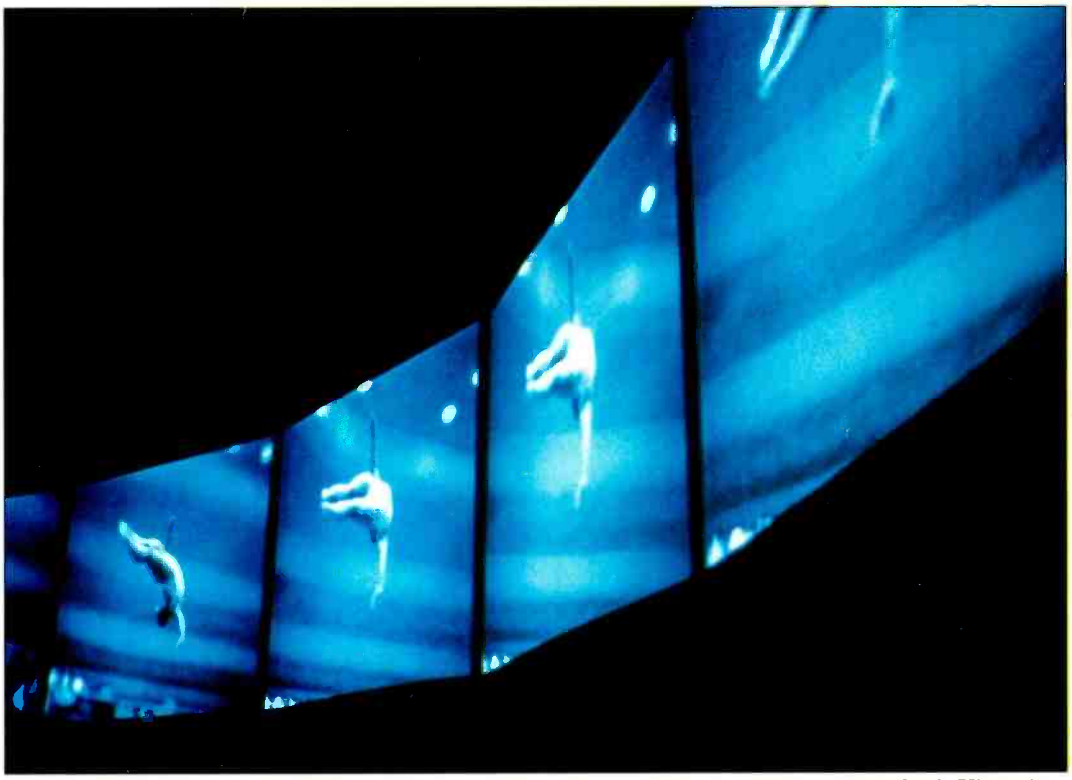
In the spring and summer of this year a number of large recording studio complexes will be opening around the world suggesting a further distancing of major studios from the rest of the commercial music studio market.

In March The Hit Factory Digital will officially open in New York City in a six-storey, nine-studio, 100,000ft² recording facility designed by Harris Grant Associates with studios equipped for recording, mixing, Dolby Surround Sound, audio-for-video work and mastering. The complex will feature one of the world's largest desks in SSL 96-input *G-Series* with 112 Ultimation faders and both *E-Series* and *G-Series* EQ. There will be nine artists' lounges, a 5000ft² restaurant and 24-hour gym facilities, including sauna and steam room.

Also in March AIR Lyndhurst opens in London with a five-studio complex equipped for a full service approach and costing nearly £12 million. Summer sees the opening of Galaxy Studios at Mol in the Flemish part of Belgium with a three-studio, four-control-room, complex. At its heart a 330m² main hall for orchestral recording and a control room featuring the AMS-Neve *Capricorn* all digital console with Sony 48-track digital recorder. Monitoring is by Genelec *1035*s and the design is from David Hawkins of Eastlake Audio.

Meanwhile Power Station International—New York Studio Power Station's worldwide network of recording studios—have established a base in France. Owner of Versailles Station studios in Paris, Philippe Besombes has been appointed French representative for the organisation which fully intends to build a Power Station France in the capital 'within a year' according to Power Station International President Nick Balsamo. Similar Power Station operations are being planned for Tel Aviv in Israel and Tokyo in Japan.

'This is not a franchise. It's for the purpose of setting up a network of new sources of talent, music and resources to make bigger and better music' said Balsamo. 'The prime goal now is to service French talent that wants to come to the US to record and produce'.



The imposing five-screen display of Virtual Vision demonstrated recently at London's Victoria & Albert Museum. Intended for presentation applications rather than broadcast, Virtual Vision sources its video from five Sony CRV recordable laser disc machines and uses a 68030-based computer platform to allow complex manipulation of picture sequences. Virtual Vision allows 125 events per second, non-destructive editing of video material and also supports a 4D sound field, an integral MIDI sequencer and a variety of sync options (including SMPTE and MTC). Virtual Vision, 7-11 Kensington High Street, London W8 5NP. Tel: 071-937 9801

Stone Court's 5-star service

The recording studio built for Jackson Five-esque UK act Five Star in Berkshire in 1989 has been reopened and renamed by new owner Joe Cooper. Stone Court studio has been refitted along with the adjoining cottage to become fully residential. New equipment includes a Trident *TSM* console, Otari *MTR90 MkII* and Urei monitors—design has been updated by original designer Neil Grant and Brian Haywood from Trident is helping with technical backup.

Stone Court offers tight security, tennis court, gymnasium and steam room and is available for bookings now on +44 0 344 28891.

New training group targets radio

On the 28th January Skillset, the new UK industry training organisation for broadcast, film and video, was launched at the headquarters of Channel Four Television. The launch, by Channel Four's Michael Grade, is a culmination of two years research into the training needs of the industry. Courses based on the Skillsearch called NVQs (National Vocation Qualification) will start at the end of year. In the meantime training surveys will continue, including one slotted for radio.

Skillset seeks to remedy the situation where most people working in a particular part of the industry is freelance and possibly undertrained for the job they are doing. The organisation was infact set up in reply to vast increases of freelancing throughout the eighties. Money raised by Skillset, nearly £1million so far, will be used to subsidise courses for freelancers. However the organisation is keen to point out that they in no way intend to replace the main broadcasters as the main trainers.

Channel Four Chief Executive Michael Grade commented, 'We must always remember that the production base underpins broadcasting in the UK and it's our responsibility that Skillset succeeds.'

Skillset, 60 Charlotte Street, London W1P 2AX. Tel: 071 927 8585

International Exhibitions

● **SCIF Sound '93.** Sandown Exhibition Centre, Esher, Surrey, UK. February 16th and 17th. ● **Musik Messe.** Frankfurt, Germany.March 3rd–7th ● **AES, 94th Convention.** ICC, Berlin, Germany. March 16th–19th ● **NAB '93.** Las Vegas, USA. Featuring **Multimedia World.** 19th–22nd April. ● **MIDI Music show.** Wembley Conference Centre, London, UK. April 23rd–25th. **SATIS.** Parc des Expositions, Paris. Featuring an audio section. May 25th–28th ● **Broadcast ASIA.** New World Trade Centre, Singapore. June 1st–4th. ● **ITS.** Montreux, Switzerland. June 11th–15th. ● **Multimedia '93.** Earls Court 2, London. June 15th–17th. ● **NAMM.** Chicago, USA. June 18th–20th. ● **APRS.** Olympia 2, London. June 23rd–25th. ● **Pro Audio & Light '93.** New World Trade Centre, Singapore. July 7th–9th. ● **British Music Fair.** Olympia 2, London. July 25th–27th. ● **PLASA.** Earls Court 2, London. Sept 12th–15th. ● **Vision'93, Film and Video Equipment Show.** Olympia, London. October 5th–7th. ● **Photokina.** Cologne, Germany. Sept 14th–20th.● **AES, 95th Convention.** Jacobs Javits Convention Centre, New York, USA. October 7th–10th. ● **SBES.** Metropole Hotel, Birmingham, UK. 4th November.

A CUTTING EDGE

To be, or not to be, that is the question : Whether 'tis *nobler in the* mind to suffer The slings and arrows of **A HORSE, A HORSE, MY KINGDOM FOR** outrageous fortune,

Or to take arms against a sea of troubles,

And by *opposing,* end them

To die,

to sleep.......

THE CUTTING EDGE

To be, or not to be, that is the question: Whether 'tis nobler in the mind to suffer The slings and arrows of outrageous fortune, Or to take arms against a sea of troubles, And by opposing, end them. To die, to sleep- No more, and by a sleep to say we end The heart-ache and the thousand natural shocks That flesh is heir to;......

...... perchance to dream

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BRITISH INNOVATION

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The video post box

The latest digital workstation to hit the market is *AudioVision* from Avid, the US manufacturer of the nonlinear video editor *Media Composer*. This link to the world of video proves to be extremely significant, as *AudioVision* and *Media Composer* are completely compatible, allowing *AudioVision* to display video sequences as a window within one of its two monitor screens. To understand the real potential of *AudioVision*, it is useful to summarise the power *Media Composer* has brought to video editors.

Media Composer works by digitising video sequences, to which the editor then has random access, as on a tapeless audio workstation. This eliminates rewind times and because the system is computer-based (Apple *Macintosh*), the editor can create as many variations of the edits as desired, with no destructive effect on the source material or earlier versions of the compilation.

The resultant Edit Decision List is then taken into the 'on-line' video suite and used to auto-conform the final programme from the original video stock. (It would in theory be possible to output direct from *Media Composer* but the quality of the digitised pictures is not of broadcast standard.) Suffice to say, time spent on *Media Composer* costs a mere fraction of time in a fully equipped on-line suite.

So what does this mean to the audio engineer in postproduction with *AudioVision*? Because *AudioVision* will accept *Media Composer* files, it means that there is no need for an external VTR or even an additional video monitor. Like the video editor, the audio engineer is freed from video rewind times and can also use the Edit Decision List from *Media Composer* to conform audio. This exchange is a two-way process, in that transferring a soundtrack from *AudioVision* to *Media Composer* requires no more than carrying the removable hard disk from one station to another.

All Avid systems are currently based on the Apple *Quadra 950*, the top end *Mac* introduced last year. *Audio Vision*, like *Media Composer*, uses two video screens. One of these is primarily concerned with source material, storage and level control. It is here that 'workreels' are kept and available effects are listed. Also on this screen is a simple set of controls for the video picture and a set of level controls with accompanying bar graphs for eight audio tracks.

A screen-shot from *AudioVision*

AudioVision is actually available in two versions, with four or eight simultaneous I-O. In either case, the internal architecture has 24 'tracks' and these are displayed on the second monitor, along with the video window. This is the screen where most of the work takes place.

A total of eight tracks are displayed at any one time and this is also the maximum number that can be played simultaneously on an eight I-O system. However, any 'track' can be called up on any of the eight track locations, meaning that there is complete freedom over which of the 24 tracks is played and to which output.

Associated with this part of the system are record and mute functions, as well as the ability to lock a track so that it can not be advanced or retarded in time. As an aid to retaining an overview of the full 24 'tracks', the bottom of the screen has a simplified display showing all sound segments for the complete duration of production. This obviously means that longer productions cram more information into the same display but there is a zoom function.

Audio sections can also be displayed as a waveform when editing. Just above 'now line' for the eight tracks is a time-code reader showing where 'now' actually is and to the left of this are a number of function buttons concerned with editing and manipulation. To the right of the video window are deck controls, with familiar transport-type functions and a section for marking in and out points. To the left are the settings and displays for overall parameters, such as sampling rate, which can be 44.1kHz or 48kHz.

Crossfades are currently linear but cover a usable range of time. Looping can be used for application such as creating endless atmos from a short section of ambience. It is worth recalling that *AudioVision* integrates digital video and audio, which means the two can be looped while comparing alternative dialogue tracks, for example.

The most experienced European user of *AudioVision* is Flemming Christiansen of Nordisk Film in Denmark, who has been testing successive versions of software since last September, just weeks after the system was shown as a prototype at IBC in Amsterdam. Primarily an audio engineer, Christiansen also has experience of video editing using Avid's *Media Composer*.

Speaking from an audio posting room at Nordisk, where the equipment includes a TAC *Matchless* analogue desk and the digital Yamaha *DMC1000* mixer, he said that the integration between *Media Composer* and *AudioVision* was its major advantage. In addition to assisting with the refinement of the software itself, Christiansen had been experimenting with various ways of using *AudioVision* in the postproduction process. Nordisk is one of the oldest film studios in the world and as such covers everything from features to commercials and television programmes. Although Christiansen worked a lot with film, *AudioVision* could be 'even more perfect' for video work, where eight-track simultaneous play out normally proved to be 'more than enough'. Alternatively, he could lay the tracks off to multitrack tape in blocks of eight, using time code synchronisation to allow successive passes.

In theory, he allowed, it would be possible to use the *Mac*-based management system for the Yamaha *DMC1000* to create automated premixes but he did not regard this as a realistic proposition from one computer, as any programme running in background could make the main application slow up. Christiansen said he had experienced no timing problems with *AudioVision*. When asked to do the impossible it dropped a sound cue, rather than lose sync.

An example was firing cues direct from the system's optical disc. Officially, *AudioVision* does not do this at all, as the optical drive is too slow. Christiansen said he had been able to take samples from the optical disc but occasionally one would drop out. Any system that allows the operator to do things that are not part of the scheme but does not crash either, should encourage confidence in its reliability. At the time of writing he had not attempted to defragment the hard disk, a process of reordering files that has been known to bring some systems to a grinding halt. However, *AudioVision* is based on *Media Composer* and that means Avid have some years experience of delivering systems that can manipulate video files, a far more daunting challenge than audio. Provisional UK prices for the system are four I-O £45,000, eight I-O £65,000 for a complete system with hardware, which includes two high resolution monitors and a specially colour-coded QWERTY keyboard for faster working.

The first systems are going to existing *Media Composer* owners but full availability is timed to coincide with NAB. ■ **Simon Croft**

Avid Inc, Metropolitan Tech Park, 1 Park West, Tewksbury, MA 01876. Tel: +1 508 640 6789. Fax : +1 508 640 1366.
UK: Avid Technology, 20–28 Kingly Court, London. W1R 5LE .
Tel: 071 434 0122. Fax: 071 434 0560.

Tri-Power condensers— studio quality on-the-road

Tri-Power Condenser Mics

AKG have introduced two new condenser microphones designed specifically for the requirements of live performance. Part of the Tri-Power series, the new *C5900* hand-held vocal mic and *C5600* instrument mic combine AKG condenser technology with the Tri-Power line. According to David Roudebush, Corporate Marketing Manager for AKG, 'AKG pioneered condenser microphone technology and is the leader with a range of condenser mics extending from our multi-thousand-dollar large-diaphragm studio models to the specialised micro-mics which give musicians studio quality wherever they are. With the *C5900* and *C5600*, we've given performers true AKG condenser sound for use in the toughest stage environments.'

The *C5900* vocal mic features AKG's new *TPC-I* condenser system with claimed exceptionally smooth off-axis frequency response, a hypercardioid pick-up pattern for high output levels before feedback, and three different switch-selectable bass contour curves. Shock isolation is provided by the AKG *InterSpider* suspension system which effectively takes the large external spider-cage shock-mount assembly used in recording studios and puts it into the microphone itself for exceptionally low handling noise on stage.

The *C5600* features the new, large diaphragm *TPC-II* condenser system and the same *InterSpider* built-in spider shock isolation assembly introduced with the model *C5900*, and comes with the stand-adaptor built into the RoadTough housing. The three different switch-selectable bass contour curves permit adjustment for proximity effect and individual instrument characteristics.

AKG Acoustics Inc, 1525 Alvarado Street, San Leandro, CA 94577, USA. Tel: +1 510 351-3500. Fax: +1 510 351 0500.
UK: AKG Acoustics, Vienna Court, Lammas Road, Godalming, Surrey. GU7 1JG, UK. Tel: +44 483 425702. Fax: +44 483 428967.

Junior Chameleon

The original Chameleon *2200S* power amplifier has become well known for providing high power from a single 1U package. Some customers have apparently commented that it is too long to fit into a standard rack, and that it is simply too powerful for some applications.

In answer to these comments and as a natural development from the original Chameleon, Malcolm Hill Associates have now introduced the Chameleon *1400S*, rated at 700W/ch in to 4Ω, and 525W/ch in to 4Ω continuous. The *1400S* is only 16 inches deep while being 1U high.

The Studio, Pro Audio Sales and Marketing, 13–16 Embankment Gardens, London. SW3 4LW. Tel: +44 71 352 8100. Fax: +44 71 351 0396.

G Series Update

The latest addition to the range of *G Series* computer software, the *G3.2*, brings a number of operational benefits of *Ultimation* to *G Series* console users for the first time, while offering full mix compatibility with *Ultimation*. New operational features include: ● 15 software groups—any channel fader can be a master to any other fader(s) ● Inverse cut—slave channels cut when master is uncut, allowing A-B switching. ● Off-line cuts—cut events can be created and stored without having to run tape. ● Insert mixing—allowing moves to be inserted into the current mix without destroying subsequent moves. ● Faders can now be locked into different update status cycles. ● Auto-takeover is now available in a New Mix. ● Safe replay status. ● The Computer now handles non-integer tach rates.

Solid State Logic, Begbroke, Oxford, UK. OX5 1RU. Tel: +44 865 842 300. Fax: +44 865 842 118.
USA: Solid State Logic Inc. Tel: +1 212 246 4290.

Hexacoil *BM10*

In direct response to the demands of broadcasters and smaller studios, DynaudioAcoustics have launched a new low cost nearfield monitor, the *BM10*.

The *BM10* is a two-way passive design with a single rear-facing reflex port to optimise low frequency output. The 7-inch bass driver uses a magnesium-silicate impregnated polypropylene cone coupled with an aluminium 'Hexacoil' voice-coil and former, to provide a rigid but lightweight assembly. Coupled with a single ABES Bass Extension System and suitable amplification, the *BM10* claims continuous SPLs in excess of 118dB from 35Hz–20kHz.

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Elkie Brooks on tour until April

ON TOUR

A round-up of the major highlights of this month's live sound reinforcement scene.

Looking for the highlights among this wintry month's live sound reinforcement scene, although confirmed contracts were inevitably scarce, turned out to be rather easier than expected.

Refreshingly, the mood in the UK PA hire industry is almost unanimously one of optimism for the year ahead. Many operators predict a particularly busy spring and summer season, although it seems likely that there will be fewer stadium acts than on the circuit than last year. Another telling indicator of the times: looking at any PA company's list of 'quoting' tours one has a hard job spotting artists aged under 25. With the recent rash of record company 'rationalisations' in favour of performance-oriented acts, there's more than a slight possibility that this will change next year, since many feel it is touring, rather than skill with a *Portastudio*, that builds a lasting audience for an up-and-coming band.

● **Audiolease** have the Zuccero tour from January to March 8th, the majority of which is in Europe but with one London date—the House Engineer being Jim Ebden. Julian Cope's tour recently finished with AudioLease partner Tim Sunderland at the FOH controls. PA was AudioLease's new proprietary *A2* system. Jesus Jones are now out in Europe with engineer Stuart Kerrison, while Bad Company's European dates also feature the *A2* system, with FOH Engineer Bill Fertig.

● **Blue Box** in Sussex, with their Turbosound systems, are no strangers to the cut-and-thrust of the rave scene, and Director Mark Metcalfe confirms that currently, most of his confirmed jobs on the books are indeed raves.

● **Britannia Row Productions**, in common with most companies, said they were 'exceptionally busy' in January providing quotes on forthcoming tours and shows. Among their end of 1992 highlights had been Madness's televised shows from Wembley Arena, mixed by Jon Lemon. At the tail end of January, their principle confirmed contract for February was Sade's world tour starting mid February in Los Angeles. With no US PA company as yet operating a Turbosound *Flashlight* system, BRP are shipping one from London for her tour. Also booked for mid February is the round-stage PA for Simply Red's six Birmingham NEC dates, including the two rescheduled from December after vocalist Mick Hucknall's laryngitis.

● **Canegreen** have Richard Clayderman in the UK, Van Morrison's ongoing European dates, Go West and The Stranglers. They are also hiring EAW stacks and racks to Theatre Projects for an Opel car launch in Barcelona (see below). Said Yan Stile: 'We've just bought a Yamaha *PM4000* which is going out with Van Morrison and Go West. Generally, the year ahead is looking great for us.'

● **Capital Sound Hire:** The B-52s are in Europe and England with Capital-supplied monitor and control systems and Malcolm Hill Associates' FOH speakers. Later in February and March there is Fish, starting at Glasgow Barrowlands, out with a Martin *F2* system, as well as Dinosaur Jr, again with an *F2* PA.

● **Concert Sound:** Like last year, Concert Sound's major booking is Eric Clapton whose 12 Blues nights at the Royal Albert Hall conclude on March 7th. Their EAW system is mixed by Mike Ponczek with Kerry Lewis on monitors. Also on the books: Elkie Brooks (UK, February to April), combining her legendary purple 'SLI' system (see pictures) mixed by husband Trevor Jordan, with desks, monitors, amps and sub-bass possibly from Concert Sound. The Ramones (February to March, Europe) with John Markovich and brother Peter at the controls. CS also have Paul McCarney's production rehearsals. Showco are the main tour contractor, while Concert Sound will supply front of house Midas *XL-3*s and House Engineer Paul Boothroyd. Finally, there's Brian May rehearsals with a monitor system and *XL-3* desk.

The purple *SLI* system

● **Encore Entertainments** have three troupes of The Chippendales —two of which are in Europe—and are currently looking after the Town & Country Club's house system.

● **ENTEC:** Steve 'Bunty' King, Sound Hire Manager, said his Martin systems are booked for TV's *The Word* until March 26th; there was also a show with Jimmy Nail. He confirmed, too, that Entec have already been asked to quote for a 'forthcoming' event—slated for Christmas '93.

● **Sound & Light Production** have Roger Whittaker out on a five-week European tour and presentations including, says John Denbigh, 'various sound services' for BBC Radio 2, and a supply of specialist microphones for Elvis Costello's tour with the Brodsky Quartet (which is using house PA's along the way). Denbigh adds he is, 'Very happy with our *EV MT-2* and new Deltamax systems—they make a great combination.'

● **SSE Hire** have Simply Red's six dates at the NEC over the last weekend in February, The Shamen's tour, the Jeff Healey Band and more WWF matches. Finally, although dates had not been decided as we went to press, Mike Oldfield's tour of *Tubular Bells II* was officially confirmed.

● **Theatre Projects Sound & Vision** finished their UK tour with *Annie Get your Gun* on January 10th, the same team (headed by top Theatre Sound Designer Rick Clarke) moving swiftly on to another 'on the road' musical, *The Sound of Music* —touring for the second time after last year's successful run. TPS&V also has a large number of Sennheiser radio microphone systems on hire to *Carousel* at the National Theatre. Lastly, there is a major car launch for Opel and 'satellite' meetings in Barcelona, through production company HP:ICM. Mick Wicks is Production Manager, Dick Sinclair the Sound Designer and Operator. Equipment is EV *Deltamax*, a *PM3000* desk and an EAW PA subhired from Canegreen.

● **Wigwam** have one of the biggest winter tours this year with Chris Rea's February and March dates, using a Meyer *MSL3-DS2-650* house system. One stage are *UPA-1*s, Nexo *TS2400* fills and Martin *LED-400* active wedges. The FOH PA is configured, says Chris Hill, as a 'Trod' system—a three-point PA with a stereo main stage system plus rear-centre cluster for surround-sound effects. Mark Kennedy engineers with John Shearman on monitors. Consoles are a Midas *XL3* on stage and a Yamaha *PM3000* plus a *PM4000* House console (46 mono, 6 stereo Ins), the desk's UK touring debut. Says Hill: 'It is wonderful—it's quieter. . . the EQ is very positive. Ergonomically everything seems to fit well.' The *PM4000M* monitor version enters production in April. ■

Live Sound is relayed by Mike Lethby

THE PROBLEMS OF STEREO BROADCASTING ARE WORTH LISTENING TO

While an ever increasing number of television programmes are transmitted in stereo, the vast majority of current viewers listen to mono audio - this can cause a number of problems for broadcasters.

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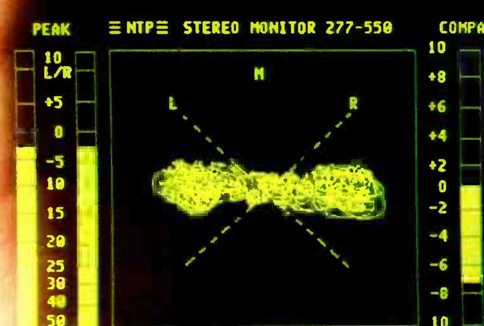

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Peavey *PC1600*—overdue and highly recommended

Peavey *PC1600*

Dedicated MIDI controllers and generators are relatively rare pieces of kit despite today's preponderance of MIDI sound modules, effects units and cheap retrofit automation systems. Perhaps because a unit like the Peavey *PC1600* does not make any noise of its own, it is often disregarded as an essential item in a MIDI setup especially when the majority of computer-based sequencers now offer the sort of MIDI generating possibilities that the *PC1600*'s existence is founded on. The difference, of course, is that Peavey's unit is presented as hardware which means significant benefits over a one-thing-at-a-time mouse approach.

The *PC1600* has 16 60mm faders, 16 buttons arranged conveniently under the faders, a data wheel, 2 x 20-character LCD with variable viewing angle, and a clump of buttons associated with the cursor movement and page scrolling required to operate the device. Two footswitch sockets can be patched to generate data externally which is then soft-routed to a MIDI function with MIDI In and Out sockets completing the picture.

A word about how the *PC1600* is organised; each of the faders, buttons, footswitch sockets and the data wheel can be assigned to a MIDI data-generating task and any one complete set of assignments can be saved in 50 presets. Additionally, the device has 100 snapshots or scenes. These are created by adjusting fader and button statuses within a preset and saving this configuration as a scene. The beauty of this arrangement is that scenes can be sent irrespective of the current 'dynamic' preset selected. This means that a preset of 16 MIDI volumes and mutes, for example, still permits access to the 100 scenes in real time. Thus you can mix MIDI volume and fire scenes of unrelated SysEx information simultaneously. This amounts to a whole lot of control in a studio or live situation and the feel and accessibility of the unit makes it easy to harness this control. The variety of function available through the device's controllers is likely to endear it to a lot of people.

A fader can be assigned to any channel continuous controller, can be disabled, can act as a master fader (with multiple groups available but no pyramiding) or it can generate a MIDI string with the fader position value inserted in it. A button can be disabled, assigned to mute a fader, solo a fader, send a program change command, send a note from C1 to G9 on-off command with programmable velocity, generate a MIDI string on a single push, generate one MIDI string when depressed and another when released or toggle between two strings on subsequent presses. And anything a button can be assigned to do can be mimicked by a footswitch, permitting hands-free starting of a sequencer, for example.

The data wheel can be assigned to control a fader, external CV input or given a floating status whereby its effect is restricted to the fader that was moved last. There was an opportunity to give the dial independence to generate data not already served by available controllers but it has not been realised.

Creating a preset configuration of controllers by editing is simple and is facilitated by moving the controller required to draw the attention of the parameter editing menus to its existence. Copy functions, from controller to controller as well as from preset to preset, make rudimentary setup quick. While the faders are smooth and feel as though they will stay that way, the buttons are, perhaps, a little too recessed but do click reassuringly. The CURSOR and EDIT, COPY, ENTER, UTILITY, SCENE and EXIT buttons interact with the LCD slightly sluggishly when editing and caused this impatient reviewer to double hit when one strike would have sufficed. It is a shame that presets cannot be scrolled through with the dial, requiring UP-DOWN button presses, while the scenes can.

Once a fader is moved the LCD shows its number and its value in real time and continues to show that value until another controller is activated. Button status is also displayed and thoughtfully a major catastrophe is averted when using these as mutes because a gentle press on a button confirms its status on the display while a harder push will activate it.

The *PC1600* can map program changes to its named presets and includes a comprehensive MIDI filter which comes into play when using the device to process an incoming stream of data. To this end the unit can replace or merge incoming data with that generated from its own individually selected faders and an approximation of a punch-in in Replace mode is instigated as soon as a fader is moved and is punched-out by pressing the EXIT button.

The greatest omission from this process is the ability to null a fader to incoming continuous controller data easily. Data can be replaced and merged with ease but it becomes difficult to rewrite sections of data smoothly purely from the information that the unit gives out. This is not a major drawback, especially as a sequencer could be used to smooth over any rough transitions, but it is worth mentioning as nulling is one of the features of the JL Cooper *Fader Master*.

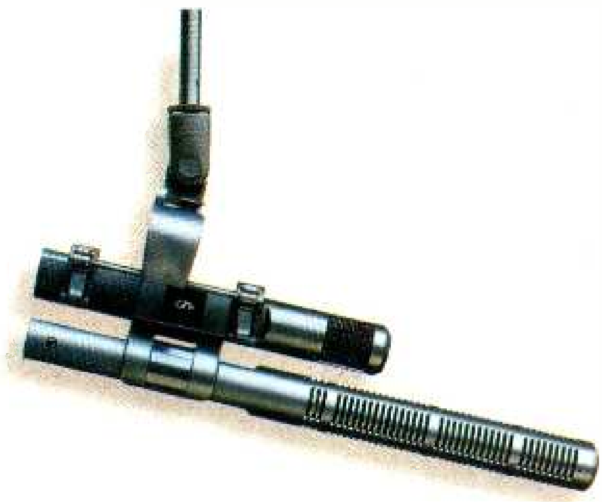
The power available through this rather innocuous-looking, sturdy but slim, wedge is quite simply awesome. At any point in time you have access to a minimum of 32 MIDI controllers available plus 100 totally different snapshots at your finger tips. This means that a very sizeable rig could be controlled comfortably. From reverb unit decays and predelays, to module volumes and brightnesses, from sequencer starting and sample firing, to MIDI control of bolt on automation packages—to name just a few basic applications. I would defy anybody to achieve similarly accessible results with any sequencer manager page.

This device is long overdue but, perhaps, it is only now that the market is ready to exploit all the facets of MIDI control that manufacturers have been including in their products for years. The *PC1600* broadens the possibilities offered by any piece of kit that has anything more than very rudimentary MIDI spec. Aside from moving faders and a stack of LEDs, which would make it ridiculously expensive as opposed to ridiculously cheap for what it is, this unit is perfect. Highly recommended. ■

Peavey, 711 A Street, Meridian MS 39302-2898, USA.
Tel: +1 601 483 5365.
UK: Peavey Electronics UK, Hatton House, Hunters Road, Corby, Northants NN17 1JE.
Tel: 0536 205520.

Music News is compiled by Zenon Schoepe

One stereotype you can't ignore.



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Room with a view—Syn Studio in Tokyo

PERFECT LOGIC

Some 12 months after its introduction, the AMS-Neve *Logic 2* is still evolving. Patrick Stapley reports on the state of the digital art

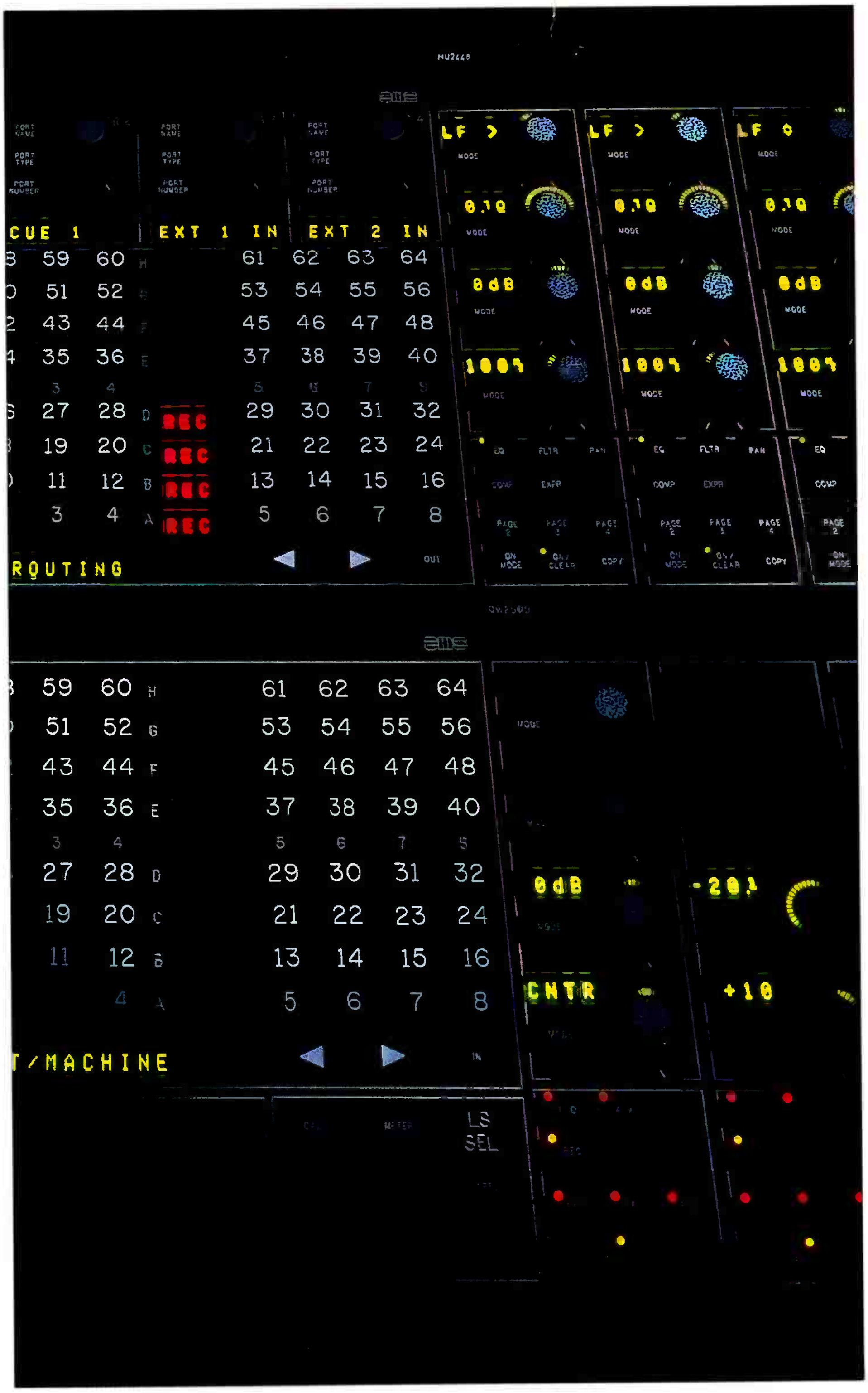
The AMS-Neve *Logic* series of digital consoles started development shortly after the merger with neighbouring Yorkshire company Calrec Audio in 1986. Calrec had already started R&D work on a digital mixer and consequently the merger produced a highly qualified team of engineers with expertise both in mixing console design and digital audio.

From the very beginning it was decided to develop a family of digital desks ranging from the small stand-alone *Edit 1* to a full-size multitrack music recording console. The transputer technology that Calrec had been experimenting with lent itself very well to this; transputers due to their parallel processing capabilities allow system expansion without slowing processing speed.

Logic 1 was designed to complement *AudioFile* as an all digital console with full dynamic automation on all functions; it was also to have a compact control surface—the plan was to get a bigger console into a smaller box. Assignability was the key, but this had to be treated very carefully; people still wanted to operate a console that was recognisable as such and they certainly did not want a mixer controlled solely from a computer screen or a console comprising of a single, assignable channel strip. A happy medium had to be struck between familiar, traditional working methods and the enormous possibilities offered by digital technology. The first *Logic 2* consoles were built in 1991. These were bigger desks with additional features, as AMS-Neve Product Manager Doug Ford explains.

'We'd always supposed there would be a *Logic* console with more than 15 faders which is the total on a *Logic 1*, and that's how the *Logic 2* arose. Two things have really happened with *Logic 2*, one is that we've added some extra controls to make it suitable particularly for multitrack work, and we've put in additional panels to make it suitable for film type monitoring and additional routing to address more buses. What we've found is that some clients are buying a *Logic 2* instead of a *Logic 1* not because they particularly require a lot of signal handling but because they prefer a less assignable control surface. Yorkshire Television, for example, have a *Logic 2*, and although they admit that they could do all the work on a *Logic 1*, they prefer to have more channels appearing on the physical surface rather than picking them up on paths.'

To date 40 *Logic* consoles have been sold splitting evenly between *Logic 1*s and *2*s. Application-wise they have a wide spectrum of use—radio, postproduction, music recording, film mixing, and there are even two installed now in opera-houses. The programmable nature of the console allows it to be tailored for specific jobs, as well as to cater for individual requirements. ▶



***Logic 2* panel detail**

This is exemplified at Air Studios where a *Logic 2* is currently being used to record and mix an album, and a second is being used for postproduction dubbing work.

An important distinction between *Logic 1* and *2* is that *Logic 1* is an integrated edit and mix system by virtue of having an *AudioFile* built-in. *Logic 2* on the other hand does not necessarily have to include the hard disk component although an *AudioFile* control surface is provided as a data terminal to set up the console and run automation. However, the *Logic 2* can be supplied with a fully functional *AudioFile* package, and so far this has proved a popular option.

'It's interesting because every *Logic 2* console we've sold has been supplied with *AudioFile*,' says Doug Ford. 'When Air ordered their first *Logic 2* for one of the remix rooms at Lyndhurst Hall, they didn't want *AudioFile* included but we actually delivered the console for convenience with one installed intending to remove it later. Air are now keeping it because they've found it so useful for things like spin-ins, track compiling, creative mix editing and so on.'

The console

The *Logic 2* system consists of three parts—the console control surface, the I-O interfaces, and the signal processing system.

The standard console provides 28 channel strips each of which can fully control four separate signal paths either in mono or stereo—thus a 28-channel console is capable of offering a total of 112 stereo channels. The standard frame measures just under two metres wide and can be increased in units of eight channels strips—the strips themselves can be increased in blocks of four up to a maximum of 70 channels. However, larger configurations can be supplied for multi-operator systems. Because the processing racks and I-O interfaces are sited remotely, the console itself is unencumbered with cables (only one control cable connects to main frame), remains relatively lightweight and outputs very little heat.

The control surface is made up from three components—the channel strips, the *AudioFile* terminal and a master section. Channel strips are physically arranged into blocks of four, and each block is sectionalised into five modular panels. Furthest from the operator are bargraphs, dynamics meters and alpha displays; followed by EQ and dynamics controls; then I-O, pan and aux controls; next are the path select, automation, mute and solo controls; and at the base are the linear motorised faders. The panels containing EQ and auxiliaries can swap position in the desk to suit operator preference.

These EQ and auxiliary panels both feature AMS-Neve's patented Logicator controls (eight per channel), which are continuous shaft encoders with a ring of light display built into the head of the knob. Light is transmitted from a circular arrangement of LEDs below the control via a network of fibre-optic filaments to translucent segments around the rim of the knob. The display can be used in many different ways to match the function of the control, and has the added advantage of taking up no extra space and is not obscured by other controls. Each Logicator also has an alphanumeric display window to the side which provides confirmation of function or parameter value.

The actual function of the Logicators is dictated by a series of buttons below each set of four, which assign control. In the case of the EQ panel the Logicators can be switched between EQ, Filter, Compressor-limiter, and Expander-gate. Once a function has been selected up to four pages of control can be accessed. So for EQ the Page buttons act as select keys for the four bands (each 12Hz–20kHz) and assign the four Logicators to control frequency, boost-cut (±24dB), Q (0.1 to 10), and bell-shelf (six settings including high, low, and notch filters). An alternative arrangement is selectable whereby a common function is assigned to Logicators—for example, Page 1 will assign boost-cut for the four bands, Page 2 frequency selection and so on.

Other buttons switch the selected processing in-out of the channel, reset the processing to a 'line-up' condition (a preset start-of-day setting), and copy across settings from one strip to another.

The function buttons use a system of integral tricolour LEDs:

Off—function is deselected from controls and is switched out of the channel.

Green—function is selected to controls but is switched out of the channel.

Amber—function is selected to controls and is switched into channel.

Red—function is deselected from controls but is switched into channel.

The tricolour system provides instant feedback of processing status without having to recall displays. It also is an efficient way of maximising communications with minimum use of space and is used in different ways throughout the console; it does, however, present limitation to colour ▶

blind operators.

The Filter function provides high and low-pass filters with a 12Hz–20kHz range operating over four slopes (6, 12, 18, or 24dB/octave). The Dynamics section offers control of two compressor-limiters featuring adjustable soft knee width, and expander with adjustable soft knee and a gate with hysteresis control. Side chain EQ is also selectable.

It should be stressed that because all signal processing is software controlled, the console can potentially be redesigned or 'customised' to offer different additional facilities. For example, it will certainly be possible in the future to offer a selection of EQ characteristics and hypothetically the desk could be setup to emulate a mixture of Neve, Massenberg and Pultec EQ-types. This kind of flexibility is one of the console's great strengths; nothing has been 'set in stone' as with analogue consoles and to a certain degree some other digital desks. The possibilities really depend on the user's imagination and the compliance of the software writer. AMS-Neve consequently have a large, client 'wish list' which is continually being addressed—at the moment software updates are being released free to existing users approximately once a month.

Setup

As mentioned earlier, each channel strip can control up to four mono or stereo signal paths (A, B, C, D). These paths can be virtually any

at the moment software updates are being released free to existing users approximately once a month

signal within the console including channels, groups, monitor returns-group outputs, main outputs, aux masters, and cue masters—it is up to the user to choose how many path types are needed and where they will be positioned. This operation is performed from a Mix setup screen on the *AudioFile* terminal. Once this has been set up the paths themselves can be configured channel by channel starting with the input and adding the other processing elements as required. To speed up the operation, configurations can be copied on a multiple basis. Depending on the amount of processing power available, it is advisable not to over order—not every channel will require dynamics, for example, and certain paths, such as aux masters, may only require two bands of equalisation rather than four.

Console setups are stored to disk providing instant reset for different types of operation or personalised configurations. AMS-Neve are also looking at the possibility of directly accessing processing elements from the channel strip itself rather than having to reconfigure the path from the screen. For instance, it would be possible to add a gate by simply accessing controls on the relevant strip, thus providing a faster and more intuitive method operation.

It may be desirable at some stage to alter path positions in a Mix Setup, for example the operator may prefer a group of tracks and their effects to be moved side-by-side near to the centre of the desk. This is simply achieved using the Desk Designer facility which includes a grid display allowing paths to be accessed and moved. The advantage of this technique over repatching is that all the channel information remains intact—that is signal processing and automation.

Paths are assigned to the channel-strip controls by locally, or globally, selecting one of four PATH buttons—paths are permanently identified by a corresponding alpha display beneath each of these PATH buttons. A lock facility is provided to prevent a selected path or group of paths from being deselected from control.

I-O interfaces and routing

Just as the amount of processing power dictates the way the console is configured, so the number of I-O interfaces fitted affects the number of channels and buses that can be physically used. If the number of ports supplied is relatively high then it will be advisable to hard wire some ports to dedicated functions; if the number is small it will be necessary to path ports.

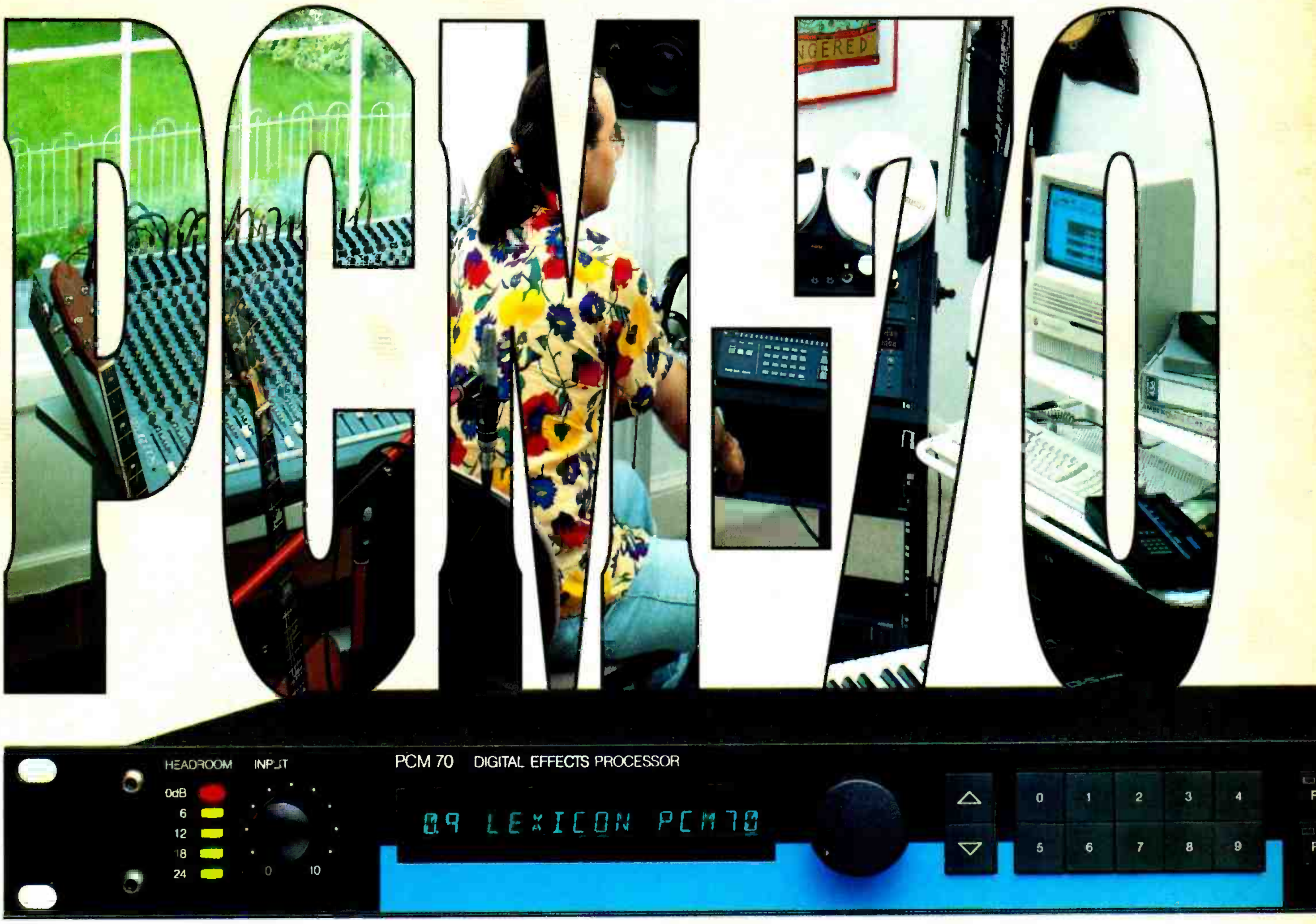
The system includes I-O interfaces for AES-EBU, SDIF 2, SPDIF, Prodigi, MADI, and 16 and 20-bit ADCs (mic-line boxes are also fitted). Inputs and outputs are selected from the path Logicator controls which will locate all possible ports including *AudioFile* if fitted. These selections form part of the MIX SETUP and are recalled along with the rest of the setup data.

If the console is configured to mimic a conventional in-line design, the channels might be arranged on Paths A and B, while the monitors-groups might be on paths C and D. Routing to tracks is then taken care of from the Master Routing panel which is divided into two identical upper and lower sections each with 64 numbered buttons. Basically speaking, the upper section deals with outputs and the lower section inputs so that destinations such as group buses are selected from above and sourced from below. Routing to Subgroups or Main Outputs can be achieved either directly from the channels or the routing panel.

The system can also be usefully interrogated to show where paths have been routed, and also provides clear indication of all available (configured) paths—for example pressing and holding a destination button on the upper routing panel will cause the lower panel to display all available sources with a green LED and those already routed to it with a red LED; in addition the corresponding PATH buttons on the channels will illuminate to indicate assigned routing.

Another use of this section is to implement Path Process Switching. This allows a path to be called up and displayed in a column of Alpha windows starting with its input and working down through the assigned processing elements. From here a new path can be created by changing the order of processing—so, for example, if a compressor were preferred post-EQ rather than pre-EQ, the original processing chain could be reordered, and an A-B check carried out to compare the two paths.

The master section also controls source selection for control room and studio ▶



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Expanded access

One of the problems with a console that relies on assignable controls and layered operation, is communicating parameter information to the user. Unlike traditional analogue designs where every control tells its own story, the assignable digital control by its very nature can conceal many. Although the *Logic 2* provides reasonably quick access to 'buried' processing, some users prefer to be able to see and operate a channel in its entirety. To satisfy this requirement, the console has been equipped with a function called Expanded Access.

Expanded Access is an instant function that operates in two ways. Firstly a selected path can have four pages of processing spread across four local channel strips—for example, four EQ bands could be displayed and adjusted. The second method allows all the parameters in a path to be spread across 12 channels. The position of these 12 channels is designated by the user, but generally speaking a central placement is preferable so that the operator remains in the stereo hot spot. Effectively this arrangement permits the console to be controlled from one central area, and rather ironically recreates the single channel assignable design that AMS-Neve were originally keen to avoid. Interestingly enough this way of working has been enthusiastically received by recent users.

Metering

The meter panels provide three vertical multicoloured bargraph displays—two audio level bargraphs for stereo signals (just the left operates for mono paths), and a gain reduction bargraph that splits to show compression in the top half and expansion in the bottom.

The channel strip audio meters are switchable between input, output, track send and return. They can also be configured to display line-in and line-out, split between left and right bargraphs. Each meter can be locally or globally switched PPM, VU or Peak and an overload bar is included for PPM and VU selections.

Below each meter is an alpha window that displays the associated I-O port, or alternatively can show user bit information, track IDs, machine IDs and so on. At the top of the meter panel is an additional indicator section that identifies the channel or group being metered.

Above the master section of the desk are four meter panels that can be individually assigned from *Logicator* controls below to meter main outputs, aux send levels, external machine inputs and so on. Assignments can also be selectively monitored if the EXT button is accessed on the LS Select panel.

Surround panning

Surround panning facilities are included in *Logic 2*, and any mono group may be designated as one of the following: Mono, Left, Right, Front, Left, Centre, Front Right, Surround Mono, Surround Left, Surround Right. This enables the console to be configured for any type of multichannel film format.

Depending on the type of film group that has been created, the available panning controls from routed signal paths will change accordingly. Each input path is equipped with three panning controls (LCR, FB< and LR) and a divergence control.

Optionally available is a twin joystick panner module that features recessed joystick (to avoid accidental movement) with LED matrix displays showing pan position. The system is fully automated and can control any number of assigned paths simultaneously.

In the near future AMS-Neve will be releasing a film record control panel which will allow simultaneous record in-out and bus tape switching for 32 machine tracks arranged in four banks of eight. Also a monitor matrix control panel which will facilitate the control of a fully configurable loudspeaker monitor matrix with up to 128 monitor sources.

Ganging

A VCA-style grouping system is available for faders and cuts. The systems operates with Master (Grandmaster), Submasters, and Slaves thus allowing group nesting. Gangs are displayed in two ways, firstly the path alpha displays will alternate between the path name and gang status to identify a selected group, and secondly, there is a permanent colour-coded display from the tricolour LED within the GANG button—red: Master, green: Slave, amber: Submaster.

At the moment gangs are limited to faders and cuts, but other channel functions should also be included in the near future.

Automation

The console is fully dynamically automated and all controls on the channel apart from solo and PFL respond to automation. Throughout the channel strip, MODE buttons are distributed which relate to the Logicator controls, function ON-OFF button, fader, and mute. These buttons are used to assign automation status to specific local controls; alternatively status can be set globally for the whole console or parts of it—that is faders, cuts, Logicators and switches, and routing—or on a channel-by-channel basis.

The usual selection of statuses are incorporated and depending on selection status will be indicated by tricolour LEDs in the MODE buttons —green: Read, red: Write, amber: Trim, and off: Isolate. Other statuses such as Takeover (causes touch-sensitive non-switch control to enter write from read on touch), and Autotakeover (causes a control in write to return to the previous read value on release at a specified rate) are indicated by flashing green and red LEDs respectively.

Automated mixing on a console where all the controls can switch to write as soon as they are touched, could be viewed as a risky business especially with a number of people involved in a mix. To guard against accidents and unintentional updates, individual controls of groups of controls may be 'armed' while others remain locked in read; also a 'Play Safe' mode has been included that can be used to rehearse moves in a similar way to Isolate although it allows Takeover and Autotakeover to be monitored.

Trim status currently applies to faders only, and there are three variations—Touch Inhibit Trim, Takeover Trim, and Autotakeover Trim. Touch Inhibit Trim will input a relative trim value at the entry point before the tape has gone into play; once playing the fader will not respond to new moves. Takeover Trim will cause the fader to change from read to trim once the fader is touched, on release it will remain at the new offset position to the end of the mix or, alternatively, by pressing TAKEOVER again it will null to the previous position at a predesignated ramp rate. Autotakeover Trim operates in the same way as Takeover Trim except that on release the fader automatically nulls to its previous position. As yet there is no null indicator for manual level matching.

A Touch To End facility is also included that causes any control that has been touched during the pass to be written to the end of the mix when the tape stops, additionally there is a TO END button that has the same effect for all controls switched to write.

Off-line functions such as editing switch-mute events and trimming dynamic data are catered for in an EDL-type display that lists events relevant to their time code positions.

Mixes are either saved manually using the KEEP button or automatically using the Autosave function which stores every modified pass unless the UNDO button is pressed to abort the current pass. Mixes are stored in a hierarchical fashion as Passes under a Mix, which in turn belongs to a Title which is part of a Project.

Mixes can be merged, spliced, copied back into themselves and so on. A graphic display shows the mixes being edited rather like crossfade editing on an *AudioFile* and allows the edit transition time to be set independently for each section.

The system supports Snapshots which may instantly be set to the console via a group of assignable RECALL buttons. The system can be used to set scene changes, build up mix presets, or as a convenient method of resetting master console status—that is Record, Overdub, or Mix. A crossfade time can be added to dissolve one snapshot into another.

The automation computer also provides full machine control and display for up to four machines, using either ES Bus or Sony P2 protocol via RS422. Record enable links from the console are soon to be implemented.

Conclusion

The *Logic* series of consoles have already proven themselves as something of a success story, and AMS-Neve have been working very hard to give existing and potential clients the features they want.

Logic 2 is an an extremely versatile and adaptable product, not only can it be personalised to suit individual preferences, but it can be configured for just about any operation whether it be music recording, broadcast, post, or theatre sound. The console's soft architecture and processing capabilities should also go a long way to 'future proofing' the design.

Although in the past *Logic* consoles have been more prevalent in the postproduction and broadcast sectors, we are now beginning to see their entry into the music recording studio. It will probably take a little time for the music industry to accept this radically new technology, but once it does and the advantages are fully realised, *Logic 2* has every chance of becoming a very familiar sight in recording studios throughout the world. ■

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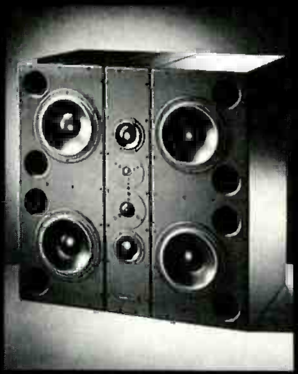
M4



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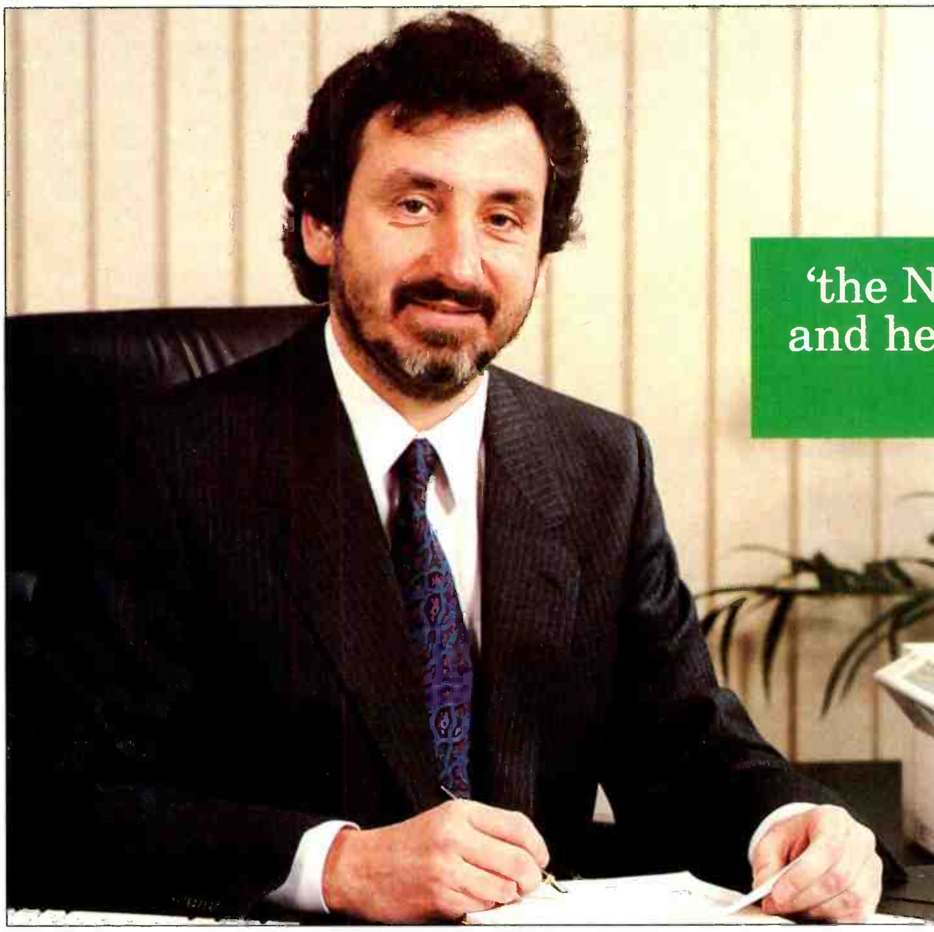


'the Neve name, products and heritage are extremely safe with us'

DIFF'RENT STROKES

Zenon Schoepe in conversation with AMS-Neve MD Mark Crabtree

Everybody had been watching the saga of Neve and AMS since the two were placed together beneath the umbrella of Austrian giant Siemens. Interest rose to fever pitch when both UK companies announced apparently competing top-end digital consoles—AMS' *Logic 2* and Neve's *Capricorn*. But the final twist was left to Siemens Chairman Hans Haider who, at the AES Convention in Vienna last year, remarked that Neve and AMS would merge before the end of 1992.

October saw the plans realised with closure and heavy redundancies at Neve's newly christened HQ in Litlington, Hertfordshire and centralisation of the new operation—tentatively called AMS-Neve plc—at AMS's HQ in Burnley supported by the Neve manufacturing facility in Kelso, Scotland and the AMS sales office in London. To the top of the pile rose former AMS MD Mark Crabtree, reporting to Chairman Haider. Non-executive directors include former Neve MD Laci Nester-Smith, Patrick Robson (MD of former Neve owners ESE prior to the Siemens buy out), and Jurgen Gehrels of Siemens plc. Further down the tree of command, Carl Lynch (originally from Calrec) is looking after the R&D side, which is what he used to do for AMS, while John de Jong (who worked at Neve for several years before moving to AMS) is in charge of the service operation. Frank Massam, who joined AMS at around the time of the Siemens acquisition, is in charge of Sales and Marketing while Ronnie Stephenson continues to run the manufacturing operation in Kelso, and Dave Allen continues to run the manufacturing operation at Burnley.

'Kelso is set up to be an efficient volume-manufacturing base.' explains Crabtree. 'In Burnley we will do first-offs of products and the final stages of commissioning and assembly of the bigger, very complicated products.

'We are trying to move the manufacturing to where virtually everything is built to order. It's something we've always done with AMS digital products. We are able to reduce decision time to delivery.'

Many have viewed the outcome of the merger with cynicism, regarding the 'Crabtree surrounded by AMS men' scenario as unhealthy. Crabtree replies: 'When you merge two companies with 200 miles between them, then, depending on the geography—never mind any other pros and cons—you are going to find that there are a limited number of people who really want to move. By just choosing one of those sites, one of those companies will be less well-represented than the other at the structural level. To some extent this has happened.'

He also counters comments that the merger has been a sad day for Neve: 'I don't see it that way. If you take my part in this then I'm a product man—Rupert Neve was a product man—and that product emphasis is something that I will be supporting very strongly. Perhaps more strongly than has been the case in the Siemens stewardship to date.

'It is terribly sad that in such a situation you have to lose people that you'd rather not lose, but my personal feeling is that we will actually come out stronger from this than if we had been left alone. Neve is the name and the faces that the market recognises. The product specialists are all still here. I think it's an incarnation of Neve that is being wept over rather than Neve itself.

'There is obviously a large degree of nostalgia for Neve and the feeling that this is an institution that we mustn't tinker with. I sense that and all I can say is that the Neve name, products and heritage are extremely safe with us. And because it has gone this way and not the other, I am going to be watched like a hawk. Even if I didn't have my heart in it, which I do, then there is no way that I could not favour the Neve side over the AMS side.'

Change have also occurred in the sales team ▶

which is built up of AMS-Neve staff and empowered to sell either.

Crabtree refuses to acknowledge that the AMS-Neve 'war' prior to the merger harmed both parties: 'There are sales people on both sides who had many opportunities to have sold the other product. Now they can.

'The one thing it did do was sharpen up both sides, certainly the R&D teams were very motivated to try and produce the better produce —and both products won. The sales people were really quite gentlemanly about it, although they were competing for business. I don't know how much better, or different, or worse it might have been if we'd done this two years ago. If we had, we might only have ended up with one digital desk and I think, looking back on it that that, would have been a disadvantage.'

Crabtree sees his role in the new company as continuing to keep his hands on product lines and developments through chairing regular Product Design Control Group meetings for each product. These comprise representatives from the marketing, R&D, manufacturing and service.departments.

'We are forming equivalent groups of people and formalising what happened informally at Neve or formally in a different way at Neve.That is how products keep their individual characters because the people in these groups are the DNA running through.'

Crabtree also maintains that a demarcation between Neve and AMS products will survive ('The products have not only functional differences but also character differences. Expect it to continue'). However, he disagrees with the notion that the merger with its shock announcement at AES Vienna was handled clumsily.

'I'm not sure whether it's possible to handle anything perfectly. I think the very positive saving grace is that we've at least provided entertainment for a large number of people in the industry. But having done that, the two companies have learnt each other's strengths and weaknesses at first hand and in the eyes of the customer. That will be very good for the future position of the business. In some ways it's a rather brutal form of market research.'

Much attention has been paid to the conflicting nature of the digital consoles of AMS and Neve, what would Crabtree say are the complimentary aspects of the two companies?

'Neve has an extremely long history in providing the best recording consoles that you can get your hands on,' he replies. 'There is great respect for the Neve name and with *Capricorn* that tradition continues. In Neve you have a set of people who fully understand music recording with complimentary areas in broadcast and some postproduction with the VRP.

'On the AMS side what we do is take technology and an application, and try to put the two together, producing innovative solutions to problems. AMS's history is in control of the process. If you take *AudioFile*, rather than say "we are trying to copy a tape machine" we said "this is what we're trying to do, these are the building bricks we have to do it with, then why don't we do it this way around?"

'When we looked at digital console requirements, from AMS it came from "we are able to capture the sound with a microphone, we can put reverb on from one of our units, we can record and edit it on an *AudioFile* but we can't mix it". We expanded our envelope of coverage by building a mixing console. We added into the picture at that point Calrec Audio to bring in the ergonomic design skills. The thrust of *Logic 2* was that it was designed to be a general-purpose console but we saw its applicability and the payback to the customer in the areas mainly of postproduction where you're dealing with a lot of manipulation and control.

'*Capricorn* has taken a more traditional approach: "let us produce something which takes the *VR* another n steps further". The original intention was to fulfil the music recording market because that is where Neve is very strong—*Flying Faders*, the Neve sound—replicating and substantially improving on the *VR* was the motivation behind the *Capricorn*.'

It could be argued that, bearing in mind the current size and health of the recording studio market, such a focussed and targeted product was a little misguided.

'That's a question I can't really answer,' responds Crabtree, 'because the decisions go back a long way and I wasn't involved. Despite the fact that *Capricorn* is an iteration of the *VR*, it still has much broader appeal than just music. Nowhere have we ever said that *Capricorn* is just a music recording console; it's finding a lot of sales in broadcast.'

Is that an area where *Logic 2* has not penetrated well?

'I think you'll find that *Logic 1* and *2* have. We've sold a *Logic 2* to Yorkshire Television, for example. But this is where the "is this a broadcast or postproduction environment?" question comes up. We're currently going through our market studies again and the old-fashioned barriers between music production, postproduction and broadcast are breaking down.'

> 'Nowhere have we ever said that *Capricorn* is just a music recording console; it's finding a lot of sales in broadcast'

Is that because of the flexibility of digital desks?

'It's the flexibility of the desk and market imperatives. Filling a music recording studio with music recording is good but people these days want a second string to their bow, they want to be able to do music to picture. *Capricorn* immediately spreads into music for picture by the addition of an analogue monitoring section. Several purchasers are using *Capricorn* for film work.'

But this is using analogue electronics, meanwhile AMS-Neve are adding panning joysticks to the *Logic 2*—surely an addition which would benefit the *Capricorn* .

'If you look at the architecture of the two consoles,' Crabtree observes, 'we built *Logic 2* as a modular system—the control surface has quad modules which you can plug-in in any dimension or order. Therefore it's very easy and quick for us to develop another quad panel for a specific purpose and that takes advantage of the transputer architecture where you can have one link going out of this panel. The software is written in such a way that you just have to define what a panel is interested in, in terms of signals from the system, and what outputs a panel has that are of interest to the system. You can write a few lines of code and you have a panel that can control anything and everything. That was because we were treading into an unknown area with a digital console and we left as many options open as possible. It's one of our hobbies.

'From *Capricorn*'s point of view, a lot of research was put into an assignable control surface which reaches in quite a few directions. That maps into the processing which is built using ASICs. It's a large architecture console built with ASICs which is another very good way of going about it—you can define your word length, you can build algorithms into silicon, and so on which has a different set of advantages from the *Logic*.'

It sounds as if Crabtree is suggesting that *Capricorn* is not an easily-upgraded system.

'It will replicate virtually any architecture of mixing desk you want,' he maintains, 'but for the more exotic aspects of postproduction work—we're talking to people not only about surround sound but up and down—the flexibility of *Logic* and the development that is already complete lends itself very closely to that. Over and above that, there is the major advantage that the architecture of *Logic* is the same as the architecture of *AudioFile*. *AudioFile* links in, which gives you the ability to do event-based automation where you can take any bit of audio and it's got full dynamic automation attached to it. And wherever you put it that event, the automation will follow. We will be delivering the first in the new year.

'That type of knitted-together system isn't readily possible with *Capricorn*. Therefore, by definition, to integrate hard-disk recording and editing with mixing capability where you need it most in postproduction, then the *Logic* has advantages. If you're recording and you want to bring each instrument up to the centre section and get it absolutely right and you want a fader for most things, then *Capricorn* is very good. If you want automation that you can use at a relatively straightforward level but takes you into the depths of event-based automation, then you can go with *Logic 2* and *AudioFile*. If you want something that's familiar to *Flying Faders* users—and there are an awful lot of those—then *Capricorn* is very good for you.'

Is Crabtree's implication that *Capricorn* will stay pretty much as it is, and will not be developed in the same way that *Logic 2* has?

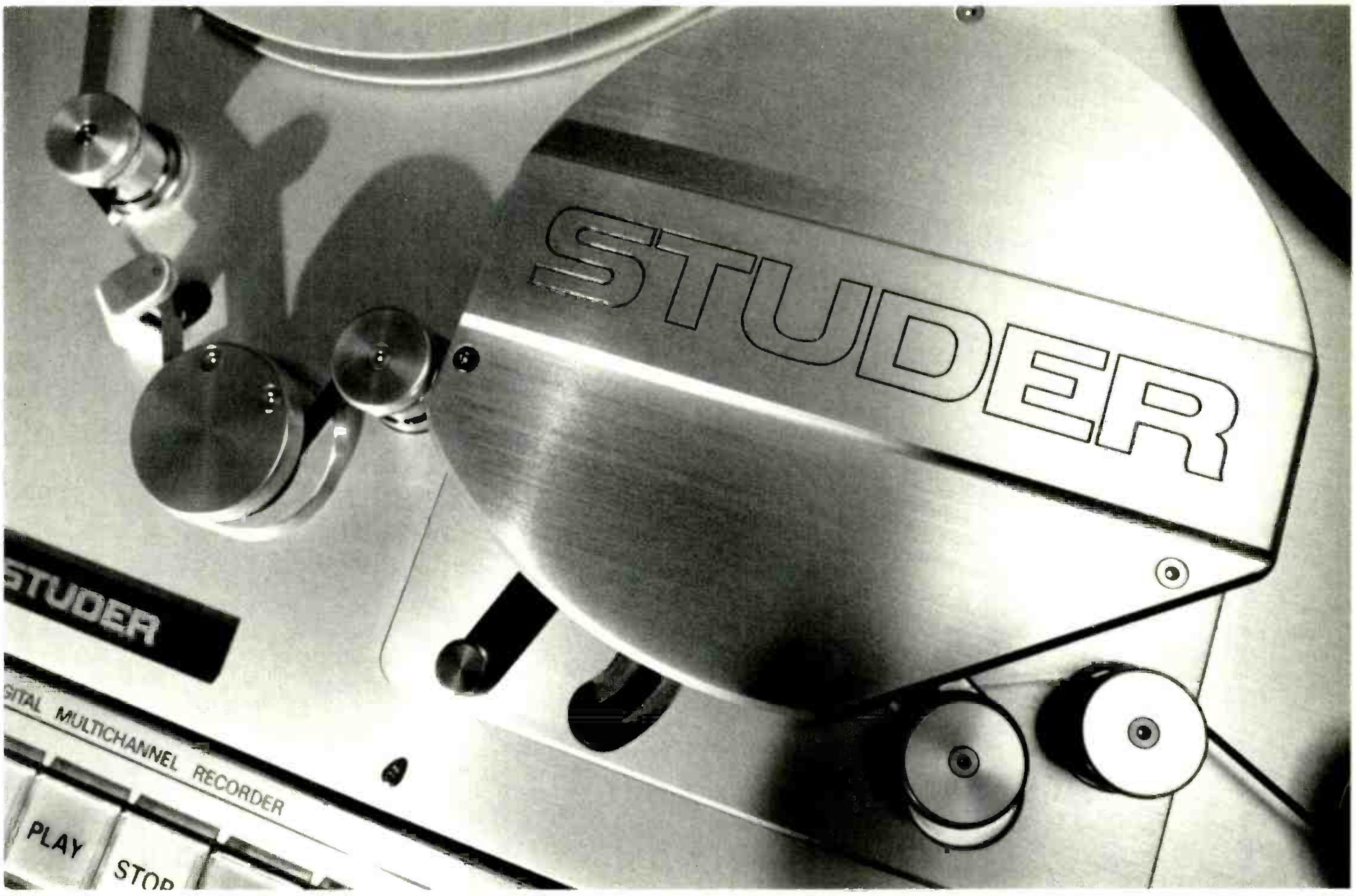
'There is no point converging the two, otherwise we end up with one product. There are distinct attractions, as our customer base is proving, to both products *Capricorn* will continue to be developed, *Logic* will continue to be developed.'

Turning his attention to the development of the *Capricorn*, Crabtree highlights the software

'The architecture is designed to be a very powerful platform and there will be continuing software effort for the foreseeable future as people begin to understand its capabilities and we listen to their requirements. Those requirements will be incorporated where it's of interest to a large number of people.'

What Crabtree is suggesting is that AMS-Neve are not interested in pushing the two desks further apart and certainly not interested in bringing them closer together—as much as the situation remains within the company's control.

'The two will approach to a degree, there is no way that can not happen. We are not proposing to push them further apart because there is a significant amount of market perception—apart from people who just want to poke a finger at us and say "what are you going to do about these consoles?" It's good fun to create a soap opera but at the end of the day both products are selling ▶



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very well.'

One question of considerable interest to the console market is whether or not AMS-Neve intend to continue developing analogue

'We intend to keep a very strong analogue presence,' confirms Crabtree. 'Because despite the fact that, even within Neve several years ago, there was the distinct belief that analogue would be dead by 1990, that's far from being the case. There is a great interest in analogue consoles. The *55 Series* broadcast console is being extremely successful internationally. We take great care to keep that capability and harness that capability in the analogue field—it's a major part of Neve's business.'

With the AMS-Neve merger came redundancies and transferral of staff. What is the size of the remaining *Capricorn* R&D team?

'We're still developing that situation,' Crabtree responds. 'There is still a relatively large number of people at Litlington working on the project. We have an agreement from a number that they will transfer here, others are making their minds up. At the same time we have shadow teams of very good staff from Burnley for people (at Litlington) who are really not prepared to move but whose knowledge we need.

'*Capricorn*'s been developed by a core team and as the project has gone through its various phases this team has been supported by contractors to a large degree.'

Is the use of contractors different to the work done by AMS?

'Yes, we use only AMS people. That will be the pattern of the future. Using contractors is not a process of development I particularly like because you lose some continuity. I prefer to have it in house, we have a fairly large team and we do what we can rather than say "at this point we need six more people" who come in, document their work, and then go.

'So, as the hardware platform for *Capricorn* is being completed, those contractors are going, or have gone, since the early part of this year. That has nothing to do with the merger, it's a planned phase out because at the height of such a project the amount of resources it consumes is enormous and you clearly can't keep that up on several parallel projects at once. From its height its now probably half to two-thirds the size.

'I don't know the exact number of *Capricorn*

> 'even within Neve several years ago, there was the distinct belief that analogue would be dead by 1990'

people we'll end up with but I expect it will be a team which will be comparable or slightly smaller in size to the *Logic* team because the *Capricorn* team is not faced with event-based automation.'

Are the technologies of the two companies close enough to share resources?

'It will start to be shared. The *AudioFile* has an MTI multitrack interface box and in the days when we were in competition with Neve, we were moderately amused by the fact that they were relying on MADI to appear. This ability to interface to digital machines we saw as very important to studios because they're not going to buy a new Sony MADI digital multitrack if they've already got an old one. Our interface was in the plans from the beginning. I think Neve got to this stage expecting Studer to do a MADI board and it's only really because *Capricorn*'s a little bit late that there are any MADI interfaces on tape machines at all. Our MTI box can interface to *Capricorn*.

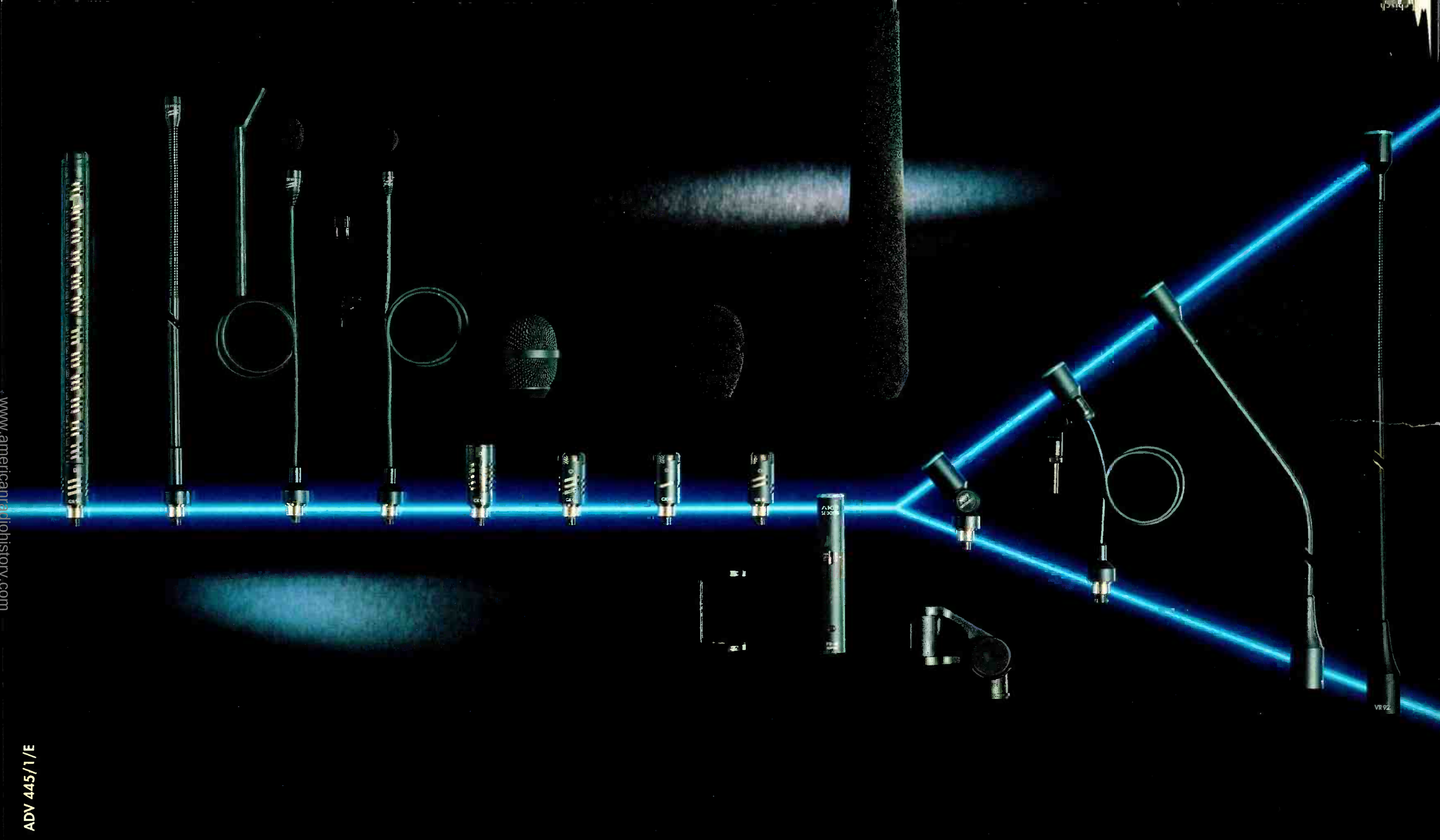
'As time goes on we will need time-code readers, machine-controller ports and already commonality is emerging between the two products and as we go through the periodic upgrades of cards within the systems we will design one card for both.'

With so much of the audio performance of a piece of equipment dependent upon the choice of its convertors, the question has to be raised regarding those used in Neve's *Capricorn* and AMS' *Logic* consoles. Are AMS moving towards using the same convertors as Neve?

'At the moment we are in the position of having a choice. There are pros and cons to each. We'll obviously make it possible for those to be interchangeable and as new convertor technology becomes available we will design one convertor for both products.'

Following the merger of these two major players in the pro audio field, it is clear that the market forces and technologies concerned have to be held in a very delicate balance. It is equally clear that this balancing act is one with which Mark Crabtree is likely to become very familiar indeed. ■

AMS-Neve, Billington Road, Burnley, Lancs BB11 5ES, UK. Tel: 0282 457011. Fax: 0282 39542.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AT4033

STUDIO CONDENSER MICROPHONE

"Audio Technica is still in its infancy in the professional market, and not having encountered it before, the 4033 Transformerless Capacitor Studio Microphone came as a very pleasant surprise. Its styling is distinctive and elegant, the finish is excellent, and the cat's cradle, again supplied as standard, is simple and effective and balances the microphone very well. Everything about the microphone looks and feels sturdy and professional. Once again the facilities are simple, the only switches are for the high pass filter and the pad, and the polar pattern is cardioid.

But the biggest surprise was the sound. On everything I tried – including a Steinway grand – the output was virtually indistinguishable from that of the 414 – open, transparent and clean, quiet and free of colouration. The main difference was in the sensitivity – the 4033 is few dB more sensitive than the 414. If this is an example of what Audio Technica has to offer, I await further developments with interest. A variable-pattern microphone with the sound of the 4033 would be a very useful addition to the arsenal indeed. As it stands, I can't imagine it will be long before this microphone is a much more familiar sight."

Reprinted from STUDIO SOUND AND BROADCAST ENGINEERING February 1992

AT4033 shown with optional shock mount AT8441

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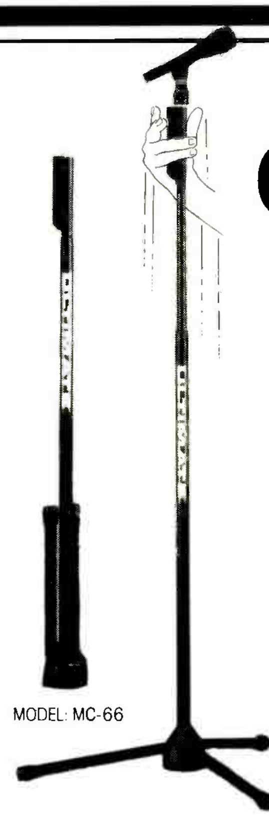

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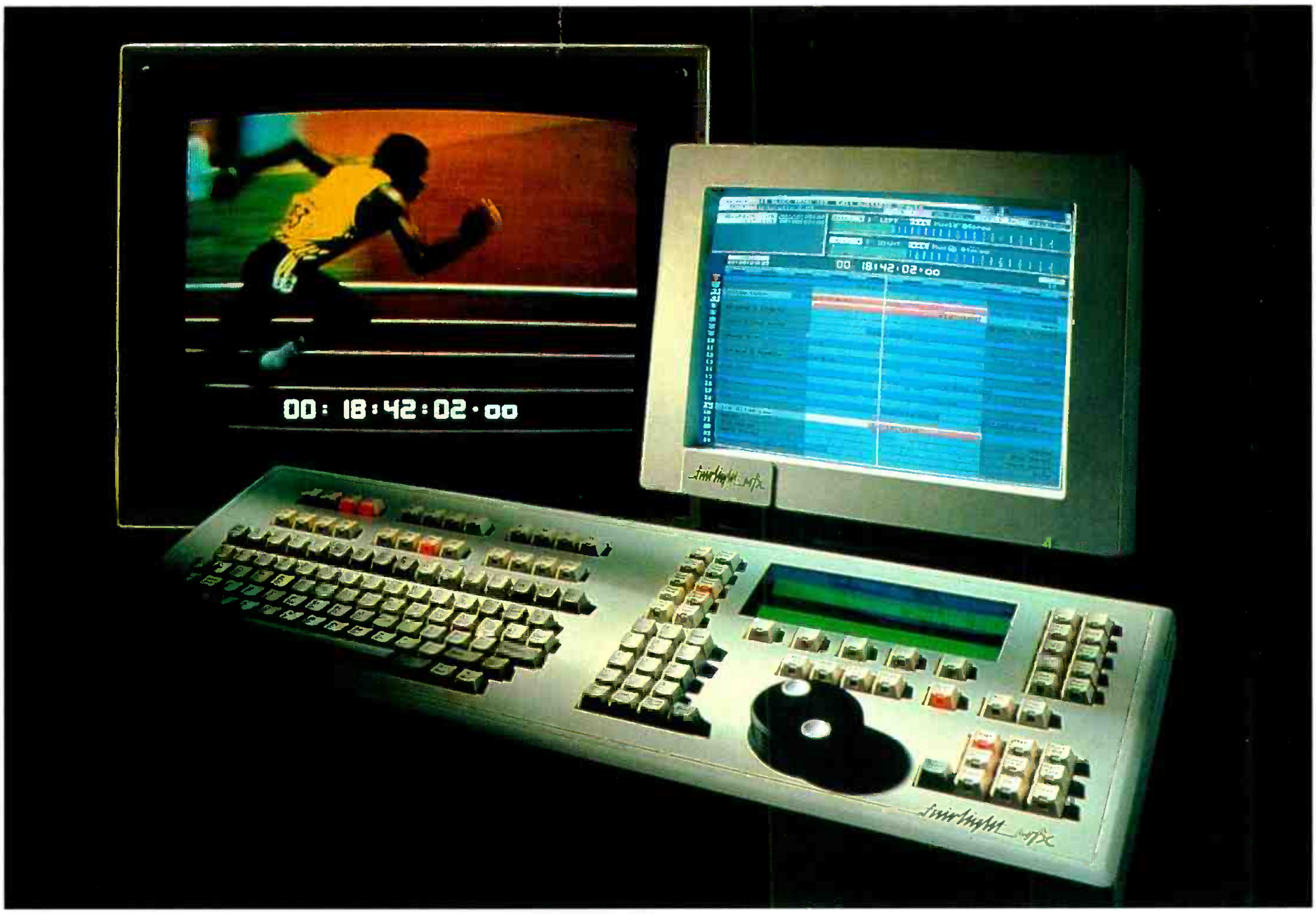

Off to a flying start—the Fairlight *MFX2*

FAIRLIGHT MFX2

After the success of the *CMI*, Fairlight failed. Now they have returned with an astute move into audio-for-video with the *MFX2*

The name Fairlight has been closely associated with music sampling and sequencing for some time. The Australian company was formed in 1978 by Kim Ryrie & Peter Vogel who launched one of the world's first sampling-sequencing systems, namely the *CMI Series I*. Around seven years ago the *Series III* was launched and to this was added 8-channel hard disk recording capabilities. In 1988, amidst a blaze of publicity, the company went into receivership. However, from the ashes, a new streamlined company called Fairlight ESP was formed by Ryrie & Amber Communications. The latter consists of a group of companies who are involved in the manufacture and distribution of professional audio and video equipment.

Fairlight ESP launched a disk-based recording and editing system called *MFX 1* in 1990. The system was aimed at audio-for-video post and supported eight continuous channels from one hard disk, with 16 simultaneous channels possible depending on the duration of the cues involved. *MFX 2* was launched in April 1992. This supports a new SCSI called 'turbo SCSI' which allows 16 continuous channels to be replayed from one hard disk. It also has improved high-definition graphics and supports Fairlight's new DSP card.

Having been relatively successful in their home territories, the company were in a position to concentrate more effort on overseas markets. They have established service centres in both the UK and Germany and appointed John Lancken as International Sales Manager and John McDiarmid as European Sales Manager.

The *MFX2*

The system supports the use of both hard and optical disc and allows simultaneous replay up to 16 continuous channels from hard disk or up to eight from optical. Each pair of output channels requires a dual channel card. The minimum number of cards supplied is three (giving a 6-channel system) and the maximum number is eight (giving a 16-channel system).

The system hardware is divided into three rack units. The digital-card cage contains all processing, turbo SCSI channel cards, system RAM and graphics controller. The analogue card cage provides 24 balanced analogue outputs (including output router) as standard, either two analogue or two digital inputs and MIDI and SMPTE interfaces. The disk drive unit supports a floppy drive, Exabyte tape-streamer and up to six drives which can be chained to provide additional recording time. Using the Fujitsu 1.9Gb drive, this can amount to a total of 35 hours.

The user interface consists of a colour monitor and a custom-designed controller which has the appearance of a rather large alphanumeric keyboard. On the left of the controller are alpha keys and functions keys which can store up to 45 macros (a macro remembers a series of operations and carries them out when it is activated). Above these are 24 track-select keys. At the centre of the controller are keys for edit menu selection, mute and solo and numeric keys.

At the top right is an LED display with five soft keys underneath, the functions of which will depend on which edit menu has been selected. Beneath these are transport control keys and a jog wheel, and to the right of the wheel are keys for JUMP, MASTER SELECT, EDIT RANGE, UNDO, MACHINE CONTROL and ZOOM functions. To the right of the LED display are keys for TRANSPORT MODE, SELECT MARKS AND RANGES, AUTO-RECORD and SET MARKS AND LOOPS. There is no mouse—all operations are achieved by pressing keys on the controller and-or moving the jog wheel.

For the main display, the standard colour monitor supplied is a 14-inch high-resolution ▶

NEC *Multisync 3FG*. However, the system will support any RGB monitor since some customers prefer larger monitors such as those manufactured by Barco.

Recording

There is one main screen for recording and editing. This consists of a track display which will show from 1 track to all 24 tracks displayed horizontally across the screen. The fewer the number of tracks displayed, the larger their size. Any of the 24 tracks can be armed for recording by selecting the Arm menu and then selecting the TRACK key. Selected (or active) tracks are highlighted in pink, whereas inactive tracks are blue. The system supports sampling rates of 44.1kHz and 48kHz and a maximum of two independent mono tracks or one stereo track can be recorded simultaneously. The source may be digital or analogue and analogue signals can be attenuated by the user. The input can be monitored, there are on-screen level meters and the system draws a waveform in real time while recording is taking place.

A feature which is particularly useful for dialogue replacement is the 'record again' function. This allows up to 4096 drop-ins to be recorded at the same location, stacking one on top of the other and automatically incrementing the take number each time. A Take display at the top of the screen lists the takes along with other useful information such as durations, location, whether mono or stereo and whether borrowed from another project or not. Stacked takes are listed vertically, with the last take appearing at the top of the stack.

Other useful recording features include 'record pre-time' which will record a handle of up to 20 frames before the RECORD key is pressed, 'record clip' which performs a recording of the same duration as a selected clip and 'record to head', which automatically drops in at the start of a selected clip but will drop out only when manually told to do so.

Editing

Each track has a fixed output and can contain up to 99 clips. Clips are displayed as coloured blocks, each containing a waveform and if a clip of a selected track is located under the vertical 'play head' its colour is changed. A track may not only contain clips sequentially, but, as already mentioned, may contain clips which are stacked on top of each other, although only the top clip will be replayed. However, any clip lower in the stack can be brought to the top by selecting it from the Take display.

The amount of time displayed on screen can be compressed or expanded very quickly and this also affects the the jog wheel, which moves across the audio at a rate proportional to how much time is displayed. Up to 1000 marks can be stored and appear as small markers in the Timebase display above the tracks. A mark display can also be called up which appears at the top of the screen and lists marks vertically along with the mark number, position and name or description. Marks can be quickly located by using the JUMP key. This will jump according to a variety of parameters such as to marks, to in and out points, by a user-defined amount, by name or to fade points.

'the jog wheel has two scrub modes, one of which imitates conventional reel-rocking'

The system will perform cut-and-paste-type operations and has one layer of undo. Clips can be defined using the 'cut head' function. This deletes the remaining audio from the current location to the start of the take, while the 'cut tail' function will do the same but to the end of the take. Should some of this deleted audio be required at some point, the trim function can be used reveal as much as required.

Simple butt edits can be achieved quickly by using the JUMP key to locate in or out points rather than scrolling through using the jog wheel. If a crossfade is required, it is performed in real time by setting a fade out and a fade in. This will, of course, take two channels for a mono crossfade, however there is the option to internally mixdown a region and the JUMP key can be used to quickly locate fade start and end points in order to isolate the crossfade for a mixdown to a single channel clip. Currently, the only level control is clip-based, static and single stage.

The system allows any clip to be freely allocated to any of the 24 tracks and also allows any combination of tracks to be selected for global (or region) editing for all edit modes except one. Furthermore, there is a function for quickly selecting a group of tracks and tracks can be soloed or muted. The user can also take advantage of the macro facility for performing global edits. For example, a region across a group of tracks could be deleted and a macro then used to automatically place a fade on all clips at either end of the region boundary.

Sync to video

A Sony 9-pin RS422 interface is used to control an external VTR such as a U-Matic. By selecting the 'master' mode, the system will control the U-Matic using the *MFX 2* transport controls and-or the jog wheel while acquiring time code via the RS422 and locking within one second. Time code values can be grabbed and used to place clips or to give a clip an offset. In addition, a useful feature for ADR is the ability to set up a loop for both the VTR and the audio.

It is interesting to note that the jog wheel has two scrub modes, one of which imitates conventional 'reel-rocking'. The other maintains original pitch by looping a frame's worth of audio at whichever frame rate is being used and can be nudged back or forth by as little as $^{1}/_{80}$th of a frame. Fairlight argue that is, in fact, a more useful way of finding an edit point than by conventional scrubbing because the audio remains intelligible, even in freeze frame. Furthermore, the audio will scrub backwards along with the picture as well as forwards.

Other special features

Using the new *96000* DSP card, the system will perform time compression-expansion from 50%–200%. This range may seem extreme, but the system performs the process in the frequency rather than time domain. The user can either ►

Operational ease has been one of Fairlight's highest priorities

set a percentage or specify a duration and can process a mono or a stereo track. The examples given during the demonstration were of music and although compressed and expanded by more than is usually recommended with time domain processes, still sounded good. There were noticeable effects at the extreme ends of the range, but these could be useful for sound design purposes.

EDL conforming is supported and was developed in conjunction with an Australian company called DigitEyes (who manufacture the *Shotlister* EDL management system). The EDL option is called MFX Interface and runs on an external IBM PC. This takes control of the *MFX* console and drives the system through the conform process. The list can be sorted for the most efficient conform and has a variety of recording and conforming options.

Work is organised in projects and a list of all projects can be called up on screen. There is no specific library structure, however Fairlight

> Future plans include sound design features, clip reversal and harmonising

suggest that projects can be created which are specially organised for library purposes and since two projects can be open simultaneously, clips can be quickly borrowed or copied from another project. This would involve copying either from the other project's take list or from the track display, where related sounds could be arranged sequentially along the same track and quickly auditioned by stepping through.

In addition, the take list can be searched alphabetically, auditioned and the selected clip (whether from the take list or track display) will be automatically placed at the cursor time on the selected track in the working project. The difference between borrowing and copying is that borrowing uses the one clip for both projects whereas copying actually records another copy of the clip onto disk, so that if the project copied from (and thus its clips) is deleted, the required clips from that project are not suddenly missing from the current one.

Once the disk is full, the user may wish to archive its contents. There is, however, the facility for disposing of unwanted audio (such as clips which are not used, tops and tails, etc) in order to create more space for further recording. The system uses high speed 8mm Exabyte tape-streamer for backup at five times faster than real time and allows selective project loading. Alternatively, optical disc can be used for backup and-or as the working medium, which may be helpful for applications which have a high turn around of work such as advertising.

Future plans include a number of sound design features, clip reversal and harmonising. In addition, the ability to increase the level of a clip will be introduced as well as varispeed and a gate function. This will allow clips to be generated automatically with user-defined thresholds and handle lengths. There will also be a global replace function and the facility for printing dub charts. Plans this year also include the ability to support 24 channels with 24 inputs and outputs with the choice of analogue or digital I-O.

Conclusion

The *MFX2* offers comprehensive and quick control and includes many useful features. In addition the synchronisation and control over external machines has been carefully thought out, is highly sophisticated, yet simple to operate. The display is easy to understand and the use of colour makes it obvious whether a track or a region has been selected. However, due to the range of control features available, the user interface may take some time to fully master and although the use of macros is particularly helpful for functions such as automatically imposing fades, this particular requirement would not be necessary if the system supported a default crossfade feature.

Nonetheless, the system terminology and operation has obviously been aimed at providing sound-to-picture editors with familiar concepts and the addition of the DSP and MFX Interface options should further increase the system's appeal to the high-end audio post market. ■

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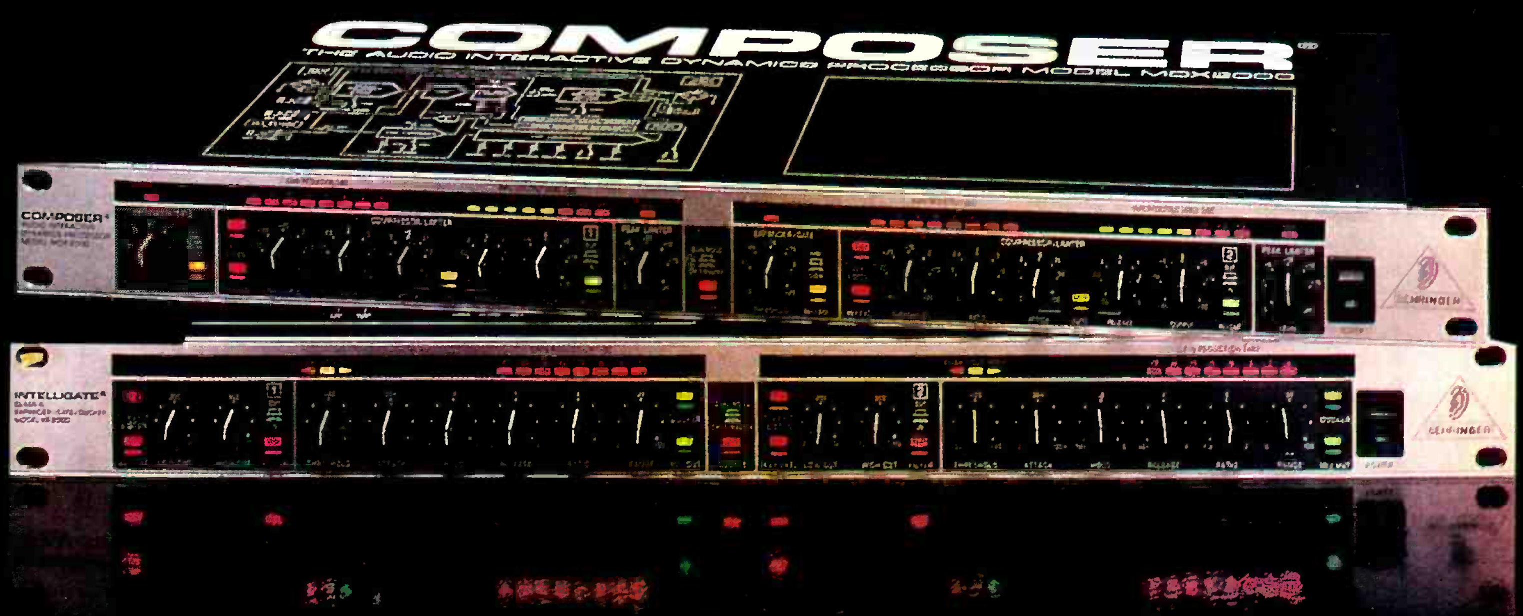

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C ◑ **calrec Audio:** booth 5.2D1. *Sabine* Feedback Exterminator DSP controlled processor which automatically locates feedback frequencies and assigns to them a digital notch filter. ◑ **CRL:** booth 5.1L3. ASSETT, a menu-based quality control system that stores excerpts from the digital master and compares the frequency content with audio cassette copies. ICE, an embedded coding system designed for automatic broadcast performance accounting and for keeping tabs on material in studio and duplication situations. ◑ **Circuit Research:** booth B-22. Real-time Event Sequencer, designed to provide up to 200 events over a seven-day period, with each event consisting of as many as eight contact closures.

D ◑ **DDA:** booth G3. New product to be announced. Also *Profile* 24-track VCA and switch automated console. *Forum Composer* entry level 24-track console. *Forum PA* and *Forum Matrix* sound reinforcement consoles. ◑ **DigiDesign:** booth R-24. Version of *Pro Tools* 4–16-track tapeless system with improved interface. *Pro Master* 20-bit system; *Session 8* Integrated recorder, editor and mixer. Alesis *ADAT* integrated system. ◑ **Digigram:** booth K8. Digital audio transmission through ISDN and PC-based workstations, now extended with Version 3 *Xtrack* and a low cost PC workstation. ◑ **Doremi:** booth B-25. Random access video playback from hard disk or M-O cartridge. From the system control panel, the operator can replay, shuttle and locate video in sync with the edited audio tracks. ◑ **Drawmer:** booth 2-L2. *L441 Quad* Auto Compressor-limiter contains four discrete audio channels with switchable hard or soft knee.

EF ◑ **Eventide:** booth M-08. *DSP4000* Ultra-Harmoniser, with facility to create effects algorithms by linking modular effects building blocks. As many as eight harmonies simultaneously, or four harmonies and stereo reverb. ◑ **Focusrite:** booth M-05. 'Cost concious' *Euroconsole*, in 40 or 48-input frame sizes. Same control and circuit layouts as *Studio* Console but starting around £165,000. Also *Red* Range of signal processors. ◑ **Fostex:** booth 5.1g1. *D10* professional DAT recorder with cueing and editing functions for less than £2,000. Two-machine editing can be accomplished with a pair of *D10* recorders without external computer or controller. Each machine has a RAM buffer and scrub control for fast location of cue and edit points. ◑ **FM Acoustics:** booth 2-H3. *ClassAmp M-1* microphone preamplifier; *FM214* precision balanced line driver and *FM 216* precision line level interface.

Lab Gruppen *SS 1300* amplifier

G

◑ **Genelec:** booth R-08. *1038A* three-way system for flush mounting or freestanding use. *1032A* two-way active system is based on the existing *1031A* nearfield but with extended bass response and increased SPL. ◑ **Girardin:** booth 5.2K4. Mixing console *CS890-892* for radio production. *CS192-CS276* television control room system, available with 18 to 36 channels, between six and 12 sends, four to eight auxiliary mixes, multitrack inputs and outputs, plus the option of bargraph metering on every channel. ◑ **Gorgy Timing:** booth H-14. Silent clocks with LED readout options including radio synchronisation, stopwatch, time signals, thermometer and outputs for ASCII and standardised binary. ◑ **GTC:** booth 2-D3. Latest update to the *Digitron* Magneto-Optical Disc recorder, specifically orientated to lipsync recording.

HI

◑ **Harrison:** booth 2-D1. *MPC* automated, assignable console for motion picture sound mixing with up to 256 channels with analogue or digital signal paths. Audio housed in separate rack, work surface can be configured for operation by one, two or three people. *Series 10* console is now showing as *Series 10B*. ◑ **Institut Fur Rundfunktechnik:** room R56. First demonstrations of MUSICAM Surround 3-2 format reproduction. Worldwide standard is based on percetual coding and is intended for applications including professional recording, computer-based multimedia, telecomms, broadcasting and consumer recording.

L

◑ **Lab Gruppen:** booth B-26. Power amplifiers and a 24-bit digital crossover, *SS1300* amplifier has DC rail voltage derived from switch-mode power supply: the magnetic energy in the ferrite transformer is controlled with a pulse-width processor and magnetic flux windings. Said to maintain output between 180V and 260V AC. ◑ **Lexicon:** booth 2-C1. *NuVerb* for DigiDesign's *TDM* standard for Mac-based workstations. A *NuBus* Card which has two DSP processors that can run together, or be separated into two independent mono devices, or cascaded for multiprocessor setups. Many of the settings are based on the Lexicon *224XL*, ▶

Focusrite *Red Box* processors

Revox on-air studio package

PCM-70 and *480L*. ◑ **LyndKraft:** booth Q3. *Tube-Tech* valve 2-channel compressor-limiter with claimed frequency response of 5hz–50kHz -3dB, said to replicate Fairchild *670*.

KM ◑ **Klark-Teknik:** booth B-04. Major new product to be announced. ◑ **Marantz:** booth B-04. *PMD-740* four-track personal studio; *CDR-610* Orange-Red book CD recorder; DCC recorder; digital two-channel real-time de-clicker and scratch suppressor. ◑ **Meyer Sound:** booth R-34. *UPL-2* bi-amped active design uses a 10-inch woofer with a 1-inch high frequency driver coupled to a symmetrical 60° horn. The HF driver is a patented design with a titanium dome and a silk suspension. *MSL-2A* high powered bi-amped system based on a proprietary 15-inch woofer and a 70° HF horn with a 2-inch throat. *MP.S Series 3* consists of *MP.S 305* compact system with a 5-inch LF driver and a piezo HF device; *MP.S 355* with two 5-inch drivers: designed to be used with the *MP.S 355* with two 5-inch drivers: design to be used with the *MP.S 3* Control Electronics Unit.

NO ◑ **Nexo:** booth R23. *PS10*, smallest speakers in range. Can be combined with *PS TD* Controller and *LSub 500*. ◑ **Omniphonics:** booth 2-L1. *Audio Resolver* preamp with level matching, balancing and stereo matrix for *Audiophile* switching and dubbing. ◑ **Otari:** booth 5.1F1. *Concept I* music and production console, a digitally-controlled system with automation. Uses a symmetrical dual-path architecture with 4-band equalisation and 100mm fader for each path. Dynamic automation will control both channel faders and mutes, allowing 96 automated channels in a 48-module frame. *Diskmix* VCA automation is standard, with moving fader option available soon.

P ◑ **Penny & Giles:** booth B-06. 'Cost-effective' conductive plastic fader designed for life-span of one million operations. ◑ **Philips:** booth 5.1P3. DCC mastering and duplication. DCC editor includes a wide range of PQ editing facilities and improved user interface. QC machines including prerecorded and blank verification of pancakes and cassettes. In-cassette duplicators and text

Pro Master Launch

Version 2 of *Pro Tools* is an integrated application in which tracking, editing, mixing and automation are performed in one computer window. Systems may have from four to 16 channels.

Digidesign are also launching *Pro Master*, a 20-bit version of *Pro Tools*, for mastering. The manufacturer is also shipping the *Session 8* system for the PC computer platform. This 8-track system includes digital recording, editing, mixing and track bouncing.

Digidesign Manager of Marketing Communications, Mark Wilcox, says: 'We refer to it as our studio in a box because, for the first time, the interface gives you the ability to patch in your effects and inserts, and route them all in software'.

Digidesign are also developing a system to integrate the Alesis *ADAT* digital eight-track. Booth R-24. ■

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Soundtracs *Jade* console

controllers for premanufacture verification. ◑ **Pro Monitor Co:** booth 5.1K2. New *MB1* transmission line monitor with removable sub-baffle which can be rotated for horizontal or vertical mounting. A 310mm radial chassis bass driver is coupled with a 75mm fabric dome mid-range and a 28mm soft dome tweeter. The 'usable frequency response' is 20Hz–20KHz. ◑ **Publison:** booth 2-D2. *Infernal Workstation 1600* enhancements including new remote control with alphanumeric keyboard, VTR control, reel scrubbing, tablet and pen, plus a wide selection of function-specific keys. Autoconforming from VTR and DAT; biphase synchronisation; optional mixing desk with moving faders and parametric controls for internal DSP; utilities including auto backup of the edit list with associated sound sources. Also: new machine, *Oceane*—stand-alone optical recorder for low-cost recording prior to *Infernal Workstation* editing and mixing.

Choice is the Theme

In addition to a 24-output *Spectra* working in conjunction with a 4-player *Logic 1* console, AMS-Neve will have a system linked to a Pioneer Laserdisc recorder.

Product Manager for postpro systems, Doug Ford, says: 'If the customer needs random access video, the machine is available; if not, they can use a U-Matic or Betacam. Choice is the theme. We have a range of different products that can be bolted together to suit specific applications.' Booth A2-A3. ■

R ◑ **Revox:** booth H-08. UK developed on-air studio package with an all-in price of £20,000 including furniture; *MB16* broadcast console; two *C221* professional CD players; *C115* cassette deck; a pair of *NF Mk 1* monitor speakers; two *M3500* mics; *PR99BV* tape machine; power amp; turntable; two cart machines; telephone hybrid; clock; mic stand; cables and connectors. New radio automation system with as many as 256 *CDC100* changers holding 100 CDs each.

S ◑ **Solid State Logic:** booth 5.2P1. *Ultimation* facilities added to *G-Series* automation including 15 Software Groups which enable any channel fader to act as master; Cut events can now be created off-line; Insert Mixing allows moves to be inserted into the current mix without destroying subsequent moves. Consoles include *SL4000* with wraparound wings. First european AES showing of *Scenaria* all-digital editor, multitrack recorder and mixer with random access video. *Screensound* and *SoundNet*. ◑ **Sonic Solutions:** booth B-17. Range of digital editing workstations with *Sonic Net* FDDI networking. *NoNoise* restoration system. ◑ **Sony:** booth 4.1A1. *DMX-S6000* digital audio mixing console for postproduction and sweetening in frame sizes of 24–64 modules. Five studio and two portable DAT machines and edit controllers. ◑ **Soundcraft:** booth 4.2A4. New automated console to be announced. Complete range of existing consoles. ◑ **Soundtracs:** booth 2-F3. *Solo Logic* console with VCA fader automation and machine control. *Jade* console with 24-group buses and parametric FdB equalisation.

TY ◑ **TC Electronic:** booth 2-Q1. *M5000* digital audio mainframe which can be fitted with four DSP or A-D-D-A modules. Programs include reverb, ambience, pitch shift, chorus, flanging and delay. ◑ **Yamaha:** booth R-27. *DMC-1000* digital mixer with new Version 2.0 software and *Project Manager* software. Version 2.0 software for *DR-8* 8-track recorder; Version 3.0 software for *DMR-8* recorder and mixer. *SPX990* effects processor with 20-bit A-D-D-A conversion. ■

EXHIBITION TIMES

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Friday	9.30am to 2.00pm

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STUDIO L'EQUIPE

Studio N showing the Nagra *T-audio TC*, *SoundStation SIGMA* and Amek *Angela*

Peter Ridsdale visits a Belgian film and video studio complex with a fascination for customised equipment

Studio L'Equipe is a Belgian film and video sound company which now occupies two buildings in Brussels both of which are not far from the RTBF radio 'city' in the Evere district. Andre and Phillipe Bosman are the directors of this company and a number of affiliated companies. Some of these share the same buildings with L'Equipe and they offer a wide range of postproduction facilities including laser subtitling, 3D effects, colour correction and duplication. Two of these companies are based in Paris.

L'Equipe was established in the 1950s by Andre Bosman (Senior), the father of the present directors, and it began life as an audio facility. Bosman was something of a pioneer and his background in mechanical engineering put him in good stead when it came to equipping the studio. Equipment was not as readily available as it is now, and if he needed something that was either prohibitively expensive or had not yet been invented, he simply manufactured his own. This tradition continues to the present day and the 'team' of 25 includes two equipment makers, one who specialises in mechanical engineering and one who deals with the electronics. Let me make it clear that we are not talking about biscuit tins and circuits from *Practical Wireless* here — there are, for example, no less than 15 shiny Set Mag machines that have been made to a very high spec by these two gentlemen. Had I been told that they were expensive Swiss or Swedish machines, I would have been none the wiser. It is as if a recording studio decided not to buy an Otari or a Mitsubishi multitrack but rather to make one themselves.

The original audio facility soon acquired a film department and Bosman continued to keep up with technological developments. His sons Andre and Phillipe eventually joined the business and Andre (Junior) in particular, shared his fathers passion for mechanical construction. When the video revolution occurred in the late 1970s L'Equipe was ready for it and began to specialise in the transfer of film to video largely at the instigation of the second generation. The house next door to the original building in the Rue Colonel Bourg was occupied as the business expanded, and today the mother building resembles a technological warren in which state-of-the-art video equipment rubs shoulders with antique film machinery. Just when you think that you have seen everything, yet another video den is revealed on yet another level. The new building is a complete contrast in that it has been custom built as a studio domain. It is modern, comfortable and as they say in Brussels, *tres chic*.

The opening of this new studio suite was something of a champagne-flavoured event, with the Mayor of Brussels and the Minister of Culture among the distinguished guests. Signs of nervousness could be detected among the staff at the thought of cheery glass-toting dignitaries, on the knife edge of sobriety, looming near the brand, spanking, new *G-series* SSL *5556* console in the auditorium studio (Studio F). Fortunately, the evening was a great success and passed without the chagrin of liquor in the mixer.

The 40-channel console is configured for film use in that its EQ pots are at the bottom of the panel below the routing switches and the pan pots. It has moving faders and Total Reset. The auditorium was designed by Emanuel Mohino to simulate aural conditions in regular cinema salons. If only the fleapits that I frequent could match up to this Platonic conception of a movie-house — one with a JBL *THX* monitoring system with 30 channels of Dolby *SR* and no coughing, spluttering or crackling of extra-loud sweet wrappers designed especially for the cinema. It is possible in Studio F to mix in *SR-D* mode with a variety of speaker configurations. One of

these, using all six digital channels, would consist of front left, right, centre and subwoofer and with left and right surround channels at the back. A moving display along the bottom of the screen enables the engineer to see forthcoming audio events before they arrive.

The projection from for Studio F contains not only a Magnatech projector but also a Sony *Dash 3324* multitrack and the aforementioned Set Mags. For those readers who are not familiar with film equipment, let me explain that, like a projector, the Set Mag unwinds sprocketed film of either the 16mm or 35mm variety, but unlike a projector it is only concerned with the soundtrack — the French word, *defileur* (unwinder) is much more descriptive. One machine may have all the birdsong for outdoor scenes, another all the footsteps (Foley in film jargon), other machines will have voices, music and special effects. Of these 15 machines five are monophonic, eight of them have three tracks and two have four, giving a total of 37 tracks. This is, of course, in addition to the 32 tracks on the Sony machine and L'Equipe consider this to be sufficient for the European market. Only films like *Terminator 2* might need to use more and there are indeed film sound studios in the States with up to 30 machines. At the head of this phalanx of filmic hardware is an off-the-shelf item; an Albrecht machine which is described by Yves Bradfer, the Studio Manager, as the Rolls Royce of unwinders.

The music for the film does not take up that many tracks on the Set Mags as the multitracking is realised, for instance, in the composers studio and delivered as a stereo master. Separate machines will, of course, be used when one piece of music cross-fades with another but that still leaves a lot of tracks for sound effects. Bradfer explains that if, for example, a bit of traffic atmosphere was needed for a film it would not be simply a matter of putting up a microphone in the centre of Brussels and recording half-an-hour of background noise that would do for all the outdoor shots in the film. Each scene has to be custom made so that each car door slam, engine revving or distant police siren fits in with the dialogue and the mood of the film. The precise second that the sound begins, its volume and its EQ are as critical to him as the layering of musical events would be for any music engineer. He eschews the use of ready made CD sound effects and is engaged in building up his own library of sounds.

Studio F is used exclusively for mixing. Studio 2 in the mother building is where the recording of Foley, sound effects and overdubbing takes place. This auditorium studio has been in use for over 20 years and has variable acoustics achieved by movable heavy curtains around the walls. Inset into the floor in front of the 32-channel Amek console are various different surfaces for recording of footsteps — dry leaves, concrete, gravel, parquet, squeaky floorboards. The Foley artist resembles a super-clean bag lady when she arrives with the flea markets of Brussels. As the studio is only used for recording the console has not been computerised. There are two 35mm recorders, one of which has Dolby *SR* and which can handle 16mm as well, and a bewildering variety of Set Mags in all the possible frame and track configurations (26 altogether). There is a Barco projector for U-Matic low-band video work and two film projectors, one of them a Magna Tech which can be operated at high speed. Studio 3, in the same building, is a video mixing studio which is also used for recording voice-overs. It has a Soundcraft *2400* console with 20 inputs and an analogue Otari *MTR 90* (16 tracks). Video mixing can be done on 1-inch, *Beta SP* and U-Matic with Q-lock synchronisation and there is also an *MWA* 3-track perforated film recorder.

The opening of this new studio suite was something of a champagne-flavoured event, with the Mayor of Brussels and the Minister of Culture among the distinguished guests

Back in the new building, Studio N (for numerique) offers a similar set of possibilities, the main difference being that it is digital. It is equipped with the 8-track DAR *Sigma SoundStation* with an optical disc and has the excellent Amek *Angela* 16-input mixer. Mixing can be direct to the master or onto synchronisable DAT, and there is also a Nagra *T-audio TC* which is something of a rarity these days. The voice-over cabin does not resemble the usual converted broom cupboard and has been designed to make the reader feel comfortable and relaxed. The mics are Neumann *U87Ai*.

To say that Belgium's linguistic divide is a great problem for the country is an understatement. Quite apart from the political and social rancour that it causes, it means that everything has to be done twice. Some people — printers, sign-writers and translators, for instance, benefit indirectly from the system and so does the film and TV industry. Foreign films have to have two sets of subtitles and commercials have to be dubbed in both French and Flemish for the different TV channels. This means that at least when it comes to commercials Studio L'Equipe automatically attracts twice as much work as would an American or British counterpart. It also seems to get a lot of subtitling work. Nearly all the films I saw while I was in Belgium were subtitled by LTI (Laser Titre Industrie) which is one of the affiliated companies mentioned earlier. The method for the affiliated company in Paris who now hold the worldwide patent for it.

As this is *Studio Sound* and not Studio Image, I will not go into too much detail concerning the more visually-orientated ▶

Studio L'Equipe's home-made Set Mag machines—*DASH* machine from Sony

facilities at the Rue Colonel Bourg. There are two digital 'telecinemas', one of which is equipped with the Rank *Ursa* system. This system is designed to treat all the broadcast video formats including D1. The *Leonardo* colour corrector is used in this room as opposed to the closely related *Da Vinci* which is used in the Rank *MK 3* room. The second telecinema also has 'Wet Gate', a process that is used in film laboratories to eliminate scratches and white spots on the film. *Regie 1* is a digital editing suite with a *Harriet* system which in conjunction with *D1* enables an unlimited number of generations. There is a *Kadenza* video mixer, a *Kaleidoscope* effects generator and a computerised paintbox with 13s of animation memory. The complementary Regie 2 is geared towards *Betacam SP* editing although it is also possible to work with the 1-inch format. L'Equipe also provide an extensive duplication service for all video formats and each copy is individually checked. ■

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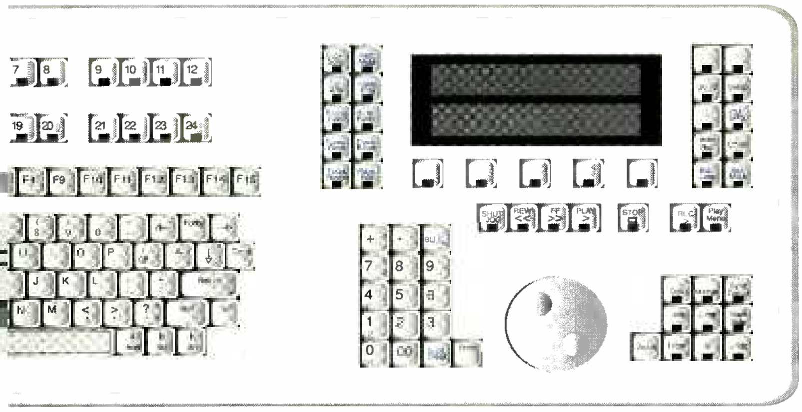

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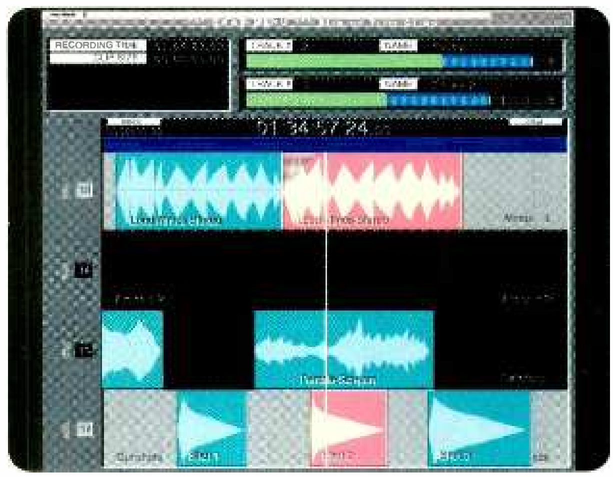

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WORKING WITHIN THE BIT BUDGET

With data reduction systems becoming an integral part of certain digital systems, Francis Rumsey looks at the possibilities and likely implications of their use in professional sound recording

There is no question about it: data-reduced audio and video are going to be major features of the future both in professional and consumer fields. This is a mixture of blessing and curse, and it will definitely be up to the end user to determine whether it ends up more one than the other. It will be a blessing because data reduction brings with it a number of benefits such as better cost-effectiveness in storage, and may make the transmission of digital sound and picture signals commercially and operationally viable; it will be a curse because it may take away forever what might be considered the one overriding merit of linear digital systems — that being signal-chain transparency. The latter 'curse', though, requires some discussion, since we are quickly moving away from the time when all linear digital audio systems were 16-bit, 44.1kHz, and did not process the sound in any way, towards a time where even linear PCM will have been through digital signal processing (DSP) of some sort, thus potentially having had its sound quality affected.

It is possible that 'transparent' linear PCM will turn out to have been a passing phenomenon of the early days of digital audio and video, and that mature digital systems, in order to be cost and spectrum-efficient, may almost all adopt some form of data reduction. In such a world it will no longer necessarily be true that keeping a signal in the digital domain throughout the production process ensures that quality is maintained. The result of this will be a reopening of the debate over subjective quality, which has taken slightly more of a back seat in recent years. It is not true that data reduction will be principally a feature of consumer systems such as MiniDisc and DCC — if the video world is anything to go by there will soon be a number of professional recording products on the market which use it. Provided that the professional is aware of the trade-offs involved, it is possible to take advantage of the benefits of such technology without incurring too many of the drawbacks.

The trade-off

Data reduction systems, as used in audio and video, rely both on exploiting perceptual phenomena such as masking, and on conventional data compression techniques as used in many computer systems. These principles were described in some detail in the article 'Aural Fibre', July 1992, *Studio Sound*, and thus will not be included in any detail here.

It is sufficient to say that, although such systems may not be fully 'transparent' in that the data which comes out is not the same as that which went in, they aim to pass the original signal with as little subjective degradation as possible. Just how far this is achievable depends on how much data reduction is attempted, and herein lies the trade-off. As the data rate is reduced, the quantising noise and distortion is allowed to rise in each of the narrow frequency bands into which the signal has been split, the intention being that this will always be kept beneath the threshold at which you can hear it, taking into account the characteristics of the audio signal at the time. A complex model of the human hearing process is used to estimate this threshold, and this is updated once every so-many milliseconds to model the nature of the changing signal. A very high listening level is assumed in the model (usually around 130dB), since this is the worst case. At lower listening levels the ear's sensitivity is poorer at the extremes of the spectrum, and thus the masking effect should be better.

The data reduction systems which we are considering work to meet a fixed and known target data rate, and thus the allocation of 'bits to bands' is really a form of short-term balancing of the 'bit budget', placing resources where they are needed most. Continuing to use the financial analogy, when there is a lot of money around (corresponding to a high bit rate) it is relatively easy to ensure that enough money is allocated to each worthy cause (each frequency band), but as the amount of money becomes smaller (lower bit rates) one has to be either cleverer at balancing the budget or more selective about where best to use the money, working on the basis of putting the money where it is needed most and withdrawing it where it will have the least damaging effects. The data reduction systems which are best at balancing the budget at low bit rates (such as ISO Layer 3) use very involved 'accounting procedures' to achieve good sound quality, but just as good accountants are expensive and their methods complex, so are such coders. It also takes longer to work out an effective strategy to allocate the budget when there is not much money, and this translates to a longer coding delay in very low bit-rate coders.

Without wishing to take this analogy too much further, there is one further parallel worth making, being that as the amount of money is reduced something eventually has to give. It is not possible to continue to supply the same quality of goods beyond a certain point, and economies must be made. Eventually people begin to notice, and the aim is to make the economies in such places that the consumers will not complain too much.

The problem of judging the appropriate trade-off between data rate and sound quality will soon be the end-user's problem, since all sorts of products will be on the market offering seemingly amazing numbers of channels or hours of recording time at ridiculous prices. It is not a new problem to the engineer, since sound quality has always depended to some extent on economics, but people have begun to get used to assigning sound quality a lower level of importance when choosing digital equipment for the simple reason that one 16-bit linear digital recorder has sounded much the same as any other, leading choice to be based more on format, features and robustness. (In case I offend those who would say that sound quality is still top of the list, I would add that, of course, it may be considered very important but when choosing between two digital audio recorders operating at the same rate and resolution any difference in sound quality is normally entirely due to the quality of convertor design and the ability of the system to provide a stable clock to the convertor, rather than being anything to do with differences in the way the data is handled.)

Back to analogue?

It is interesting to consider that what we may witness with data-reduced digital signals in the years to come represents a return to the problems of the days of analogue audio. By this I mean that it will become important again to assess whether a particular recorder or broadcast product will give the required sound quality both in the first generation and after many generations, since this will depend on the data reduction algorithm ▶

and the accuracy of the psychoacoustic model used to code the signal. Whereas with linear PCM one may copy a signal digitally without losing quality (provided that the source and destination operate at the same resolution), with data-reduced signals the subjective quality may become degraded by copying. Whether or not it does in practice depends on whether the coded signal is copied, or whether the coded signal is converted back to linear PCM first (see **Fig.1**). In order to allow copying of the coded signal there will arise the need for standard digital interfaces which handle data-reduced audio, since current interfaces work only with linear PCM. Alternatively, in recording and editing systems based on computer storage media, a simple copy of the compressed file transmitted over a network to its destination would have the desired result.

There are parallels here with analogue noise reduction. Dolby noise reduction systems rely on masking phenomena when processing an analogue signal to reduce the subjective annoyance of noise. Dolby *A* and *SR* treat the signal in a number of frequency bands, relying on masking to hide the noise where the signal is at a high level and only processing those bands where the noise will not be masked. If you copy a Dolby-encoded tape then there is less loss of sound quality if you copy the encoded signal, rather than decoding it and then re-encoding it, since every stage of coding and decoding may degrade the signal slightly.

The parallel with analogue noise reduction should be investigated further, though, because there is an important difference between the way it is used in normal practice and the way in which digital data reduction is used. On the surface it could be said that the two have entirely opposite effects, since the analogue noise reduction system is used to make sound quality better, whereas digital data reduction is taking a good signal and perhaps making it slightly worse! This is because analogue noise reduction is normally used without changing the characteristics of the channel (typically a tape track), and thus the processing has the effect of improving quality, whereas the aim of data reduction is to allow the use of lower bandwidth channels and lower resolution recording — in other words the processing is used to maintain quality while allowing the use of a poorer channel. A noise reduction system, though, could be used simply to maintain sound quality while allowing the use of a poorer analogue channel, just like the digital system, so there is really less difference between them than might appear from a first look.

Although quality losses can be minimised by ensuring that as few generations of coding and decoding as possible are involved in the production and transmission chain, there will inevitably be stages at which the signal must be converted back to linear PCM for processing. At the moment there is very little postprocessing that one can do in the data-reduced domain, since any operations like filtering or effects would require entirely different digital filters to those used on linear PCM signals, but it is possible that such products may be developed if a need is perceived, and this has already been proposed, along with a suggestion that considerable savings in DSP power could be gained by operating on data-reduced signals.[1] The question of whether to convert a coded signal back to linear PCM is similar to that of converting a linear PCM signal back to analogue, since one has to weigh up the need against the potential quality reduction. If a certain process is only available in another domain then there is really no alternative to converting the signal.

Sound quality

The answer to the question, 'Just how much will sound quality be affected?', is almost as difficult as the proverbial question, 'How long is a piece of string?', since it depends on the data reduction process used and by how much the data rate has been reduced. The concept of 'headroom' is important here, although it does not mean the same thing as headroom in the conventional sense (the number of dB between a reference level and the peak recording level). Perhaps a different term is more appropriate, 'coding margin', since it refers to how far below the hearing threshold the unwanted side effects of low bit-rate coding lie).

For example, in an ISO Layer 1 system operating at the relatively high data rate of 192kbit/s per channel there is plenty of coding margin in the case of nearly all audio signals that one could throw at it. In other words, the additional quantising noise products generated by the coding process are a long way below the masking threshold and in nearly all circumstances will not be heard — therefore there could be said to be a good coding margin. Each successive generation of coding and decoding will gradually raise the unwanted products in level until they appear above the masking threshold, at which point they will be heard. In a Layer 3 system operating at 64kbit/s the noise is much closer to the masking threshold and thus sound quality will be noticeably affected after only a small number of generations, and even in the first coded generation may be audible.

Brandenburg[1] has stated that so-called 'noiseless' or lossless coding may be used on audio signals with a maximum benefit of around a factor of two in data reduction, and in many cases less than this. Such techniques would allow perfect reconstruction of the signal but with a reduction in data rate that might not be worth having. He proposes that the better solution for professional studio applications is more likely to be a perceptual coding method, similar to the ISO processes already described, but using a transform coder with a greater coding margin and 'perfect' reconstruction of the spectral bands. Using such a technique he claims that reductions in data rate of a factor of four to six are quite realisable, while still allowing sufficient margin for a number of generations of coding-decoding and some postprocessing without a noticeable degradation in sound quality.

The moral of this story is that systems offering large amounts of data reduction, say greater than a factor of six, will be much more likely to show up coding artifacts after copying and postprocessing operations than systems offering only modest reductions of say a factor of four. In current terminology, then, systems operating at around 192kbit/s per channel or greater (provided that they have been designed properly) will be most suitable for professional recording applications.

Options and temptations

For the professional looking to cut costs it may be tempting to consider adopting consumer recording systems such as the MiniDisc (MD) and the Digital Compact Cassette (DCC) for original recordings, just as consumer formats such as R-DAT were adopted in the past. For speech this might be acceptable, although such recordings would normally have to be copied to another ▶

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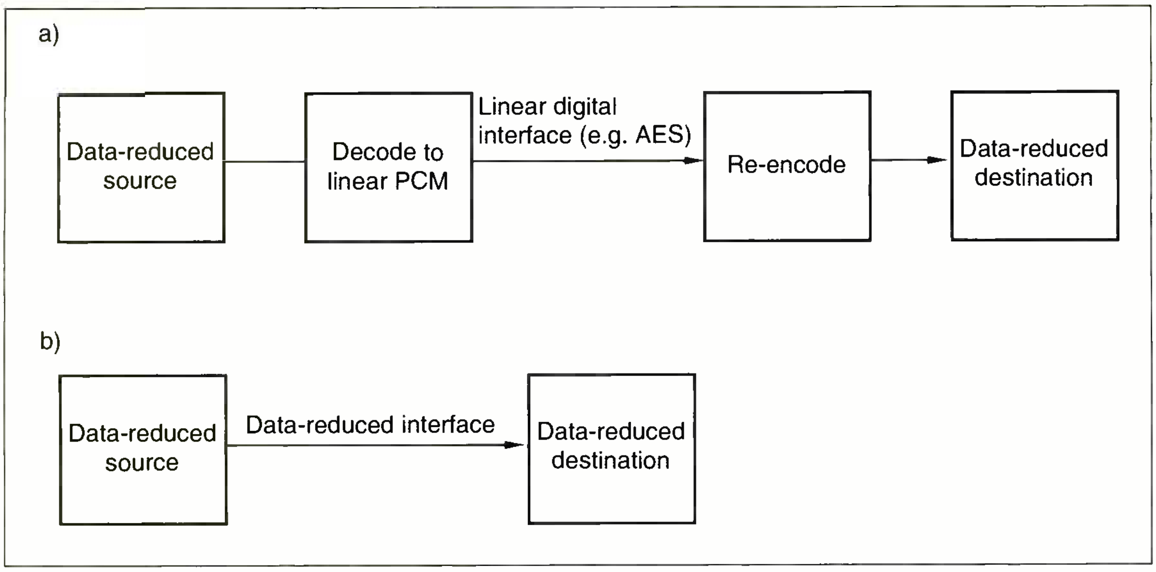

Fig.1: At (a) the data reduced source is decoded to linear PCM before being transferred via a conventional interface, and quality will be affected. At (b) the data reduced information is transferred without reconverting to linear PCM and no quality is lost

format for editing, but for original high quality music recording one would quickly find that side effects became noticed, especially since the recording might well have to be recoded for release, and possibly on the other format. (No-one knows yet what the effects of tandeming a DCC codec pair and a MD codec pair would be.)

It is likely that semiprofessional recording systems will begin to adopt data reduction — so look out for budget 24-track machines offering an hour of recording time on a Video 8 tape and assess them with care. It may be that the trade-off between cost and sound quality will be acceptable for some applications, but be clear about what you actually want to do with such a machine first.

Be on your guard for data-reduced hard-disk or optical disk editors, again offering large numbers of tracks and long storage time from disks of only modest size. Often the fact that they use data reduction is hidden away in the specification somewhere, where only the eagle-eyed may find it. Again it will all come down to paying your money and making your choice, but be sure that the degree of data reduction used is not going to leave you 'up the creek without a paddle' when it comes to postprocessing, and check that the resolution of editing is fine enough for your needs, since data-reduced editors often only allow editing to block resolution (24 ms in ISO Layer 2 and 3 systems).

Real benefits

The real gains from data reduction will come where the trade-offs between cost and sound quality are known and controlled, and where the number of codec pairs in the signal chain can be predicted. Unlike recording studios where sound quality tends to be the top priority, the broadcasting world has long been in the business of working with trade-offs between bandwidth and sound quality, since it is a feature of the communications industry. If by using a data reduction system more channels can be carried down a given line with little or no change in sound quality, then the system may well be adopted. If the tried and tested but long-in-the-tooth cart machine can be replaced by one using computer disks with almost instant cueing, then all the better. If all the day's programmes can be archived to a computer data cartridge with acceptable sound quality, again there is a strong argument for adoption.

These latter areas are the ones where data reduction is being quickly adopted. Tandberg Data recently launched the *TDC 9200* series system for archiving broadcast programmes onto computer ¼-inch cartridges (QICs), offering over 24 hours of storage on a single 1Gb QIC. This system uses the *MUSICAM* compression algorithm on which ISO Layers 1 and 2 are based. There are also a number of broadcast cart machines using computer disks or even solid state memory cartridges, from companies such as 360 Systems (using Dolby *AC-2* coding) and Barco-EMT (using *MUSICAM*). Such systems are not for production or postproduction as such, they are for a specific purpose where a single generation is the norm, and where the sound quality of the data-compressed digital cart is better than that of the analogue magnetic tape it replaces, and more operationally flexible.

Digital audio broadcasting (DAB) is the big application for which a lot of this work has been done, and we should begin to see services appearing in Europe by around the end of 1995 or early 1996 if things go to plan. Again one is dealing here with data reduction used for the specific purpose of carrying a lot of stereo channels in a small spectrum space, and where the number of generations of coding and decoding is known and controlled. Here ISO Layer 2 coding is used at a rate of 128kbit/s per channel, giving a sound quality better than that of FM radio and with vastly improved immunity to noise and interference.

The techniques will also be used widely in digital surround sound for film (such as in the Dolby *SR-D* format), and for the sound channels accompanying digital HDTV transmissions. (In passing it is worth noting that the BBC recently demonstrated digital HDTV pictures compressed by a factor of 45 times such that the picture could be transmitted over a conventional 8MHz TV channel with seemingly minor effects on picture quality, although for the time being the coding for seven seconds of video has to be done on the mainframe computer overnight due to the amount of processing involved!) Don't be surprised also if you begin to see consumer video recorders with surround sound digital recording using data reduction, as a spin-off from Dolby's *AC-3* technology.

Professional music recording has yet to feel the benefits of data reduction — although it may yet do so. Provided that users are prepared to accept relatively modest reductions in data rate it may be that significant improvements in operational flexibility and cost-performance ratio may result. For example, even if the rate reduction were only a factor of four, which would offer plenty of coding margin and high sound quality, it would be possible to turn a four-track optical disk recorder into a 16-track optical disk recorder. Such technology makes the concept of cost-effective disk-based multitrack machines with plenty of recording time a realistic proposition. ■

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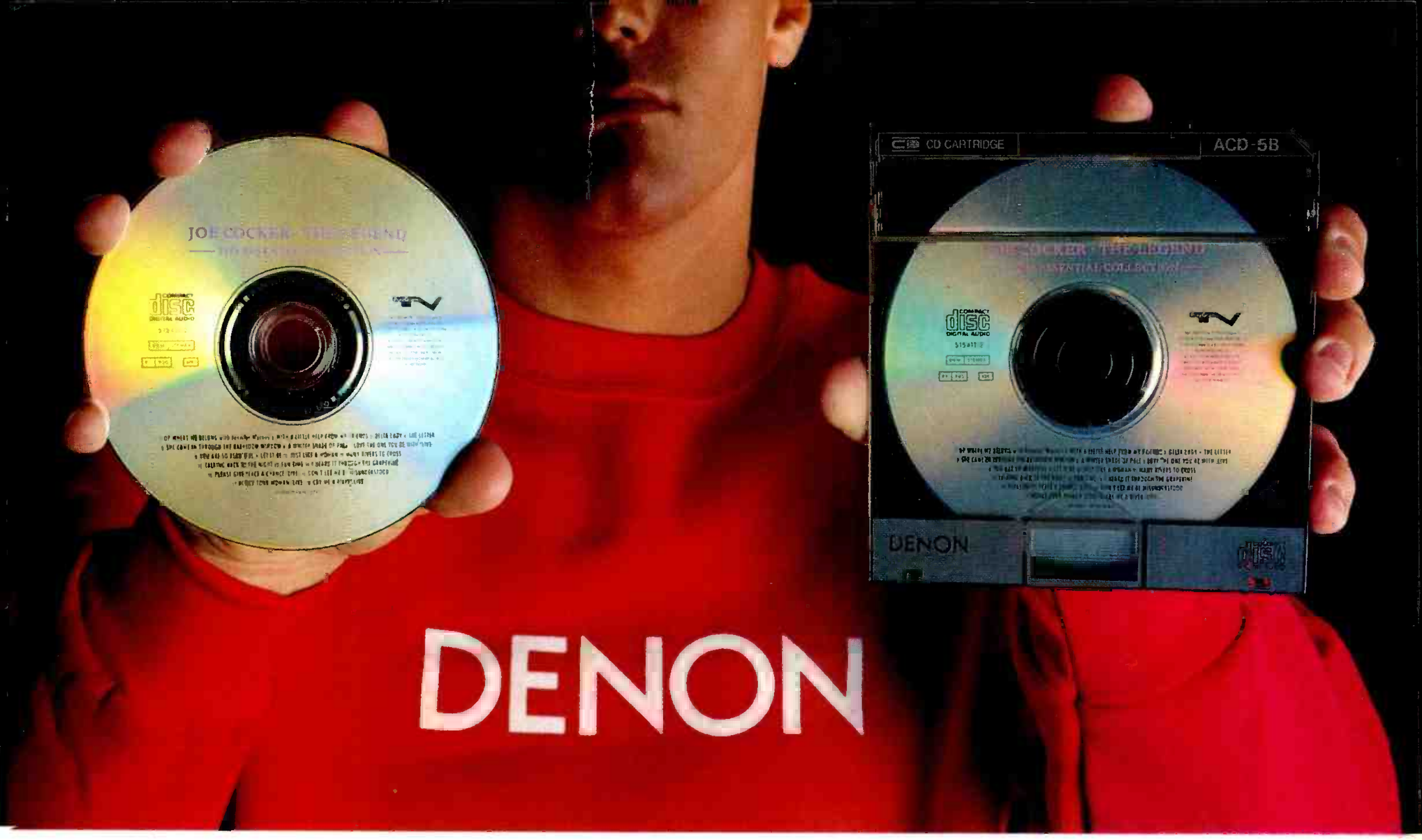
1. Brandenburg, K. and Herre, J. (1992) *Digital audio compression for professional applications.* Presented at the 92nd AES Convention, March 24–27th, Vienna.



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Remember the Armchair Record Store, the plan to use digital audio broadcasting channels for piping scrambled music into the home? Listeners pay to unscramble the signal and record it on digital tape or blank disc instead of going out to buy records from a retail store. The same music is broadcast on more than one channel, each running a few minutes behind another to give people the chance to buy on impulse after hearing a preview.

The television industry is now thinking along similar lines. With digital video and data compression, one analogue TV channel can carry four or more digital TV channels of comparable quality. Already there are plans for 500-channel cable services. Stations cannot afford to put a different programme on each channel, but they can put the same film on several channels, each delayed by a quarter of an hour. So viewers get an Armchair Video Store.

Technically there are no long-term bars to implementation. But ARS and AVS customers would miss out on the artwork and liner notes that come with retailed discs and tapes. Simple, says an engineer friend I met in a record shop recently, as soon as the customer downloads the material and pays for it by account or credit card, the station mails out the liner notes and artwork to arrive a few days later for tucking into the jacket of whatever blank medium was used to record the programme material.

In the USA, Bose run an enormous mail order outfit called Music Express which claims to offer 'all recordings currently available' on CD, compact cassette and video cassette. It would be very easy for someone like Music Express to tie in with an ARS or AVS, and mail out liners and artwork.

While the consumer market endlessly debates the relative merits and demerits of MiniDisc and DCC, professionals are locking into a debate on the best way to get sound between digital editing workstations. One way is by shanks's pony—dump data from magnetic fixed disc to removable magnetic or magneto-optical disc (or even tape), then carry it down the corridor, or put it on a bike. The alternative is to send it by network wire, or better, optical fibre. Sonic Solutions were early into optical storage, but have gone for the network option. Sonic's Bob Doris calls the rejected option a 'sneaker net'. But it's horses for courses.

Networking will allow several people in different parts of the same building, or in different buildings, to work on the same sound recordings, at the same time, without interfering with each other, or altering the master recording. So several editors can produce several versions of the same record, and let the producer judge which is best. Film studios can take sound effects from one store and mix them with dialogue and music taken from other stores.

But the option does not come cheap; Sonic's editing stations cost between $15,000 and $85,000, depending on the number of sound channels to be handled. NoNoise puts the price up to around $90,000. Connecting the station to a SonicNet adds around $6,000. It all makes sense if you are planning to spread the net wide, with a large number of workstation nodes, for instance round a

Barry Fox

Are you sitting comfortably? Then prepare to buy records and videos from home and edit Studio 1's recordings from Studio 3

studio complex. But it makes little sense in a small operation where there is no need for two people to work on the same material at the same time.

If you are planning ahead it may pay to understand how technology like SonicNet works and where it will surely lead in the future. The net handles sound like computer text in an office complex and shares it round a ring of optical fibre which can stretch 200km, through 1000 workstations, each up to 2km apart. The sound signals circulating in the ring are clones of the original master recordings which remain untouched, in stores round the ring. So each workstation can reproduce sound from the same store, completely out of sync with each other and with different parts of the recording cut and spliced to alter its playing time.

The network follows the FDDI, Fibre Distributed DTA Interface, standard set by the computer industry for high speed data transfer. Data streams at 100Mb/s which is fast enough to carry 80 channels of CD-quality sound simultaneously. FDDI works on the token ring principle. At any given moment only one workstation is allocated an electronic token which lets it transmit data into the ring. So there can be no collision of data from different stations. Other ring systems, such as Ethernet, wait for collisions to occur and then resend the corrupted data. This is acceptable for text but may make the ring too slow for digital audio. Because FDDI is an industry standard, there are now three chip suppliers and prices are falling. Bob Doris predicts that by the mid 1990s it will cost only $1,000 to make an audio workstation a net node.

All this ties in neatly with recent developments in off-line video editing. These let a producer play

Networking audio will allow different people to work on the same recordings at the same time without altering the master recording

around cheaply with edit points in programme material by using only sub-broadcast-quality copies of the source material. The final selected edit points are marked with time-coded instructions. An on-line editor then uses the time code instructions to assemble the selected sections by copying them from the source tape to programme master tape.

The time code instructions are stored on floppy disk, or sent by wire between the off and on-line systems. Until recently the off-line systems have worked with tape. What the new systems, like Avid and Lightworks do is store a working copy of the source material either in solid state memory, or on a magnetic or optical disk. The store is controlled by a computer, either (as in the case of Avid) an Apple *Mac*, or (as Lightworks) dedicated hardware. The computer pulls sections of the recording out of its memory of disks and displays them as moving video on screen, either in a small window, or as a full screen display. By using small windows, the system can show a mosaic of picture sequences, like a menu.

When broadcast-quality television pictures are converted into digital code, the data stream runs at over 200Mbits/s. Divide that by eight, and you get the number of 25Mb/s. So drastic compression is needed to get useful lengths of movie material on a hard disc.

The MPEG (Moving Pictures Expert Group) standard is used to bring Full Motion Video to CD-I. MPEG compress the video signal by a factor of over 160, to around 1.2Mb/s. The new systems do not, however, use MPEG. Instead they use the JPEG (Joint Photographic Experts Group) standard for still-picture storage. This is because the MPEG standard compresses moving video by comparing each picture in the motion sequence with the pictures which come before and after it. This lets the coder throw away information, like the still background behind a moving object, which remains much the same through the sequence. But this means that the decoder can only work by comparing whole strings of pictures to rebuild the information that was thrown away. So MPEG is no use for applications where the user wants to zip quickly through programme material, both backwards and forwards, an display still pictures. The decoder does not have the information it needs to rebuild the picture. But a JPEG system treats each picture as a still. It compresses the image by comparing different parts of the same picture, and throwing away information which remains the same across the frame — like a wash of blue sky or a white wall. The decoder just codes changes in the one picture.

With any compression system, the greater the ratio of compression, the greater the risk of picture quality loss, but the more storage time you get from a disc.

At one extreme each picture is compressed down to around 5Kb, which—with stereo sound—allows over an hour of storage per Gb of disk space. At the other extreme, each picture gets 80Kb. The system them manages only around ten minutes storage per gigabyte, but quality can be good enough for on-line editing.

All this is still well within the 100Mb/s capacity of a fibre ring. So the day is not far off when one ring can be used both for sound and picture networking. ■

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Martin Polon

The new music formats: good enough for the pros, or just the public?

As we find ourselves in the digital world of the 1990s, the battle for the quality of recorded music may well be decided by factors other than technological superiority. What happens in the home could decide the future of the audio recording workplace. Will home delivery centre on new, innovative music responsive to the consumer? Or will some compromise of old and new technologies and record company back catalogue be accepted as 'good enough'?

The battle between Digital Compact Cassette and MiniDisc may be the deciding engagement in describing the direction of the audio industry. The issue that is of such concern to the studio community is the potential for record company acceptance of a technologically inferior (to CD) standard for recording and reproducing digital audio; albeit in the home. This technology, common, though in different implementation, to both DCC and the MD, is that of data compression necessary to reduce the storage space occupied by the recorded signal.

It might be sufficient to say that the new formats will be 'good enough'. And every indication is that for the home, the car and for recreation, they will be very good indeed. The technology and chips used to reduce the amount of data placed on the recording medium for DCC operate with 24-bit precision in 32 sub-bands. The MD system utilises variable time segments and encoding driving a 200°C laser. Very sexy stuff. The new formats offer to supply the portable digital audio needs of the consumer better than anything else to date. But will they be 'good enough' to justify their usage and influence on other segments of the audio and recording industry?

The encoding (and some other technologies) used for these systems is not the same but, generically speaking, there are more features in common than not. Both DCC and MD digitise an incoming digital signal; both then quantise again to code the signal —Precision Adaptive Sub-band Coding (PASC) in the case of DCC and Adaptive TRansform Acoustic Coding (ATRAC) for MD. These protocols utilise psychoacoustic masking techniques to remove those portions of the recorded signal that will be essentially inaudible. By not recording sounds that would not be heard, the amount of data to be recorded is reduced (by 75% for DCC and by 80% for MD).

Much pro audio concern focuses on the limits experienced by both systems in capturing all of the information present at the input (analogue or digital). The effect of increased quantisation noise inherent in digital recording on reduced-size formats is also of concern. Designers insist that masking reduces audible quantisation artifacts to levels equal to, or better than those on CD. The possibility for synergistic degradation of signal quality due to the multiplication of the quantisation process in making multiple copies using the same system also is of concern.

For many in the audio community, A&R practices are also a cause for concern. Most feared is an extension of the current 'out-of-studio' evaluation of a mix or session on home or in-car equipment. It is controversial today even though it has become commonplace. Philips raised the Pro-DCC flag at the 1992 San Francisco AES convention—presumably to capture this studio 'outboard' marketplace—although several Philips' reps in attendance were already discussing the 'advantages' of Pro-DCC.

Another side of the argument is presented by the former owner of a successful recording complex: 'This could, and probably will, mean that each studio will have to have DCC, MD, DAT and Dolby *S* to provide tapes for clients. Each format will have to be premastered to compensate for the differences between systems and you will have the same kind of psychology that dominates the recording process, in that groups who have used Dolby *SR* for their session will want to hear the results on Dolby *S* rather than on DCC.'

The threat voiced most often by those in the record companies not directly connected to either the Sony or Philips digital hardware formats (and even some who are), is that the concomitant new software technology could threaten CD. On the other side of the equation, there is much energy being expended in evolving a 'downwardly-compatible Super CD'. This would offer 20 to 24-bit quality to enhance the quality of compact disc without requiring new playback hardware.

The impact of new music formats to be stocked by record retailers further exacerbates the problem of insufficient stock held by the mall or high street record retailer. The single most important change in the population base (and audience) since World War II is the aging of the baby-boomers, with their median age approaching 40. Two-thirds of the population controlling three-quarters of the 'disposable' income are over 30, with significant percentages over 40, 50 and 60. Yet the majority of new record releases are focused specifically on the under-30s. The small stocks of CDs (4000–10,000 on average) carried by mall or high street record chains who do 80% of the total retail business mandate that about 80% of their CD titles focus on their under-30s. They do not have the space to hold in-depth stock.

It is curious to note that the current ratio of 'old' music to 'new' or previously unreleased music is approximately 2:1. This means that for every record or tape containing new material, two units of previously released material are made available. Some critics think the ratio may be as high as 5:1 or even 10:1. In any case, the assumption in the record retail industry is that at least 50% of all music sold during 1992 was a rerelease. That means the emergence of the two new formats would most likely see 'catalogue building', similar to that taking place on CD. The impact of existing back-catalogue release and distribution on studio time can be seen from the fact that the income figure for all studios in 1977 derived from recording projects was about 80% of the total billings for that year. In 1977, the back catalogue was nowhere as powerful a force as it is today. In 1992 by contrast, the figure for recording project time as a percentage of total billings has moved down to the 25% range.

'at least 50% of all music sold during 1992 was a rerelease'

The duplication industry also faces risks in embracing one or both of the new formats before a clear indication exists as to which format has the legs to go the distance. DCC and MD manufacturers would prefer duplicators to use mass storage devices based on dynamic computer memory rather than any kind of moving-tape system. There also is significant disapproval of duplicators using banks of single MD or DCC units to produce copies—a method still used for copying analogue cassettes and VHS video tapes. This requires duplicators to make considerable investment in duplicating plant for MD and in equally expensive technology for DCC.

Said one investment banker approached to fund new facilities for DCC and MD technologies: 'It is a speculative investment at this time. If it were not for the courage of a few independents and of the CD system supporters Sony and Philips, plus the Philips' relationship with chemical giant Du Pont, there would have been no successful introduction of the CD. And there was only one system. With MD and DCC competing, all that you have from an investment point of view is a formula for potential financial disaster if you pick the wrong system!'

In the past, consumers have waited for a clear choice. Nothing could be worse for the various industries involved. The consumer electronics market place would see no significant economies of scale in production to lower retail prices. Record labels would not be able to broaden their catalogue for the new formats. Record retailers would have to reduce stock of existing formats to carry software for the new systems that would not move off the shelves. Duplicators would have to accept the high price of mass duplicating equipment for production demands that might not justify the investment. And the recording studio community would continue to suffer from a frozen audio and record industry.

The bottom line is that there can be no compromise with quality or the consumer's sense of long-term stability. Consumer confusion over analogue cassette, Dolby *S*, DAT, DCC, MD and CD-R is not 'good enough'. Let's not lose sight of what has been achieved with the analogue cassette; a standard for the marketplace that has produced sales of billions of units over the last 20 years. Let's support a similar digital standard of prosperity for the recording industry. ■

DIVIDE AND CONQUER

Sam Wise charts the mysteries of the Yamaha *D2040* digital channel divider

The Yamaha *D2040* is a self-contained processing system for multiway loudspeaker systems, contained in a two-unit-high, rack or table-mounting, steel enclosure. Internally, all operations are performed in the digital domain, while inputs can be either digital or analogue. Except for the potential requirement for a 1/3-octave or multiband parametric equaliser at the front, for room equalisation, everything is provided to align a loudspeaker: including delay, electronic crossover, two-band parametric equalisation, and a compressor-limiter. The *D2040* is not the first digital loudspeaker processor, having been preceded in the marketplace by the modular TOA *SAORI* system. It is, however, cost-effective for most applications, being less expensive than a similarly equipped system using a digital delay plus high quality crossover and some basic equalisation. In addition, it matches an analogue system in dynamic range.

The device is essentially stereo, with two inputs, each having four outputs. Since each output is electronically identical, this provides quite a degree of versatility in operation. See the block diagrams in **Fig. 20**. Though only two inputs are provided, the related outputs can be configured to provide four two-way outputs, or two three-way outputs plus two additional full-range or band-limited outputs, or two 4-way outputs. This is very useful for sound reinforcement systems as we have found on recent designs.

The *D2040* is not only appropriate for sound reinforcement. It is already in use for three and four-way studio monitoring systems. In a well designed room, with well designed monitors, even extra equalisers are not likely to be required.

The main filter and equalisation effects are quite useful. **Fig.1** shows a typical four-band crossover setup, while **Figs.2** and **3** indicate the capabilities of the two bands of parametric equalisation in each channel. Virtually any reasonable loudspeaker crossover system can be implemented with the *D2040*.

Construction

On arrival, the *D2040* appeared to have a fault. From the left channel input, outputs 1 and 2 were not properly operational. Suspecting internal disorder caused by shipping, we examined the unit's interior. The top cover is easy to remove by extracting only seven screws. The bottom cover is similar. Internally, there is a steel front-to-rear divider separating the power supply section from the rest of the electronics, and providing stiffening for the box.

The power supply is contained on one PCB, with substantial filtering in evidence between the mains power supply input and the switch-mode power supply providing the internal DC power. It looks like it is designed to meet the new EMI-EMC regulations in the EEC. The mains power lead is captive, and as reviewed the unit has a fixed mains voltage of 240V. The chassis and internal ground are connected to mains safety earth. This did not produce any symptoms of earth loops on our measurement or listening test setups.

A large PCB accessible from the top covers virtually the entire remaining area of the unit. This handles the A–D and most digitally related functions. Removing the bottom cover reveals a similarly large PCB with the D–A, analogue level control, regulator ICs for the analogue circuitry and output amplifiers.

At the front, there is another PCB with eight motor-driven rotary level controls, a PCB for the front-panel-mounted, channel-related switches, and another for the display and master switch section—this latter resembling many from other Yamaha products. At the rear another PCB holds the analogue output connectors.

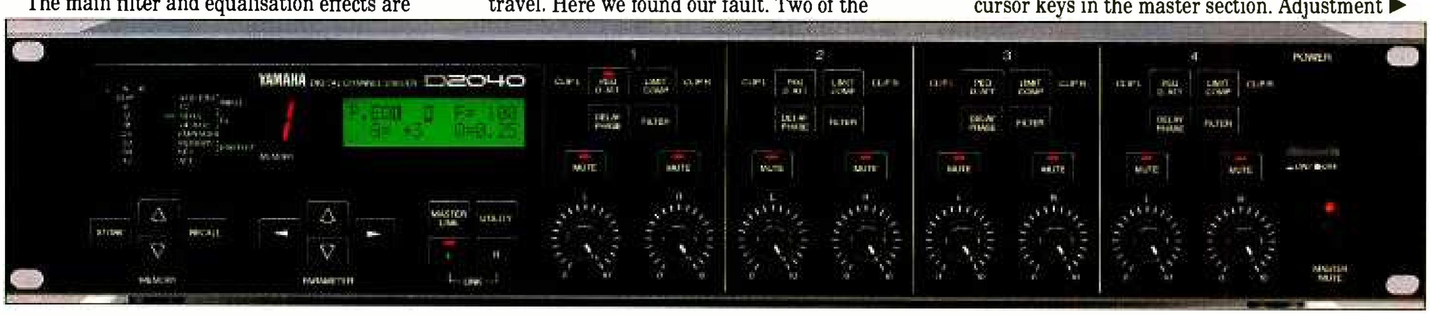
Most PCBs are made of fibreglass for increased strength and are thoroughly legended, easing servicing. All wiring is very tidy, and connected by multi-pin plugs and sockets. Servicing is easy, while all PCBs are well braced to withstand travel. Here we found our fault. Two of the connectors linking to the motorised level controls were hanging off. Once these were pushed back into place, they appeared to be well retained. We can only suppose that they were left off or not pushed fully home on assembly.

Operation

The front panel has five main sections. At the left are the master switches, level meters and two-line text display. To the right of this are four identical sections, each controlling a stereo pair of outputs. In each section, a cluster of four push-button switches allows selection of the parameter of that section which is to be adjusted. These are self-explanatory, having no more than three menu selections linked to each switch. Adjustment can be made individually to either left or right outputs as selected by master L and R switches. Pushing both L and R together allows the stereo pair to be identically set. The first switch allows adjustment of PEQ (two-band parametric equaliser), and D.ATT (control of internal digital attenuators from +6 to -18dB). The next is LIMIT-COMP which controls ratio, attack time, decay time, and threshold for the compressor-limiter section. Below is DELAY-POLARITY which is used to set the path delay in 21µS steps and to invert the polarity. And last is FILTER which sets the slope and frequency of the crossover filters, along with the loss at the selected frequency. Adjacent to the switches are a pair of CLIP led indicators, detecting clipping in the digital processing stages of the channel.

Below the channel-switch clusters are individual mute switches and output level attenuators for the left and right outputs. The attenuators are motor driven and operate on the analogue portion of the output, therefore they reduce channel noise as they are turned down. A FADER LINK switch in the master section of the *D2040*, used together with the L and R switches allows left, right or all channel gains to be altered together by turning the channel 1 controls. Any offsets in gain from channel to channel are retained as gain is adjusted.

When any of the switches are selected, its function is shown on the alphanumeric display, and adjusted by selecting the correct subparameter using left and right PARAMETER cursor keys in the master section. Adjustment ▶



Yamaha D2040—used by Andy Munro in Air Lyndhurst's Studio 1

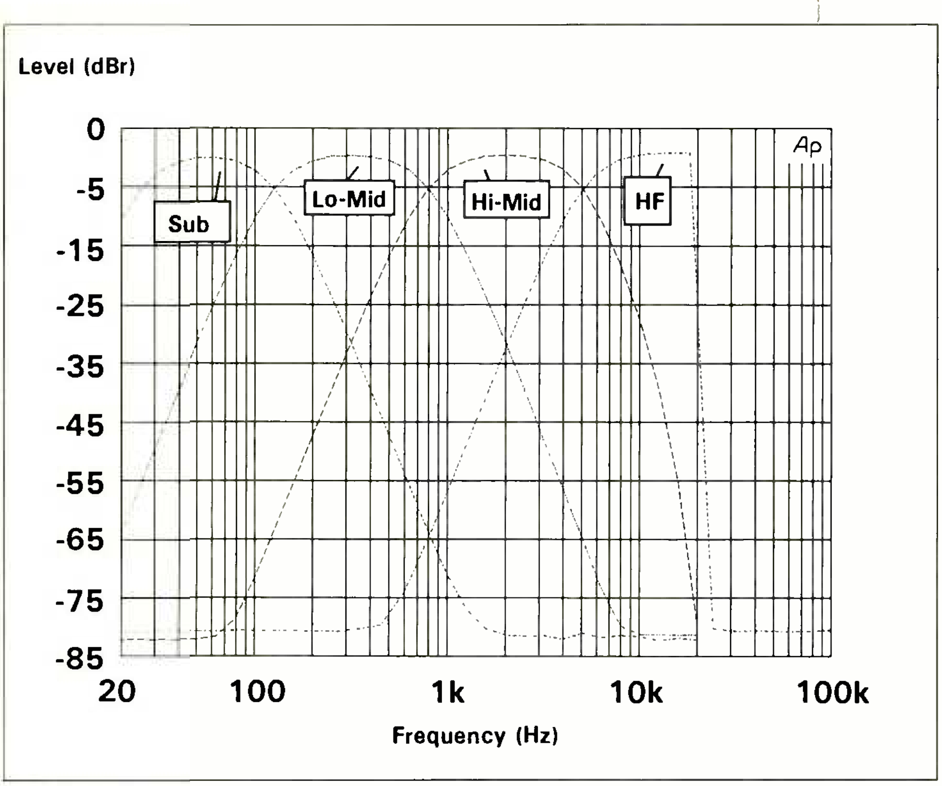


Fig.1: Typical four-way crossover characteristics

is then made using the UP and DOWN cursor keys. In use, these are not so nice as the BSS wheel, but neither is it quite as easy to get confused.

Filter adjustments are by numbers, and do not give a graphics display of shape. For the purpose, that is good enough.

The remaining control functions, all located in the master section, are UTILITY and the MEMORY cluster of controls. UTILITY steps through a large number of menu selections, being the 'catch-all' switch for everything not regularly accessed. It provides step control of output gains, 16-character titling for memory settings, software protect mode (which just prevents alteration due to fiddling but is readily overridden), input source and emphasis selection, second-meter-feet display of delay settings, parameter copy, and finally RS485 port addressing and baud rate selection. Parameter copy allows settings to be transferred between left and right outputs or vice versa, within one output channel only.

Operationally, the RS485 interface is somewhat MIDI-like. Each *D2040* has two address settings, its own local address for receiving information, and a remote address to which its front panel controls can send information. This allows one *D2040* to act as a master, controlling the detailed settings of up to 31 other *D2040*s individually, in groups, or all together. When recall of presets only is required, 128 program-change numbers are provided, which can be linked to select any of the 16 internal memories of the *D2040*. BULK DUMP allows all internal settings of the *D2040* to be transferred to a remote device.

Electrically, RS485 is utterly different from MIDI. This is a true serial bus system, where each device can just sit on the bus. No loop-throughs are required and therefore there can be no build-up of delays. In addition, RS485 is a balanced system and can run long distances given that the correct cable is chosen. RS485 is the underlying technology for twisted pair LAN systems. Unfortunately, as implemented on the *D2040*, there is no 'multimaster' communications method provided. In real LANs, software can sense bus activity and avoid having messages collide. For the *D2040* no such system exists, so bus control is crude, but definitely usable.

The MEMORY switches are easy to use, with UP and DOWN cursor keys used to select a memory ▶

MANUFACTURER'S SPECIFICATION

Analogue Inputs
2 channels, electronically balanced, XLR
+4dBm nominal input, +24dBm maximum
20kΩ input impedance
Analogue Outputs
8 channels, electronically balanced, XLR
+4dBm nominal output, +24dBm maximum
150Ω source impedance

Digital Inputs
AES-EBU-PRO balanced input with XLR
Yamaha Y2 (8-pin DIN connector)

Sampling Frequency
48kHz to 44.1kHz, 32kHz

A–D Convertors
19 bits x 2

D–A Convertors
20 bits x 8

Frequency Response
20Hz to 20 kHz at 48kHz sampling

S-N Ratio
110dB typical (emphasis ON)
107dB typical (emphasis OFF)

THD
<0.005% (emphasis ON) at 1kHz
<0.007% (emphasis OFF) at 1kHz

Memories
15

Control Interface
RS-485, 9600 to 38400 baud, XLR connectors

Analogue Output Attenuators
Motor drive, servo

Digital Level Control
-18dB to 6dB, 0.5dB steps

Filters (per channel)
HPF frequency: 20Hz to 16kHz (per channel)
HPF slope: -24, -18, -12, -6dB/oct, THRU
HPF gain at cutoff frequency: -6,-5,-4,-3
LPF frequency: 20Hz to 16kHz
LPF slope: -24, -18, -12, -6 dB/oct, THRU
LPF gain at cutoff frequency: -6,-5,-4,-3

Parametric EQ (per channel)
2 bands
Frequency: 20Hz to 16kHz, ⅙-oct steps
Gain: -18dB to 18dB, 1dB steps
Q: 0.5 to 10

Polarity (per channel)
Normal-Reverse
Delay (per channel)
0 to 1,365,313ms, 21µs steps (at 48kHz sampling)
0 to 1,486,054ms, 23µs steps (at 48kHz sampling)
0 to 2,047,969ms, 31µs steps (at 48kHz sampling)

Compressor-Limiter (per ch)
Threshold: 0dB to 20dB
Compression Ratio: 1:1 to ∞:1
Attack Time: 1 to 20ms
Release Time: 0.01 to 5s

Muting (per channel)

Level Indication (per input)
7-segment LED indicators x 2

Clip Indication (per channel)
Red LED, pre D–A convertor

Dimensions (WxHxD)
480 x 101 x 389.6mm (18⅞ x 4 x 15⅜ inches)

Weight
8.5kg (18lbs 12oz)

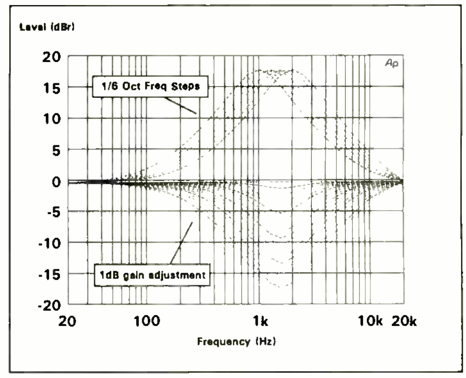


Fig.2: Parametric equaliser—frequency and level variation

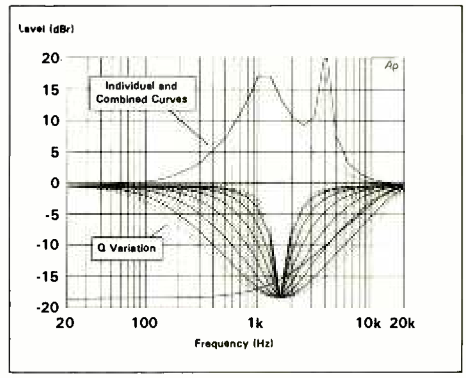


Fig.3: Parametric equaliser —combining effects and variation of bandwidth

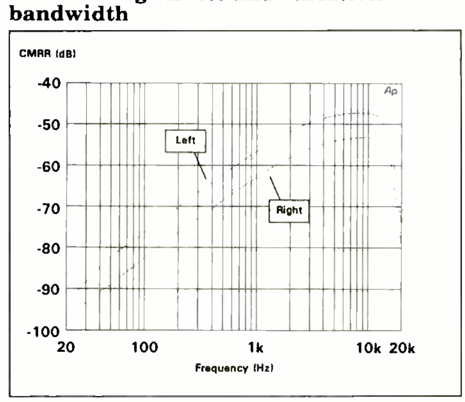
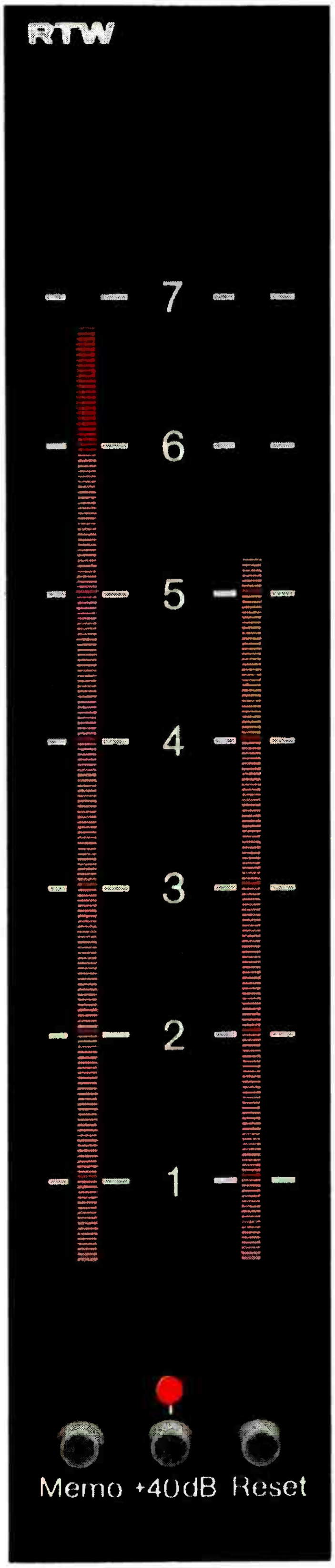


Fig.4: Input Common Mode Rejection Ratio

Peak Viewing



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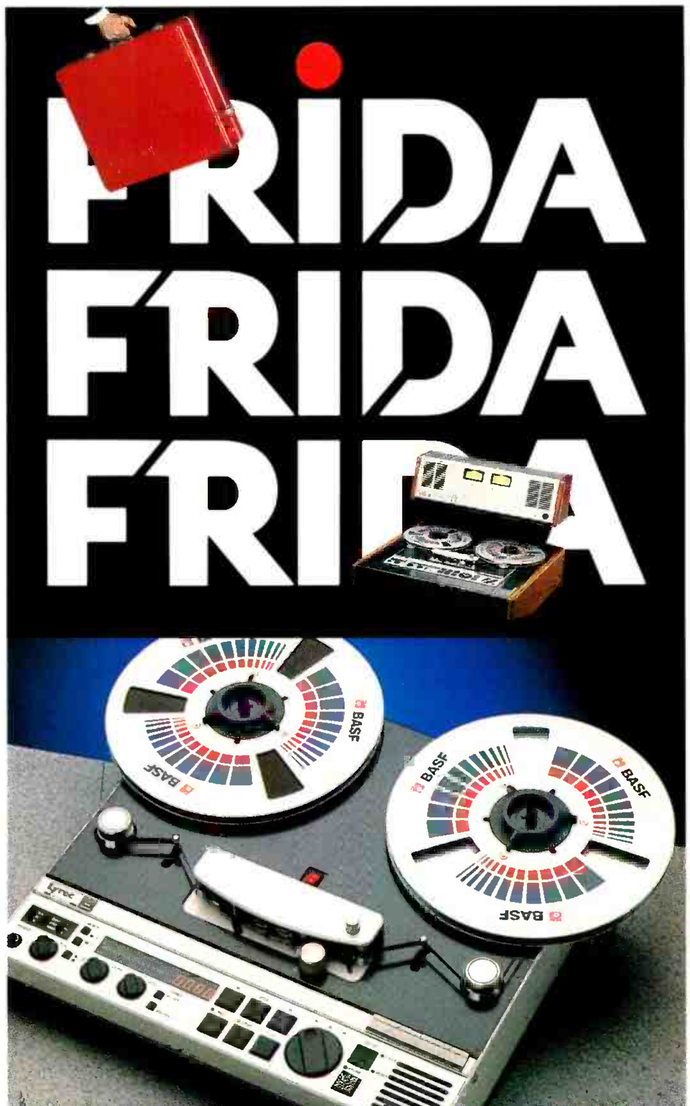

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TABLE 1: WIDE-BAND NOISE MEASUREMENTS

Inputs	Noise Levels dBu 22Hz to 22kHz RMS	400Hz to 22kHz RMS	22Hz to 22kHz RMS A wtd	22Hz to 22kHz Q-Peak	CCIR 468-3 Wtd	CCIR-ARM
Analogue L	-86.4	-87.0	-87.5	-82.2	-75.5	-86.5
Analogue R	-86.3	-87.0	-87.5	-82.2	-75.5	-86.5
Digital L	-85.0	-85.2	-87.1	-80.0	-74.1	-85.0
Digital R	-84.4	-84.6	-86.7	-80.3	-73.8	-84.5

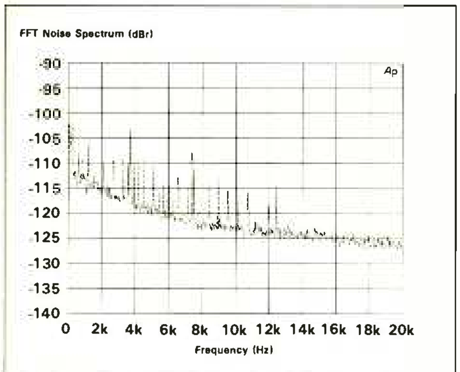

Fig.5: Noise spectrum, analogue input terminated in 50Ω emphasis ON

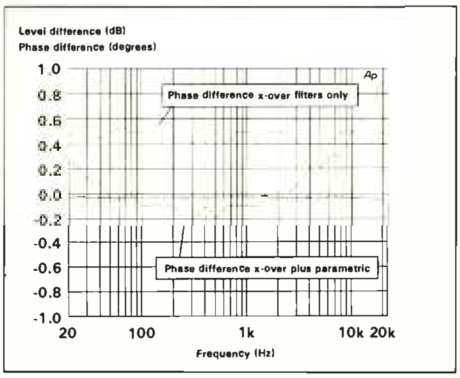

Fig.6: Phase and level difference, single channel left and right outputs. Crossover 800Hz and 5kHz, 24dB/oct slopes parametric equalisers: +6dB at 1.6kHz, Q=1 and -6dB at 3.2kHz, Q=7

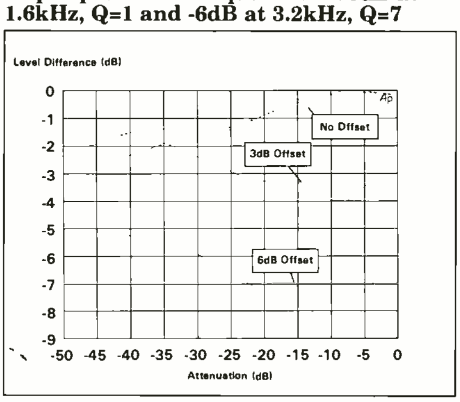

Fig.7: Tracking of output level attenuators

number. The description of the contents as set up in the utility mode, are shown on the alphanumeric display. RECALL of a memory can take up to 10 seconds, mainly waiting on the motorised output attenuators to reach their new positions, but the transition is smooth. A minor software bug was noticed here. Pressing RECALL without changing the memory number results in a momentary alteration in system configuration, temporarily affecting levels. Pressing STORE asks for confirmation by another press before changing memory contents.

There are only two rear-panel controls. Real 'fiddle' protection is provided by a recessed PROTECT switch. This has an Off position which allows access to all functions; a Memory position which allows front panel fiddling but prevents the overwriting of memory; and a Key position which locks the entire front panel and prevents external control. In this position, the level controls will automatically return to set positions if moved. The other slide switch sets nominal output level between +4 and -6dB. It actually reduces the level by 9.6dB in the -6dB position, reducing output noise by the same amount—therefore maintaining dynamic range. This is a useful feature.

The stereo input meter is calibrated in dB below clipping, and is virtually instantaneous peak reading, displaying the correct level for a one cycle burst of 4kHz.

All in all, we found the unit easy to operate without reading the manual. Only the RS485 information needed to be read before correct operation could be made. The manual itself is clear and concise.

Inputs and outputs

All analogue and digital audio inputs and outputs are balanced and provided on XLR type connectors, as are the remote control RS485 connections. Analogue input common mode rejection is shown in **Fig.4**. Performance is good, exceeding 75dB at 100Hz. The input clip indicator illuminates at a steady-state level of 23.8dBu and a clipping level for 0.3% THD is reached at +23.9dBu, within measurement accuracy of the specified +24dBm.

Maximum output level is +23.8dBm (600Ω load) or +24.9 into 100kΩ at 1kHz, more than sufficient and again meeting the specification.

Crosstalk was measured from each of the two inputs to their respective channel-3 outputs, using an 10kHz input level of +20dBu. The result is crosstalk of -73dB from right to left and -79dB from left to right. This figure is not specified but should be adequate for all practical loudspeaker applications.

Noise and dynamic range

With all output volume attenuators set to maximum, and digital gain set to zero, the analogue input to output gain of the system is 0.1dB with a 100kΩ load—in other words, unity gain. Measuring noise band limited from 22Hz to 22kHz, RMS, unweighted, gives a result of -88.9dBu with emphasis ON. With a maximum output level of +23.8, this gives a dynamic range of 112.7dB, an excellent performance.

Reducing the output level by 20dB using the output attenuators gives a noise level of -100.7dBu, retaining a dynamic range of over 100dB. This noise level is due to the output stage alone, and does not decrease further as the output level is reduced.

With a digital input of 0dBFS (0dB relative to full scale), the output level is 23.1dBu. Reducing this to the level at which many power amplifiers reach maximum output, say +4dBu, by using the output attenuators, results in an attenuator ▶

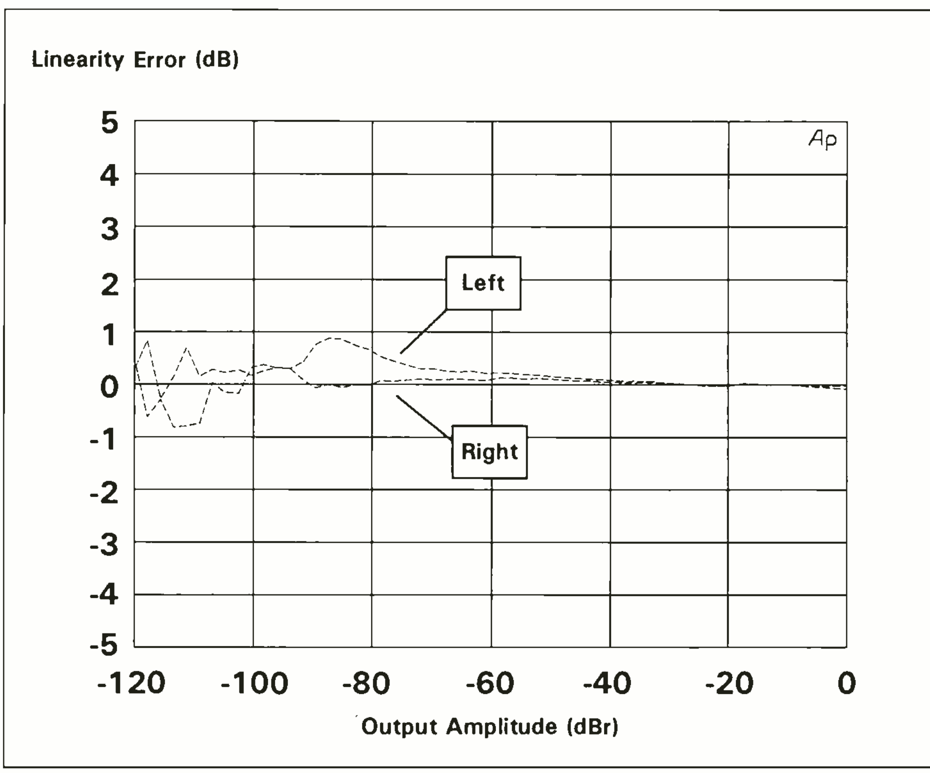

Fig.8: A-D plus D-A convertor linearity error

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"After using 996 for over 12 months, I remain very impressed with its consistency and performance. 996's low noise floor makes it ideal for most applications, even without noise reduction, and its high level capability copes with almost anything we throw at it without any saturation".
- Callum Malcolm, engineer and producer. Castle Sound Studios.

"The performance is excellent. You can push it very high indeed, yet it still retains the clarity needed for CD's, combining the best of analogue warmth with a good crisp quality - real competition for digital".
- Craig Leon, producer.

"I've been using 3M 996 tape at 30ips without noise reduction, and it sounds terrific. It's analogue like analogue ought to be - with digital, all you can do is get the level right but 996 gives you far more control over getting the sound right. It's the only tape I use now".
- Chris Kimsey, producer.

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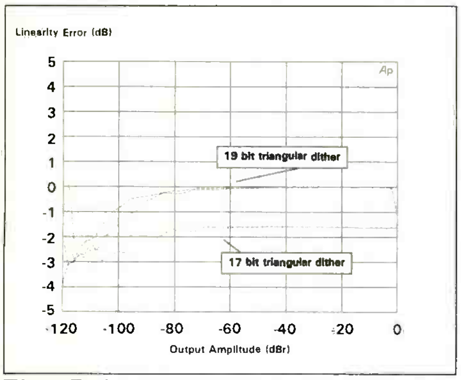

Fig.9: D–A convertor linearity error — note variation with dither level. Curves are offset for clarity

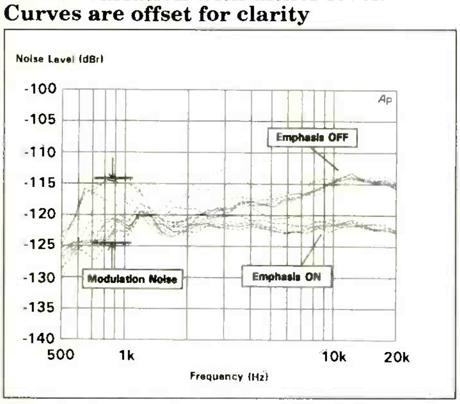

Fig.10: Modulation noise—difference between upper and lower curves

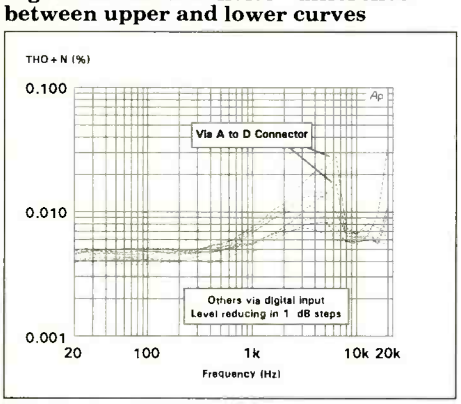

Fig.11: Total harmonic distortion plus noise, no filtering, emphasis OFF

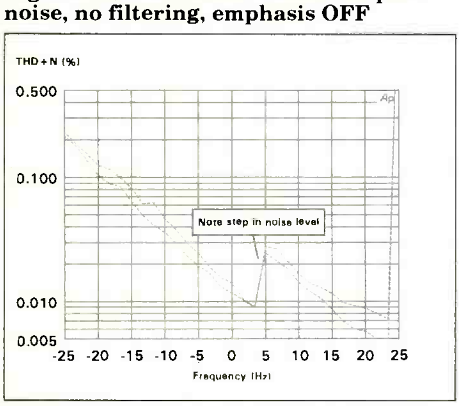

Fig.12: THD+N versus signal amplitude. Noise step indicates possible level ranging around basic A–D

position of about halfway. With this setting, the digital oscillator was turned off, and the residual noise measured as above. The result is -100dBu with emphasis ON, or -99dBu with emphasis OFF, giving a typical dynamic range of 104dB, again an excellent performance.

As mentioned above, the rear panel nominal level switch gives a 10dB gain range adjustment while maintaining the dynamic range.

Table 1 gives the full range of noise measurements to international standards, while **Fig.5** shows the noise spectrum from analogue input to analogue output.

Channel phase and level difference

An important quality of any crossover system is matching between stereo channels. While it is easier to match the electronic systems than it is the two loudspeakers themselves, it is obviously nice to know that the electronics are not introducing any problems. In this respect, the *D2040* excels itself. **Fig.6** shows a left to right level and phase difference for channel 3. This is set to crossover frequencies of 800Hz and 5kHz, with 24dB/octave slopes. There is a constant phase error of 0.4°, and a constant amplitude error of 0.03dB.

Introducing a nominally identical equaliser in each channel, with a 6dB boost with Q of 1 at 1.6kHz, and a 6dB cut with Q of 7 at 3.2kHz, gives the second result in **Fig.6**. Here, the channel amplitude difference is within 0.02 dB, and phase goes out by a maximum of 0.7° at 20kHz. This is superb matching, undoubtedly better than any analogue system could achieve. It is almost impossible that anyone could hear a difference between these outputs due to anything electronic.

Pot tracking

An unusual feature of the Yamaha *D2040* is the motorised output attenuator pots. These come with an audio taper-track controlling the audio level, and a linear track which is used to detect and control the pot rotary position. A usual problem with audio taper-tracks is that, though they have the right control law relating position to loudness for audio, they do not work very well when supplied as offsetable, ganged, stereo pairs such as was found on the Fostex *PD2*. Users will often try to offset the controls to correct for a channel imbalance, only to find that the offset is not at all accurate when the pair of controls are rotated together to adjust the overall level. Yamaha make intelligent use of microprocessor control to overcome this limitation. They have an internal table or equation which relates the pot position as measured by the linear track, to the dB attenuation resulting from the audio track. A little maths and now the pots can be made to track even when offset. How well did they do? Well enough to help a lot! **Fig.7** shows the result, where typical errors are ±1dB over a 40dB range compared with an expected 10dB or more error with mechanically ganged and offset pots over a similar range. This may not be perfect, but it is a good compromise.

Convertor performance

The *D2040* has two analogue inputs provided with 19-bit A–D convertors and an AES-EBU stereo digital input. Selection of these is made from the Utility menu selected by the UTILITY push button on the front panel. Since the outputs are only available analogue (which makes sense until direct digital power amplifiers become freely available), there are a further eight D–A convertors at the outputs, having a specified 20-bit resolution. In **Fig.8**, the results of a linearity test can be seen. Using an analogue input signal, input to output linearity is superb, being virtually error free over a 120dB dynamic range. When a ▶

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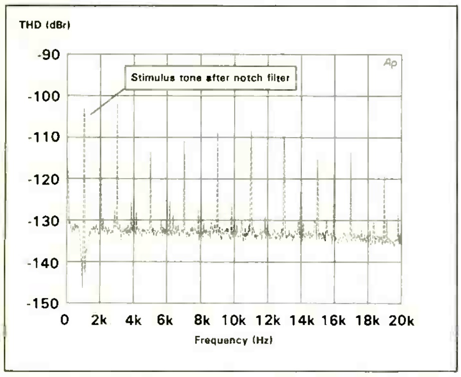

Fig.13: THD Spectrum with 997Hz digital stimulus at -1dBFS

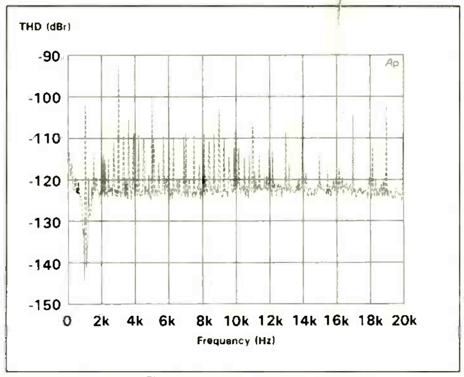

Fig.15: THD Spectrum with 997Hz analogue stimulus at +20dBu. Note modulation noise products when A–D is in use

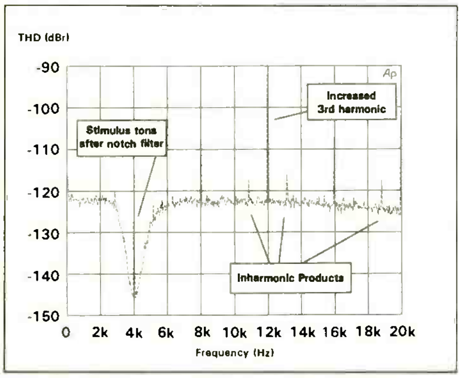

Fig.14: As figure 13 but with 3982Hz stimulus characteristics

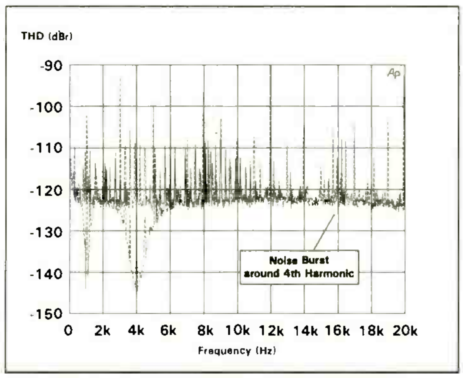

Fig.16: As figure 15 but with 3982Hz stimulus

digital input signal is used, as shown in **Fig.9**, the results are not as good. This indicates possible limitations in device trimming during production. There are no visible adjustment facilities for the A–D convertors, but three are provided for the D–A convertors. It is likely, therefore, that errors in A–D linearity are adjusted at the D–A stage. When an analogue to analogue signal path is used, this method creates no disadvantage. But, for digital input signals there is a very minor performance loss which may result. One may ask whether linearity errors at signal levels of -100dB are likely to be heard.

Fig.10 shows the modulation noise created by a 60Hz stimulus tone. If these curves are compared to earlier reviews, it is clear that the overall noise level is lower. However, the relative modulation noise is not as good as the best convertors we have seen and indicates some nonlinearities. The overall level of these are so low that audibility is not very likely, since they are masked by signals 60dB greater.

Distortion

Total harmonic distortion plus noise versus frequency at a level 0.5dB below clipping is shown in **Fig.11**. The peak in distortion shown at about 6.3kHz using the analogue inputs is genuine, and appears identically on each output channel, indicating an effect caused by the input circuitry. This is marginally higher than specified at 1kHz. When using digital inputs, the result is better, rising to a maximum of 0.014% at 20kHz.

Using the analogue inputs, **Fig.12** shows the variation of THD+N with input signal amplitude. The glitch in the curve at about +4dBu indicates range switching around the A–D convertor, similar to the BSS dynamic floating window convertor used on their TCS804 delay unit. The output D–A convertors showed a similar effect at about 18dB below full scale. This also confirms the variation seen in the linearity tests using various dithering levels, indicating basic convertor resolution at 16 to 17 bits with some sort of ranging circuitry surrounding it. The output D–A is a Burr-Brown device which is specified to 20 bits. Certainly the greatest benefit of these early generation very high resolution convertors is the resulting wide dynamic range. Absolute accuracy has yet to be achieved at a marketable price.

Finally, **Fig.13** shows the distortion spectrum via the digital inputs from a 997Hz tone, and **Fig.14** from a 3982Hz tone. The latter again shows the increased distortion evident in the upper mid-band, compared to lower frequencies. **Fig.15** shows the distortion spectrum from a +23.0dB input 1kHz sinewave via the analogue input. This shows a considerable amount of nonharmonic products which would appear as modulation noise if it was audible above the stimulus tone. In **Fig.16**, the stimulus frequency is raised to 3982Hz. The generally increased distortion is again evident, with a burst of noise surrounding the third harmonic at about 12kHz. This again confirms deficiencies in the analogue to digital convertor stage. Although this looks bad on the graph, the actual levels mean that audibility is unlikely. IMD tests are similarly good, with 0.001% for CCIR twin-tones at 10kHz, and 0.005% for a SMPTE-type test.

Compressor action

The last process within the system to examine is the compressor-limiter response. Here is where we found some disappointment. Good quality loudspeaker processing systems may incorporate a limiter to protect the output device from accidental overload. Alternatively, they incorporate some analogue or digital memory and are placed within the gain loop of the power amplifier to sense that it is getting uncomfortable or driving the loudspeaker into predefined danger conditions. In practice, an overload is most frequently caused by a temporary and unexpected

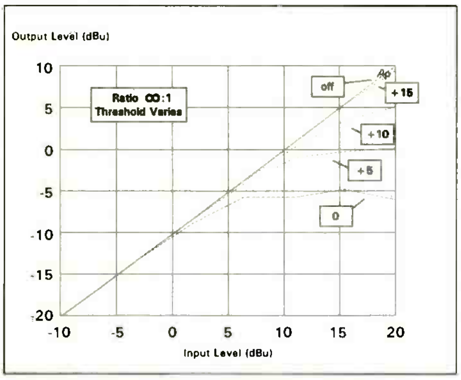

Fig.17: Compressor-limiter—effect of threshold variation

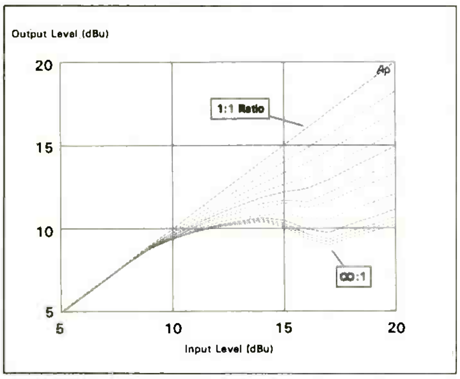

Fig.18: Compressor-limiter—effect of ratio variation

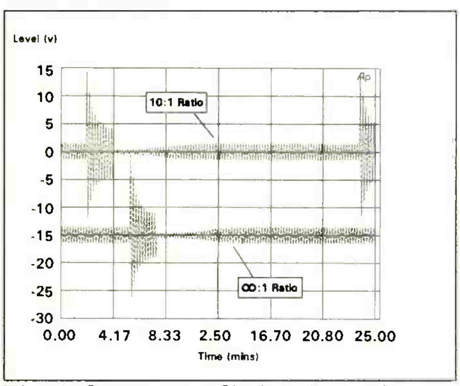

Fig.19: Compressor-limiter dynamic characteristics. Thresshold +5, attack time 1.0ms, release time 0.015

feedback, but there are, of course, many other possibilities, including an overeager sound mix engineer. The processor is usually there to save the loudspeakers from either overexcursion (literally moving too far and tearing itself to pieces) or overtemperature (resulting in a burnt out voice-coil).

A good compressor may have a hard or soft knee, but once the knee is reached things are definitely controlled in an orderly manner. On the *D2040*, something happens, but not as I would personally like to see it happen. In **Fig.17**, the slope is meant to be a constant 1:1, that is after the threshold, no further increase in output should occur as the input level increases. With +15 as a threshold, there is almost no action at all; reducing the threshold to +10, we get a slope, with a total gain reduction at +20 of 5dB, but certainly not a flat slope. When threshold is wound down further to +5, something approaching the expected response is achieved. At a threshold of zero, the response is similar again, but begins to rise again at levels above +20. My opinion is that a device like this should come with a hard knee characteristic, particularly at higher compression ratios where limiting is intended. Alternatively, Yamaha ▶

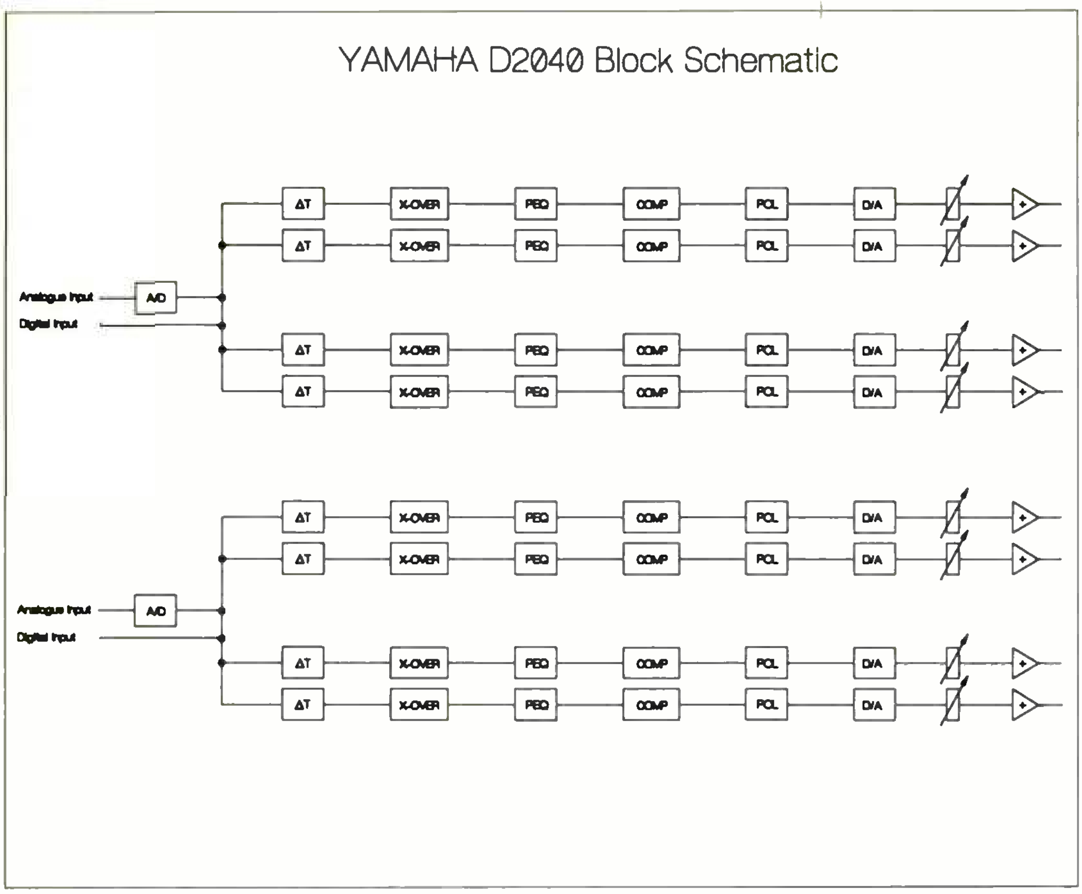

Fig.20: Block schematic for Yamaha *D2040*

could make the knee characteristic selectable. On a loudspeaker processor, I want to know exactly where limiting will occur, to allow the loudest sound levels while protecting my loudspeakers.

Slope adjustment is more correct, as shown in **Fig.18**, where slopes range from 1:1 to limiting. Dynamically, the unit also operates as required. **Fig.19** shows the response to a sinewave burst input signal. Attack is fast enough to protect loudspeakers from heating problems, while overexcursion may still be possible, but at least not repeatedly.

Unfortunately, there is no indication of compressor operation at all, so to get an idea of what is happening, further instrumentation is required.

The sound

Unfortunately, we did not have a multiway active-type loudspeaker in our lab at the time this review was written. Therefore, we were unable to compare alternative processors or crossovers. But we were able to listen to the basic characteristics of the device, convertors, dynamics and equalisers. Initial listening tests were made using the digital inputs direct from a CD digital output.

The compressor does its job, but when active produces a stridency and thinning of the sound when used with a full-range loudspeaker on the output. This seems evident regardless of slope or threshold settings. When the signal is below the threshold, there is no audible effect. If the compressor section is only used as loudspeaker and power amplifier protection, there will be no audible effect on the output signals.

Maximum signal delay is over 1.4 seconds, and, of course, the left and right outputs from one channel can be offset to produce some very strange effects. However, as soon as left and right parameters are linked and the delay or any other setting (except output level) are changed, the left and right outputs are made identical again.

When a memory is recalled, the unit mutes while it makes its internal adjustments. However, when the output level faders are controlled, they are silent in operation so the audio path is preserved, and good image position is maintained while the levels are adjusted. Use of any other controls in real time produces an intermittent audible effect, but nothing which will cause any harm—the level just temporarily drops. So, considering that the *D2040* is really a presetable loudspeaker crossover with optimised output level adjustment, there are no side-effects worth considering.

Listening via the analogue inputs and their A–D convertors, there is a change in the quality of sibilants and the bass seems to reduce in body compared to the direct digital inputs. If we had more time it would be interesting to try to quantify these apparent effects, since, though the ear is the final arbiter, it is notoriously fickle. Certainly, without a direct comparison, it is unlikely that I would be aware of any change in quality due to the input convertors.

Summary

The Yamaha *D2040* is a well-designed and thought out addition to a relatively small collection of adaptable loudspeaker processors. It will certainly find use on systems installed for performing arts, and also, if the loose connectors prove to be a fluke, for hire company use, since many loudspeaker configurations can be stored. When driving larger and more complex systems, the *D2040* is very cost-effective.

As a crossover-processor for studio monitoring, the requirements are sometimes more stringent on sound quality. My guess is that any imperfections in the *D2040* will be inaudible compared to other crossover-processors in use, so many will find that the *D2040* brings a significant improvement to their systems at modest cost. ■

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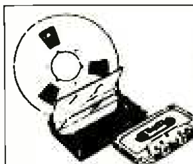

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Keith Spencer-Allen

How to handle the client —or perhaps not

My recent 'Craft' column on the subject of studio politics (the relationship between engineer and client) drew a lot of comment. Some asked for the answers to the real-life examples of potentially difficult situations I used, but these are not really that useful. Firstly, many do not have answers—they were, in retrospect, handled wrongly at the time. Secondly, without knowing the personalities involved, it would be wrong to suggest solutions.

For example, look at one of the simplest cases: the client whose expensive multitrack session tape was deposited down the side of the multitrack when the take-up motor died unnoticed at the start of a mix session. The first action is to recover the tape safely and inspect it. For this you need several quiet minutes to manually rewind it with great care. How you gain this time depends on the client's personality. Some clients might accept a motor failure on an otherwise well-maintained machine as a fact of life and help in recovering the situation. Others might be more like the client in this case—a producer barely more than 'a bad mix away from a nervous breakdown'. It would have been detrimental to everyone and everything if he had been openly informed of the facts.

The tape operator actually noticed the problem as the tape neared the end of the track and managed to pass a note unnoticed to the engineer. The engineer decided to remove the client from the room on a very flimsy excuse —something akin to a comedy farce, like showing him a new coffee machine. The tape was swiftly rewound undamaged, and maintenance called. When the client and engineer returned, they found the technician under the deck and was told that the deck had not been handling properly and—rather than risk the client's tape—the tape operator had called in the technician. Result: praise all round. And while I do not like the idea of deceit, there are times when a distraction saves much grief.

One of the most difficult sides of engineering to acquire expertise in is this relationship with clients in general. How sympathetic should you be to their indecision? Should you insist that a certain sound is correct for its situation or agree to try alternatives and use up lots of studio time? If the client is experienced enough to know that you really cannot fix everything in the mix, then the correct position to take would be somewhere between the two extremes: 'Okay, let's just try one more mic position'. In some cases, being flexible can be construed as inexperience, just as being 'over flexible', certainly is.

When to call in the maintenance team during a session is another frequently misread sign. If you call them at the first sign of trouble, then you introduce frequent intrusions from the creativity-dampening multimeter and soldering iron; the studio gets a bad reputation for repair and you gain the reputation of not being able to do anything without calling in the technicians.

If you decide to take a good look at a fault before calling for help then you can be accused of time wasting, again slowing the creative atmosphere and generally working against the session. A catastrophic failure is one thing but it is probably better to work around a smaller fault until a session break.

Often it is the client's wish to try an undue range of experiments which the engineer already knows will prove stressful to both sides. The client will not feel happy until he has explored every possibility and the engineer knows that in 24 hours time no one will tell the difference. Over-long sessions can prove stressful all round unless the engineer is in-tune with the client—everybody should be able to handle an overdub session of this type, but a six-month album session requires a special sort of person.

Then comes the problem of the degree to which you let the client get involved in the engineering. While there are cases when you wish they would keep their fingers to themselves, real involvement in EQ selection for example can be very helpful in the creative process. However, anyone fiddling with equipment when you are not aware of it is trouble. The real problem is 'idle fingers', and it is here that the studio recreation room pays dividends.

On one occasion, I had to replay a ¼-inch mix on an old Telefunken machine. It was one of those with the built-in editing scissors that unfolded swiftly and snipped the tape when an 'innocent' knob on the head block was pushed lightly. The inevitable happened in play mode and quite a lot of damage was done to the tape.

A much-mentioned technique, although I believe only in jest, is to give the idle hands 'dummy' faders to operate—faders with no signal running through them. This is only really to be considered in the same sense as letting a novice drummer; in on the 'secret' that real professionals hit the kick drum by hand with a boxing glove rather than the beater! Far better is the use of spare hands to keep a mix log of the musical differences between takes. More creative input can come from operating a fader with a fine adjustment capability on an existing mix channel signal—split the signal through two faders, one providing most of the signal and the other under idle fingers control having a little gain and little damage potential.

Some ideas of the client are certainly worth indulging. If a musical producer suggests recording a particularly instrument in a specific acoustic situation, they may be right no matter how odd or inconvenient it may seem. On helping the tape operator carry some sessions tapes to the tape store, the producer notices that the small room acting as the short-term store is wood panel lined, including ceiling and floor. Although nothing was said at the time, the next session was booked with the request to record piano accordion in the tape store. Despite the complaints of those who had to move and replace the tapes thinking it was a typically daft idea, it wasn't. The effect was a massively reinforced sound, thick and almost a complete track by itself.

Less effective was the non-musician producer charged with producing a slightly country-tinged album. He had spent days listening to country albums for inspiration. The day before the session he rang up asking if we knew how to get the 'Nashville Sound', a perceived separation between the instruments giving a feeling of space and air around the instruments. We promised that we would consider what we had to do to achieve what he wanted. Discussions in the studio concluded that the 'secret' was simple: great sounding instruments, superb musicians, sympathetic recording and an intuitive approach to playing that left musical space in the orchestration, all adding up to an 'airy' quality.

The producer, however, was convinced we were missing a trick and I think that there would have been nothing we could have done to change his mind about the feasibility of his intentions. This studio was below ground level and he had some idea that you needed daylight to make it work!

There comes a point, however, on a stressful session that you have taken about all that you can for a single day. A story came to light about an engineer who had just finished a particularly difficult session having worked a continuous 36 hours. He was exhausted but there was no sign of the clients doing anything other than this at this very session. The studio manager was unsympathetic — the extra income was doubly welcome in a period of slack bookings. The engineer appeared the next day far more cheerful and ready for his session. The studio manager chuckled thinking that it looked like the engineer's complaints had been overcome. Even the engineer was smiling as he sat behind the only recently vacated board.

Only he knew that deep in the machine room was a mains supply timer switch sitting across one of the console power supplies! ■

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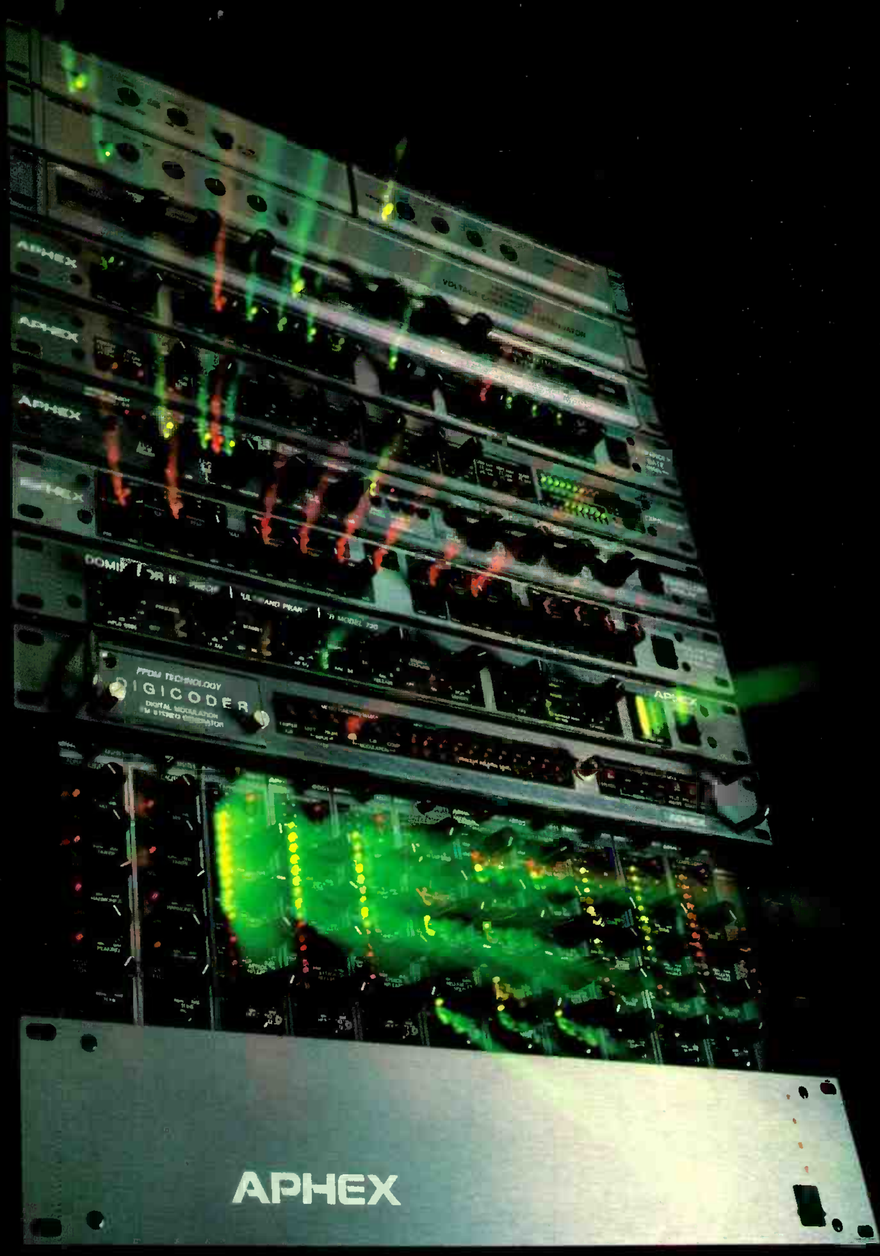

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