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STUDIO SOUND

AND BROADCAST ENGINEERING

INTERNATIONAL PRO AUDIO AND POSTPRODUCTION MAGAZINE

Russia

Raising the curtain on the new East

Loudspeakers

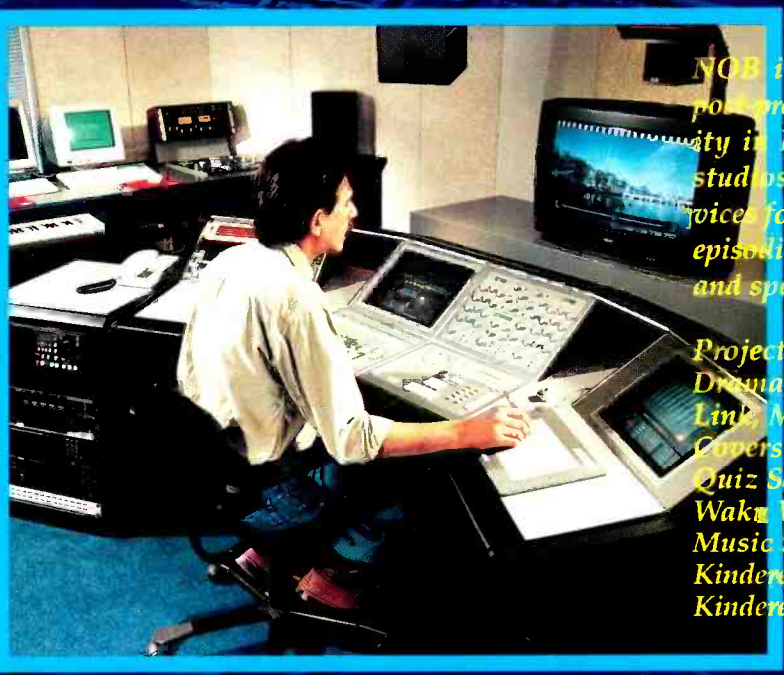
A design revolution

Scenaria & OmniMix – The First Choice for Audio Post-Production

NOB, Holland

"We wanted to have the most technologically advanced systems at NOB to reflect the importance of the facility. Scenaria is the first system to combine the advantages of digital audio production with digital video. Scenaria is now widely accepted by top facilities throughout the world. Our clients are delighted from their first experience with the system."

Frits Paeper, Production Manager, NOB, Hilversum



NOB is the largest post-production facility in Holland. The studios provide services for a variety of episodic programmes and specials.

Projects include:
Drama Series:
Link, Myxomatose, Coverstory.
Quiz Series:
Waku Waku.
Music Specials:
Kinderen Voor Kinderen, Pink Pop

Sun Studio offers a broad range of recording and post-production facilities in a multi-room complex, which includes the first OmniMix installation in Denmark, plus a stand-alone ScreenSound V5 with VisionTrack™.

Projects include:
Feature films and animated series for Walt Disney and Warner Bros

Sun Studio, Denmark

"Working on five different language versions of *The Lion King* was made much easier with Scenaria OmniMix. Matching pan to voices, for instance is no longer a problem. We save all the pan data, and then all I have to do is make sure that I have the character voice on the same channel for each language. It's much quicker and OmniMix matches the pan exactly."

Brian Christiansen, Engineer, Sun Studio, Copenhagen

STUDIO SOUND
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In touch with T[2]
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Meeting the THX cinema standard. See page 68

10 YEARS

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We have never forgotten that sound is above all a vibration

our role simply revive and transmit it

1995 : **Digigram** celebrates its 10th birthday
1995 : **Digigram** launches a new line of products
1995 : **Digigram** exhibits at AES, stand n° 4T46

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STUDIO SOUND
AND BROADCAST ENGINEERING

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PPA PERIODICAL PUBLISHERS ASSOCIATION

Total average net circulation of 18,834 issues during 1993. UK: 7,357. Overseas: 11,477. (ABC audited)

un A United Newspapers publication

Manual Labour

For electronic publishing, the future's so bright the publishing world is wearing shades. As they say.

The future is so bright, in fact, that those shades are obscuring almost all of the view. When publishers look carefully they can just discern the outlines of CD-ROM and on-line publishing services. The real shapes of these apparitions, however, remains obscured.

It is already easy to quote examples of electronic publishing which give some idea of both their scope and the levels of interest being expressed in them. Expert systems, such as those used in medical diagnostic work, have proven relatively ideal (and therefore obvious) applications for CD-ROM. And in the UK, a leading national daily newspaper recently published a sample of its vision of future on-line dailies proving the medium to be something more than simply a sign of the electronic times. In the wonderful world of audio, CDs of both audio and ROM variety have established themselves as logical extensions to the printed word—although I suspect this is something of a transitional stage on the way toward a tighter integration of words and other media.

Studio Sound's Interactive Disk (*PDI*) initiative represents a 'first' for pro-audio in that it offers us the opportunity to apply the power of electronic processing to the practical exercise of attending a trade show and evaluating the deluge of information with which it threatens us—and new opportunities in the presentation of advertisements.

Opportunities abound.

In the light of this simple analysis, it seems strange to me that pro-audio has missed a golden opportunity in electronic publishing. Perhaps it is the fact that this opportunity is singularly lacking in glamour that accounts for its oversight. Alternatively, it may be just too obvious for our highly-tuned minds.

Manuals. Equipment manuals. The same manuals that self-respecting engineers pride themselves in not having to read, yet know enough about the contents to rant about the negligence of the manufacturer in compiling—or translating—it.

I am not suggesting that the 'hard manual' be abandoned (it could, after all, be printed from its 'soft' parent), but a 'soft manual' might offer far higher levels of information with a level of cross referencing prohibited on paper. Interrogation of even an ASCII file by wordsearch would allow unprecedented freedom of approach to a manual that would simultaneously conceal from the user apparently daunting amounts of information and make that information more efficient in use. The fight for suitable manuals has long been a 'cause' with certain pro-audio equipment reviewers, with good contents pages, indexing and general layout being the best that could be hoped for. Wordsearching would go a long way to answering such criticisms forever.

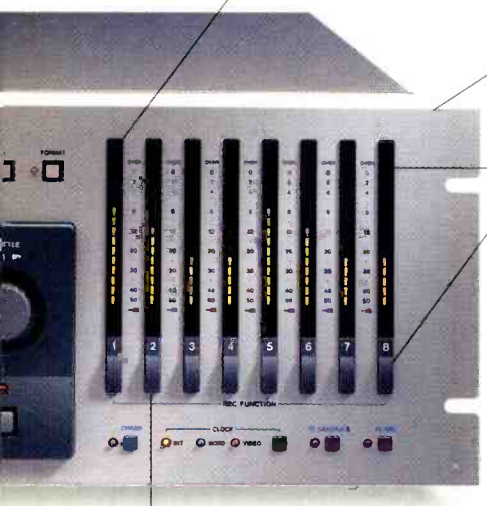
More than this, a recent 'roundtable debate' hosted by *Studio Sound* to form the basis for a forthcoming feature revealed just how crucial manuals are likely to become as increasingly sophisticated equipment finds its way into an expanding market place. Although it is a long way from an ideal situation, what can be learned from equipment manuals is frequently adopted in lieu of any more thorough—though not necessarily more formal—training.

The golden age for electronic publishing is still a way off, but certain opportunities are already being overlooked in all areas of business. If there is a lesson here, it is that we must hope to be more responsible in our attention to future audio technologies. ■

Tim Goodyer

Cover: One of two identical 80-channel SSL *SL 8000 G+* film dubbing consoles at the Warner Bros. studio facility in Burbank, California.

What do you like best about your DA-88?



"Built-in headroom. It makes a big difference when you're trying to track quickly."

"MIDI synching is so easy and clean. It's hard to know where the virtual recording ends and the taped recording begins."

"It sounds great. Especially in the upper frequencies."

"Bouncing. I can bounce forever, it seems. The DA-88 just doesn't distort like other digital decks."

"We really like the convenience of the Hi8 format. You don't get tape stretch, you get much more time per tape, and it's really cost-effective."

"The Hi8 format is a superior recording medium, and it's a TASCAM."

"I can't help but notice the difference in the sound. Unbelievable."

"The TASCAM unit is clearly more dependable."

"I've had a lot of experience with TASCAM... the dependability and the value. It was a no-risk investment for us."

"The sync card has built-in SMPTE, video sync and Sony 9-pin. Perfect for our video house."

"I like having the A/D and D/A converters on an interface card because you can drop in a new card without having to disassemble the whole machine."

"With the sync card it's so easy to synchronize our audio both to video and other audio source, including our DAT."

"The seventy minutes of recording time means we can do something we never could before — get an entire performance on a single piece of tape."

"The sheer number of professionals using the DA-88 system put paid to any reservations I might have had about moving into digital recording."

"It locks up a helluva lot faster than our other digital multitrack recorder."

"It's trouble-free. All I have to do is clean the heads. I'd call it the stress-free modular digital multitrack!"

"I needed over an hour of recording time. The DA-88 is the only digital recorder to offer that."

"The frame accuracy is so fantastic, I can edit voice-overs and guitar parts as small as two frames using the Absolute Time capability."

"Punching in and out is so simple. That fact alone made it worth buying a DA-88."

"The punches are very clean and accurate. A dream."

"I've been on the DA-88 nonstop for three weeks now. The transport is phenomenal."

"I like the size, the editing capabilities, and the price. I even like the way it looks."

"I just finished scoring two movies on it. Unbelievable machine."

"Two things. First, the DA-88 gives me a full one-hour and thirteen minutes of record time. Second, the wind times are so wonderfully tight and quick... it takes my other digital recorder a day and a half to rewind."

"I like the auto-locator and rehearsal modes, and of course we're doing our next album on it."

"I can now offer my customers digital recording at analogue prices."

"It's just faster. Speed counts. Time is money."

"It just feels better than any other recorder in its price range."

"We love the jog/shuttle wheel. It's working out great."

No wonder the DA-88 is the preferred digital multitrack.

It features the latest generation in digital multitrack recording technology. According to users, it's the best-sounding, best-built, most functional and affordable digital multitrack on the market. You're going to love the DA-88. Get to your dealer now and see why!

"It's the name. They make the best recording equipment, period."

"I can lock it to video and my analogue machines with no hassle. Life is so easy now that I have my DA-88."

"I wanted the serious machine for music production. That's TASCAM."

"I'm focused on the TASCAM. It will become the standard. And I don't want to be left out."

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World Events

February 1995

● February 7th–9th, **ISDN User Show**, Olympia 2, London, UK. Tel: +44 1733 394304.
● February 12th–15th, **Siel 95** and **Theatrical Services Exhibition**, Porte de Versailles, Paris, France.
● February 14th, **British AES Section Lecture: Digital Frontiers**, Imperial College, London, UK. Tel: +44 1628 663725.
● February 14th–16th, **The Video Forum '95**, Wembley Conference Centre, London, UK.Tel: +44 1203 691169. ● February 15th, **Conference: Still Protecting the Media**, Le Meriden Hotel, London, UK. Tel: +44 171 637 4383.
● February 20th–23rd, **Communications World '95**, Hong Kong Convention & Exhibition Centre, Hong Kong. ● February 20th–23rd, **Digital Hollywood: The Media Marketplace**, Beverley Hills Hilton Hotel, Beverley Hills, California, USA. Tel: +1 212 226 4141.
● February 23rd–24th, **Conference: Video on Demand**, Langham Hilton, London, UK.Tel: +44 171 637 4383.
● February 25th– 28th, **98th AES Convention**, Palais de Congrés, Paris, France. Tel: +32 2 345 7971.

March 1995

● March 8th–12th, **Frankfurt Pro Light & Sound**, Messe Frankfurt, Frankfurt, Germany. Tel: +69 75 75 6415/6907. ● March 8th–12th, **ITA Seminar: The Converging World of Entertainment, Information and Delivery Systems**, Westin Mission Hills Resort, Rancho Mirage, California, USA. ● March 9th, **Sound Sense Show**, Swallow Hotel, Gateshead, UK. Tel: +44 1491 838575.
● March 15th–17th, **The Television Show**, London, UK.● March 25th–27th, **The Pro Audio Show**, Karachi, Pakistan. Fax: +21 493 2535.
● March 26th–28th, **ECTS**, Olympia Grand Hall, London, UK. Tel: +44 181 742 2828.

April 1995

● April 3rd–5th, **Cable & Satellite 95**, Olympia, London, UK. Tel: +44 181 910 7849.
● April 4th–5th, **Television Distribution Technology 95**, Olympia, London, UK.Tel: +44 171 637 4383. ● April 4th–6th, **REPLItech Europe**, Austria Center, Vienna, Austria. ● April 4th–7th, **Communications Tokyo Exhibition**, Tokyo International Trade Fairgrounds, Tokyo, Japan. Tel: +1 301 986 7800.
● April 7th–12th **MIP-TV 95**, Cannes, France. Tel: +44 171 528 0086.
● April 9th–13th, **NAB 95 Symposia**, Las Vegas Convention Center, Las Vegas, USA. Tel: +1 617 965 8000.
● April 10th–13th, **NAB 95**, Las Vegas Convention Center, Las Vegas, USA. Tel: +1 617 965 8000.
● April 21st–23rd, **MEMS 95**, Olympia 2, London, UK. Tel: +44 1225 442244.
● April 26th–29th, **Broadcast Technology Indonesia**, Jakarta, Indonesia. ● April 26th–28th, **5th Australian Regional AES Convention: Making Waves**, Sydney Exhibition Centre, Sydney, Australia. Tel: +61 3 534 5755.

May 1995

● May 1st, **IEE Audio Engineering Colloquium, IEE Head Office, London, UK. Tel: +44 171 240 1871 x2206.**
● May 9th–12th, **Pro Audio, Light & Music China 95**, Beijing Exhibition Centre, People's Republic of China. ● May 13th–21st, **MultiMediale 4**, ZKM/Center for Arts and Media Technology, Karlsruhe, Germany.
● May 15th–20th, **Expo Comm Moscow Sviaz 95**, Krasnaya Presnya Fairgrounds, Moscow, Russia.
● May 23rd–25th, **Midem Asia**, Hong Kong. Tel: +44 171 528 0086.
● May 30th–31st, **Leipziger MedienMesse Hörfunk**, Leipziger Messe, Leipzeig, Germany. Tel: +37 41 2 230.

June 1995

● June 8th–10th, **2nd Annual South American Pro Audio Expo**, Centro de Extension, Santiago, Chile. Tel: +56 2 635 1994. US. Tel: +1 914 993 0489. ● June 8th–12th, **China Sound Light & Music**, Beijing Exhibition Centre, People's Republic of China. ● June 8th–13th, **International Television Symposium/Exhibition**, Montreux, Switzerland.Tel: +41 21 963 3220.
● June 10th–12th, **12th ShowBiz Expo**, Los Angeles, USA. Tel: +1 714 513 8400. ● June 13th–15th, **REPLItech International**, Santa Clara Convention Center, Santa Clara, USA. ● June 19th–20th, **Radio Festival Trade Exhibition**, International Convention Centre, NEC, Birmingham, UK. Tel: +44 1491 838575.
● June 21st–23th, **Audio Technology 95. Formerly APRS**, National Hall, Olympia, London, UK. Tel: +44 1734 756218.
● June 21st–23rd, **7th Japanese Regional AES Convention: Advanced Audio Technologies for Audio-Video and Multimedia**, Sunshine City Convention Center, Tokyo, Japan. Tel: +81 3 3403 6649.

July 1995

● July 12th–14th, **Pro Audio & Light Asia 95**, World Trade Centre, Singapore. Tel: +852 865 2633.
● July 17th–19th, **WCA 95. Wireless Cable Association Show**, Washington Convention Center, Washington, USA. Tel: +1 202 452 7823. ● July 20th, **British Music Fair**, London, UK.

August 1995

● August 17th–20th, **Popkomm**, KölnMesse, Köln, Germany. Tel: +221 8210. ● August 25th–28th, **Beijing International Radio & TV Broadcasting Equipment Exhibition 95**, World Trade Centre, Beijing, People's Republic of China.

September 1995

● September 6th–9th, **1995 World Media Expo**, New Orleans Convention Center, New Orleans, USA. Tel: +1 202 429 5350.
● September 10th–12th, **ECTS**, Olympia Grand Hall, London, UK. Tel: +44 181 742 2828.
● September 10th–13th, **PLASA**, Earls Court 2, London, UK.
● September 14th–18th, **IBC 95**, RAI Centre, Amsterdam, Holland.
● September 19th–24th, **Live 95**, Earls Court, London, UK. Tel: +44 181 742 2828.
● September 21st–24th, **Nordic Sound Symposium XVII**, Bolkesjø Mountain Hotel, Norway. Tel: +47 2 79 7730.

October 1995

● October 5th–8th, **99th AES Convention**, Jacob K Javits Center, New York, USA.
● October 17th–19th, **Vision 95**, Olympia, London, UK. Tel: +44 181 948 5522.
● October 19th–23rd, **9th International Audio, Video, Broadcasting and Telecommunicationas Show. IBTS**, South Pavilion, Milan Fair, Milano/Lacchiarella, Italy. Tel: +39 2 481 5541.
● October 24th–26th, **REPLItech Asia**, Singapore International Convention & Exhibition Centre, Singapore. ● October 25th–28th, **Broadcast Cable and Satellite India 95**, Pragati Maidan, New Delhi, India.

November 1995

● November 1st–5th, **Audiovideo 95**, Lenexpo Exhibition Complex, St Petersburg, Russia. Tel: +7 812 119 6245.
● November 2nd–4th, **Broadcast India 95**, World Trade Centre, Bombay, India. Tel: +91 22 2151396.
● November 7th–9th, **Wireless World Expo 95**, Moscone Center, San Francisco, USA. Tel: +1 301 986 7800.
● November 9th, **20th Sound Broadcasting Equipment Show. SBES**, Metropole Hotel, NEC, Birmingham, UK. Tel: +44 1491 838575.
● November 21st–23rd, **Visual Communications 95**, London, UK.

December 1995

● 5th–9th December, **Expo Comm China South 95**, Guangzhou Foreign Trade Exhibition Centre, Guangzhou, Peoples Republic of China. Tel: +86 1 841 5250 or US. Tel: +1 301 986 7800.

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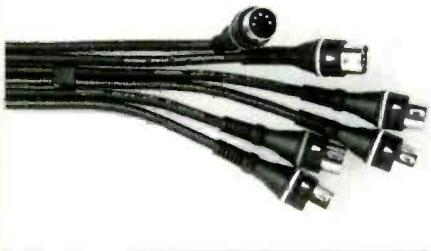
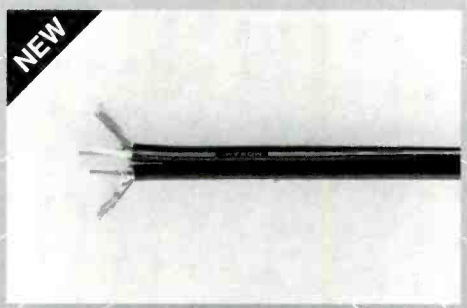


MIDI SYNCHRO CABLES

NEW



STEREO MIC. CABLE AWG #24 (0.226M²)

NEW

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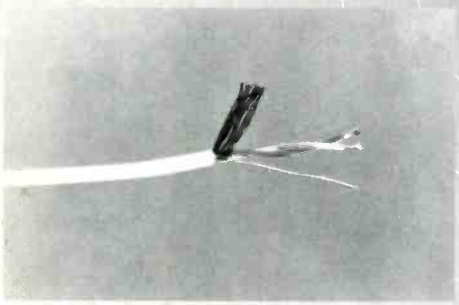
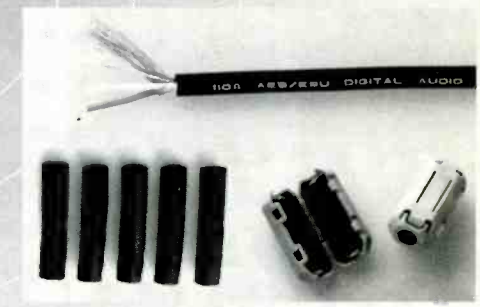
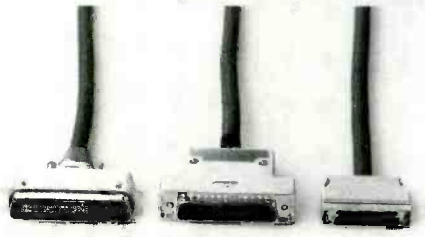


CONSOLE WIRING CABLES

CL2 RATED SUPER-FLEXIBLE STUDIO SPEAKER CABLES

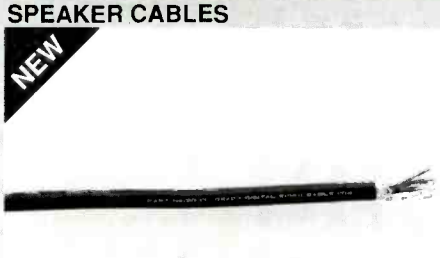


AES/EBU 110Ω DIGITAL AUDIO CABLE

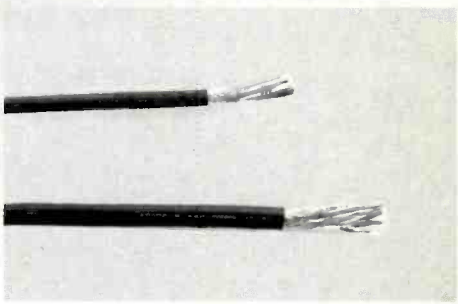


SCSI -II SINGLE ENDED A CABLE

NEW



PARALLEL DIGITAL VIDEO CABLE



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UK and Scandinavia: Phone ++44 1908 696880, Fax ++44 1908 694020 • Mainland Europe: Phone ++31-3463-53730, Fax ++31-3463-51544

International News

In-brief

● National French broadcaster begins disk-to-air transmissions

French broadcaster M6 have started transmitting news programmes direct to air from hard disk via Avid Technology's *AirPlay* digital playback system. *AirPlay* is used to broadcast hourly news programmes, including live presentations, with shorter bulletins every half hour. M6 also use four *NewsCutters*, Avid's digital news editing systems, to prepare news, sport and general interest stories.

Avid Technology Inc, and DigiDesign Inc, have also announced that the merger of the two companies has successfully been completed. Digidesign will operate as a wholly-owned subsidiary of Avid.

Avid Technology Inc, US. Tel: +1 508 640 3345.

● Time-code *Portadat* now shipping

HHB are now shipping the *Portadat PDR1000TC*, the time-code version of the portable machine launched last year. The first 100 customers include the BBC, Anglia TV, the National Film Board of Canada and the Defence Research Agency, who are using the recorder to carry out acoustic measurements in fighter aircraft.

HHB Communications Ltd, UK. Tel: +44 181 960 2144.

● Kerridge to chair APRS

From 1st January 1995 Adrian Kerridge took over from Dave Harries as Chairman of APRS. Director of CTS and Lansdowne Studios in London, Adrian Kerridge has over three decades experience in the recording business and has been closely involved with the APRS for the past six years, serving for most of that time as Chairman of the Administration, Finance and Legal Committee.

APRS Ltd, UK. Tel: +44 1734 756218.

● 3M, Ampex tape awards

Erasure are the latest recipients of 3M's Visionary Award for their album *I Say, I Say, I Say*. The project was recorded over a six-month period between Vince Clarke's private studio in Amsterdam, Dublin's Windmill Lane Studios and The Church in London, and mixed at the Strongroom on to *996* ½-inch (at 30ips without noise reduction) by Engineer Phil Legg and Producer Martyn Ware.

Meanwhile, Pink Floyd have been presented with the Ampex Golden Reel Award for the *The Division Bell*, recorded onto *499 Grand Master Gold*.

3M US. Tel: +1 612 736 9599.
3M UK. Tel: +44 1344 858614.
Ampex US. Tel: +1 415 367 3888.
Ampex UK. Tel: +44 1734 875200.

● Turnkey to distribute Manley

Turnkey Studio Systems have been appointed exclusive agents for Manley Laboratories professional processors and microphones, which includes Manley valve mics and processors and Langevin EQ, limiter and *CR3A* mic.

Turnkey. Tel: +44 171 240 4136.

EuroDab Forum

Europe has decided to unite to launch digital-audio broadcasting services in the greatest possible number of countries this year. A European platform (the EuroDab Forum) for the introduction of these services in Europe and all over the world has been established at the EBU headquarters in Geneva. The countries represented were Belgium, Denmark, France, Germany, The Netherlands, Poland, Spain, Sweden, Switzerland and the UK, along with the European Community, the European Association of Consumer Electronics Manufacturers (EACEM), the European Digital Radio project (EDR) and the Community Radio Association. Consumer DAB equipment is expected to be commercially available in the autumn of 1995.

The EuroDab Forum will hold its inaugural meeting in Geneva on 22nd March. A number of points, such as marketing issues and service launch strategy, coordination of national plans, and user requirements for receivers will need to be addressed as a matter of urgency. Forum membership is open to all existing and future national DAB groupings, manufacturers, service operators, broadcasters, and international organizations. Countries outside Europe will also be welcome.

European Broadcasting Union, Switzerland. Tel: +41 22 717 27 32.

Otari OEM agreement

Otari Corporation and TG Systems have announced an agreement giving Otari world-wide exclusive rights to market and distribute a new line of products called PicMix—prototypes of which were on display for the first time at November's AES. PicMix products are hardware and software peripherals for converting any console into a mix-to-picture board and will initially consist of multichannel, multiformat panning and monitor systems that are compatible with all surround-sound formats.

TGS is a newly-formed company whose principals are Paul Galburt and Michael Tapes, the former owners of Sound Workshop, which was purchased by Otari in 1989. Since then, the two have helped design Otari's *Premiere*, *Series 54* and *Concept 1* consoles.

The PicMix system can interface with most console automation systems and MIDI sequencers for full automated panning effects. Up to six dual-input panner modules can be installed in one rackmountable unit, with each module configurable as two L/R/S panners, two LCRS panners, an LCR and LCR-LS-RS panner, or a single 7 or 8-channel panner for SDDS and custom applications. Control is by a stand-alone or console mountable controller that features both dual joysticks and rotary knobs.

The monitor system will be compatible with Dolby Stereo, Dolby Digital (SR.D), DTS Digital and Sony SDDS digital formats.

Otari Corporation, US. Tel: +1 415 341 5900.

BSI DAT standards

The British Standards Institution have introduced new standards for various aspects of DAT technology, contained within BS EN 61119. The content, format and other parameters of calibration tapes for checking machine performance, while the second specifies methods of measurement and minimum requirements for DAT cassettes themselves. Alongside these is a third defining requirements for SCMS. All three are completely new standards, not superseding previous ones.

At the same time, another new section, this time of BS 7434, defines the methods for measurement of the FM audio tracks on nonbroadcast VTRs.

Copies are available from BSI Customer Services, Publications.

British Standards Institution, UK. Tel: +44 181 996 7444.

Commercials by design

Mingles Music have launched a new company offering the services of established artists to the advertising industry. Artists signed to Music By Design will be available to write and perform music tracks specially for commercials, and the company has already signed more than 50 artists across a wide range of musical genres. Country is covered by Faith Hill, Clint Black and Dwight

Euphonix *CS2000* consoles have recently been installed in two major American studios —Manhattan postpro house, The Audio Department (above), and Chicago postpro house, the Chicago Recording Company. The Audio Department's *CS2000* has gone into Studio A and will run in conjunction with an Otari *MTR90* and Digidesign *Pro Tools*. In Chicago, the *CS2000* will keep the company of an AMS *AudioFile* and *Synclavier* system. Both houses major in sound for television, with The Audio Department specialising in high-quality commercials and Chicago Recording on general TV work and motion pictures.

Euphonix US. Tel: +1 818 766 1666. Euphonix Europe. Tel: +44 171 602 4575. Fax: +44 171 603 6775.

Yoakam, with soul represented by Al Jarreau, Smokie Robinson, Wilson Pickett and Barry White. Also on the list are Status Quo, Madness, Daryl Hall, The Beach Boys, Dave Stewart and Stereo MCs.

The company estimates that the costs of having music written for a campaign are equitable with the payment of copyright on an old tune coupled with the cost of writing a script to fit the music. In addition there is no chance of two different campaigns duplicating through licensing the same music track, while the back-up technical services of the company will enable Music By Design and the agency to provide cut-downs, copy tapes and instrumental versions even if the artist is unavailable through recording or touring commitments.

Music By Design, UK.
Tel: +44 171 437 7418.

Dolby surrounds BBC Radio 4

Radio 4 have produced their first Dolby Surround drama, scheduled for transmission this month. A dramatisation of the Len Deighton novel, *Bomber* traces the story of a Lancaster raid over Germany during the Second World War. The programme will be broadcast in four episodes during the same day, corresponding to the actual time that the operation took place.

Dolby Surround has already been used on Radio 1 productions including *Batman: Knightfall* and *The Adventures of Superman*. Other programme formats currently using Dolby Surround include video, television, CD and computer games, and Dolby say there are now 15 million suitably-equipped systems world-wide.

Dolby Labs, US.
Tel +1 415 558 0200.
Dolby Labs, UK.
Tel: +44 1793 842100.

Micropolis-led consortium

Micropolis are to lead a consortium involving five other high-tech companies to accelerate the development of high-density disk drive technology. The project's focus is to produce a prototype 1GB capacity 1.8-inch, single-platter magnetic disk drive, which can be used in a wide range of applications from portable devices to large-scale video servers.

Half the estimated cost of the $20m, two-year project will be met by the Advanced Research Projects Agency (ARPA) of the US government, which have specially selected the Micropolis project for R&D funding. The research team which will provide the remainder of the funds includes Read-Rite Corporation, Silicon Systems Inc, Adaptec Inc, Tulip Memory Systems Inc, Intevac Vacuum Systems Division and the California Institute of Technology, UCLA. Each member of the group is responsible for one or more of the core technologies required for advanced hard-disk drive development.

Micropolis Corporation, US.
Tel: +1 818 709 3300.
Micropolis, UK.
Tel: +44 1734 751315.

The Tallis Scholars recently gave a concert in the Basilica of Santa Maria Maggiore in Rome to mark the 400th anniversary of the death of Palestrina, and Gimell Records were there to record the event for release. The audience included cardinals and other representatives of the Roman curia, and the concert was also filmed for video release. Both the CD and the video soundtrack were recorded on *Nagra Ds*, of which engineer Philip Hobbs used three—one plus back up for the CD and one for the video soundtrack with back up provided by the 20-bit audio tracks on the Digital Betacam recorder. The decision to use separate recorders for the two purposes was made in order to avoid the degradation of sample-rate conversion; the CD was recorded at 44.1kHz and the video sound at 48kHz, all recorders being synchronised with time code generated from the Digital Betacam. Microphones were two pairs of Schoeps *Mk6s*, with Jensen mic amps and Prism Sound *Dream* 20-bit A–Ds positioned near the microphones. Nagra Kudelski, Switzerland. Tel:+41 21 732 0101. UK. Tel: +44 1727 810002.

ITS Digital Code of Practice

The International Teleproduction Society's UK Chapter is to produce a Digital Code of Operational Practice, which will be co funded and supported by the ITC, Channel Four and the ITV Network Technology Centre. The Code underlines the ITS's dedication to maintaining professional standards within the independent facility sector.

Its objectives will include the standardisation of operational sound and vision line-up levels and procedures, the question of video signal legality and validity, standards for monitoring and declaring tape-error rates, and recommended methods of evaluating definitions of 'broadcast quality' in the digital domain.

ITS (UK Chapter) Ltd, UK.
Tel: +44 1707 260216. ▶

In-brief

● ARX drum up support in Tokyo

Tokyo rental company Lark Inc made prominent use of ARX equipment at the opening of the Japan *Stone Fair International 94* at the Makuhari Messe to reinforce a performance by the Wa Dai Ko drummers. Some of the group's drums are 12-feet across, placing great demands on the LF capabilities of a sound system, and ARX's *925* Sub-Bass cabinets were used along with *212* Mid-High cabinets and *SX* series amplifiers.

ARX Systems pty Ltd, Australia.
Tel: +61 3 555 7859.

● Noise Control for Disney

When the mobile ADR unit working on the new, still secret, Disney project at Bray Studios, Maidenhead, Berkshire, needed some acoustic work done quickly the job fell to The Noise Control Centre. Two days after being called in and surveying the site, and without disrupting the production schedule, acoustic treatment was installed in both the voice booth and the control room complete with coordinating colourways. The acoustic package was based on Melatech absorbing panels and tiles, with pyramid profiled Melatiles for the ceilings.

The Noise Control Centre, UK.
Tel: +44 1664 60203.

● Soundtracs move

Ten years at the same address is to come to an end for Soundtracs when they move their head office on 20th February 1995. The move will provide additional space for R&D, production, sales and marketing, and 'social facilities'.

Soundtracs plc, Unit 21-D, Blenheim Road, Longmead Industrial Estate, Epsom, Surrey KT19 9XN, UK.
Tel: +44 181 388 5000.
Fax: +44 181 388 5050.

Contracts

● First *Flashlight* in New Zealand

The first Turbosound *Flashlight* system in New Zealand has been delivered to PA specialist Oceania Audio. The order comprises 16 *Flashlight* units together with 32 *Floodlights*, plus full flying hardware and system managers. The rig has already seen service with Jose Carreras and a 92-piece orchestra, as well as Billy Joel on the Auckland dates of his world tour.

Oceania Audio, New Zealand.
Tel: +64 9 849 3114.
Turbosound, UK.
Tel: +44 1403 711447.

● Soundcraft go up to North Down

North Down and Ards Institute of Further and Higher Education, in Bangor, Northern Ireland, have installed a Soundcraft *DC2000* as the centre piece of their prestigious new studio complex. With a 70 m² live room and five adjoining rehearsal rooms the facility will be the largest of its kind in Ireland. Walker Audio supplied the desk to the studio, which is housed in a new £11m extension to the college.

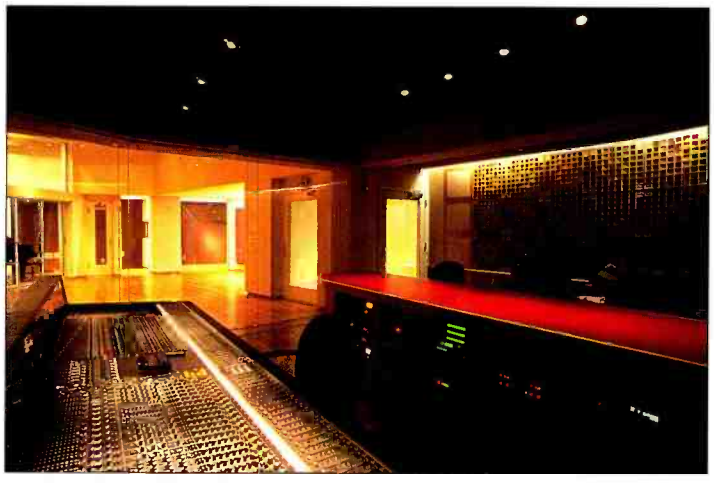
Soundcraft Electronics, UK.
Tel: +44 171 372 7871. ▶

Tonmeisters available for placements

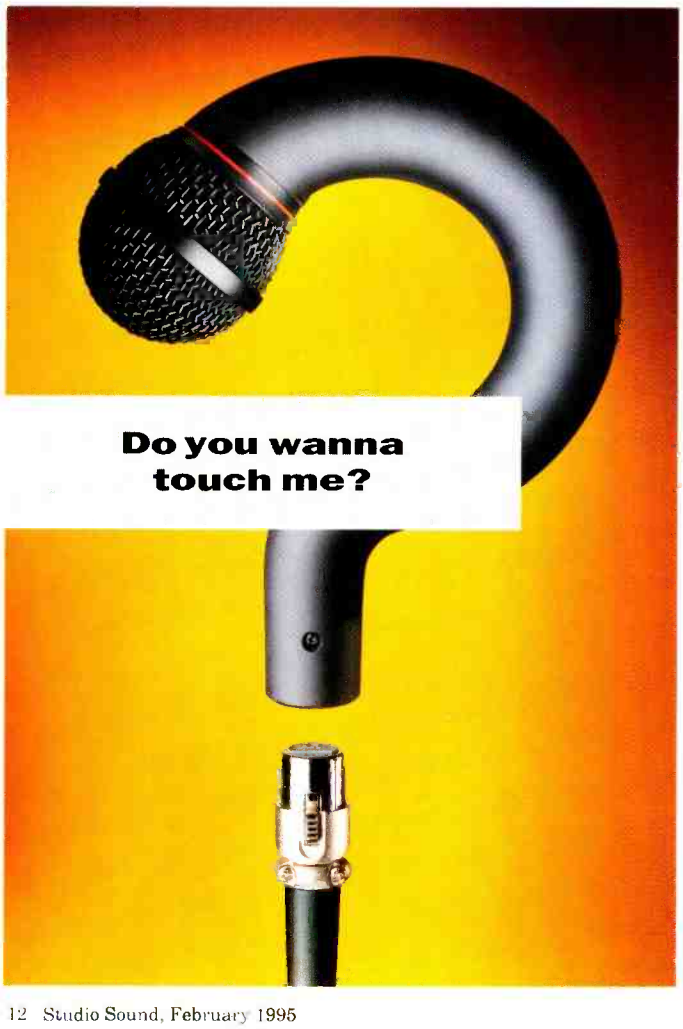
Following the expansion of its Tonmeister degree programme in Music and Sound Recording, the University of Surrey is increasing the number of students available for professional training placements. Students spend the third year of their four-year course working in the audio industry, and have been successfully employed in a wide range of fields, including top recording studios, postproduction facilities, product support, service and QA, consultancy, broadcasting and theatre. Tonmeister students have been well received by their employers, and many employers have found that the scheme provides an ideal means of recruiting experienced qualified staff. Students have a comprehensive background in music, as well as experience in recording and editing all types of sound using industry standard equipment. They also have a broad-based knowledge of acoustics, electroacoustics, electronics, microcomputers, MIDI and laboratory test equipment.

Any employer interested in joining the scheme is encouraged to contact Francis Rumsey at the University's Music Department.

University of Surrey, UK.
Tel: +44 1483 259317.



Sony-Tree, major publisher of country music, recently moved into new corporate headquarters on Nashville's Music Row. A recording studio designed by Russ Berger Design Group, Inc. is a highlight of the $3.5 million renovation and expansion. The project nearly doubled the size of the previous facility and is intended to allow the company to develop more and better writers.Russ Berger Design Group Inc. Tel: +1 214 661 5222



New sound courses in Sheffield

The Northern Media School, Sheffield Hallam University's centre for postgraduate study in film and television production, is offering Postgraduate Diploma and MA courses in Film and Television Production, including Sound Design and Postproduction. This will offer sound specialists the chance to see through films from location recording with industry standard equipment to the final postproduction on film, broadcast video and Digidesign digital audio systems. Sound students this year won the prestigious Fuji Award for best sound on a student film. The school is equipped to full broadcast standard for film and video, and tutors and visiting fellows are practising professionals with proven track records.

Northern Media School, School of Cultural Studies, Sheffield Hallam University, The Workstation, 15 Paternoster Row, Sheffield S1 2BX. ■

Contracts

● **Finnish *Gold***

Finnish broadcasters YLE have just taken delivery of a SoundStation *Gold* integrated audio-production system from DAR, making them Europe's largest DAR user to date. The addition of *Gold* takes YLE's total up to eight *SoundStation* systems, with their first acquisition dating back to 1989. The new system is to be installed in a new studio within YLE's TV1 postproduction complex, and will be commissioned early in 1995.

DAR, UK. Tel: +44 1372 742848.

● **Soundtracs *Jade* successes**

Soundtracs' top of the range *Jade* console has sold around the world recently; buyers include members of Roxette and Ferger Studios in Sweden, and studios in Singapore, Bangkok, Indonesia, Switzerland and France. In the UK, the score for ITV's new *Kavanagh* series was composed and produced by Roger and Anne Dudley on their new *Jade*, as was Anne Dudley's album *Ancient & Modern*.

Soundtracs plc, UK.
Tel: +44 181 388 5000.

▲ ***Soundfield* for Stock**

Mike Stock's new multi million-pound London studio complex (see *Studio Sound*, January 1995) have opted for the SoundField *Mk V* as their main studio microphone. Chief engineer Mike Picking was responsible for tie-lining each of the studio rooms with 12-core *SoundField* cable and is exploring the possibility of having a customised *SoundField* bar-graph display added to the studio's SSL *SL 4000G* console for more local viewing of the microphone's directional information. Mike Stock famously used the *SoundField* extensively during nine years at PWL, crediting the microphone on album sleeves.

Soundfield Research Ltd, UK.
Tel: +44 1924 201089

● **Drawmer Valves for Korean TV**

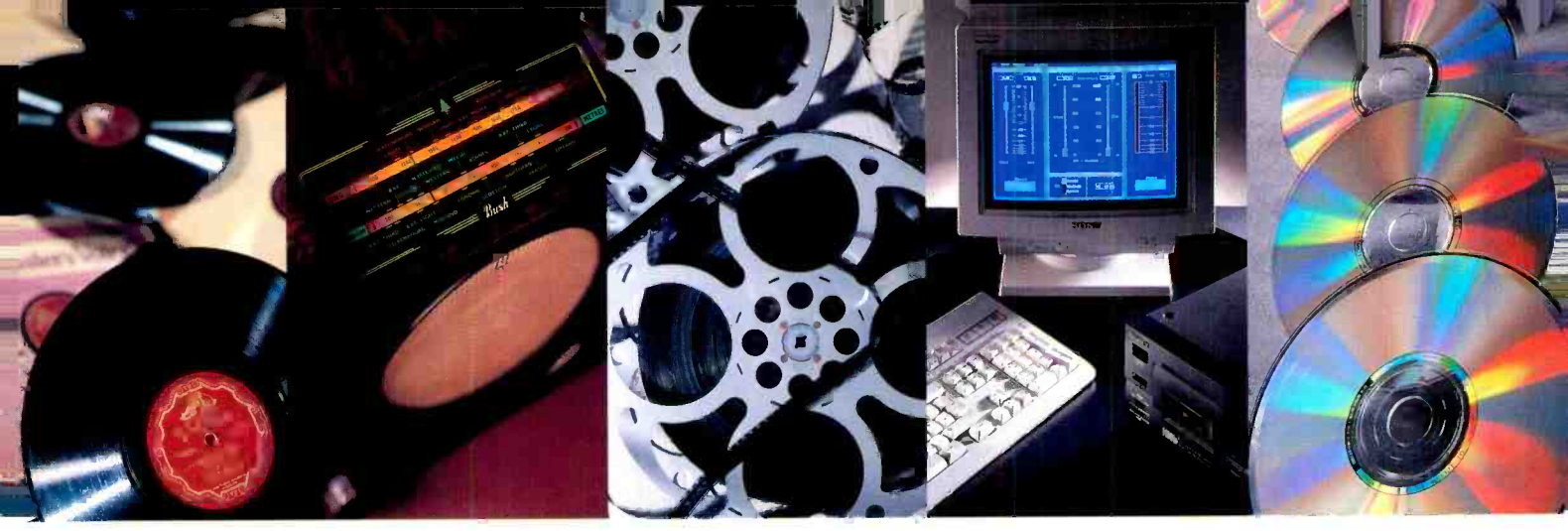
The Hyundai corporation have installed five Drawmer *1960* Mic Preamp-Valve Compressors in their cable TV station which will serve Korea's new cable TV network, due for completion in March.

Drawmer Distribution Ltd, UK.
Tel: +44 1924 378669.

● **Tetra on Parr**

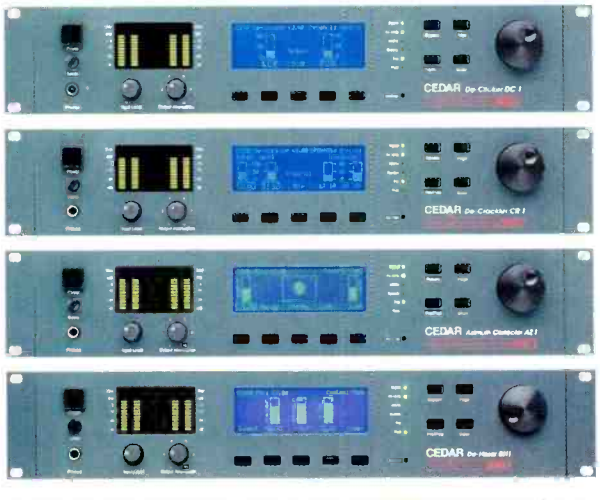
John Parr—songwriter and composer responsible for the *Three Men and a Baby* soundtrack—has opted for *Optifile Tetra* automation for his Amek *2500* console.

Ad Systeme, France.
Tel: +33 42 53 3118.
Home Service, UK.
Tel: +44 181 943 4949. ■



FIRST CAME NOISE REDUCTION NOW COMES NOISE REMOVAL

The CEDAR DH-1 real time stereo de-hisser



Call today for details of important SERIES 2 upgrades to CEDAR's world beating family of real time audio restoration devices. The constantly expanding CEDAR range now includes the DC-1 De-Clicker, the CR-1 De-Crackler, the AZ-1 Azimuth Corrector and the revolutionary new DH-1 De-Hisser.

The new DH-1 De-Hisser from CEDAR Audio is the most important breakthrough in the fight against hiss since the advent of noise reduction.

It removes the broadband noise from hissy recordings virtually instantaneously, restoring the original signal in real time with little or no loss of transients or ambience. It's that easy – there's no need for spectral fingerprinting or encoding / decoding, and with both analogue and digital I/Os, it's easier to use than a reverb.

The DH-1's ability to transform noisy recordings makes it invaluable for use in mastering, film, broadcast, archiving and industrial applications.

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Independent Audio, 295 Forest Avenue, Suite 121, Portland, Maine 04101-2000, USA Tel: 207 773 2424 · Fax: 207 773 2422

Artisan AmpPacks

The new Artisan *AmpPacks*, to be shown at AES, are designed to convert a standard passive speaker into a powered monitor. Three models are available, all with balanced-unbalanced XLR inputs and level controls to handle levels from -10 to +4dB. Apparently designed to be attached to the back of the cabinet, the models range from the *120*, delivering 30W RMS into 8Ω, to the *480*, rated at 150W RMS.

From the same worldwide distributor come new broadcast delay lines from Bel Digital, including the *7300* synchroniser-delay featuring an autotracking facility to adjust the audio delay in real-time automatically, maintaining sync with video. Additionally, Bel's *BDE-1000* Digital Signal Modifier allows control of the gain, phase and routeing (including optional mixing) of the four audio channels contained in two AES-EBU signals without external mixers or switchers.

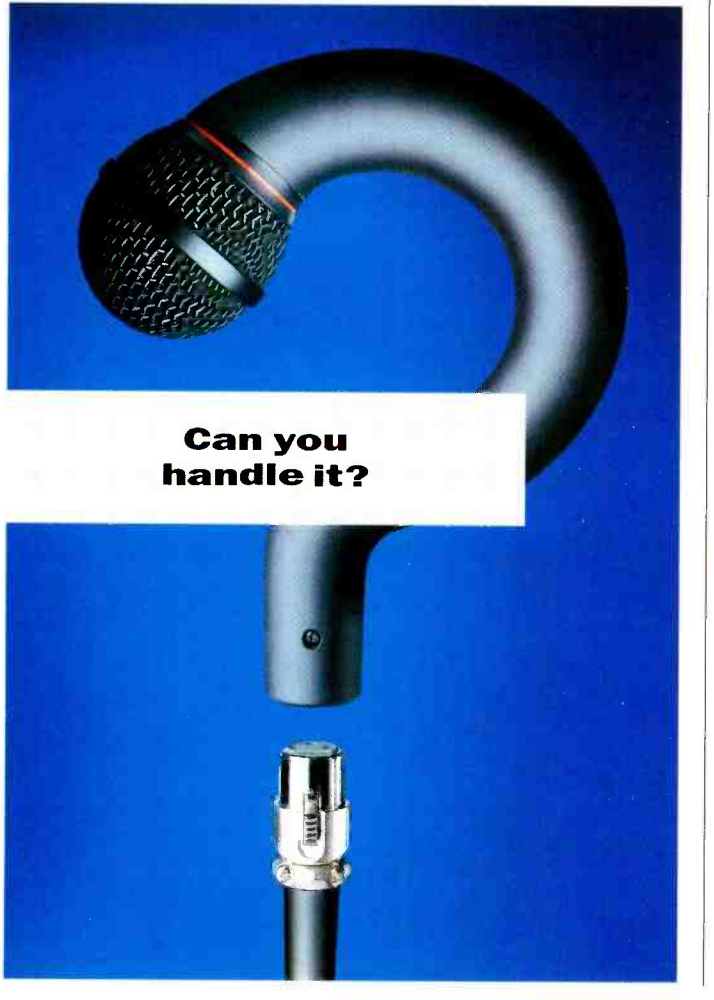
Michael Stevens & Partners, UK. Tel: +44 181 460 7299.

KM 184

The Neumann *KM 184* is the successor to the much-missed *KM 84*, combining a pressure gradient transducer with the transformerless circuitry of the *KM 140* in an all-in-one body, unlike the modular *KM 100* system. Cardioid in pattern, the microphone has an equivalent input noise level of 25dB and a maximum SPL handling of 138dB, giving a 21dB increase in dynamic range over its predecessor. The *KM 184* is fully compatible with the whole range of accessories for the *KM 100* series, except, of course, for those involving detaching the capsule from the body. The integral design, say Neumann, has simplified the mechanical construction enabling the price to be kept to mid-size budget levels.

Georg Neumann, Germany. Tel: +49 30 41 77 24 0.

***KM184* air to the *KM84* crown**



dB Technologies

dB Technologies, manufacturers of the *dB3000* Digital Optimizer, have announced the introduction of the *AD122* A–D Convertor. Utilising proprietary technology, the *AD122* converts analogue signals to a 22-bit digital-audio data-stream. dB claim a combination of superior linearity, fast and accurate transient response, extremely small quantisation steps and low-noise performance, and say that full 22-bit operation resolves signals down to -160dB from full scale, 'approaching the smoothness of analogue.' Further specifications include 0.00009% THD+N, built-in Acoustic Bit Correction re-dithering to 16–20-bit formats, and a switchable digital soft-knee limiter, 'simulating gentle analogue tape saturation.'

Audio Intervisual Design, US. Tel: +1 213 845 1155.

APT codec

Audio Processing Technology will be launching the *MCE-MCD800* digital audio codec at AES in Paris. The unit enables up to eight channels of 24kHz bandwidth to be passed over digital networks such as T1 and E1.

The *DRT128* Digital Reporter Terminal is a new compact portable unit designed to deliver high quality audio commentary in outside broadcast and limited bandwidth applications. It incorporates a two-channel ISDN terminal adaptor with internal IMUX.

APT, UK. Tel: +44 1232 371110. ▶

In-brief

● Audio Ease *Rex* software

The new Dutch software group Audio Ease have recently released *Rex*, a Macintosh utility that copies regions from a *Sound Designer II* file to new *Sound Designer II* files, a process they call Region Extraction. They expect *Rex* to be of assistance when editing long voice-over takes into short fragments, in the production of soundfiles for CD-I or CD-ROM, and in getting CD sound-effects libraries on line. *Rex* is shareware, runs faster than real time in the background and does not occupy sound hardware.

Audio Ease. B.F. Suermanstraat 19, 3515 XK Utrecht, Holland

● New modules for *Seeport*

Seem Audio of Norway have developed a new range of studio modules for their portable mixer Seeport. The modules can directly replace the standard modules, allowing a 30% price reduction for small on-air studios or film-and-video editing suites who do not need all the standard functions for outdoor use. Input configurations are mono mic-line and stereo line, and there are also new telephone hybrid modules for the system.

Seem Audio, Norway. Tel: +47 66 797730.

● *ADAT* System 4.0

Alesis have announced *ADAT* System 4.0, a major software upgrade to the complete *ADAT* system. The important features are over an hour of recording time, single-button record, improved external synchronisation and compatibility with Digidesign's *SMPTE Slave Driver*. An upgrade requires changes to all elements of the system– an EPROM for the machine (except those running software pre v3.00 which need a hardware update), an EPROM and microprocessor change for the BRC, and an EPROM for the AI-2 interface.

Alesis Studio Electronics, US. Tel: +1 310 558 4530.

● XTA *DP100*

Following news of Beyerdynamic France's appointment as their French distributor, XTA Electronics are launching the *DP100* Audio Delay Processor at AES Paris. The *DP100* is a 2-input, 4-output, assignable audio delay with 11-microsecond delay increment, 80 memories and a compensation function for ambient temperature change. XTA's proprietary Audiocore DSP technology supports a range of features including 3-band parametric EQ for each output.

XTA Electronics Ltd, UK. Tel: +44 1299 879977. ▶

Spectrum™ Organ contains 128 presets including classic rock, jazz, gospel and pipe organ sounds. Each preset includes individual vibrato, distortion, reverb, key click and release click settings. These settings can be globally altered from the front panel, or using MIDI controller messages. In addition, each preset contains four drawbar waves which can be accessed in real time using the PC™-1600 MIDI Controller.

- *1 Mb 16-bit Classic Organ Sample Wavetable*
- *128 Presets*
- *32 Oscillators*
- *32 Voice Polyphonic*
- *4 Part Multi-timbral*
- *Voice Pedal Input*
- *Leslie Speed Pedal Input*
- *Stereo Audio Outputs*

Spectrum™ Synth contains 256 (64RAM/192ROM) classic synthesizer presets including analog, digital and hybrid sounds. With 24 dynamic resonant filters, hard sync and pulse width modulation, the Spectrum Synth emulates classic analog synthesizers better than any other digital instrument. Presets can be edited and saved to RAM locations using the PC-1600 MIDI Controller.

- *2 Mb 16-bit Classic Synthesizer Sample Wavetable*
- *256 Presets (64 RAM /192 ROM)*
- *24 Oscillators*
- *12 Voice Polyphonic*
- *12 Dynamic Resonant Filters and 24 LFO's*
- *Poly and Legato Receive Modes*
- *Hard Sync and Pulse Width Modulation*
- *Stereo Audio Outputs*

Spectrum™ Bass contains 200 presets including classic analog and digital synthesized basses, as well as electric, acoustic, fretless and slapped sounds. The Spectrum Bass includes sustained and legato versions of most presets sounds. Up to 4 presets can be layered on separate MIDI channels to create incredibly fat combination sounds. Individual presets can be edited using the PC-1600 MIDI controller

- *1 Mb 16-bit Classic Bass Sample Wavetable*
- *200 Presets*
- *8 Oscillators*
- *8 Voice Polyphonic*
- *8 Dynamic Resonant Filters and LFO's*
- *4 Part Multi-timbral*
- *Poly and Legato Receive Modes*
- *Stereo Audio Outputs*

Spectrum™ Analog Filter is a true programmable analog filter system which can be used to process any sound. It offers a 3-channel input mixer followed by a classic voltage controlled resonant 4-pole filter and voltage controlled amplifier. The filter circuit includes an ADSR envelope, velocity and key track amounts, and is MIDI controllable. The amplifier circuit also offers an ADSR envelope and master volume. 100 program locations allow settings to be stored in memory.

- *Classic Analog 4-Pole Filter Circuit*
- *100 Programmable Locations*
- *3 Audio Inputs*
- *MIDI Note Triggering*
- *Audio Trigger and Envelope Follower*
- *Filter Frequency Velocity and Key Tracking*
- *MIDI Controllable*
- *Mono Audio Output*

PC™ 1600 MIDI Controller This general purpose MIDI controller offers 16 sliders and 16 buttons that can be programmed to send system common or system exclusive MIDI messages. In addition, 2 CV pedals and the data wheel can be used as alternate controllers. The PC-1600 has many uses including programming and controlling any of the Spectrum series sound modules. The PC-1600 comes with 50 presets offering a variety of synth editors, sequence controllers, fighting system controllers, etc. All presets are fully programmable, so as other needs develop, they can be programmed by the user very easily.

- *16 Programmable 60mm Sliders*
- *16 Programmable Buttons*
- *2 Programmable CV Pedal Inputs*
- *Multi-function Data Wheel*
- *50 Programmable Memory Locations*
- *100 "Scene" Memory Locations*
- *Setup String Send on Patch Recall*
- *MIDI Dump/Load Capability*

GET WHAT YOU NEED. . .
AND WANT. . . AT AN AFFORDABLE PRICE!

In a world of keyboards and sound modules which claim to offer "every instrument sound known to man," Peavey realizes that you probably don't want, or need, all of that! **The Peavey Spectrum Series** sound modules are each designed to do one thing--offer specific instrument sounds you do want. The Spectrum **Organ, Synth, and Bass** units offer unique features and capabilities needed to produce the most realistic reproduction of its particular instrument family.

Complementing the Peavey Spectrum Series sound modules are the **Spectrum Analog Filter** and the **PC-1600 MIDI controller**--offered to make the Spectrum Series modules even more powerful. The Spectrum Analog Filter will add that fat, classic and true analog sound to whatever you plug into it. And the Peavey PC-1600 MIDI controller allows programming and controlling of any Spectrum module.

Amazing sounds, amazing simplicity, amazing flexibility, and truly amazing prices! The only thing about the Spectrum Series that is not amazing is it's from Peavey...the company dedicated to giving musicians everything they need...and want!

For complete Peavey keyboard information, pick up the latest edition of *KEY ISSUES* at your local Peavey dealer.

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PEAVEY

Otari *Status*

Launched at the recent NAMM show in Anaheim, Otari Corporation was showing the first in a new family of consoles. The new line of digitally-controlled analogue consoles, dubbed the *Status*, carries forward the design philosophy Otari began with the *Concept 1*.

The first console, the *Status R*, has on-board automation and a set of computer-controlled features that Otari claim has never before been available in the price class. Designed for the music and postproduction markets, each input module has two independent signal paths and a 4-band EQ that may be assigned to either path or split between the two. Most of the console's routeing functions are under central digital control, allowing console-wide master switching and storage of module presets for future recall.

Otari Corporation, US.
Tel: +1 415 341 5900.

The Otari *Status R* music and postproduction console

Dancing Faders

The latest addition to the Fostex *Foundation* line is the *DFM* Dancing Fader Mixing System. This is an assignable control surface with ten long-throw moving faders used for mixing audio, setting DSP parameters and controlling automation. The DFM is designed to complement the *Foundation* Edit Controller, with which it enables all the recording, editing and mixing features inside the *Foundation 2000*.

Besides the faders, seven large rotary encoders give quick access to parametric EQ, panning and compression-limiting. All DSP and mixer parameters can be saved in up to 999 'total recall' snapshots, and all fader and pan positions can be continuously automated.

Fostex Corporation, Japan.
Tel: +81 425 54 611.
Fostex, UK. Tel: +1 310 921 1112.

Sabine FBX-*Solo*

Sabine have expanded their line of Feedback Exterminators with the introduction of the *FBX-Solo*, a miniature model designed for use on individual microphones. Intended to be inserted into problem channels, the *Solo* is aimed not only at individual vocalists but churches, auditoriums and theatres. Two versions are available, one operating at line level and the other including a microphone preamp with switchable phantom power, and six solos will fit into a 1U-high rack tray. Both models use Sabine's patented 1/10th-octave adaptive digital filters to automatically sense feedback and reduce the problem frequency as much as is necessary to remove the feedback. The *Solo* models include six such filters and a filter locking feature which freezes the fixed filters created during setup so that they cannot go deeper than their original setting.

Sabine, US. Tel: +1 904 371 3829. ▶

● ARX *MicroMAX 1*

Australia's ARX Systems have announced the release of the *MicroMAX 1* full-range loudspeaker system. The speaker features a new sculptured, radial-front, mesh-baffle design and trapezoid-angled sides for easy arraying and placement. Drivers are an 8-inch for LF and a 1-inch dome radiator. The *MicroMAX 1* is designed to be used as a system with the MicroPRO Loudspeaker Processor, a dual-channel unit that supplies EQ trim, subcrossover functions, phase correction and ISC speaker protection.

ARX Systems Pty Ltd, Australia.
Tel: +61 3 555 7859.

● Optifile *Tetra* moving faders

The first moving-fader console automation system from AD Systeme, makers of the VCA-based Optifile *Tetra*, will be launched at the Paris AES. Known as Optifile *Tetra FM*, this latest release is the result of collaboration between VCA and software specialists AD Systeme and Audiomation, manufacturers of moving fader hardware.

Ad Systeme, France.
Tel: +33 1 42 53 3118.

● Denon *DN-995R*

Based on the *DN-990R* MD cart recorder, the *DN-995R* offers a full automation and synchronisation package and adds a range of enhancements requested by end-users. New features include External sync I-O, allowing multiple machines to be controlled by the clock in the master machine, switchable RS-422A/232C, a *Windows* software control package, calendar Autostamp recording and improvements to some of the transport functions.

Hayden Laboratories Ltd, UK.
Tel: +44 1753 888447.

● Continuing Dialog

German networking experts Dialog4 are to extend their product line at the Paris AES. New arrivals are the *MusicTAXI VP* ISO-MPEG audio networking codec, the *PC-MPEG Decoder* and the *MT-Reporter*.

Dialog4 System Engineering, Germany. Tel: +49 7141 22667

● Soundcraft *K1*

The *K1* is a compact four-bus multipurpose mixer specifically aimed at the professional installer. Stereo inputs, six auxiliaries (switchable pre-post) and a matrix are fitted as standard, and the modular blocks of four inputs allow mono sections to be replaced with stereo channels.

Soundcraft Electronics Ltd, UK.
Tel: +44 1707 665000. ▶

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AT THE MUSIK MESSE, FRANKFURT
MARCH 8th - 12th 1995

USED & ENTHUSED

Put yourself in front of a Soundtracs Jade Production console and you know immediately you're in for a treat.

If the clear and concise module layout and big feel don't make your mouth water, turn it on, play some audio and listen.

It's then that you'll discover the sparkling highs, the distortion free lows and the smooth transparency of the patented FdB Parametric Equaliser which has helped make the Jade console 1st choice in so many of the world's most successful recording studios.

The engineers who work in these busy environments enthuse over the exceptional audio quality and time saving features built-in to every Jade.

Features that include the innovative Assignable Dynamics Processor which provides DSP control over gating, expansion, compression, limiting and modulation on every channel; and the efficient VCA fader automation available on a combination of up to 64 inputs with mix information stored along with Dynamics settings for speedy recall.

Features that demonstrate the unique understanding Soundtracs have as to the design and manufacture of quality recording consoles for the demanding sessions of today's music industry.

Unparalleled sonic performance with reliability designed in and problems designed out.

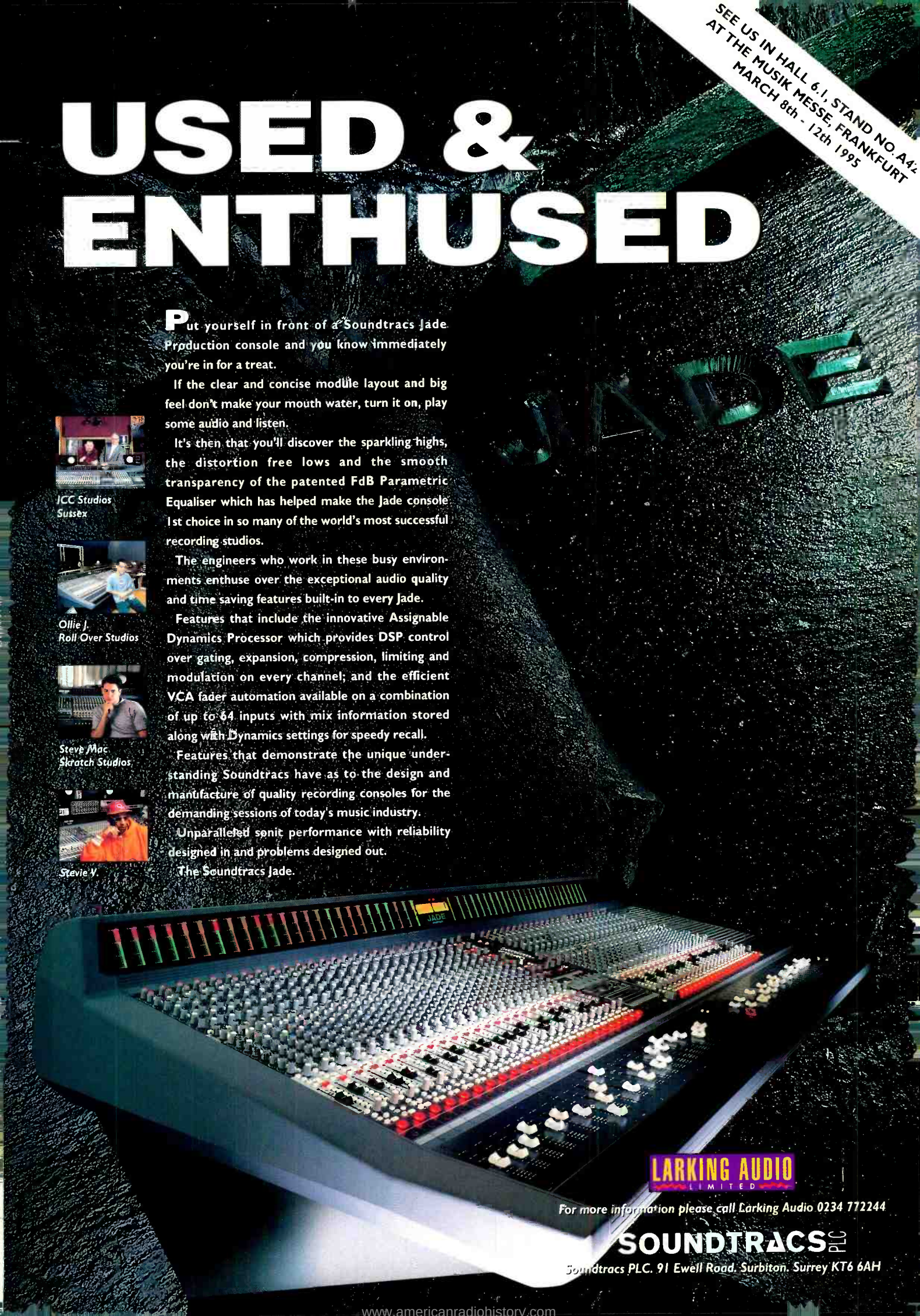
The Soundtracs Jade.

ICC Studios Sussex

Ollie J. Roll Over Studios

Steve Mac Skratch Studios

Stevie V.



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D&R *Airmix*

At AES and Frankfurt D&R will be showing their new broadcast console, the *Airmix*. It features VCA-controlled faders, a minimum of control knobs and switches and a maximum of intelligent digital control. The *Airmix* comes with two input-module types, a triple-input module accepting mic signals as well as line and RIAA equalised inputs, and a dual Telco module accepting incoming signals from two telephone lines. This module handles extensive signalling from incoming calls and communications with directors, announcers and technicians. Up to 16 modules can be loaded into the chassis, which also has a central master section with a script space below, and the console is designed to be equally at home with preproduction work and on-air broadcasting.

Also on display will be a production model of the new automated *Merlin* dual line-input mixer, minus the initially planned recall system which was not considered sufficiently attractive to end-users; the price has been decreased accordingly.

D&R Electronica, Netherlands.
Tel: +31 2940 18014.

JRF replacement *A80* heads

JRF heads

JRF Magnetic Sciences have introduced new Flux Magnetics $^1/_2$-inch 2-track replacement record-and-playback heads for the Studer *A80*. The heads are built using the latest generation of high permeability, high saturation, low coercivity, metal core materials, and the design is claimed to give increased tape stability, increased bias and record headroom, and 20Hz wavelength playback response at 30ips. JRF recommend the heads for critical, quality, music recording and mastering applications, 'particularly those requiring near-digital dynamic range combined with the warmth and 'air' of the analogue format.'

JRF Magnetic Sciences, US.
Tel: +1 201 579 5773.

Dream Broadcast

Digital Music's *Dream Broadcast* is a multitrack (2 stereo) hard-disk recording, editing and mastering system running on an included 486/66MHz VESA local bus PC with an 850Mb fast hard drive. The system runs under *Windows* and handles 16-bit audio at sampling frequencies up to 48kHz. This Broadcast version is specifically aimed at applications such as voice overs, jingles, ENR editing and cart machine replacement.

From the same company comes CD-R Pro, a fully integrated mastering system complete with built-in Yamaha *Quad Speed* CD recorder and Corel *CD Creator*, suitable for CD Audio, CD-ROM and mixed mode CDs.

Digital Music, UK.
Tel: +44 1703 252131.

Apex *CDR 2000*

Launched at AES Paris will be the Apex *CDR2000*, which Apex say is the first of a new generation of CD recorders and the first stand-alone machine which can make frame-accurate master recordings. The unit is a modular audio CD recorder which has the facility for expansion through free slots. Standard features include a sample-rate convertor, a DAT interface with a delay feature for translating start IDs, a MIDI interface, ISRC code handling, quality control, and double-speed copies through the standard optical I-O.

Expansion possibilities include a SCSI board to turn the *CDR 2000* into a fully-featured SCSI writer for direct recording from hard disk systems, and a SMPTE board allowing for exact PQ encoding, frame accurate against time code.

Apex NV, Belgium.
Tel: +32 89 306313. ■

● **Electro-Voice *RE2000***

The *RE2000* studio condenser mic from American manufacturer Electro-Voice has been annouced for launch at the Paris AES Convention. The mic uses a short-D transducer and features a low-mass diaphragm, transformer-balanced output, external 'computer grade' power supply, 10dB capsule attenuator and switchable 12dB-per-octave roll off at 130Hz. The *RE2000* also employs a Constant Environment System to combat adverse effects of the operating environment.

Electro-Voice Inc, US.
Tel: +1 616 695 6831.

● **CEDAR OEM**

Set to debut at the Paris AES is the CEDAR *ProDSP*. This OEM DSP board is derived from the powerful CEDAR system hardware and uses the AT&T DSP32C processor. to support up to 512Kb, zero wait-state RAM per DSP, an FPGA chip, on-board AES-EBU and SPDIF I-O, two 6.144Mbaud expansion connectors for *ProDSP* daisy-chaining, and daughter board connectors for aux I-O. *ProDSP* has been developed for multimedia applications and is compatible with D3EMU development programming. A development kit is also availbale and offers fully developed software libraries for the host and ProDSP.

CEDAR Audio, UK.
Tel: +44 1223 464117.

● **ART limiter and gate**

The new ART *MDM-8L* is an 8-channel limiter optimised for use with tape-based modular digital recorders, while the*Eight Gate* is an analogue 8-channel, noise gate featuring channel linking, balanced inputs and outputs and visual display of all parameters.

ART, US. Tel: +1 716 436 2720.

▼ **Sony *DTC60ES***

Aimed squarley at the project studio, the Sony *DTC60ES* DAT recorder offers 44.1kHz sampling, optical and coax I-Os, 1-bit D–A convertors with 20-bit resolution. Significantly, the *DTC60ES* incorporates Sony's Super Bit Mapping system and offers extended performance over conventional 16-bit systems.

Sony Corporation of America.
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ADV 469/1/E

Jünger *d01*

As the digital-audio signal chain becomes more complete in music production, broadcasting and postproduction, there arises an increasing requirement for easy to install, external, digital, signal processing. In particular dynamics processing that will maximise levels while preventing any overs to tape or transmitter.

The German company Jünger have responded by introducing a range of three digital dynamics processors called the *d* Series. The three 19-inch units are similar sharing identical circuitry: the *d01* reviewed here) is a compressor-limiter-expander; the *d02* has the same configuration but also offers a 20-bit A–D convertor but fewer types of digital I-O; and the *d03* offers sample-rate conversion in place of expansion.

One of the most striking things about these 1U-high stereo units is the lack of user-controllable parameters—the *d01* has just four front-panel step switches that control dynamics settings: COMPRESSION RATIO; EXPANDER THRESHOLD, INPUT GAIN, and a PRESET SELECTOR. All other parameters, including attack and release times, are controlled automatically by what Jünger refer to as Multi-Loop processing.

The result of this is that the unit requires very little in the way of setting up, and is consequently extremely simple and quick to use. However, this is something of a double-edged sword—although operational simplicity is obviously a plus point, the trade-off is limited user control which will undoubtedly be viewed as restrictive by some users.

However, the *d01* should perhaps be seen in a different light compared to conventional dynamic devices—for a start this is more of a corrective and protective tool than an effects device, and although it can do a good job of thickening programme material and making things appear louder, it does not colour or distort the sound in any way. In fact the most impressive thing about the *d01* is that it processes dynamics with very little perceived change to the original signal—and often it is only the meters that give away the fact that processing is actually taking place.

Unlike conventional analogue devices, the *d01* compresses signals evenly across the entire dynamic range rather than affecting just the portion above a set threshold. This means that the processed signal retains its original dynamic relationships (although somewhat condensed) which plays a significant part in the transparency of the processing.

Also because the system does not work with fixed attack and release times but adapts these values depending on the nature of the input signal—long attack times during even source programme, and short attack times for transients—the unit manages to intelligently deal with processing and avoid artefacts such as pumping, breathing and distortion and general coloration. Apart from adding to the overall transparency, this also does away with the 'one setting suits all' approach required in setting up conventional dynamics units.

The *d01* claims that it will prevent all digital over levels, and during test this proved to be quite correct; with the limiter switched into circuit (the default condition), the unit managed to catch every stray transient irrespective of speed. This has been achieved by introducing a 2ms delay into the signal path which allows time for a preview circuit to analyse the input. Control data is then passed on to the limiter, thus enabling corrective levelling to take place in absolute time. Although this guarantees no overshoots, it does, of course, mean that the processed signal will be delayed, and if this is to be mixed back with other signals, time alignment will be necessary to achieve proper sync. However, in the majority of cases where the *d01* has applications, the signal will be a mixed output, so this will not present problems.

The four presets—Universal, Classical, Pop Music, and Speech—have been added to provide 'optimised settings' for different progamme material. These settings affect the attack and release times, and the way control parameters interact. In practice, however, the difference between presets appears extremely subtle, and in many cases totally indistinguishable.

The gain make-up of the compressor is automatic, and the lower the signal the greater the gain will be. The maximum amount of compression gain, though, can be adjusted between 2dB and 15dB, thus making allowances for any obtrusive background noise that may be elevated. Noise is also effectively dealt with by the expander.

Apart from dynamic processing the unit can also provide some other functions. The first of these is Format Conversion: because all digital outputs are available in parallel (regardless of the input format) the unit can simultaneously output AES-EBU. SPDIF, Yamaha Y2, and SDIF-2. The unit can also perform as an D–A convertor using its 18-bit 64x oversampling stereo D–As to provide a balanced analogue output with adjustable level. Another function is De-emphasis: a De-emphasis Filter, which is switched from the rear of the unit, can be used separately without dynamic processing. De-emphasis is automatically switched for AES-EBU and SDIF-2 signals, but must be manually switched for SDIF and Y2. During dynamics processing it is essential to avoid the elevated high frequency will be compensated for.

The *d01* offers an easy to use, all-digital processor with truly transparent dynamic processing. the unit has applications where an all-digital chain is required, and maximises digital full-scale resolution without clipping. It also successfully levels programme material from track to track or programme to programme, and is effective at making signals louder and denser without introducing unwanted artefacts. On the downside are the lack of user adjustable parameters, and the delay introduced by the preview circuit which makes the unit unsuitable for track laying or overdubbing. ■

Patrick Stapley

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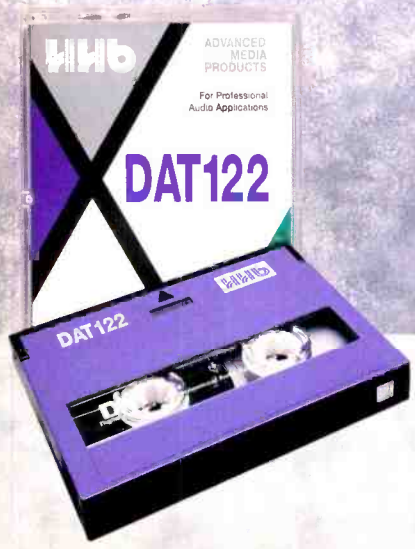

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MIDI-Scope

With the popularisation of MIDI as a means of equipment control, there comes a very real requirement to be able to monitor accurately MIDI data streams. Ideally, we would all like to be able to hook up a familiar computer sequencer and view what is happening on something resembling an edit page, but in reality the call is for something a little more portable and specialised. The *MIDI-Scope* promises to be be just such a box.

British manufacturers Artistic Licence are perhaps better known for the *Lamp-Tramp* lighting console —with its 1,024 dimmers and MIDI remote control which saw action on recent tours by Pink Floyd and The Rolling Stones—but the *MIDI-Scope* addresses many of the functions that a hand-held analyser ought to—it is small, it is battery-powered and it is quite clever.

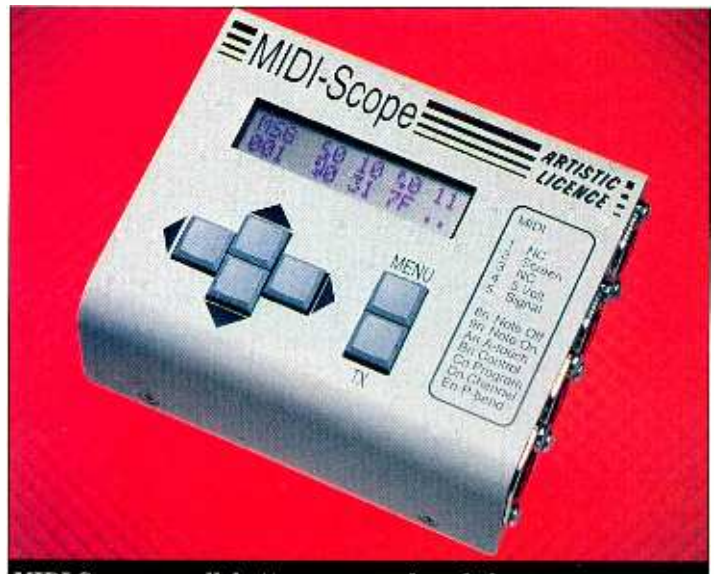
Operation centres around a MENU key which selects the operating mode of the unit, a TX button used for data transmission purposes and four cursor keys which take care of cursor movement around a 16 x 2 LCD as well as parameter selection.

MIDI-Scope uses an external PSU which recharges the internal batteries giving around 30 hours of continuous use for around 3-hours of charge time. The complete package is presented in a sort of leatherette-style handbag.

Heavy-duty connectors are found on the right-hand side panel with MIDI In, Out and Thru plus a Boost output (also switchable to Thru operation) which supplies a RS485 signal for driving longer cable lengths. The box reverts to a power saving Standby mode if no key is pressed for ten minutes.

There are eight Menu modes to choose from which outline the *MIDI-Scope*'s capabilities. It displays received MIDI data in hexadecimal or as command icons, while errors are displayed as icons. MIDI information can also be captured and copied to one of eight memories, displayed and edited and then selected for transmission. Additionally, *MIDI-Scope* can check for shorts between pins 4 and 5 of a MIDI lead connected to the Boost out without disconnecting the other end.

The fact that *MIDI-Scope* permits data to be transmitted at the same time as it receives it means that it is able to test merge, Thru and filter functions on units that perform such roles. Error messages are shown for framing and for overrun errors. The choice of hex or icon display is considerate but I reckon that most users will prefer the latter simply because of its immediacy. Even using icons, the display takes some getting used to because of the number of icons and the amount of data involved and the manual fails to communicate the true potential of this device.



MIDI-Scope—small, battery powered, and clever

Individual icons represent note on and off, aftertouch, control change, program change, channel pressure and pitch bend in a manner which is still an altogether more satisfactory way of interpreting the MIDI stream than hex ever is.

The capture buffer—which can be activated by a user-defined sequence of four characters called a capture trigger—is 128 characters deep and is shown in blocks of eight characters on the LCD that can be scrolled through and saved to one of eight messages. These in turn can be selectively transmitted from the device's MIDI Out with the additional benefit of a Continuous mode in which the message is sent repeatedly with the TX button functioning as an ON-OFF switch. There is also the facility to send consecutive characters from a transmit message or to send the entire message whenever the programmed capture trigger is received.

MIDI-Scope avoids the common pitfall of cheaper MIDI analysers, which only show you what is happening, by adding the ability to effectively test with the set and get proactive. Admittedly, MIDI data type is not as instantly recognisable from the *MIDI-Scope*'s LCD as it would be with something like an illuminated LED, but it also displays values.

With many MIDI anomalies the fact that a MIDI event is occurring is generally only half of the problem and the amount that is occurring often helps to complete the picture of the extent of the problem and consequently aid its isolation and cure.

The fact that the *MIDI-Scope* can also transmit data is its biggest bonus as you can store custom sequences in the unit's non volatile memory which will permit testing of equipment. You capture a string which will enable you to ascertain whether a connected piece of equipment is responding correctly and you can loop it in to the *MIDI-Scope* to simultaneously check whether what is coming out of it is what you expect. Some form of switchable back lighting for the LCD would have been welcome for use in low-light situations—particularly as many MIDI problems do tend to involve crawling around in near darkness at the back of racks.

The device is definitely priced with the professional user in mind (£400 UK) but its build quality suggests it will run and run even when subjected to professional misuse. A useful device that achieves what it sets out to do. ■

Zenon Schoepe

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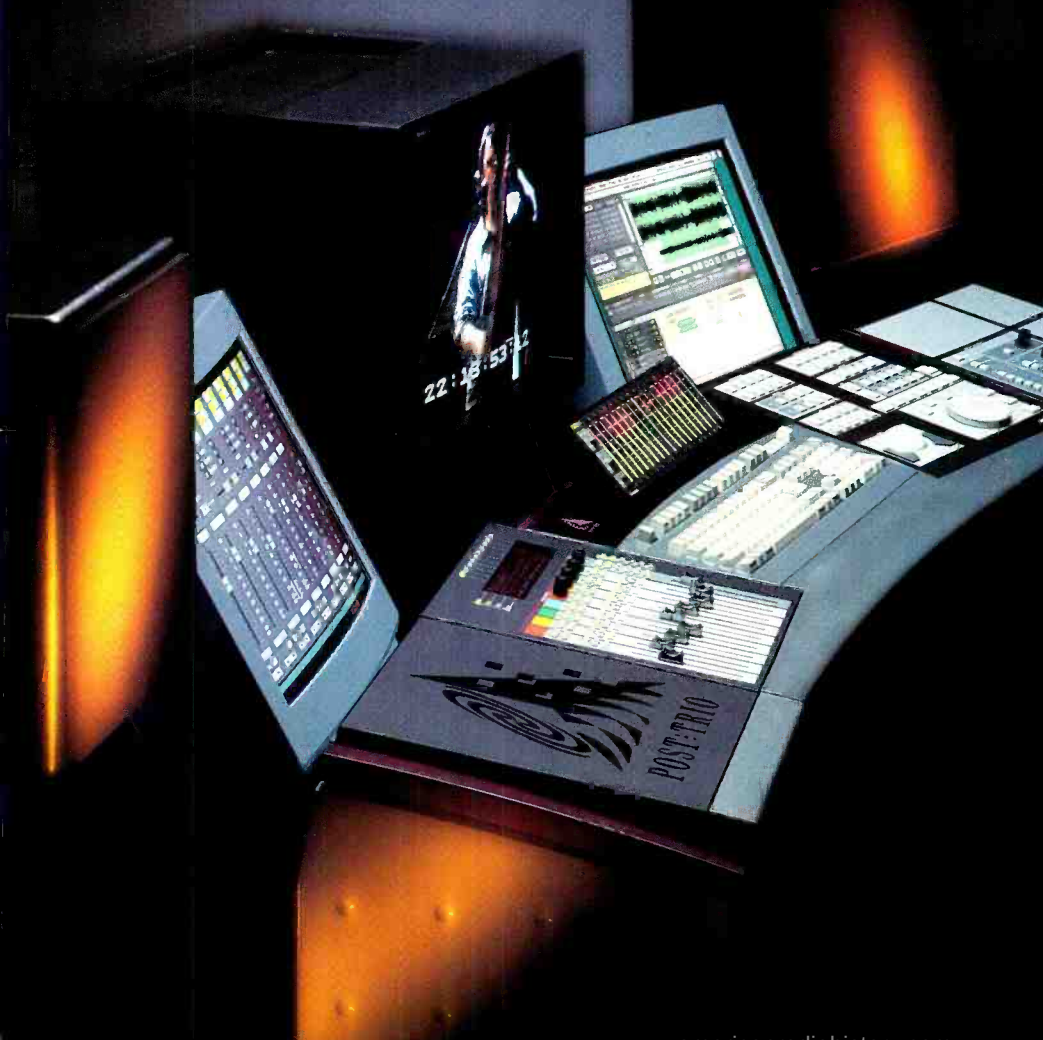

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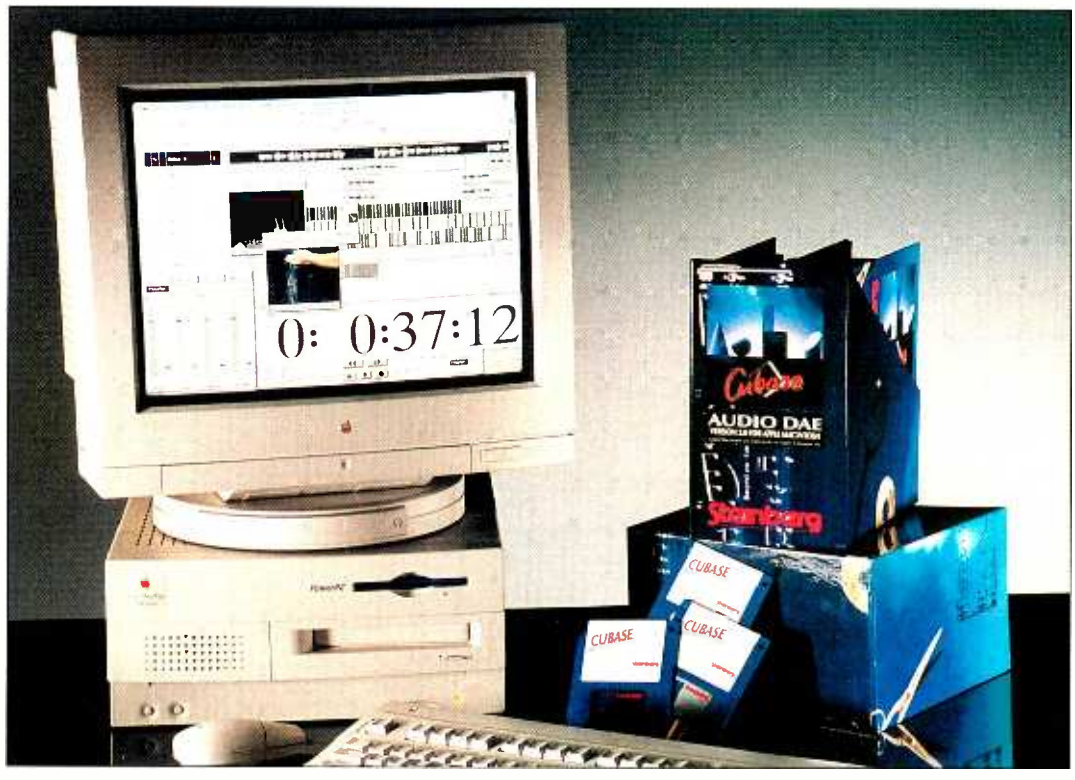

The obvious increase in operational ease and 'funkiness' of computer-based MIDI sequencers has largely obscured the true growth potential and value that this software has as a means of education. It is in this area that the manipulative possibilities afforded by a sequencer—if harnessed wisely—can be used to illustrate musical points and to give examples which can be modified, altered and rearranged by a good teacher to make the musical experience a little more immediate and, dare I say, fun for the pupils.

Steinberg, now in their tenth year of software manufacture, have embarked on a remarkably powerful drive into the educational sector. The company argue the case that people are more likely to buy a system that they have learned on—and, no doubt, they also recognise the bulk buying potential that educational establishments represent.

The Steinberg Education Division was established late last year and builds on the company's almost unrivalled market dominance in certain territories in which they now offer a policy of 'multipack' buying, allowing educational establishments to purchase multiple software packs at reduced rates. Thus, such institutions can buy a pack of five *Cubase Score* programs with one manual and educational tutorial or a 10-program pack with two manuals and tutorials.

Cubase Lite and *Score* for the *Macintosh*, IBM PC or Atari are produced in educational packs and contain the standard version of the program plus a 2-tutorial booklet with disk of musical examples. This is in addition to the more advanced features of the company's *Cubase Audio* range. *Cubase Lite* features can be learned by students individually or in groups through the first Tutorial which includes worksheets and multiple choice questions on notation and basic music theory. *Cubase Score* Tutorial 2 features a near-interactive approach based on popular musical passages with on-screen information about the composers, musical styles and forms.

This Tutorial is perhaps the more interesting of the two being aimed at the 14-year-old and above and it starts with the premise of whether the course of musical history would have been changed if the great composers had had sequencers in their day. The answer is quite obviously a resounding yes but we should be grateful that they did not. However, this Tutorial does hook computer interest into music in a way that, while it may not be highly modern in musical style, is decidedly hands-on and well planned.

The ball starts rolling with *Greensleeves*, in which students change the tempo and record a bass line followed by variations imposed on the recorded bass line. Program changes, mutes and solos are demonstrated on a piece by French composer Arbeau followed by work by Handel and JS Bach. (The latter, we are informed, still managed to knock out 20 offspring despite a hectic musical schedule.) *Score* displays at this point demonstrate the different constituent parts of a piece and the student is encouraged to edit, name and move parts around with Copy, Transpose and Paste functions.

The editing theme continues with master tempo alterations and demonstrations of key signatures and split staffs for piano scoring followed by quantisation and note moving.

A Haydn quartet illustrates clef changing, note moving and velocity changes in addition to revising copying, transposing and the naming of tracks. Beethoven gets the treatment for showing the difference that velocity makes to a passage along with step entry and adding dynamics on the score. Dvorak's *New World Symphony* is used to show transposition and the exploding of parts for proper scoring.

We then leap to the 20th Century —by way of ragtime—to demonstrate syncopation and enharmonic shift. A brief flirtation with Schoenberg, meanwhile, demonstrates the sequencer's ability to reverse play. The Tutorial is capped by practical outlines of how to arrange and print out a score with percussion staves and chord symbols.

All along, the session requires the student to have at least a basic ability to get around the sequencer and (presumably) this must be addressed by the teacher at an earlier stage to bring them up to a level where the exercises will be meaningful.

General computer literacy is likely to be more of a requirement for the student than natural musical flair and, consequently, the balance is about right for the intended age group. However, it is worth pointing out that the tutorials are as good at teaching *Cubase* as they are at teaching an understanding of music and it is up to the teacher to decide which side of this thin line the lessons should lean towards.

As an observation, it is perhaps unfortunate that the organic music chosen for the examples should be given the cold technology treatment on one level in analysing its substances as just a series of copies, edits and sequencer functions. However, anything that drags young people away from computer games and back to music in general has to be applauded.

While these tutorials may not seem to be the universally ideal training package for all budding musos, it does show the flexibility of sequencing software in teaching music. As is always the case, the ability and skill of the teacher to harness the pupils' energy and their abilities with a sequencer could produce a near limitless number of musical exercises for entertainment while educating. The teacher's skill at adapting this method of learning is critical and to get the most out of it they will have to be computer and sequencer literate themselves. It raises the matter of teacher training, something that Steinberg supports through its own regional teaching courses.

The fact that Steinberg are promoting the educational aspect of their music software must be welcomed. It is up to the teachers to present it as elegantly and suitably as possible.

The big question is not whether the course of musical history would have been changed if the great composers had access to sequencers but if the course of musical education would have been changed if schools had had these sort of tools sooner. ■

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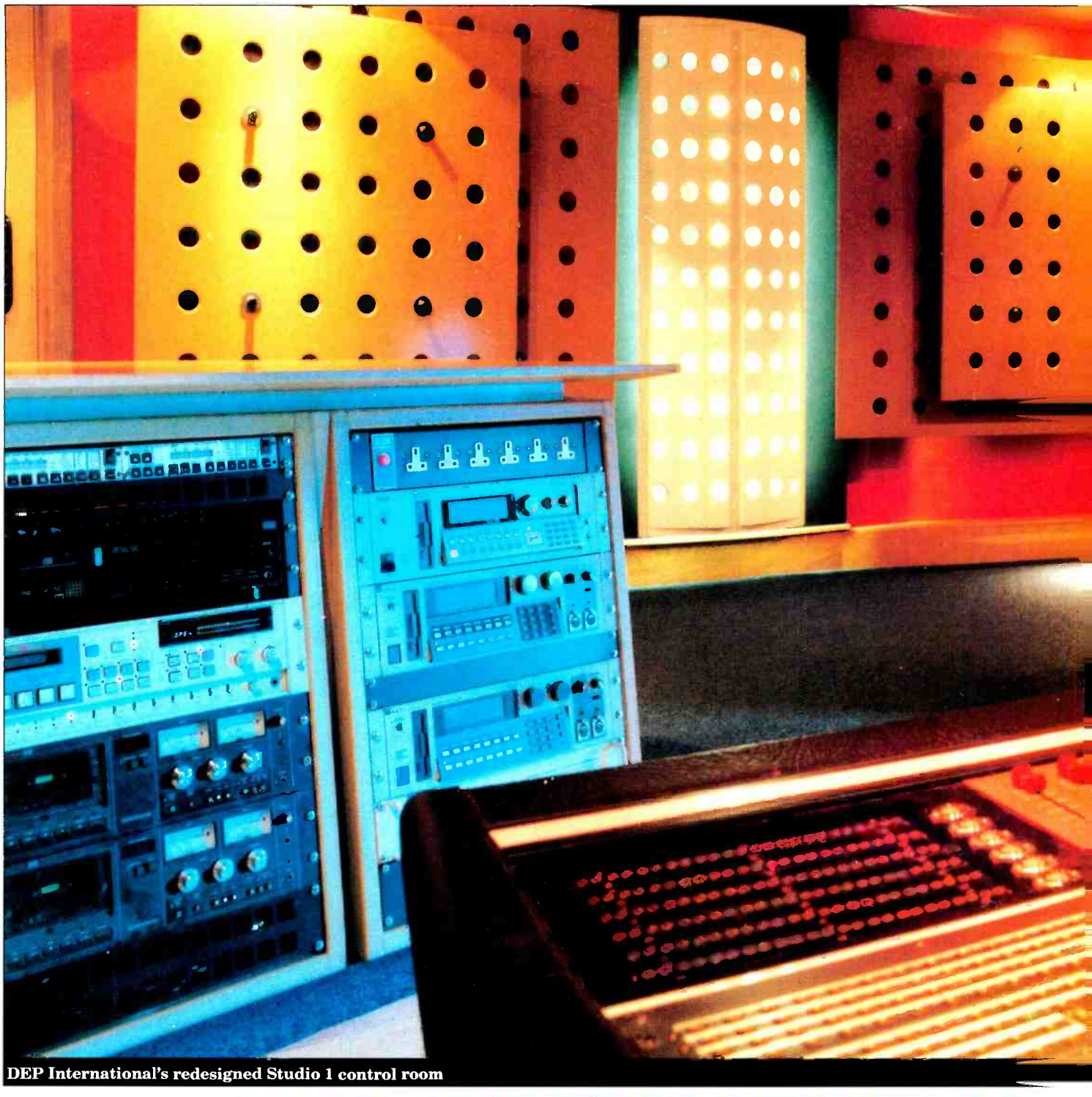


DEP International's redesigned Studio 1 control room

DEP INTERNATIONAL

Bringing an independent studio into the commercial arena has proved to be a challenge for pop artist-owners UB40. Caroline Moss visits the British studio with a new design, a new name and a new agendum

It is common knowledge that for the past 12 years UB40 have owned a studio in the Britiain's midlands. Less well-known, however, is that the studio is also run as a commercial facility. Now, DEP International Studios have undergone a complete renovation, and there is a drive to cast off the 'band's private studio' label—and to lose the marketing-unfriendly label of the Abbatoir, the studio's former name reflecting the building's original use.

There are two main principles behind the renovation of DEP International: to provide the band with a well-equipped and versatile studio which can run as a commercial facility when they are not using it and to provide Birmingham's local musicians with a good, affordable track-laying facility.

'The studio has primarily been adapted to take into consideration the band's requirements for their changing recording techniques,' says Studio Manager Nick Phipps. 'However, it has always been a commercial facility also, and that has not been widely known in the market place.' Consequently, Phipps admits that the studio has become a 'bit of an anomaly,' with potential clients, especially locally, unaware that the studio could be booked commercially.

DEP International lends itself to being a fully-functional commercial facility due to UB40's working patterns—six months in the studio every couple of years followed by between one and two years on the road. Most major labels have booked the studio at some time in the past, but only for relatively local acts such as PWEI, The Charlatans and Julian Cope. Now the studio is being actively marketed nationwide as well as internationally.

Constant renovations have been made throughout the building over the last four years, mainly in administration and recreation areas. However, in early July last year the basement studio, Studio 2, underwent a full refit and facelift. This was completed within a month, and major work began on Studio 1, resulting in Birmingham's only world-class 48-track commercial facility.

Much of the work has been implemented by the studio's strong technical team. Technical Manager Ron Pender, Chief In-House Engineer Mike Exeter and Assistant Engineer Programmer Dan Armstrong have worked together for over a year and have been involved in all stages of the refit. With the band away in Australia for the duration, this meant that responsibility for the project lay with the staff. 'It was to a certain extent an act of faith on the band's behalf,' says Pender. 'They don't know Mike and myself very well and here we were saying "We think you should do this, and it's going to cost you quite a lot of money".'

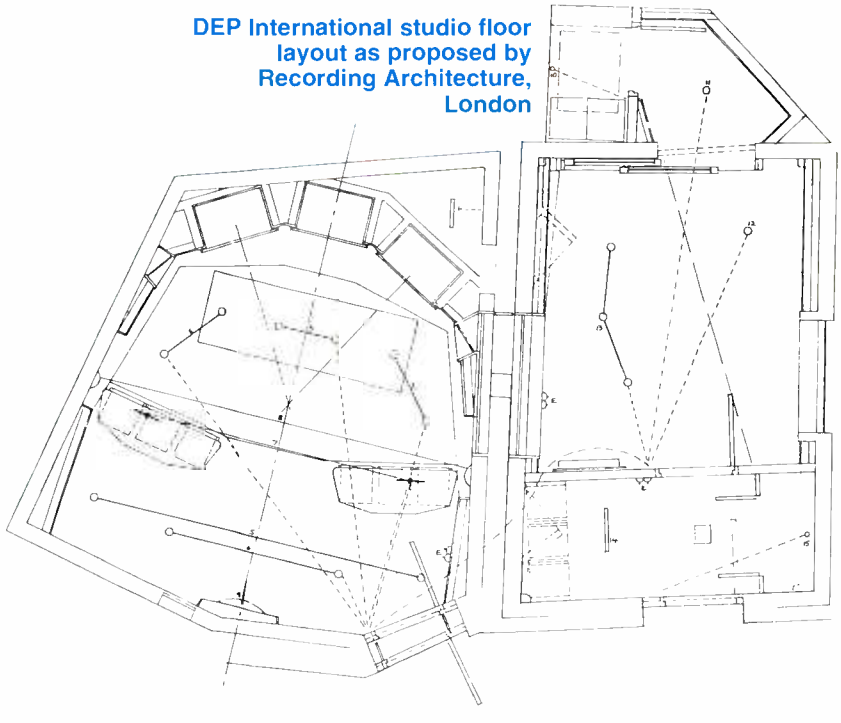
The project proceeded with the help of 'a tremendous accountant' who sold the band on Pender's and Exeter's abilities as much as on the need to spend the money. This extraordinary trust was fully rewarded when the band members returned to their refurbished studios. 'They've all been stunned and amazed by it,' says Pender. 'The whole atmosphere of the place has changed.'

Studio 1

The main changes have taken place in Studio 1, which was gutted, and the control room and recording areas reversed. Roger D'Arcy from Recording Architecture designed the studio, with Nick Whittaker responsible for acoustics. 'We very much left the visual aspect of it in their hands as much as we did the acoustic,' says Pender. 'They agreed with us that it would be better if the rooms were swapped over, and after listening to our requirements and doing some measurements they came to the sort of conclusions that we had reached. They've come up with a pretty amazing scheme for using the space, because it's not a huge control room but we've got the space where we need it.'

The decor in the control room is light and simple, giving a feeling of space. Much of the constructional material is ▶

DEP International studio floor layout as proposed by Recording Architecture, London



The live room with Neuman *TLM 170* mic in the foreground

economical and ecological MDF, although some hardwood finishes have also been used. The room has been designed around the ATC *SCM-300* monitors—a subject of considerable enthusiasm for Pender and Exeter. 'Mike and I are amazed that we took the decision to go all the way from a horn-based system to essentially a soft-dome-type of system, because of the latter's response and much wider dispersion,' says Pender.

'We had to try to change the band's way of thinking away from the Ureis which they'd previously had in here,' adds Exeter. 'With eight members in the band, dispersion's a real consideration—everyone at the back of the room wants to be able to hear as well as those nearer the speakers.

The decision to purchase the ATCs was made after a visit to an installation of the monitors in BBC Pebble Mill's Recording-Architecture-designed room. 'They just blew us away,' says Exeter. 'They are incredibly detailed.' So converted is he to the ACTs that he even uses them to mix on. 'There is a widely-held belief that you can't mix on main monitors, but I think more and more people are finding that you can,' he says. 'You can play stuff through the ACTs, go down to the *NS10*s, transfer over to boom boxes, and it translates really well. The ATCs downstairs are very similar sounding speakers so work done downstairs translates very well onto these as well.'

The main monitors represent the largest purchase decision in Studio 1 to date, as the desk, a DDA *AMR 24* with Optifile *3D* automation, has not been replaced. 'We've decided to stick with this desk and see how we get on,' explains Pender. 'Because there are so many changes within the industry we thought we'd just hang fire. The band tend to do all the tracking that is necessary downstairs, so this is more of an overdubbing and mixing environment anyway. What we have done, however, is to make provisions within the structure of the building so that when we do change over it won't be too horrendous a job, we'll have the spatial and wiring requirements to deal with it.'

The Studio's tape machines are two Otari *MTR 90II*s with *Lynx* synchronisers and 48 channels of Dolby *SR*. Recording rates are a reasonable £350 per day for 24-channel (plus engineer and tape) and £450 for 48-channel.

Studio 1 is particularly well equipped with outboard equipment, some of which was in the original studio, some sourced from companies such as Music Lab and HHB, and some bought secondhand from the Audio Toyshop. The selection includes Drawmer *1960* mic preamp-compressors, Focusrite *Red 2* and *Red 3*, Amek *9098* mic preamp-equalisers, Tubetech *LCA 2B* compressors, Eventide *H3500* and *H3000* harmonisers, Roland *SDE-3000*s, Yamaha *SPX900* and *990*s and much more. then there are the three 32Mb Akai *S1100*s, and UB40's extensive range of touring instrumentation can also be drawn upon. Not many London studios of a similar standard are as well equipped, so does this self-sufficiency highlight Birmingham's isolation from the UK recording scene?

'We're not entirely devoid of studios in Birmingham, although that is one of the reasons why we have tried to provide an extensive range of outboard gear,' says Pender. 'However, this is one of the attitudes we want to overcome. Birmingham isn't as far away from the rest of the living world as people might imagine, and I say this with feeling having worked for much of my life in London. There's a lot of local talent but we're also aiming to attract business from outside the area and indeed the country.' In fact Birmingham is very accessible, being less than two hours away from London by train with good rail links to the rest of the country and an international airport 15 minutes from the studio.

So the new Studio 1 at DEP International is a well-equipped, designed and laid-out facility promising to service local, UK and international recording needs at reasonable prices. But how far have UB40's own recording requirements dictated the studio's refit? 'The band have obviously worked in studios around the world, and wanted their studio to be as good as those they have used' says Exeter. 'They left it to us to provide them with a commercial facility of the calibre they are used to.' Does this mean that clients will benefit from UB40's high requirements? 'Absolutely. All the extras are included, for instance lot of studios charge extra for things like Dolby *SR*, but it's here because the band use it, and it's always available at no extra charge.'

As the studio was only completed in mid December, a commercial session has yet to take place. However, it has been used for the final preparation work on UB40 member Ali Campbell's solo album before it is mixed. There are several other projects lined up for the studio, commercial and otherwise, including a live UB40 video.

Studio 2

Downstairs, however, Studio 2 has been booked solidly since the refit was completed in late July. The local Bhangra scene provides much of the work, a freelance engineer uses it on a regular basis and unsigned Birmingham artist Mark Porter, who moves upstairs in early February to become Studio 1's first commercial client, is comparing a lengthy recording project.

One of the first projects was a competition run in conjunction with enduring Birmingham music free sheet, *Brum Beat*. Four bands were chosen out of around 400 who sent in demo tapes, and each recorded and mixed one song at DEP with Exeter at the controls. The four tracks were issued as a limited edition of 500 CDs, 300 of which were distributed around the industry. 'We hope that this will stoke the fire for them, and that at least one of them will get some recognition or a deal out of it,' says Phipps. 'And each band have got 50 well-presented and recorded CDs for their own use.' Three ▶

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of the bands, Flower House, Big Trouble and We'll Always Have Paris, have since been featured on Radio 1, and fourth, Unit 213, has reported interest from a couple of major labels.

This confirms DEP International's promised support of the local music scene, and it is hoped to lure other competition entrants into the studio at reduced rates. 'We're holding an open day in February for two members of each band to look around the facilities, and they will each be given a voucher for a reduced recording rate of £200 per day including engineer,' says Phipps. As the usual rate is £250 without engineer this represents a good deal for bands unused to recording as well as being a good promotional exercise for the studio.

The refit of Studio 2, the complex's original facility which had been up and running for 12 years, came about almost on a whim and paved the way for the redesign upstairs. 'Ron initially came on board for three months to give the equipment an overhaul,' says Exeter. 'We decided that the downstairs studio deserved some attention so we ripped all the equipment out, gave it a facelift and carried out a complete reinstallation,' Pender points out, 'This not only gave us the opportunity to do a real installation but also introduced us to the ATC *SCM-150A* monitors which are very similar to the large ones in Studio 1.

The refurbishment of Studio 2 has been largely cosmetic, although Nick Whittaker executed a few acoustic tweaks which took care of the high-frequency reflection that was causing harshness. Apart from new monitors all the original equipment, including an Amek *Angela 3624* console with Optifile *Tetra* automation, was reinstalled. an old MCI ½-inch machine which according to Pender had been 'stuck in the corner gathering dust,' was installed, together with some extra outboard equipment. The result is a brighter, more professional looking and sounding studio, as Pender testifies with this story:

'One of our clients was booking some work in and asking when the new studio was going to be ready because he needed to do it in there. But when he saw the changes we'd made to this one he was quite happy to do it here instead because of the way the atmosphere's changed.' This might be a budget basement studio but the lighting and professional finishes have brought it much further up-market.

Studio 2 now has a larger live area than Studio 1, and it is intended that the band will use this as a rehearsal room, their previous space having been usurped by the Studio 1's new control room. Tie lines for audio and video are in place in both studios, which, like the wiring throughout, was carried out by Pender and Exeter working with one or two contract wiremen. Pender, who has a background in design through years of work at companies such a Dolby, Abbey Road and SSL, has even custom-built some pieces of equipment such as level matching interfaces for the DAT machines, saving the studio a substantial amount of money.

This high level of technical expertise among the staff is obviously a confidence-building factor when it comes to launching DEP International as a seriously commercial studio facility. Round the clock maintenance is vital for a busy studio and there are clear advantages in having to hand the team which was instrumental in refitting the entire studio. 'Knowing where every joint is we're well placed to fix, modify and upgrade everything,' says Exeter.

'We're now staffed in a manner which means we can work commercially,' says Phipps. 'We also have 24-hour security so the building is always accessible. There are no limitations, no restrictions.'

Reliable staff and great-sounding, well-equipped studios apart, the *piece de resistance* at DEP International is its lounge, a glass penthouse perched eyrie-style at the top of the building. Shaded with thin black Venetian blinds, its huge windows and sloping glass roof look out over the Birmingham skyline. 'At night, when it's dark in here, you can look out over the lights of the city and you could be anywhere in the world', says Exeter. And that's just what DEP International is hoping its refurbishments will have accomplished. ■

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SSL SL 9000J

Perhaps last year's best kept secret belonged to SSL. Confidentiality is not generally one of the pro-audio industry's better qualities, nevertheless those in the know (and there were quite a few) managed to keep under their hats one of the most significant product launches in Solid State Logic's long and highly successful history.

To introduce one major product at a trade show would be considered a *coup* by most companies, but to release three could be regarded as ambitious in the extreme. However, SSL were in something of a quandary—if they were to launch a new analogue console, opinions would be that they were not supporting digital and, of course, vice versa. The most effective way of putting over the clear message that SSL had feet firmly placed in both camps was to simultaneously introduce a new analogue console, *SL 9000j Series*, and a digital console, *Axiom*.

The third unit, *DiskTrack*, bridged the two consoles being equally viable in both cases, and fitted in perfectly with SSL's philosophy of product integration. From the start SSL have believed in integrating satellite functions and facilities into their consoles—witness dynamics, automation, machine control and so on. It is not so much of a surprise then to find a multitrack, random-access, storage-editing system now becoming part of the 'Total Studio' concept. In this and the next issue of *Studio Sound* we'll be taking a close look at this trio of products, starting with the SL 9000j analogue console.

SL 9000j

A first glance at the *SL 9000j* belies the amount of work that has gone into it and one could be forgiven for thinking this was a spruce-up rather than a major update. However, on closer inspection, new features such as the large, recessed, colour monitor, pen and tablet, redesigned centre section, enlarged channel dynamics section, and new style meters begin to stand out, making it obvious that the console has undergone some very important changes and taken a considerable leap forward.

There are also plenty of changes that are not visible, including enhancements to the circuitry that have greatly improved the sonic performance of the desk, and the adoption of an entirely new automation computer which apart from offering new levels of power and control, also allows the desk to integrate and control SSL's range of digital products including, of course, *DiskTrack*.

The desk retains the in-line approach and offer 48 multitrack buses, four stereo buses, a main stereo bus (optionally LCR), and eight auxiliary buses (six mono sends and a stereo cue). Sizes range from 48 to 128 channels.

I-O module

Sonic improvements to the I-O signal paths have been achieved in all areas with better signal-to-noise performance, reduced distortion and extended bandwidth. The signal path is DC coupled immediately post line-input all the way through to the main output, which in terms of performance improves distortion, and significantly extends bandwidth with 1dB down points now being at 3Hz and 50Hz—the core path through the mic amp is even more impressive being 1dB down at 150kHz.

Working from the top of the channel down—the output section offers four possible outputs paths: main stereo bus, four stereo subgroups, 48 multitrack buses, and a direct group output.

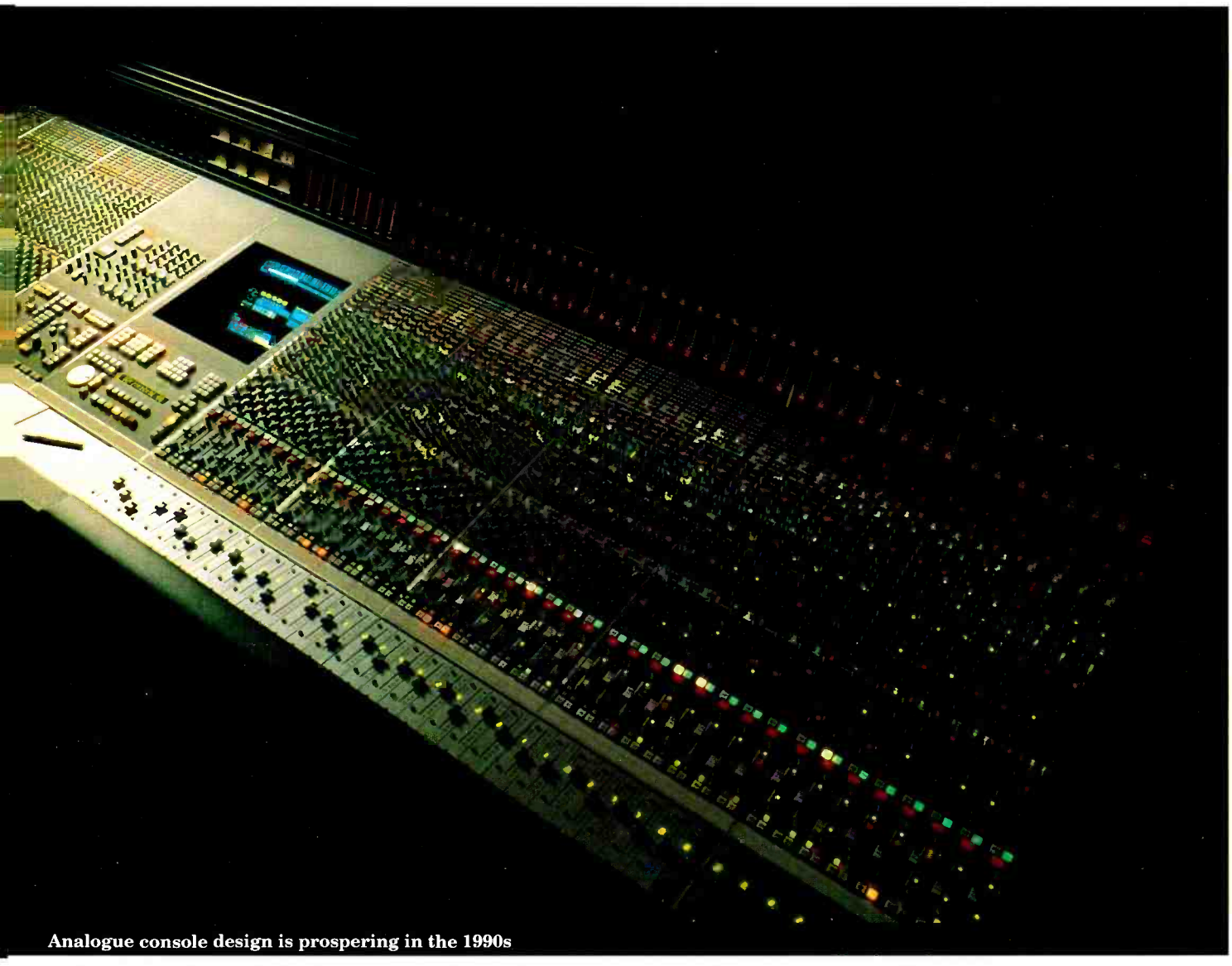
The 48-track routeing is on paired routeing keys—1-25, 2-26, 3-27 and so on with twobank selector switches allowing access to 1–24 and-or 25–48. As before source selection follows console status—so, for example, if the desk is in a mix configuration, the small fader will feed the multitrack matrix and the large fader the mix bus.

Source selection can locally be overridden by using four keys—LARGE FADER, SMALL FADER, EFFECTS ODD, and EFFECTS EVEN (more on the EFFECTS keys later). For an operation like track bouncing, it is now simply a matter of selecting LARGE FADER to access the multitrack; this replaces the original FLOAT switch, but unlike the *4000* console there is no need to copy across pan positions as the feed is taken post monitor pan; also the small fader remains operational rather than being disconnected. The arrangement offers greater flexibility and make things like creating subgroups using the MT buses very simple.

As well as the MT buses there is the main stereo bus which can optionally be LCR) and four stereo subgroup buses (A, B, C, and D). Like the MT buses the stereo subs can input the same four sources again maximising on flexibility.

The MT group trim control and DIRECT OUTPUT switch have been moved from the Group-Tape switching area to the top half of the module, and these also can source the Effects Odd and Even buses.

The surprise launch of two high-end studio consoles and a nonlinear recorder made SSL the talking point of the San Francisco AES Show. Patrick Stapley gets an exclusive hands-on session with the *SL 9000j*



Analogue console design is prospering in the 1990s

The input stage includes the familiar SUB GROUP, FLIP, PHASE and 48V phantom power switches with separate controls for mic and line gain. The mic gain has been changed from a stepped to continuous (+15dB to +75dB) and now has a 20dB pad. Making the control continuous was in response to users wishing to ride the gain directly from the mic amp—with a stepped control this was impossible to do inaudibly. Another addition to the mic amp is a HIGH-Z (high impedance) switch that allows direct patching of synthesisers and so on without the need for a DI box.

The Compressor section remains very similar to the *G-Series*, the major change being that the RATIO control has a pull-up position which changes the compressor from RMS sensing with a soft knee, to peak sensing with a very sharp knee—providing quite brutal processing. The Expander-Gate has a new Hold control allowing the onset of gating to be held-off by a set time. This control also has a pull-up switch which will switch response from gating to expansion. As with the *4000* the dynamics section can be keyed from the monitor path, but it can now, additionally, be keyed from the Insert Return, thus no longer tying-up the monitor path for a side chain.

The 4-band equaliser in its normal form has curves based on the *G-Series* EQ, but by popular demand the x3 RANGE switches that appeared on the two mid bands have been replaced by the previous arrangement of BELL switches on the HF and LF bands. Although based on *G-Series*, the equaliser is not a direct copy, for example gain control gearing has been changed so that initial gain adjustment from the centre point has greater resolution and is a little less abrupt. Also, and this is probably due to a large extent to the increased bandwidth of the console, EQ generally sounds smoother and in the extreme high end is noticeably more effective.

For users who are less keen on *G-Series* curves, the equaliser will locally transform into an *E-Series* EQ by the press of a button. Comparatively this produces rather more gentle slopes on the HF and LF shelving bands, and the mid bands have a constant bandwidth so that Q increases as gain increases—as opposed to bandwidth increasing as gain is decreased. By supplying both types of equaliser SSL have avoided the situation, particularly in America, where desks would be configured with a mixture of *E* and *G-type* modules to suit client's tastes.

The EQ *IN-OUT* switch is now automated, and a with *G-Series* consoles the filters and EQ can be split between channel and monitor paths. The filters may additionally by switched into the dynamics side-chain.

The auxiliary section now has two additional mono sends making a total of six mono and one stereo. As before, sends are switched on and off using the individual push-pull gain control which have become automated switches. Sends can be individually sourced from channel or monitor, although pre-post switching is now a paired function. Apart from increasing the number of sends, the versatility of the auxiliary section has been further improved with the addition of the Effects Send Reassign System (EFX for short).

EFX buttons beside each send allow an Odd and an Even mono send or the stereo cue to be disconnected from their respective buses and used as sources for the 48-track routeing, four stereo subgroup buses and-or direct group output. Each output routeing section has EFX ODD and EFX EVEN buttons that source the reassigned sends, thus increasing the auxiliary capability ▶

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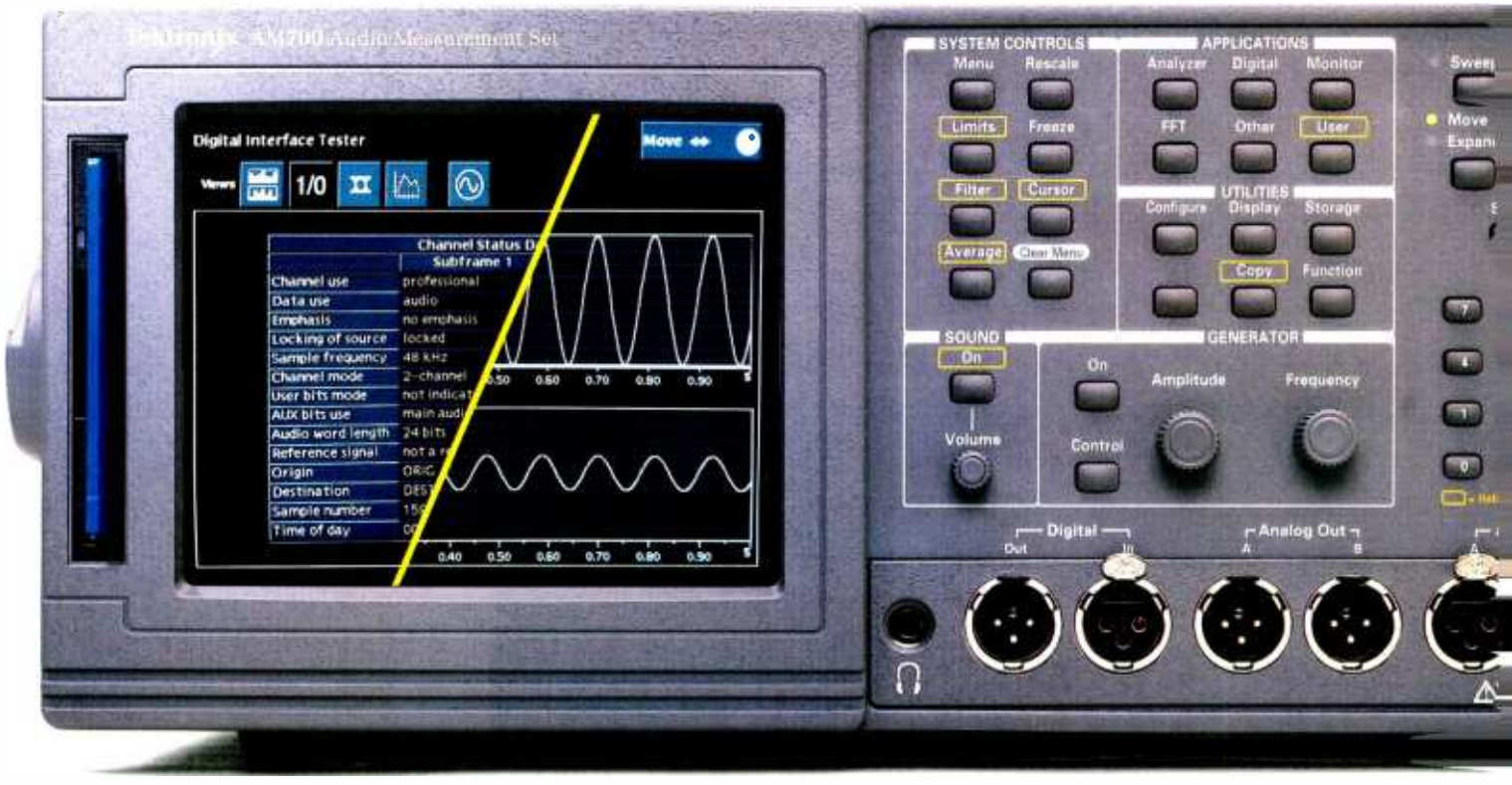

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SL 9000j centre section

during mixing to a very impressive 64 possible sends and this is without making use of the small fader.

The same GROUP-TAPE and RECORD ENABLE switches are fitted as with previous consoles and these function identically. The small fader, however, does see some changes; perhaps the most significant being that both fader and cut are now automated. Audio is only routed via the VCA with the automation switched on, and fader can be individually switched out of the VCA at any time. Unlike the *4000* the small fader pan control is permanently associated with it, and likewise the large fader's pan will remain attached irrespective of switching. Both small and large fader have individual routeing buttons to allow their respective paths to be sent-disconnected to the main stereo mix bus.

The solo system offers four identically selected modes for both small and large fader—stereo AFL, PFL, destructive SIP, and SIF (Solo in Front). SIF mixes together the AFL signal with a dimmed (user set) main mix output—this is useful where an engineer wants to keep an ear on where he is in the track, or to make 'up-front' adjustments while still relating to the mix. All selections can be cleared by central switching—additionally, solos can be globally switched from latching (default) to intercancelling (Alt), momentary (Fleet), and linked so that large and small solo-cut buses operate in unison.

Automation status switching on the console is now found in three places—large fader STATUS switch, small fader STATUS switch, and two buttons (MATCH and PLAY) for the 11 automated switches in the channel. The large *Ultimation* fader is motorised and can connect to eight VCA 'hard' groups in the centre of the console by using the selector switch at the bottom of the fader (a short press increments the number in the fader display window, a longer press decrements it).

Centre section

The console's central control area is divided into two main sections, each eight modules wide. The left-hand section contains all the master audio facilities plus meter contnow slides neatly below the pen and tablet.

The main console output is 4-channel with a 4-channel fader and compressor. A new feature for the main fader is automatic VCA bypass when the fader is at the top of its travel—there is also a ±20dB offset trim and STATUS button associated with the fader. The Quad compressor is the same design as *G-Series* although a new feature is the ability to key it from an external source.

If the console does not have the LCR Film Pan Option fitted, any two of the four stereo subgroups (A,B,C and D) can be configured as the Centre and Surround channels. The 4-channel main monitor output also has a switchable insert (pre the monitor level control) designed for Dolby Surround encoders-decoders.

The four stereo echo-returns remain similar, although they now contain routeing switches to LR, Centre and Surround outputs, as well as the headphone outputs. Auxiliary masters have been simplified to level controls only, with level and balance for the stereo cue.

The design of the studio monitoring section has changed considerably. There are now three separate studio headphone outputs and one stereo studio loudspeaker output, each having independent level control, AFL, cut and three source selectors. Source selection is additive and for the headphones it is taken from a stereo patch signal (normally the stereo cue bus although this may be specified by the customer), and two External Source Selectors banks. For the studio speakers, the primary source is always the main mix output. Likewise the control-room monitor source is selected between the main mix (2-Chan ▶

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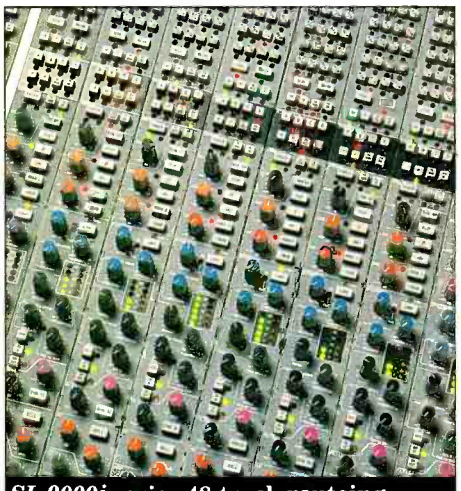

SL 9000j series 48-track routeing selectors and improved dynamics section

or 4-Chan), and the two Ext Source selectors.

The External Source selectors themselves are arranged as before into two columns of buttons, but because they are no longer split into dedicated studio and monitor selectors, they can contain different sources, allowing for great choice and flexibility. Two additional buttons—SUM and LINK—further enhance operation by allowing sources to be selected additively rather than intercancelling, and enable the two banks of buttons to linked producing a 22 into 1 selector.

Channel meters may be switched to read tape return, group output, channel input, and fader VCA levels (both large and small fader). The eight Main meters may be switched between desk output, the two external source selectors, and follow monitor. Peak hold functions are included, and meters can be switched between digital peak and VU.

The meters themselves are a new back-lit LCD design providing a digital peak scale and a VU scale. The digital scale uses a very short 100µs integration time enabling very fast transients to be registered. they can also be recalibrated using a terminal so that 0db equals between 16dBu and 24dBu, but the factory setting will be at 18dBu to match Sony digital meters. The VU scale can also be recalibrated but this will be set at 0VU = +4dBu mechanical meters can optionally be fitted.

Automation

The *J-Series* computer is completely new and is based on the same hardware and operating system that SSL have used on their digital products like *Scenaria* and *Omnimix*. The advantage here is that rather than starting totally afresh, or building on a system that was probably near its limits anyway, the console gains a system that has been developed and proven over the last three years. Also, by integrating the product family onto one operating system, developments for one product can directly enhance another, thus increasing efficiency and ultimately benefiting the customer by speeding-up implementation time of new facilities.

The automation system is now faster, controls more console functions and provides comprehensive session management. All data is stored on hard disk (System Disk) and can be archived to 3½-inch floppy or M-O disc. The system can import *G-Series* and *G+* mixes on either Bernoulli Cartridge or 3½-inch floppy, and all drives apart from the Bernoulli are supplied as standard.

A new user interface has been installed—the pen and tablet—which again originates from the digital product range. There will undoubtedly be some users who will be put off by the idea of this, but as SSL rightly point out, the pen is a much more intuitive interface than a mouse or track ball, simply because most of us have used one since childhood. However, once the system has been set up, the use of the pen becomes quite limited as all commonly used functions have been duplicated on hardware switches. Although a computer keyboard has not been built-in as part of the control panel, it can as previously mentioned be found in a slide-out drawer beneath the tablet.

As far as organising session data, the *J-Series* introduces a new system of Project files. A Default Project will first of all be set up which contains all basic setup information including console-computer setup parameters, tape-machine enables, function-key data and so on. From this setup file subsequent Working Projects can be created which will additionally contain ▶

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Automated small fader can be used for a variety of functions, including automated panning

mixes, automation modes, hard and soft grouping, *Total Recall* data, track lists, event (Cue) lists and so on.

As mentioned above, automation now controls more than large faders and cuts, and will additionally control the small fader, small fader cut, aux send on-off, insert in-out, and EQ in-out. The system has been designed to operate with a maximum of 120 channels, which amounts to a total of 249 faders and around 1,300 switches all controlled to a 1/4-frame accuracy.

Although the automation is new, similarities do exist between the systems with the same kind of statuses being employed, and SSL's *Ultimation* VCA-motor-fader system being used for large faders. The system operates with seven different mix modes—Normal, Rollback, Static, Clip Fill, Clip End, Cycle Fill and Cycle End. All these modes apart from Normal were developed for SSL's digital postproduction products, and Normal remains the closest to the *G-Series* mix system.

Irrespective of the selected mode, there are two fader write statuses –Absolute and Trim—and a Replay status where faders playback previously recorded data. In addition there are a number of update modes including Renull, Snap, Autoglide, Autotakeover, and immediate Pickup. Renull is basically the Revise or Join function for Trim faders that will be familiar to *G-Series* users; Snap causes the large fader to enter the selected write mode once it is touched and jump back to its null point and continue replay when released; Autoglide returns the fader to its null point at a user-specified rate (up to ten seconds); Autotakeover is the same as before allowing easy manual nulling of faders; immediate Pickup is similar to Snap although rather than activating write status on touch it does so as the fader is moved and thus is a useful facility when large fader motors are switched off and of course for small faders. there are also two protection modes for faders —Protected Manual allows new fader movement to be monitored but not recorded, and Protected Replay disconnects new fader moves from both the monitors and the computer.

Automated switches are independently status selected, and can be switched to either Play or Record. MATCH and PLAY buttons are included on each channel (and for the group faders) to allow editing of existing switch data. When MATCH is selected the switches on that channel are primed so that when a switch is pressed it will drop into write matching the previous state. Pressing the switch again will cause new data to be written. Selecting the PLAY button will prime any switches in record to revert to replay at the moment they are next pressed. If the M and P buttons are selected simultaneously, switches will automatically return to their previous state as soon as their current state matches it.

The system uses a new mix pass structure whereby every time a rollback is performed after updating the mix, the computer stores a new Mix Pass. Up to six Mix Passes will be stored in memory allowing six levels of undo. The system will only store a Mix Pass that contains updated information—thus rolling back and listening to the mix will not create anew pass. However, because the saving of updated passes is a continuous process, earlier passes will eventually be deleted to make way for new ones, so it is advisable to periodically save mixes to the hard disk.

Up to 40 snapshots can be stored and recalled with automation enabled or disabled. The system also features a Pre-Enable snapshot that is automatically stored as the mix system is enabled—this acts as a safety feature storing a desk-wide pre-automation snapshot in the event that by enabling the automation an old mix inadvertently reset destroying the current balance.

An overview display is included that graphically represents channels and shows audio clips, automation data and location points in relation to time. The display scrolls giving a unique perspective on how the component parts of the system are ▶

functioning and can be zoomed in to view fine detail. Additionally, mix data can be edited off-line directly from the Overview display, and it can also be used to locate cue points.

The automation allows an unlimited number of switch groups to be set up as well as 32 fader-cut software groups for both large and small faders. Cuts can appear in both switch groups and software groups but faders can only appear in one group. As with the latest *G-Series* software, individual slaves in a software group can be configured in six different ways—Fader and Cut, Fader Only, Cut Only, Cut Inverted (inverts slave cut with respect to master cut), and Status Only where the slave follows the fader status of the master. There is now an additional mode provisionally called 'One of N Uncut' which has been designed for vocal composites. In this mode all channels in a group apart from one remain cut, so that each time a channel is uncut the previous one will mute.

The *Total Recall* system operates in the same way providing manual rematching of switches and rotary controls to a 0.25dB tolerance. However, the system has now been extended to function with centre-section controls, which has been a much requested facility. *Total Recall* is now a standard feature—it was a *G-Series* option.

The Machines menu allows connection and control of any SSL digital product via Ethernet. Apart from products such as *DiskTrack*, *VisionTrack* and *Omnimix*, the system can also interact with SSL programmable patchbays, a remote keypads, switchers, M-O disc drives and so on. A Network page shows all the devices currently connected to the system.

Direct serial control is provided for up to four machines; for machines without serial ports, the desk will interface to TimeLine *Lynx* modules or Audio Kinetics *ES 1.11/12* modules. Full track arming, autolocate, jog, and varispeed functions can all be performed from the desk. The console outputs both linear and MIDI time-code allowing it to act as a virtual master during random-access operation.

The *SL 9000j* also includes an expanded function keys system providing up to 66 user-definable macros. Unlike *G-Series* where function keys are saved as part of the keyboard information, they are now saved as part of the Project information enabling them to be installed on another *SL 9000j* console along with the rest of the Project data.

DiskTrack

DiskTrack is an option for the *SL 9000j* console and as already mentioned can be shared with other *SL 9000j* or *Axiom* consoles. All control is executed directly from the desk, and there is no requirement for any additional user interface.

The system is made up of an array of hard disk (6–16 disks), a main processor which communications via the Ethernet network, and a number of highway node-cards which provide a high-speed serial audio link to remote I-O units providing 48 analogue channels or 95 digital channels per link. Digital I-Os are via AES-EBU and MADI.

Extended auxiliary section with automated in-out and soft grouping

The system offers up to 95 tracks of concurrent record and replay with a maximum storage capacity of 56 track-hours at 16-bit, but can also operate at 20-bit depending on user preference. Recording is nondestructive and over-recorded segments will stack-up rather than erasing original material. This also means that drop-in/drop-out points can be adjusted after recording.

A complete range of nonlinear editing functions are provided which are similar to the facilities found in *Scenaria* and *Omnimix*. The system also features background loading-restore projects to be turned around in the quickest time possible.

Conclusion

Without a doubt the *SL 9000j Series* represents a major achievement for SSL. Not only does the console offer a whole host of new facilities that enhance operational flexibility, and help simplify operation; but it also brings new levels of integrated control that opens an enormous wealth of possibilities.

Attention to the sonic performance has resulted in significant improvements to bandwidth, noise and distortion, making the desk's technical spec truly impressive and top-ranking.

The *DiskTrack* option brings the exiting addition of nonlinear recording-editing completely integrated within the console, which has the potential of radically shaking-up traditional methods of working.

The *SL 9000j Series* presents studios with the best of analogue, full integration of facilities, proven comprehensive automation and system management, renowned ergonomics, plus new levels of audio manipulation. Who said the analogue super-console's days were numbered? ■

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Series 900

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- 6 Echo returns, with EQ and level fader
- Aux's can be routed to groups, for extra 24 Aux sends
- Auto mute groups on channels
- Penny and Giles faders throughout (optional on 900)
- Mosses & Mitchell patchbay as standard (optional on 900)
- Factory or Retrofit of any Automation, to customers choice

Other Series 980 Owners

Charlie Daniels, Nashville - USA
DK studio - Belgium
Billie Ray Herns - USA
Studio 45, Connecticut - USA
Audio Playground, Florida - USA
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Cole Street, LA - USA
Dino MKII, LA - USA
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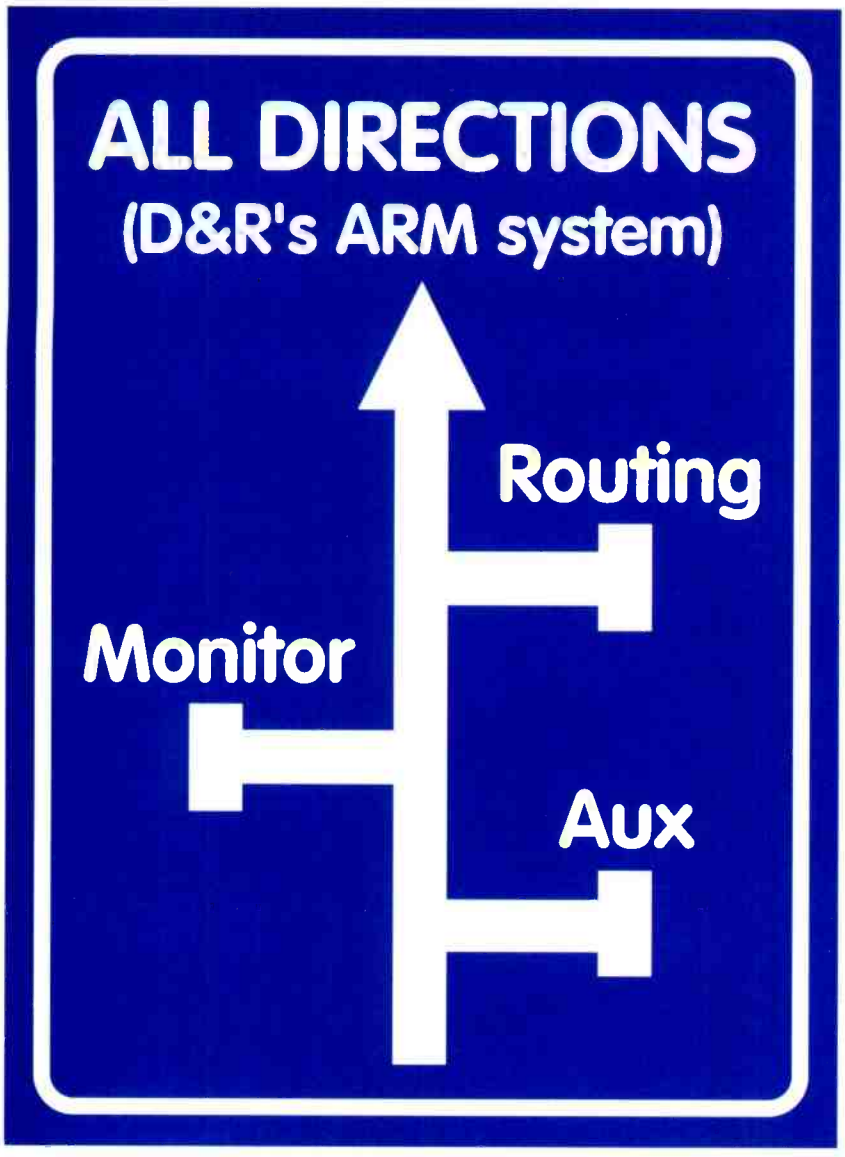
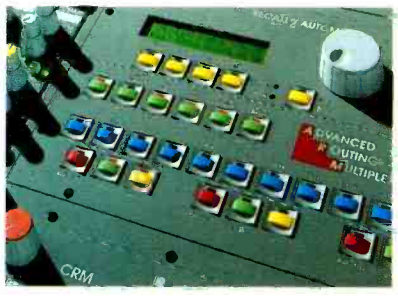
ALL DIRECTIONS
(D&R's ARM system)



NO MORE DETOURS

No more back roads. No more one-way streets. No more silly antics to make your console jump through audio hoops. Allow us to present Merlin-a melting of incredible sonics and features that delivers a new and very powerful recording console.

Merlin is D&R's answer to the high-tech, studio world of the 90's. A recording console with superb electronics. A true dual input channel and highly automated signal paths. Merlin is about to change the way you think about recording consoles forever.



Much of Merlin's power lies in its ARM-the unique Advanced Routing Multiplexer. Easy to understand yet incredibly powerful, ARM revolutionizes the way that consoles are designed. With the ARM system, your signal routing no longer encounters any detours or one-way streets.

ARM allows access to 36 Aux sends during mixing and access to the 24 group busses from both signal paths, simultaneously or individually during tracking. It also allows access to signal routing possibilities that even we haven't envisioned yet. Add to this Merlin's automation system and you will begin to understand how Merlin already has begun to weave a spell over you.

The Merlin is much more than an in-line or dual EQ console. It is the first serious dual input console offered at an affordable price. You can be sure that a studio equipped with a 32 frame Merlin has the equivalent of 64 truly assignable and highly automated input modules.

TWO VCA or optional motorized faders, TWO automated mutes, TWO automated insert in/out switches, TWO automated Aux send mutes, and automated access to 12 additional functions per input module.

I'm sure by now you are thinking that it must take a rocket scientist to operate all this, but through the magic of Merlin, anyone with just a bit of studio engineering experience can be up and running a session in a matter of moments.

So when it's time to create some magic of your own, call today for the complete story of Merlin. It'll have you spellbound.

D&R

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Merlin

RAISING THE CURTAIN

Philip Newell pays a visit behind the former Iron Curtain to report on the new state of Russia, and its place in modern professional audio

Like many other people, when I hear talk of the Third World, I think of a less privileged and possibly overpopulated section of the world. Rarely do I think of where the Second World stood in relation to all of this—hidden away as it was behind the Iron Curtain. At the hub of this Second World, however, is Russia, a country which boasts some of the greatest classical composers in history. The Soviet Union, of which Russia has been a key part, has launched more space rockets than any other nation, and Russian scientists have been recognised at the highest level of physics, chemistry and biology since the first awakenings of the sciences.

Recently, however, since the Russian borders have opened to a free market economy, there have been many western press reports of drunkenness, violence, street and organised crime. So when I was first invited to visit the Audio Video 94 exhibition at the Lenexpo Centre in St Petersburg (see Conference Report, *Studio Sound* January 1995), I had conflicting expectations. Rumours were earlier circulating around the APRS Exhibition in London that it would be dangerous to attend—I consulted the British Consulate in St Petersburg—their response was that anybody fool enough to visit such uncivilised areas of the planet as Chicago, Washington DC, New York, Los Angeles, Detroit, Miami or many other cities in the Land of the Free should have a comparatively relaxing time in St Petersburg or Moscow.

After spending a week in the city and making extensive use of public transport, I had seen no fights and encountered no crime whatsoever; nor did I hear of any involving any of the people who I met. Even the organised crime which one reads about does not affect foreigners as long as they follow their Consulate's directions. Only one of the (few) drunks I saw was causing any nuisance, and he was being escorted by a large policeman, who appeared very controlled and educated. In fact, I spent the week searching hard for the Russia of crime and depravity of which I had read in the western press, surprisingly enough, even in *The Times* (London). Perhaps there is another Russia somewhere.

One thing which spurred my interest in Russia was my first encounter with some power amplifiers—their build was solid and competent and the sound quality impressed everybody who heard them. So much so, that I have now adopted these amplifiers as my first choice for my monitor installations. In August 1994, I invited Mikhail Borisovitch Matusov to visit London and Lisbon so that I could find out more about this equipment first hand. We had been communicating for some time on technical matters, and I wanted him to cast an eye over some of my recent work and to have an opportunity to meet him face to face. He brought with him a selection of microphones from four different companies which were comparable with the bestwestern mics available. Some friendly studio owners who tried them were almost reduced to tears when they discovered a 4:1 price differential between the Russian and western mics.

Technology

Once in Russia, I was treated to a new surprises on a daily basis. I saw compression drivers whose ultra- lightweight diaphragms could withstand a frenzied attack with the sharp edge of a screwdriver. They can withstand a crossover failure and full-range musical signal up to their rated RMS power handling, yet respond wonderfully from 800Hz to over 23kHz. (I brought some back for my own tests, and they sound very smooth, clean and open.)

A Russian called Alexander Vagin showed photographs of some huge low-frequency drivers. He also claimed that he had developed new classes of loudspeaker radiators, capable of 109dB–122dB/W/m sensitivity, yet using traditional materials. He was interested in a joint venture for development of his ideas, which included high sensitivity, low distortion, wide, smooth frequency range transducers, cardboard loudspeaker cabinets (said to be surprisingly effective) and more. The list of his claims is too long to publish here. Can he back all of them up? I do not know, but for 20 years he was a scientist at the very secretive Soviet State Laboratory of Powerful Electrical and Acoustic Radiators.

The architecture of St Petersburg is breathtaking. The buildings never go up above about the fifth floor, except for church spires and the like, but though the buildings are massive and imposing, they never feel overpowering. Hidden away in one of the many areas of park land, situated on Stone Island is the AS Popov Scientific Research and Development Institute for Radio Broadcast Reception and acoustics. The Institute works in collaboration with the Russian ▶

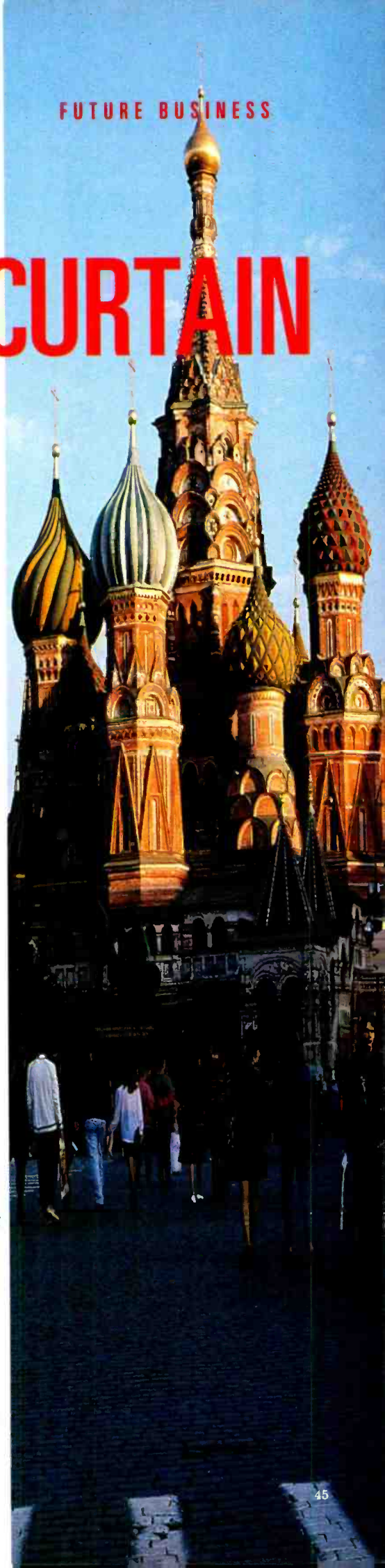

PHOTO: ZENON SCHOEPE

The Lutherian church housing CMS Studios still holds services on Sundays

Academy of Sciences, and was situated on the island, which is within the city of St Petersburg, because its freedom from ground borne vibrations from heavy traffic allowed extremely low background noise levels for measurement in its three anechoic chambers, the largest of which are among the biggest in Europe.

I was shown round the Institute by its deputy director and Chief Science Manager, Dr Irina Aldoshina; the list of papers on electroacoustics which she has published is quite awesome. The Institute is for the development of all aspects of electroacoustic equipment, from domestic television loudspeakers to mixing consoles and microphones. In their listening room I was shown high-frequency dome loudspeakers which have cellulose diaphragms. Now, cellulose has long been known to be strong, light and glossy but as with so many natural materials, consistency has always been a problem. Not so this cellulose, it was said to be highly consistent from batch to batch. Was this a synthetic cellulose? 'No, bio-cellulose', I was told. 'But how do you achieve consistency?' I asked. After being told that they use genetically engineered bacteria, it was a sort of checkmate.

The Institute is also looking for international cooperation. It can offer a research facility with a staff of over 800, and has a total area of about 12,000m², some of which it would rent to collaborating companies for either testing or production facilities. It produced the sound systems for the 1980 Olympic Games in Moscow, and already cooperates with Sony and several European research centres on the development of AM stereo and high-quality multiprogramme digital audio broadcasting systems.

As with so many aspects of Russia, an air of competence generally exists, but without any of the arrogance and hype which often prevails elsewhere. The Popov Institute can even undertake manufacturing, and while they are unlikely to ask for peanuts, the price of R&D and manufacture is likely to be very competitive. The totally distorted value of the rouble makes the proposition of Western collaboration highly cost-effective.

The openness with which the Russian people speak, and their willingness to share ideas almost seems naive by the standards of the capitalist world, but at the same time, the Russians are no fools. I visited the factory of a company producing power amplifiers, and though conditions were a little bleak, I noticed that all of the production line ladies had fresh air fed to them, and exhausters to remove the fumes directly from the soldering positions. There were no 'sweat shops' and the communist system had obviously taught people to consider the welfare of the workers. In fact, overall, I was quite impressed by just how many positive legacies the old USSR had left behind.

An air of competence generally exists, but without any of the arrogance and hype which often prevails elsewhere

This openness extends to the chief designers and engineers, who spoke freely, but entirely without exaggeration. I got the impression that not one word was wasted, but the thoroughness of their thinking was as exemplary as their testing was rigorous. Again, many of the staff had spent years in the Soviet military services, and their training has been both highly disciplined and 'no expense spared'. There is no sloppy or fuzzy thinking, yet the general attitude is one of humility and self-criticism. Complacency seems to be an unknown concept. ▶

'The totally distorted value of the rouble makes the proposition of Western collaboration highly cost-effective'

GALAXY STUDIO, BELGIUM

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If you work in audio for motion pictures, post or multi-track recording (or perhaps yours is a facility that services all three industries) then there are 3 extremely good reasons why you should visit stand number 4T30 at this year's AES.

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Traditional acoustic spaces provide new opportunities for Russian recording studios

'These people are looking for co-operation to rebuild their country, not charity'

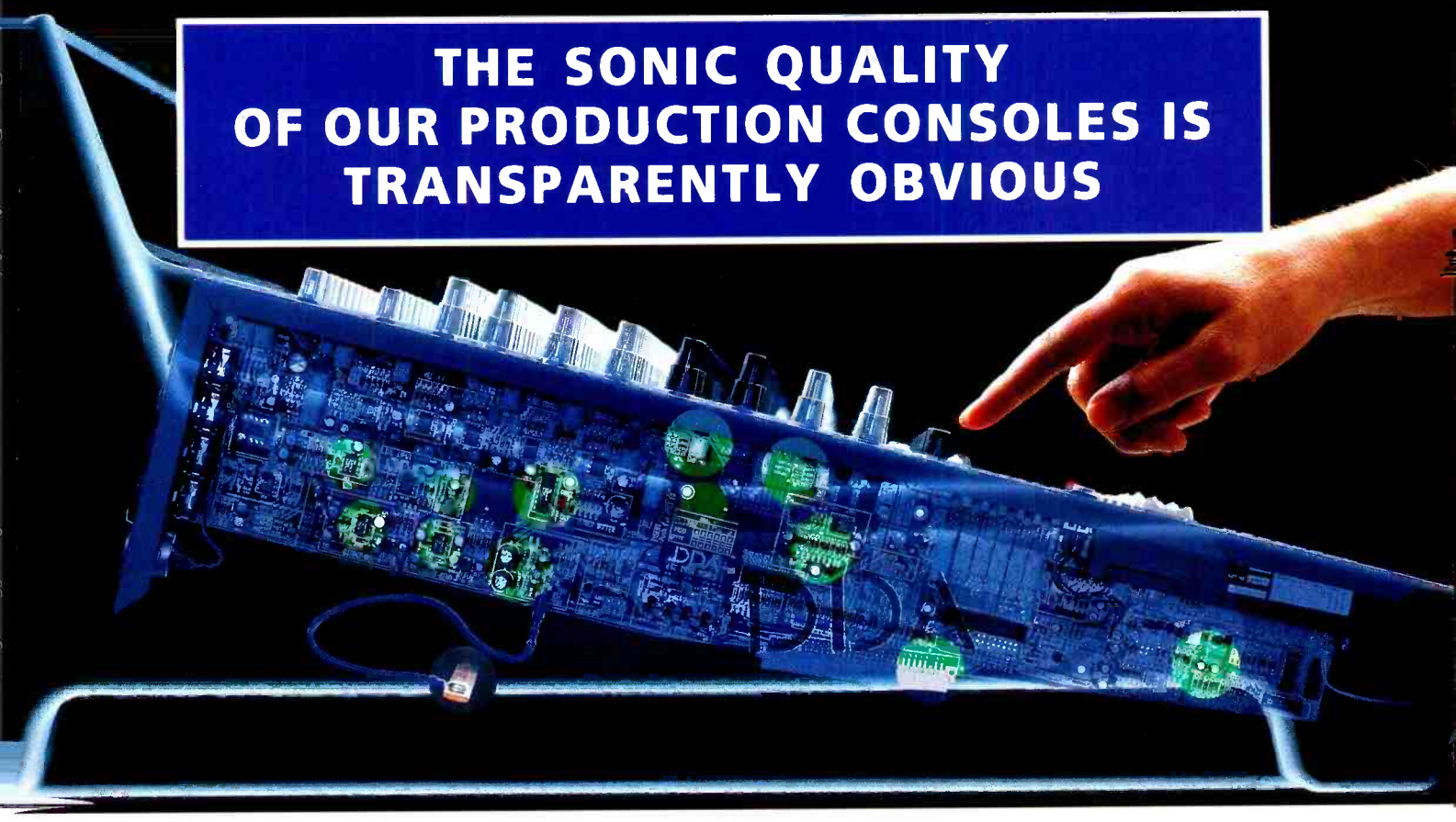
Survival

A good way to invite malnutrition in Russia is to eat in its restaurants, where many items are probably unavailable and portions of whatever is available are minuscule. If all the home comforts and excesses are a fundamental part of your life, then Russia is not the place for you. My Russian hosts brought home-made food to me each day that I was at the exhibition, and although many things are in short supply, I never actually went hungry.

I had asked before arriving that if I could stay in a Russian hotel, and not one of the faceless international chain hotels. I stayed in the Oktoberskaya across from the station for the trains from Moscow. It is a huge rambling affair, where one has to pass desk after desk before arriving at one's room. We had lots of fun in totally alien surroundings, but though I had never seen anything like this 1,000-room monster in the west, the lacklustre facilities did provide a 24-hour snack bar, several bars, an empty fridge in each room for any private 'stocks' and satellite TV. Though rather brown, the hot water was plentiful, the rooms were adequately warm, and for £20 (UK) a night, one could not complain. However, I would advise any persons considering visiting Russia to do so with an entirely open mind. I was advised not to use the cafes at the Lenexpo centre, as prices were 'severe'. As it was, I allowed myself the convenience of their use, and happily was 'ripped-off' all day with excellent cups of tea at 15p each (three for a US dollar) and large cups of fresh orange juice at four for £1. To the locals, however, these were the extortionate prices.

There are Russian PhDs earning $100 (US) per month, so life is tough, but these people are looking for co-operation to rebuild their country, not charity. Capitalist vultures looking for quick profit and exploitation should beware; these are strong and intelligent people. In the early 1940s, the Third Reich besieged St Petersburg and cut it off completely. For over 900 days, the people of the city resisted, suffering over 2.5 million dead, either from shelling, bombing, starvation, or from the -40°C winter cold. Not only did they not surrender, but when the siege faltered, they broke out and were amonge the numbers who subsequently took Berlin itself. As then, the Russian are not selling out, but they do have a truly enormous amount to offer any collaborators who will help to develop what they have. ▶

THE SONIC QUALITY OF OUR PRODUCTION CONSOLES IS TRANSPARENTLY OBVIOUS



DDA's approach to console design is simple.

We believe that where audio electronics are concerned, less is definitely more. The less we put in the way of your signal, the more your mix will shine through.


FMR

But making this concept practical is quite a design feat.

With so much demanded of a production console, most tend to fall short of one major quality.

Transparency.

At DDA we put a lot of thought into making our electronics sound less. (Even while our facilities give you far more.)

The entire audio path gives you the freedom of an elegant gain structure with over 22dB of headroom throughout and a low noise floor.

Our minimal signal path topology, where unused circuit blocks are completely by-passed, and state-of-the-art *Analog Devices chips are two more keys to accurate audio.

High quality controls and switches, distributed decoupling and gold plated connectors subtly improve signal integrity.

Meanwhile, everything *around* the signal path is designed to protect that quality throughout the desk.

A full-length copper earth bar, balanced line level interconnects and ground-planed PCBs improve noise, RF immunity and crosstalk.

So while we broaden your creative horizons, you'll hear virtually nothing from our electronics.

You'll find all these attributes in the FMR production console's Forum Standard Input and Master modules and its new Bus/Tape Monitor module.

They also draw on the legendary AMR24's overall >100dB dynamic range and incredibly low noise floor. Along with more intelligent features like 4-band EQ and complete aux, solo and mute facilities.

DDA's range of production consoles include the Interface, QMR, DMR12, Profile and DCM232 desks.

As for their sonic quality, we know that as soon as you work with a DDA you'll agree with us: It's transparently obvious.

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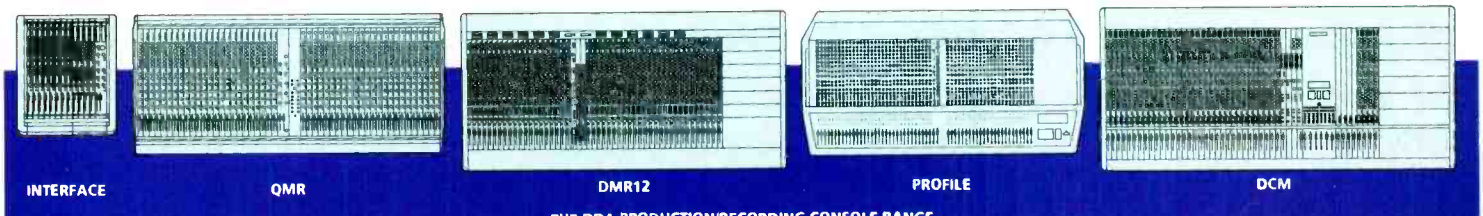

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'Some of the gear may have been old, but they had the staff who knew how to keep it all going'

Studios

I visited a number of recording studios, most of which were reminiscent of late-1960s to early-1970s studios in the UK—in terms of their rooms at least. There were many Hungarian tape recorders and the odd Studer *A820*. Amek desks seemed quite popular also. What was also noticeable was that tucked away in corners of most rooms were impressive sets of Soviet or East German test equipment; much more comprehensive than that which would be found in all but the biggest western European facilities. Some of the gear may have been old, but they had the staff who knew how to keep it all going, and also how to get the best out of it.

The people working in Russian control rooms are producing music that is well written, well arranged well performed, and recorded by staff who are musically and technically competent, and possess good ears for musical balance. There is no evidence of the western European failing of reaching for equalisers before hearing the music. The approach is more akin to what one still finds prevalent in the US, where good rooms, good choice of microphones, and a good pair of ears are the most important tools of the trade.

In western Europe, I have encountered engineers with books of equalisation settings for different instruments—sometimes cribbed off somebody who didn't know the full story in the first place.

Some of the recording spaces, such as at Melodija or CMS in St Petersburg are breathtaking. The former is a Lutheran church , still operating as such on Sundays, and the latter is in what appears to have been a theatre, and later an officers meeting place. The church is resplendent in newly restored gold-leaf work with a beautifully typical church-like reverb which can be augmented by the old EMT *251* digital reverb-room heaters, like those which I used to use with Mike Oldfield to such good effect in the early 1980s. The huge room at CMS had an acoustic which is full and smooth, yet which dies fairly rapidly. It adds an extraordinary fullness to the sound, yet without in any way obstructing its clarity. One would be very hard pushed to match the sound quality of these rooms with any artificial devices.

I met some people with an unfulfilled project —an 80% built sound recording and video postproduction studio, sited in a perfectly good building, and with a roster of around 150 musicians. but whose Russian financial backers have pulled out—who are looking for western partners to help them complete the work and share in the development.

One of the surprises of the visit was that I encountered Sergio Castro in a back room at House of Radio. There before him was a digital audio workstation more advanced than many western equivalents, yet costing about 30% of the price. The Russians seem to have it all except money.

The first signs of western invasion are Coca Cola advertisements, and the disgusting opportunism of the big tobacco companies who are decorating the city with advertisements now outlawed in the

west. On television there is evidence of overdubbed Brazilian, Venezuelan and Mexican soap operas, together with American-style game shows, such as *Wheel of Fortune*. Heaven forbid that McDonalds are ever allowed to contribute to the nation's architecture.

Television

The Russian produced programmes are beautifully made. The standard of music and dance performances is exemplary—never did I see an out of focus zoom, a poorly chosen camera angle or shoddy production. As with many other aspects of Russian media, there is an evident lack of the necessary hardware, but training and general ability goes a long way towards making up the shortfall in this and other areas. When the Russians eventually obtain the level of equipment that their abilities warrant, we can begin to expect to see work of exceptional creativity and realisation making its way onto the international circuit.

If anybody is contemplating a visit to Russian, I can thoroughly recommend it. Certainly the people of St Petersburg will be found to be warm and receptive. Admittedly, there are petty bureaucractic inconveniences such as having to pay to get your passport back from the hotel reception on your first visit, but this is a state surcharge. However, once the demand for 2000 rubles is converted into your native currency, the extortion no longer seems important. The courtesy and apologetic nature of the lady who confronted me with these demands, also made them more palatable

Customs and immigration persons are thorough, but not unreasonably rigid. Bags are X-rayed by both security and customs, yet surprisingly, I saw people taking in rifles and ammunition. I was told before I left that Russian 'Filmsafe' X-ray equipment was not as safe as the name implies, but when I requested that they search my camera bag by hand, I met with no objections. All in all, their front line is quite impressive. Believe me, I would prefer to have these people as friends rather than enemies. Above all, don't even think of adopting any sort of patronising stance, they don't need it and they won't appreciate it. They are looking for cooperation and partnership, and they have a lot to offer. ■

Awaiting discovery by the West – the *MKE-113* condenser mic

THE AMAZING TECHNICOLOR ROADSHOW

It was in 1968 at the Colet Court School, Hammersmith, that Andrew Lloyd Webber premiered his first musical, *Joseph and the Amazing Technicolour Dreamcoat*. Twenty-seven years on—and with four Tony Awards, four Drama Desk Awards, three Grammys, five Laurence Olivier Awards, a Star on the Hollywood Walk of Fame, and a knighthood to his name—Lloyd Webber has resurrected the show that launched his fairy tale career and transformed it into a lavish 1990s production.

The new *Joseph* has opened in the West End, Broadway, Canada and Australia. But it has also embarked on a UK tour which in true LLoyd Webber style makes it the largest touring musical ever to be staged in Europe. No compromises have been made in comparison to the original London production, and as a result a fleet of 15 40-foot trucks is required to transport set and equipment—the audio side alone consisting of 20 tons of gear valued at over £0.5m. In fact, the scale of the production is more akin to a major rock tour than repertory.

So far, *Joseph* has played Blackpool, Bristol, Birmingham and Dublin staying between eight and 12 weeks at each. The latest stop in the sell-out tour is Manchester, and it was here at the city's Palace Theatre that I talked to the team of people responsible for the sound to get an insight into the logistics of touring a production of this size.

'It's pretty hectic really,' understates Production Sound Engineer Mike Walker. 'The show closes on a Saturday night at one venue and opens at the next on a Thursday evening. That gives us just three days to install the system, so that on Thursday morning we get a band call, and a dress rehearsal in the afternoon. Basically we have a team of 16 people and 72 hours in which to get everything full up and running including all the communications side of things.'

Compared to a rock 'n' roll tour, or one of the American bus and truck theatre tours, the time scale might at first appear rather generous. However, unlike US venues, the traditional UK theatre has not been designed for fast turnarounds, nor does it easily accommodate the kind of sound system being used for *Joseph*.

'We treat every venue as though it were a permanent installation and try and keep equipment as low profile and as unobtrusive as possible,' explains Walker. 'This involves running all cables out of sight and fitting speakers to wall and ceiling mounts. There are obviously a few compromises—for example some of the cable runs around the proscenium probably wouldn't be allowed in the West End—but generally speaking, everything that happens in the auditorium, such as the desk area, delay and surround speaker, is kept as discreet as possible.'

Initially the show was rigged by splitting the team into two shifts—one working from 10am to 11pm, the other from 10pm to 10am. According to Walker, this worked perfectly well but did cause a few 'practical and political problems.' At Manchester, the decision was taken to use one large team working three long days—although this arrangement was generally preferred by all concerned, organising the bigger group called for much closer supervision by Walker and assistant Brian Beasley.

'Coordinating that number of people is a major task and involves continual problem solving and a lot of running around.' comments Walker. 'However we've been very lucky in terms of continuity, and this time well over half the people involved had worked on at least three previous moves. It means that you don't have to explain everything from scratch, although of course there are things like cable routes and exact positioning of equipment that people won't know about. What we have found, though, is that as we do more moves the rigging has become more efficient, and at Manchester the job was done with fewer people and in record time.'

Equipment requirements vary slightly from venue to venue, and about six weeks before a scheduled move the next theatre is thoroughly studied by the production team. This results in a detailed sound system plan showing equipment positions and cable routes; any additional equipment is carefully itemised allowing plenty of time for it to be added to the standard rig. Also at this point clearance is requested for any damage that prerigging might cause—although this is usually minimal and generally only involves a few holes to be drilled for fixings and cable access.

The setup

Equipment for the tour, including a 74-input *J Type* Cadac console (the largest to tour in the UK), has been supplied by London-based Theatre Projects (see Equipment List). The man responsible for specifying the gear is sound designer Martin Levan who since 1982 has worked on virtually all Andrew Lloyd Webber's musicals worldwide.

Levan set up the first show at Blackpool and then handed over subsequent installations to associate Sound Designers—at Manchester the job fell to Richard Ryan, a previous

PHOTOS: MICHAEL LE POER TRENCH

'It's pretty hectic really,' understates Production Sound Engineer Mike Walker. 'The show closes on a Saturday night at one venue and opens at the next on Thursday evening. That gives us just three days to install the system...'

employee of Levan well practised in his working methods.

'When it comes to setting up a sound system, Martin is not a great believer in analysers and flat response curves,' Ryan reveals. 'He's also not very keen on deadening the acoustic with loads of drapes; his philosophy tends to be 'work with rather than against the acoustic you're presented with, and use your ears rather than your eyes to get results.'

Accordingly, there are no pink noise generators or spectrum analysers to be found anywhere in the theatre. Instead the entire system is setup by ear, with the sound designer walking around the auditorium radioing instructions for EQ, level and delay to operator at the desk and amplifier areas.

'This is where theatre sound differs from a rock concert.' remarks Ryan, 'Rather than placing the audience in front of what effectively is a giant hi-fi system, we put speakers all over the auditorium to try and make the sound as uniform as possible. That's why it's really important to do a thorough reccie (study) to make certain there's sufficient equipment to provide total coverage.'

The proscenium speaker arrangement remains standard from venue to venue, with left and right towers and an expandable truss. Ryan explains how this has been configured.

'Each tower has identical speakers —two Tannoys, two Meyer *UPA*s, two Bose *302*s and two Meyer *USW*s. The stalls and circle are covered by the *UPA*s and Tannoys, and we then takeover using a *UPA*/Tannoy combination from the truss for the balcony. There is also an inner set of UPAs and Tannoys that point down to just beneath the front of the circle and these are used for imaging purposes to bring the sound into the centre. It has the effect of focussing things to the stage, and together with the front fill and delayed JBL *Control 1* auditorium speakers, helps give the impression that a singer is actually singing from the stage rather than appearing a a disembodied sound from left and right systems.'

Stage monitors are integral to the set design with front fills and foldback fitting into specific positions. The show also features a surround sound section as a grand finale, and for this eight Bose *802*s are placed at the rear of each theatre level, being bolstered with Bose *302* subwoofers and Servodrive Contrabass speakers. The total number of speakers at Manchester, excluding fold back, came to a massive 118.

Obviously with a system of this size, substantial amplification is required and at the back of the musicians' pit is a 25-foot row of flight cases containing over 40 amplifiers. ▶

Patrick Stapley joins Andrew Lloyd-Webber's *Joseph* on the road to report on one of the most sophisticated mobile musicals to date

PHOTO: MICHAEL LE POER TRENCH

'The amp area goes together in four hours flat,' says Mike Walker, 'We've packaged controllers and together in the same racks so that all linking and sensing is integral. this means that all you have to do is connect audio inputs and speaker outputs, everything else is already in place. All crossovers and phasing remains the same from venue to venue, the only thing that changes is relative levels and we use PIP ATN cards, which slot into the back of the Amcrons, for adjustment because the knobs on the front are not accurate enough.'

An area of theatre production that has escalated over recent year is communications, with video comms in particular playing an increasingly important roll. According to Mike Walker, the use of video monitors on this production has help cut down of fold back requirement.

'By including a good video comms system, it means that wherever people are positioned on stage they can easily receive a visual cue from the Musical Director rather than having to reply on foldback. It's something that has become expected and it gives directors much greater freedom. The other advantage is that performers don't have to appear as if they're looking at the Musical Director all the time—although, of course, that's exactly what they are doing. In this production we have a large children's choir and the kids have been trained to take direction from the monitors rather than looking down into the pit. It works really well and gives the impression that they're singing out into the auditorium whereas really they're focussing on a couple of video monitors on the edge of the circle. We even place video monitors off stage because quite often people will be singing while they're changing.'

MAIN PERSONNEL

- **Production Sound Engineer:** Mike Walker
- **Associate Sound Designer:** Richard Ryan
- **Assistant Production Engineer:** Brian Beasley
- **Sound Operator:** Steve Brierley
- **Assistant Sound Operator:** Janet Moorhouse
- **Sound Operator No. 3:** Sarah Sendall
- **Theatre Projects Production Manager:** Dave Perry

The mics

As well as speaker positions, microphones have also been built into the set—in particular PZMs which have the advantage of being easy to fit as well as easy to disguise. However with a fast and furious show like *Joseph*, microphones are inevitably at risk, and since the start of the tour six Crown PZMs have been written-off and many more have need major repairs. This record, though, looks quite good when compared to the lavalier mics worn by the performers.

'We probably lose two or three lavalier mics a week through "sweating-out",' says Assistant Production Engineer Brian Beasley. 'At £200 a time it's a real problem, but it's accepted as part of the expense of putting on a production like this where there's a lot of action and people get very hot. We try various tricks like putting tape around the mic or painting it with clear nail varnish to seal it, but although this helps it's not a long term answer. The situation used to be better when you could get lavaliers with replaceable capsules, but there doesn't appear to be anyone making these any more. I think it's a shame that microphone manufacturers don't pay more attention to the requirements of theatre sound, after all if you look at any major musical anywhere in the world they'd be using a similar system and will be having identical problems.'

The lavaliers are normally attached to the performer's hair with elastic and looped so they point down towards the forehead; the radio packs are then worn around the waist, although ▶

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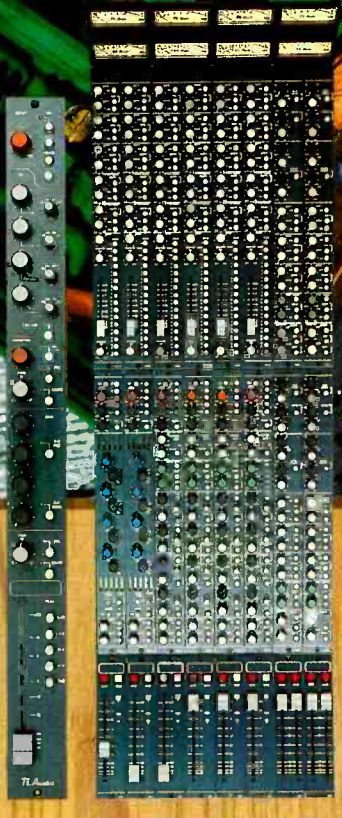

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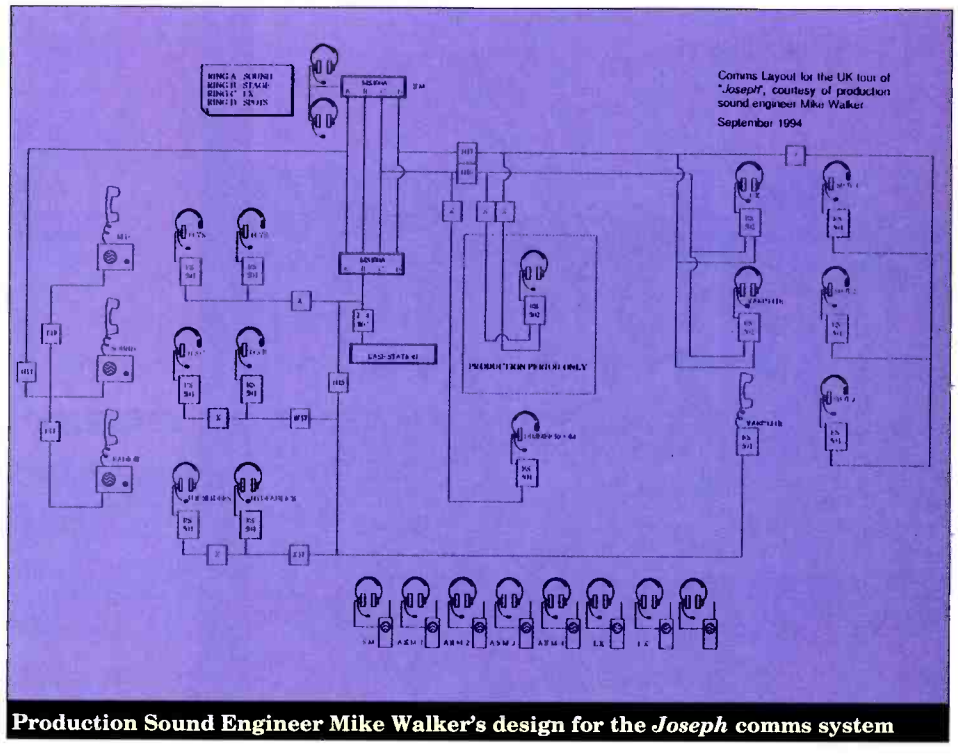
PHOTO: PATRICK STAPLEY

Sound Operator Steve Brierley (left) and Associate Sound Designer Richard Ryan

there are exceptions like the protagonist who —because he is bare-chested for most of the show—has a complete radio-mic system built into his wig.

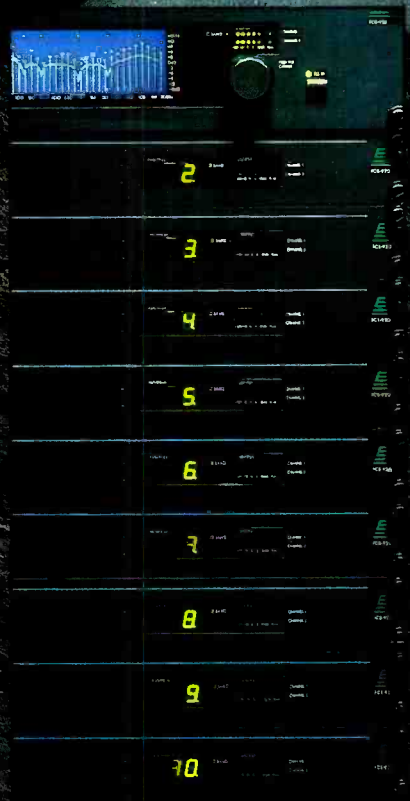
During the performance, a close check is kept on radio mics by the Assistant Sound Operator, Janet Moorhouse, and Number 3, Sarah Sendell. Their job is keep an eye and ear on the system making sure that signals are strong and output quality high. Any suspect mics are changed at the soonest opportunity and mics especially repositioned after custom changes. It's also up to the girls to enforce the show rule that mic packs remain switched on at all times—something that performers do not always remember, as Janet Woodhouse points out.

'People see an on-off switch on the mic pack and have a tendency to switch it off if they're having a private conversation or going to the loo. Of course what happens then is that they forget to switch it back on when they go on stage and there's no audio. You actually have to get quite stroppy with people and get it through to them that as long as they're miked-up the pack stays switched on ▶



Production Sound Engineer Mike Walker's design for the *Joseph* comms system

...how times change ...



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—whatever they're doing!'

The production also includes a couple of hand-held Sennheiser radio mics which apart from delivering more level than the mini-mics, make good props during a rock-and-roll sequence and the finale. A couple of spares are also kept which can quickly be handed to performers in case their lavaliers fail.

Radio mics are transmitted on 32 separate frequencies, but these inevitably change as the show moves from venue to venue requiring new frequencies to be licensed.

'It's been a bit of a nightmare touring this because we're looking at a lot of radio frequencies,' admits Mike Walker, 'Here we're having to take away the Channel 35 receivers that we had licences for at other venues and add Channels 22, 24 and 25, because Channel 35 is being used in the Manchester area for test transmissions of Channel 5 TV. It means that we have to place ahead and change parts of radio systems accordingly. Similarly there is radio talkback as part of the comms system which we have to license everywhere we go.

'Although we haven't experienced any external interference problems with the radio mics, we have had problems with the radio comms at Manchester,' continues Walker. 'It's from a fairly high powered local transmitter which we think is probably a taxi company. Apparently it's also been interfering with the comms system at the Opera House where they're doing *Phantom of the Opera* under a mile away from here.'

'The board has been laid out so that the operator controls vocals with his left hand and the band with his right,' explains Richard Ryan

The desk

The communications system being used for *Joseph* is a sophisticated Clear-Com package which has been supplied by Autograph Sales.

Mike Walker: 'We've ended up with quite a sizable system, which has grown substantially from the original specification. The setup is a 4-ring system with two rings for lighting, one for the stage and one for sound. The radio talkback is a duplex system that interfaces with Clear-Com providing stage management with up to eight sets. wiring is generally integral to the multicore, but also to make the installation easier. There are only a few single cables we have to run, to the fly floor or the follow spots for instance.'

Also designed for the rigours of touring is the Cadac *J Type* console. built specifically for the tour, the desk is in two frames forming an 'L shape with 38 modules in one and 56 in the other. The console has 12 VCA groups and a 12 x 24 matrix which is used to feed the Tannoy system left-right, Meyer system left-right, out-speaks on the truss, down-speaks, delays, frontfills, and surround system left-right. All 74 inputs are being used and the desk has been configured so that radio mics appear on the left and band mics on the right.

'The board has been laid out so that the operator controls vocals with his left hand and the band with his right,' explains Richard Ryan. 'Most of this is being done using VCA group masters—the 16-piece band for example being split across three VCAs—Rhythm, orchestral, and Keyboards. Faders that are furthest away from the operator tend to be things that don't need tweaking through the show, like reverb returns and hand-helds that are individually controlled by VCAs.'

The console is fitted with Cadac's proprietary automation system which control VCA levels, mutes, routeing, MIDI programme change and relay ▶

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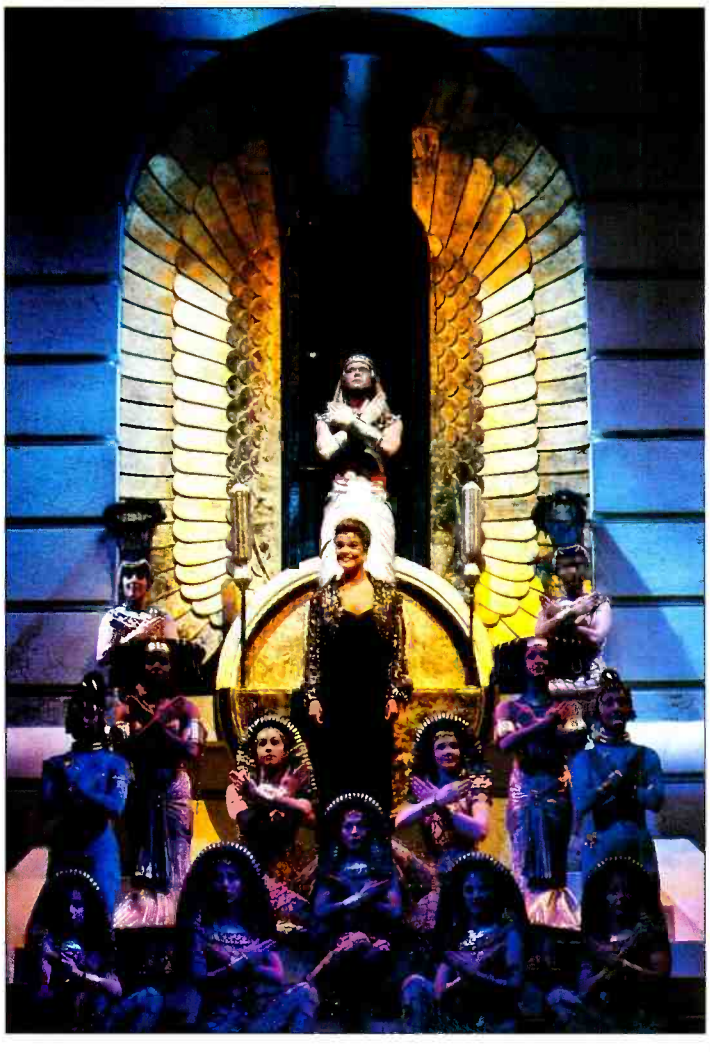
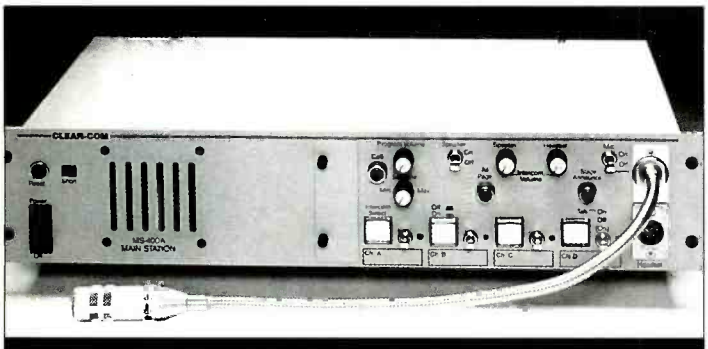


PHOTO: MICHAEL LE POER TRENCH

events by storing settings as a series of cues. These are then stepped through at key points through the show. apart from the desk functions the automation also controls a rack of Yamaha *SPX 1000*s, four Yamaha *DMP 11* mixing processors (used to submix children's mics, drums, and percussion), BSS delays for vocal mics, and an Otari half-inch 8-track which contains the prerecorded 'mega-mix' used at the end of the show in surround sound.

'This is actually a very small show as far as cues are concerned,' comments Sound Operator Steve Brierley, 'We're only using 52 cues whereas on the last Cadac show I did, *Aspects of Love*, we used over 80 which probably about average—although I know it go as high as 130.'

Another use of cues, and one that fits in very well with a touring production, is for system checks. ▶



Clear-Com *MS-400A* Main Station, the heart of the comms system used for *Joseph*

'Because the desk has programmable routeing modules we can use cue to provide a walk around system-check,' says Brierley. 'By adding step times between cues it gives the operator time to move from one area to another and make it possible for one person to check the whole system by themselves without relying on someone at the desk to do the switching. We also have a set of cues that are designed to check that all switching and MIDI functions are operating correctly, and this can provide a useful tamper check.'

So far the tour hasn't suffered from any tampering problems, but as Brian Beasley recalls the West End production of *Joseph* was less lucky.

'It went on for a few months and it was obviously someone who knew that they were doing. On one occasion all the gains on the reverb return had been cranked-up—unfortunately the operator didn't notice it during his checks, and the first anyone knew about it was when he switched cues at the start of the show and the whole system went into colossal and quite dangerous feedback. After that incident a video camera was placed above the desk to try to snare the culprit, but this was found the next morning in pieces. However, from then on the tampering did stop, although everyone remained very wary and very nervous for a long time after.'

A feature of the *J Type* console that seldom gets used, is the ability to change modules without powering down the console.

'It's a very useful facility,' claims Richard Ryan. 'Although we haven't had occasion to use it yet on this tour, I have done so in the past when a module's become noisy midway through a show, and it's proved very quick and easy to change. We carry a spare of each module in the desk, but generally speaking it's rare that you get any problems.'

And the same can be said for the *Joseph* tour as a whole, which thanks to the hard work and professional attitude of a small group of dedicated people, has gone impressively smoothly. The show's next stop is Edinburgh in Scotland, where the show will run until March 1995—after that the tour may call at some additional UK theatres before moving into Europe. ■

PHOTO: MICHAEL LE POER TRENCH

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TOURING EQUIPMENT LIST

Sound Desks
(including tape playback):
- Cadac 74-input *J-Type* Console
- Otari *MX5050*
- Yamaha *DMP11s* (4)
- Yamaha *HA8* mic preamps (4)

Amplifiers
- Amcron *MA2401* (35)
- Amcron *MA500* (3)
- Yamaha *PC2700* (3)
- PIP ATN cards (41)

Loudspeakers
- Bose *302* subwoofers (11)
- Bose *802* (24)
- JBL *Control 1* (64)
- Meyer *UPA-1A* (7)
- Meyer *USW* subwoofers (4)
- Servodrive Contrabass (4)
- Tannoy *3836* speaker-chassis drivers (7)

Processing
- BSS *TCS 804* delays (5)
- BSS *FDA 360* stereo crossovers with 24dB/oct Linkwitz Riley x-over cards (2)
- BSS *FDS 310* stereo crossovers (2)
- Meter *M1A* controllers (4)
- Meyer *B2* controllers (2)
- White *4700-2* stereo graphics (2)
- Yamaha *D2040* delays (4)
- Yamaha *Q 2031* stereo graphics (8)
- Yamaha *SPX1000* (4)

Radio Mics
- AKG *D112*
- AKG *C414 ULS* (2)
- AKG *C460/CK61* (5)
- AKG *C402* (2)
- AKG *C408* (4)
- Beyer *M160* (3)
- Beyer *M201*
- Crown *PCC 160* (13)
- EV *RE20* (2)
- EV *PL 88*
- Neumann *KM 140* (4)
- Neumann *TLM 140*
- Neumann *U87A* (2)
- Sanken *COS11 PT* mics (13)
- Sennheiser *MD441 U* (5)
- Sennheiser *SK2012-9* VHF transmitters (10)
- Sennheiser *SK2012-* UHF TV transmitters (18)
- 10-way receiver system for above based on *EMC 1036* chassis system
- Sennheiser *EM1051* Tx-Rx (4)
- 18-way *1046* receiver systems
- *MKE2-2R* (red dot) mics (pink) (28)
- *MKE2-2R* (red dot) mics (black) 28)
- Sony *ECM77* mics (13)
- TP DI box

Comms and Video
- Beyer *DT108* headset (10)
- Grundig 27-inch monitor (2)
- Grundig 9-inch monitor (6)
- Ikegame ICD camera
- Motorola *P60* Duplex walkie talkie c/w headset (8)

Pit Systems
- JBL *Control 1* (6)
- Yamaha *MV 802* mixers (4)
- Yamaha *P2075* (3)

Miscellaneous
- 23 signal multicores (8, 12 and 17 pair)
- Over 6000m speaker cable
- Revox *C221* rackmount CD player
- (1) Sony *PCM2500*
- IBM PC (2)

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TACTILE TECHNOLOGY M4000

James Douglas evaluates the American *M4000* console and identifies the elements of a new hybrid analogue-digital console design

The current paradigm in console design is simple: technology springs from the furtive imagination of talented individual that responds to a new market need, or one that can extend the envelope another couple of centimetres. Normally this creative process takes place within a well organised company that can gather together the intellectual and financial resources necessary to perform the required research and product development. Seldom, if ever, do these resources spring into being simply to develop one-off products which, through more than a degree of serendipity, proceed to appeal to a far wider market than its inventors ever envisaged.

One such company is Tactile Technology which, according to company President Mark Cohen, was set up to design and develop the kind of console that he required for his own music projects. 'Yes', Cohen recalls, 'the *M4000* that we now produce came into being because I wanted an easy-to-use, yet very powerful console that I could use to memorise every front-panel setting. I needed a mixer that would let me return to a project at a later date, with everything set to exactly the same values that I had been using before.

'Many of us need to prepare several different versions from a master mix, maybe for a video-game version, or perhaps a movie score. The ability to reset an entire console—including, with appropriate MIDI-controllable outboard units, reverb and related effects —means that you can get back to the exact mix you had up weeks or months ago. Also, during preproduction I like to be able to store several mix-EQ settings so that I can some back to them as the project progresses.

'I looked at the Euphonix *CS2000* and Trident *DI-AN*, but decided that I needed a dedicated control per function, although both of these boards were extremely well designed, they seemed just a little too complex to use. Like a lot of engineers, producers and musicians, I do not want to have to press several buttons to reach the correct assignable control. I want to go to it straight away, anything else just slows you down.'

The result of nearly three years of R&D, the new *M4000* Automated Mixing System from Tactile Technology was intended to 'be fully automated, fast to use and relatively small,' Cohen offers. 'Also, it does not need an external PC to operate, and, unlike other reset designs, the *M4000* does not require the user to manually reset every control against system setups displayed on a video screen.

'We also realised that practical expandability was an important factor; users are tired of buying one mixer, only to have to buy a larger one a year later. Additionally, our market research showed that people need such hard-to-find features as event controllers —which are extremely rare on consoles, but very useful for triggering outboard gear —plus MIDI effects automation, programmable inserts, and the ability to swap from mono to stereo input channel configuration with just a keystroke.

'We were also posed the Digital Question. How do you accommodate digital stereo inputs from, for example, a hard-disk recorder or editor, add analogue-signal input in the digital domain through mixdown? To solve this problem, we added a digital-audio mix bus, which allows a two-channel digital input to be blended in the digital domain with the *M4000*'s stereo output, and the result laid directly to DAT, for example. Who else offers that function?'

Overview

The *M4000* Automated Mixing System comprises the following system components:

Model M410 Assignable Hardware Controller, which provides access to 24, dual-channel signal paths, divided into upper and lower sections. (The lower section of 24 signal paths offers a 100mm fader per channel; the upper bank of 24 signal paths are equipped with a shorter 60mm fader per channel.)

Model M420 Mixing Engine, or APC, to use Tactile's vernacular, which is in reality a digitally-controlled rack of analogue circuit elements, including gain control, EQ and summing functions; all parameters are updated every subframe to produce the designated console layout. Internal system routeing is provided to a total of eight bus outputs, six auxiliary sends, two independent monitor/two-mix paths, plus solos, PFL and AFL outputs.

Settings of all assignable front-panel knob and switches are continuously being scanned and their values sent to the analogue-digital engines; they can also be saved as snapshot settings, or stored dynamically against a MIDI or time-code reference. All *M4000* systems are supplied with a diskette drive for saving snapshot automation setups and loading system updates; dynamic automation versions feature a 250Mb hard drive. Since all system functions are MIDI-compatible, a MIDI sequencer can be used to edit automation data, as well as providing control of external effects boxes units. An Events Controller provides four TTL-compatible logic ports, eight relay closures and four opto-isolated outputs; predesignated events can be triggered either against time code or MIDI Scene Charges.

Three variants of system automation are available: the basic *M4000A* configuration provides snapshot-scene automation of all routeing, level and EQ functions (with 'next-previous scene' sequencing for live sound applications): the *M4000B* adds dynamic automation of all fader settings, mutes and channel pan against internal-external EBU-SMPTE time code; and the advanced *M4000C* features servo- controlled faders for all upper-channel and lower-channel bus and master outputs. Up to five Mixing Engines can be controlled via a simple Local Area Network from a single user interface, for banked control of a total of 240 assignable channels.

Scheduled to be made available by

Tactile Technology's Model *440A* main control unit

The *M4000* automated mixing system

late autumn, an all-digital version of the M420 Mixing Engine —designated the M450—will offer AES-EBU-format individual inputs and outputs, plus *ADAT*—and TDIF—compatible I-O ports (the latter for Tascam *DA-88* modular digital multitracks). Both analogue and digital engines will be controllable from as single *M4000* Digital Controller, for mixed-format applications.

In terms of cost competitiveness, a basic, 48-input *M4000B* with storage-recall of up to 480 time-code-based, snapshot-scene automation (faders, mutes and pan) plus control surface sells for $36,000 (US), additional 24-channel sections with companion power supplied sell for $19,000. Moving-fader automation adds another $9,000 to the cost of a basis remote controller. (So long as they are reusable, existing owners can simply exchange a non-servo M410 controller for the M411, which features servo-driven faders, for the current price difference.) For larger installations, a pair of engines plus a single moving-fader controller is capable of providing 96 simultaneous input paths, routeing to eight buses, stereo and monitor buses, and sells for just $64,000.

M490 PSUs can power up to three system units; one possible configuration run from an M490 might include a Control Unit, M420 input and M430 Group-Monitor units. A relatively simple RS-422 serial connection is used between the various M440 units and the master M400-410 Digital Controller. For short runs up to around 100 meters, the firm recommends twisted-pair cabling, although a fibre-optic link is available for longer runs.

Each M420 Input Unit offers connections for 24 mic, line and tape inputs, plus send-receive inserts and a direct output per channel. The M430 Group-Monitor Unit offers eight group inserts; auxiliary sends-receives; eight group outputs with parallel outputs for tracks 9–16 and 17–24; stereo L/R send/receive; and monitor outputs.

Control surface

The *M4000*'s control surface features a bank of 24 upper faders and a bank of 24 lower faders, plus eight group, stereo monitor and stereo master-L-R faders. Each signal path boasts a dedicated LED meter, plus others that are assigned to the eight bus outputs and stereo buses. A small LCD window with controller buttons are provided for selecting specific functions, and displaying numerical and EQ settings, or a graphic (amplitude versus frequency) display of channel EQ.

Each signal path features a dedicated level fader, plus solo and mute switches. An assignable Section that runs left-to-right across the top of the user interface is accessed via 'active' buttons located below the upper fader bank. Stereo mode can be activated by selecting two adjacent buttons; settings from the left-hand/odd channel of the pair are mirrored into the right-hand/even channel.

Either mic, line and-or tape inputs can be made active on a selective channel, with 48V phantom power, 20dB pad, phase reverse and a 75/150Hz low-pass (12 dB/octave) filter for the mic source. Separate gain trims are provided for mic and line inputs, plus a peak-reading meter for monitoring the selected signal source. An insert send per channel can be set to follow the mic, line or tape input signal; the insert point is normally post-EQ, post-fader.

The Monitor section sets up assignments to the selected upper or lower monitor fader bank, with pan. A Group-Stereo section assigns the post-fader signal between a total of the eight group outputs, or during remix to the Stereo LR output. Panning is across left-right, or odd-even buses. An Auxiliary Send section selects either mic, line or tape signal path to one of six auxiliary outputs.

A single EQ section per channel strip is assignable to either the Input signal (mic or line) or the Monitor signal (normally Tape input). The assignable 3-band EQ section per channel features a peak/shelf LF band with a roll-off between 45Hz and 1.5kHz, a fully parametric MF band continuously variable between 400Hz and 12kHz, and peak-shelf HF band with a roll-off between 800Hz and 20kHz. cut-boost for each section is ±15dB. An EQ NULL button sets the gain settings of all three bands to 0dB.

A dedicated Input COPY key enables EQ, fader ▶

BEHIND THE SCENES

The tale of Tactile Technology, is the story of three paths converging. Yoshiharu Abe, founder and past President of Fostex Corporation, left that firm in 1990 to set up his own R&D and engineering consulting firm, whose client list soon read like a 'Who's Who' of the audio and video community. (Abe was also one of the founders of TEAC, and founder of Tascam division).

Coincidentally, a few months after Abe left Fostex Corporation, Fred Huang, the firm's General Manager, also left to manage his investments. Prior to working with Fostex, Huang had been Executive VP at Maruchan, and Director of New Product Development at Tascam.

In mid-1992, Mark Cohen retired as VP of Sales and Marketing with Fostex Corporation to return to writing and producing music. 'As the work started to pile up,' he offers, 'I needed a new console. Having taken a look at what everyone had to offer, I realised that none of them really focused on my particular needs—and I had the money to purchase any console I might desire.'

'Eventually I came up with a design that would handle my needs, and then some. At this point, I contact Phil Ramone and Roger Nichols to ask for their 'wish lists' of console features. They were both extremely enthusiastic about the design I had come up with, and gave me their suggestions and support.

'At this point however, I still didn't have a 'commercial' vision of my design; I figured it was such a personal vision of console design that only a few people would want it.'

During a reunion dinner with Yoshiharu Abe and Fred Huang, Mark Cohen expressed his frustration with the design of current mixers, and told them about his new design concept for mixers with a different style of control surface. Abe and Huang showed interest in the new concept, and suggested a way in which it could be manufactured in limited quantities. Thus Tactile Technology was born (The name was chosen, Cohen concedes, because it relates to the company's desire to make products with better interfaces. 'In other words,' he says, 'the controls should be 'user-friendly' instead of 'expert tolerant!')

The firm currently employ some 25 permanent staff, 7% of whom are engineers or system designers. All manufacturing takes place in Japan. Sales are currently approaching 40 units per month. To reduce end-user prices, Tactile Technology sell direct to customers through North America and Latin America; agents are used throughout the rest of the world.

'We consider that one of our $64,0000 systems would end up cost $110,000 if it was sold through a conventional dealer network,' Cohen says, 'with the various margins we'd need to offer. This way, we can pass the cost savings directly to our customers.'

In the autumn of 1994, Tactile Technology first displayed the resultant *M4000*, a new concept in assignable front-panel design that appears to have received overwhelming acceptance. 'Since that time,' Cohen remembers, 'we have added many features to allow use of the *M4000* in many different production environments, including studio music recording, video postproduction, live sound, and theatrical applications.'

'I believe that the *M4000* is the fastest automated console to use, and certainly one of the most flexible,' Cohen adds. ■

and switch settings to be duplicated to other channels, EQ copying relays only equalisation information. In addition, a LINK key allows a pair of input-channel or group faders to be assigned as a stereo pair; solos and mutes are also interlocked.

A pair of Multiply keys—one located on the left, and the other on the right-hand side of the Controller—accelerate the action of rotary controls by a factor of four. During system power-on, the default is high-speed or a Turbo-boost mode that speeds up control such parameters as pan, auxiliary sends and similar functions. For more precise of, for example, mic-line gain or EQ centre-frequency, the user can begin in high-speed mode, and then throttle down to normal for find adjustments.

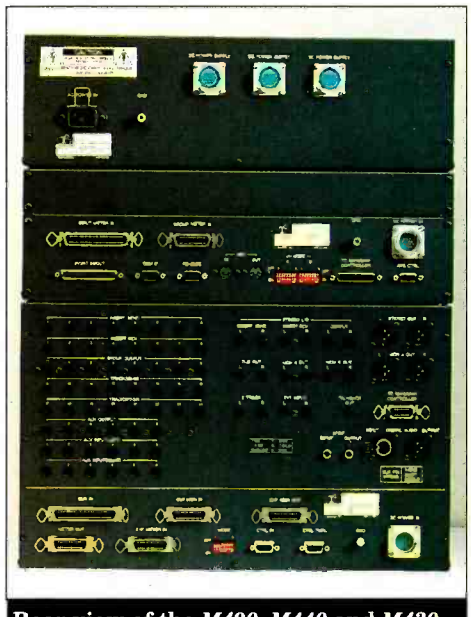
Resetting of channel levels to match previously stored snapshot or dynamic mix data can be achieved via a pair of LEDs located above each fader, the position in which both LEDs are dimmed indicates the fader's true position.

Several Global Functions enable user or factory-set snapshots to be recalled instantly.

Tracking mode, for example, automatically selects mic-line to the lower bank, and tape inputs to the upper, with output routing from the lower bank to group buses, and from the upper bank to the stereo bus to emulate an 'in-line' recording-overdub configuration. Mixdown mode selects off-tape signals to all faders, with output routeing to the master stereo bus. Allmix selects mic/line inputs to all faders during, for example, live-mix applications.

To eliminate 'double printing' of outboard special effects during mixdown, an insert I-O can set to function during Tracking Mode yet be inactive during Mixdown.

A quartet of user-programmable keys enable storage of favourite snapshot configurations, while two 'scratch-pad' areas are available for holding temporary EQ settings and so on,. for copying from one channel to another, comparison during mixdown, for example.



Rear view of the *M490*, *M440* and *M430*

Rear viewof the Model *M420*

Enhanced functionality

By the end of this year, Tactile Technology are scheduled to unveil the new *M6000* automated Mixing System, which will offer a total of 48 dual-channel strips (for a total of 96 simultaneous inputs), routeing to 24 buses and 12 aux sends. 'Unlike the *M4000*,' Cohen explains, 'the *M6000* will be laid out with 24 channel strips to the left and 24 to the right, with the master-control section in the centre.'

The new M450 all-digital Mixing Engine, scheduled to be made available by the end of this year, is expected to cost between $60,000 and $70,000 (US). To ensure direct compatibility with current outboards—and also reduce processing delays—all inputs to the engine, as well as inserts, will be as Analog ports, as well as AES-EBU-format I-Os. A Dynamics Package is also planned for later this year. The 3U-high rackmount unit will offer eight channels of compression, limiting, expansion and gating; price is expected to be less than $5,000.

'Also on the immediate horizon,' says mark Cohen, 'is special *Macintosh*, *Windows* and SGI compatible software that will provide both off-line editing and on-line operation of the controller and multiple mixing engines. We are also preparing new system software that will enable two controllers to be on-line simultaneously, a development that should expand the *M4000*'s range of potential applications into the film and video arena, by providing concurrent access to mix functions.' ■

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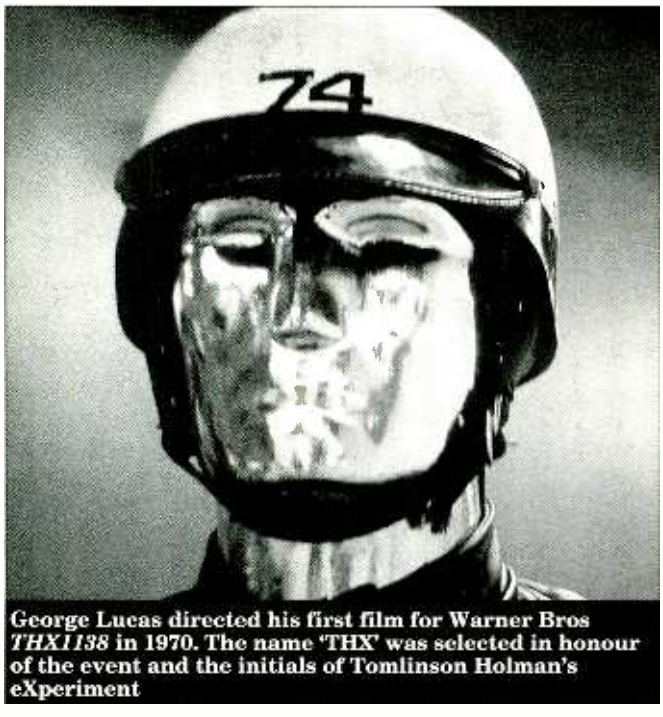
THE EXPERIMENT

James Douglas examines the Lucasfilm THX Sound System specification and looks at the Apogee Sound triamplified Motion Picture Theatre System One for smaller rerecording stages and dubbing suites

Arguably, two technical developments, more than any others, have helped advance the state of the art in motion-picture sound: Dolby Surround, and the Lucasfilm THX programme. While Dolby Surround provided a standardised encode-decode technique for carrying at least four channels (LCRS) of matrixed information to the viewing audience, Lucasfilm's THX Division ensured that the audio signal could be reproduced faithfully in the movie theatre or home playback environment. In essence, the sound department now has a simple matrix-encoding 4:2:4 channel system for creating highly realistic surround-sound audio. And theatre operators have no excuse for depriving their audiences of the high degree of art that the sound designers and rerecording engineers practised on behalf of the film's directors.



George Lucas directed his first film for Warner Bros *THX1138* in 1970. The name 'THX' was selected in honour of the event and the initials of Tomlinson Holman's eXperiment

Contrary to popular belief, THX is not just a patented design licensed by Lucasfilm to movie theatre operators and rerecording facilities. It also represents an installation process and playback specification involving, among other parameters, carefully defined frequency-response linearity at specified playback SPLs throughout the environment. If these carefully defined specifications are met, then the movie house or dubbing suite will, for a fee, be measured by Lucasfilm and can then describe itself as being THX-Certified. To date, more that 800 such environments have met the exacting design criteria developed by Lucasfilm's Theatre THX Sound System programme, including some 120 dubbing theatres around the world. THX-certified motion-picture theatres are retested each year to ensure that optimum playback quality is being maintained.

In addition to system design, Lucasfilm's Theatre Alignment Program (TAP) was set up in 1983 to offer a variety of services to film-makers and movie studios. TAP services include a review of 35mm and 70mm film prints before release; technical alignment of projection and sound equipment at theatres; and on-site evaluation during regular showings of a film to the general public.

The THX programme

Historically, the THX Programme has its origins in George Lucas' desire to create 'the ultimate theatre experience', which, in practical terms, means clearly hearing all of the elements of a carefully crafted and recorded soundtrack, as well as seeing a bright and correctly projected picture. In 1980, Lucasfilm hired Tomlinson Holman, a leading sound-system designer and audio 'guru' to build a postproduction facility in preparation for the making of *Return of the Jedi*; Holman was eventually named in the position of Lucasfilm's Corporate Technical Director.

'After examining the entire film-sound process, from recording on the set to playback in the theatre,' Holman recalls, 'we developed a new theatre sound system consisting of a unique combination of loudspeakers integrated with specific room acoustics that affects the actual design and construction of the theatre. The system was created to complement advances being made by Dolby Laboratories in soundtrack encoding and decoding, which concentrates on the A chain. The THX Sound System concentrates on the B chain of a theatre or dubbing stage's sound system.'

Although the first Holman-designed room was built for in-house use at the Lucasfilm postproduction facility in San Rafael, California, it was not long before other studios and commercial theatres asked if it was possible to install the new patented system. As a result, a name was selected—'THX' in honour of Lucas' first film, *THX1138*, and the initials of Tomlinson Holman's eXperiment.

As Holman explains, THX Theatres provide several sonic advantages for film- goers and rerecording engineers:

1. Extended frequency range from sub-bass to beyond 16kHz.
2. Smooth, linear frequency response over the specified range.
3. Even sound that uniformly reaches every seat in the auditorium.
4. Enhanced dialogue intelligibility, allowing audiences to understand what the on-screen performers are saying.
5. Enhanced dynamic range performance, enabling playback of all sounds—from the softest whisper to the loudest explosion—without distortion.
6. An enveloping surround-sound experience.
7. Carefully defined background noise and isolation from external noise sources, including leakage from adjacent theatres.

By using the same basic design in both the rerecording theatres and movie theatres, mix engineers now have a calibrated environment in which to place sounds throughout the 360° soundstage, safe in the knowledge that they will be heard at the same SPL and frequency response by the viewing audience.

Key to the THX Design, Holman offers, is the patented installation process that involves the use of a baffle wall for the behind-the-screen loudspeakers, whereby the bass response is based on a 2π boundary condition. In addition, because high-

PHOTO: KEITH HAMSHERE © 1991 UNIVERSAL CITY STUDIOS

Radioland Murders **directed by Mel Smith. A romantic comedy involving many new sound editorial and mixing techniques**

frequency signals are liable to be reflected strongly between the rear of the screen and the barrier wall or baffle, absorbent material is added to minimise HF reflections. The current design is a two-way system, with a crossover frequency of 500Hz.

THX-approved components

As has been mentioned, THX is a performance specification, not just a system design—theatre operators and dubbing facilities are pretty much free to use any equipment they like from a list of tested components; it is only the end result that is measured for THX certification. 'THX really is a specification criteria,' Holman stresses; 'not a specific set of components'. The only noncommercial product that needs to be used in a THX room is Lucasfilm's patented 2-way crossover. (The THX *Monitor 3417* is a combination booth monitor and crossover power frame.)

During the past several years Lucasfilm's THX Division has gained a great deal of experience in the evaluation of a large numbers of completed systems. To assist designers to focus their attention on system components that have more that a fighting chance of providing the results that they need, Lucasfilm regularly publishes lists of subwoofers, surround speakers, power amps, and so on that the division has tested and approved for use in a THX-Certified room.

'One of the main criteria for selecting screen-system components is that they must offer a directivity match at the crossover point,' says Holman. 'After more than a decade of evaluating THX-Certified rooms, we have determined which ones will work with our two-way crossover and which will not.'

The list of THX-approved components includes power amplifiers, replay-decode processors (including Dolby *CP-200*, *CP-65*, *CP-55* and *Ultra Stereo JSX-1000*), compression drivers, horns and LF woofers that will produce the required SPL and linear frequency response from the two-way THX design. Also included on the approved list are subwoofers that have been found to offer suitable performance in the Dolby Surround 5.1 or 'Baby-Boom' 70mm and discrete, multichannel digital-sound formats.

Finally, the list provides details of complete loudspeaker cabinets that can be incorporated into a THX-approved design, including Altec *A10/MR945A*, Electrovoice *TS900D-LX*, JBL *4675B-8LF* and *4675C-8LF*, plus KCS *S-3001*s. Such systems can also be configured for discrete 6-track playback, with five, full-range channels behind the screen, plus surrounds—as well as the newer digital multichannel playback formats, including Digital Theatre Sound (DTS), Dolby's SR.D and Sony's SDDS configurations. ▶

MOTION PICTURE THEATRE SYSTEM ONE

Based on the highly successful approach taken in over 800 installations of the THX Sound System Programme throughout the world, the new Apogee Motion Picture Theatre System One is a triamplified system designed specifically for smaller rerecording stages and dubbing suites, where limited space might preclude the use of a larger system. In addition, the MPTS-1 is said to provide mixes that are in every other respect fully compatible and comparable with sound balances prepared in larger THX-certified mixing and rerecording environments, albeit from the use of a smaller, more compact system. The MPTS-1 is approximately a one-third scale model of the larger system.

Separate cabinets are provided for left, centre and right screen-associated loudspeakers; in addition, the MPTS-1 includes a single subwoofer for very low-frequency information, plus multiple surround cabinets. The three main left-centre-right playback-channel cabinets can either be mounted behind a perforated screen in conventional mix-to-film facilities, or located either side and below a video projection screen. Only 18 inches of depth is required for behind-the-screen installations.

Designed by Lucasfilm's THX Division, each main-channel system consists of two enclosures: the *L-1* Woofer Cabinet houses a 12-inch Apogee *DD1202* low-frequency driver; the *MH-1* mid-high cabinet contains a pair of 5-inch Apogee *CD103* mid-range direct-radiator drivers, plus an 2-inch Apogee *CD201* high-frequency compression driver mounted on a 90° by 40°, controlled-directivity horn. An adjustable bracket for the *MH-1* mid-high cabinet allows precise tilt-and-pan adjustment relative to the *L-1* woofer cabinet. Main system crossover frequencies are 160Hz and 1.5kHz.

The *S-1* subwoofer cabinet houses a pair of 12-inch Apogee *DD1202* low-frequency drivers in an optimally vented enclosure. The Surround Cabinets (a minimum of four per system installation) comprise dual Apogee *CD103* 5-inch MF cone drivers, plus a single Apogee *DD101* 1-inch titanium dome tweeter. An optional dual 18-inch subwoofer, code named *S-2*, is available for larger rooms that require higher output power at very low frequencies. The *S-2* provides a similar frequency response to the *S-1*, but with 6dB greater headroom.

A companion *MPTS-1* Electronic Signal Processor includes active equalisation for both film and video playback response; screen-compensation loss (for behind-the-screen installations); time and frequency-domain correction; and fourth-order frequency-dividing (crossover) networks, with acoustical 24dB/octave, linear-phase filters. Each channel is triamplified, and shares a common subwoofer output. Separate left and right-surround outputs are also provided. Quoted overall frequency response from the system is 20Hz–18kHz, ±3dB, 1m on-axis. SPL is quoted at 118dB continuous; 124dB peak at 1m. ▶

Smaller environments

Responding to the need for a THX-approved system that might be used in smaller, more compact environments—possibly a video-style mix-to-picture room, or a one-man rerecording stage —Lucasfilm recently unveiled a design that is functionally a one-third scale model of the larger, two-way system—now dubbed Type 1. Designed by Lucasfilm-THX and manufactured by Apogee Sound, the new Type-2 or 'Junior' design has now been released as a commercial product in the form of Apogee's Motion Picture Theatre System One (MPTS-1), see next page.

'For smaller venues,' Holman explains, 'we needed to rethink the basic two-way operation of our Type-1 THX system. For rooms with less than around 12,000 ft³, we don't need as much sound power to reach the same SPL at the mixing console, and can scale back the power-handling capabilities of the systems; we still retain the baffle design to ensure 2π boundary conditions.

'Having examined the technical possibilities, we opted for a design that retained a major achievement from the Type-1 THX system—that it produces no discontinuity in directivity index at the crossover point—but in a smaller, more compact arrangement. Discontinuity in directivity at the crossover is a major problem with all loudspeaker designs; as we approach the crossover frequency, we need to ensure that the horizontal and vertical propagation patterns of one drive match that of the unit you are crossing over to. In the case of the Type-1 system, we could adopt a two-way design, with 90° x 40° constant directivity horns.

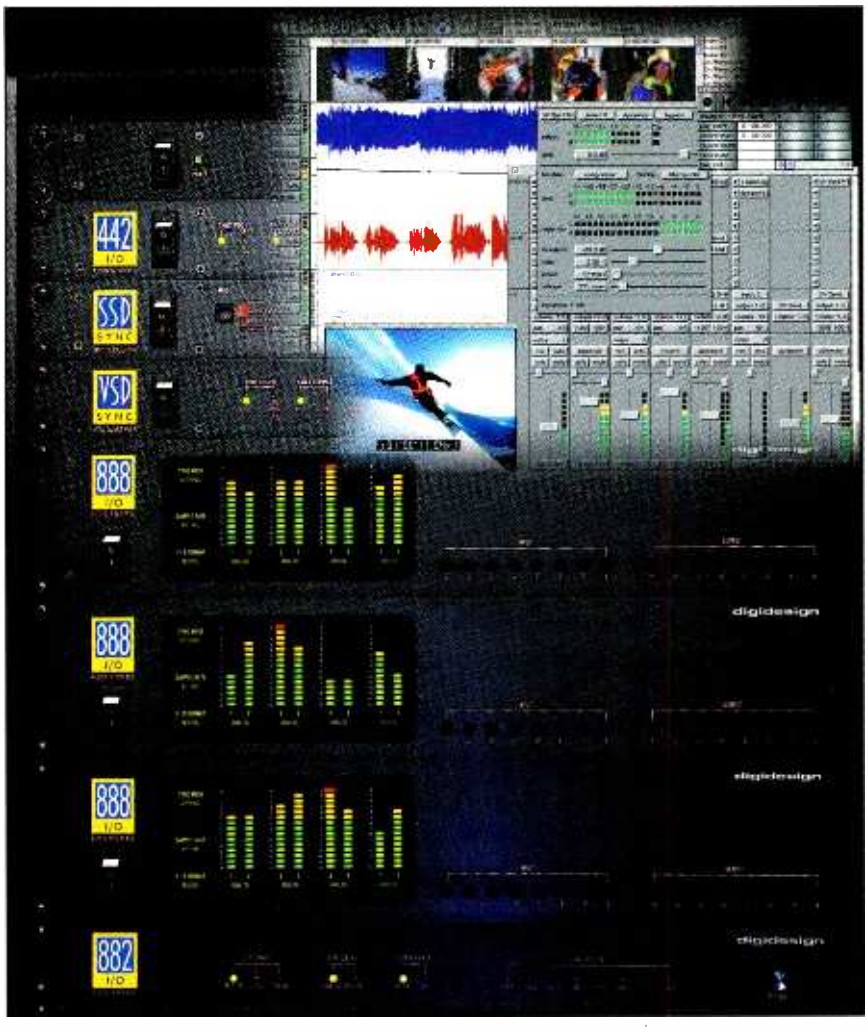
'For the Type-2 design, however, we needed to reduce the component sizes by around a third, which meant that the mid-range units would be five inches in diameter, and that the MF/HF crossover point would rise from 500Hz to 1.5kHz. In this way, we could maintain the same 90° x 40° dispersion pattern through the mid to high-frequency crossover. And for the LF-MF transition, we chose a crossover frequency of 160Hz. The electronic crossover is quite elaborate—much more than even the Type-1 system—to stitch everything together perfectly.'

Holman compared the performance of a prototype Type-2 THX design with a 'benchmark' Type-1 system installed in the main house theatre at Lucasfilm's Skywalker Ranch postproduction facility. 'We mounted the Type-2 cabinet next to the centre channel and performed A-B comparisons,' Holman continues. 'We found that the results were very good indeed, with high degree of compatibility in sound quality and frequency response throughout the pass-band.'

During subsequent listening tests between complete Type-1 and Type-2 THX systems at Skywalker Ranch—an Apogee *MPTS-1* system was installed recently in Studio A—Holman recalls that 'we found comparable results from mixes produced in our larger mixing stages to those from Studio A. While a mix may sound 'larger' in the big room' —there is, after all, more sound energy being propagated into the room 'we do maintain the same 45° subtended angle between the left and right speakers, and a 50° field of view to the picture —parameters that we now use as standard references for dubbing and playback suites. The loudness, timbre and localisation all match well from a room that measures less than 12,000 ft³ room, to one that's 120,000 ft³.' ▶

PHOTO: KEITH HAMSHERE © 1993 LUCASFILM

Action from Sean Patrick Flanery in an episode of *Young Indy Chronicles*. The series represented several landmark achievements for its producers, Lucasfilm



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***Radioland Murders*: The Miller sisters charm the studio audience**

PHOTO: KEITH HAMSHERE © 1994 UNIVERSAL CITY STUDIOS

MPTS-1 mixing of *Radioland Murders*

Production of the American television series *The Young Indiana Jones Chronicles* represented several landmark achievements for its producers, Lucasfilm. Company founder George Lucas elected to see if it was possible to produce an hour-long TV programme that had a 'look and sound' of productions that had been mixed in a large, movie-style rerecording stage, but which in reality had been prepared in a more cost-effective environment. Lucas, in conjunction with some technical expertise from Lucasfilm's Skywalker Sound and THX Divisions eventually developed a technique that allowed complex Dolby Surround soundtracks to be produced quickly and economically by just a single dubbing engineer, working with sound elements premixed to analogue multitrack machines rather than interlocked 35mm mag transports.

Since those ground-breaking achievements with *The Young Indiana Jones Chronicles*, which secured several Emmy Award nominations, and won an award for Best Music, George Lucas has been examining similar cost-conscious techniques during production of a major release film. According to Tom Bellfort, a Sound Supervisor at Skywalker Ranch, *Radioland Murders* involved use of 'many of the sound editorial and mixing techniques that we pioneered on *Young Indy*, but using digital storage and sound assembly'. *Radioland Murders* is a new film from English Director Mel Smith, and coproduced by George Lucas.

Working in Studio A at Skywalker Ranch, Sound Mixer Bob Edwards, with Tom Bellfort as the project's Supervising Sound Editor, spend nine weeks premixing and then rerecording *Radioland Murders*. The comedy follows the broadcast of an evening's radio entertainment in 1939 by WBN, a fictitious American fourth radio network.

To ensure that the multichannel soundtrack recorded in the smaller Studio A rerecording stage suite would translate to both large and medium-sized movie houses, Skywalker Sound installed one of Apogee Sound's MPTS-1 Motion Picture Theatre Systems.

Despite the fact that the premixes and rerecording were being performed in a narrower and shorter dubbing stage, Bellfort continues, 'the surround-sound mix [from] the Apogee MPTS-1 system translated very well to other, larger rooms here at Skywalker Ranch, as well as commercial movie houses'.

'George [Lucas] wanted sound to feature prominently in the film,' Bellfort offers. 'He envisioned using the same types of sound techniques as those used on *American Graffiti*.' But, in contrast to the 1950s music that served as an audio 'thread' throughout the soundtrack of *American Graffiti*, for *Radioland Murders* the continuing radio programme provides a continual backdrop for the action, as well as a sound link for the film's dramatic action.

Radioland Murders shares other conceptual techniques with *American Graffiti*. As Mixer Bob Edwards explains, 'we also used a 'Worldizing' technique of re-recording certain music and dialogue elements to simulate the on-screen environment. In that way, I could blend the 'Worldized' sound with the dry, original elements. The ability to place the audience into the atmosphere of the radio play, or bring them back to reality, can be done by carefully balancing the sound of each contribution in the final mix.'

Edited dialogue, Foley and ADR sound elements were premixed to a Sony *PCM-3348* digital 48-track, and then re-mixed in Studio A to 6-track DTS-format Surround Sound. Music was edited to a 16-track New England Digital *PostPro* hard-disk recorder, which was then synchronised to a video work print for the rerecording process using Studio A's automated Solid State Logic *SL-4048* console. Multitrack mix stems were laid back to another 48-track reel. During dialogue and sound-effects editing, a second *PostPro* was used to assemble and then prelay the elements in sync to the 48-track.

The emergence of Apogee Sound's THX System has successfully proven that a motion-picture feature can be mixed in a smaller environment than the industry might previously have believed to be possible. This new approach not only saves space, but also costs far less than a traditional dubbing stage—yet without sacrificing the quality of the end product, nor compatibility with commercial movie houses. ■

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Apogee Sound Inc, 1150 Industrial Avenue, Suite C, Petaluma, CA 94952, USA.
Tel: +1 707 778 8887. Fax: +1 707 778 6923.

Design philosophy

The MPTS-1's crossover frequencies and cabinet configuration are said to simplify system installation. Mid and high-frequency enclosures can first be placed correctly for best stereo imaging, and then the associated low-frequency enclosure placed either below, above or to the sides of the corresponding MF-HF enclosure. The system is designed for essentially flush mounting in a 2π boundary baffle, with the MF and HF enclosures rotated and tilted as necessary for uniform coverage of the listening environment.

A principal feature of MPTS-1 operation, Apogee state, is that its direct-field response is set to the standard 'X' curve of ANSI SMPTE Standard 202M-1991 and ISO 2969 frequency response curves, and that the directivity of the system is controlled by matching the dispersion of the elements at each of the crossover frequencies. According to Apogee Sound, this important advantage means that both the direct-field response, or first arrival, and the later arriving reflected and reverberant energy within the mixing environment are minimally affected by abrupt changes to directivity that plague other designs. Directivity changes can typically lead to coloration around the crossover frequencies.

Two switchable functions enable use in a variety of film-and-video-style operations:

For best sound-to-picture image synchronisation, the MPTS-1 system was designed to be installed behind a perforated motion-picture screen. However, for installations where it is not possible able to place the loudspeakers behind the screen, Screen Loss Compensation Equalisers can be disabled during system installation. (Circuit equalisation needs to be added to overcome the inevitable HF loss when using a perforated screen is used; obviously the EQ boost needs to be removed if the speakers are not covered by the screen.)

A Film-Video Electronic Switch is provided for day-to-day operations. In Film Mode, the system meets the standard 'X' curve of SMPTE 202 and ISO 2969, and is suitable for all monitoring operations associated with the film industry, including Dolby Surround dubbing and quality-control operations. In Video Mode, the response curve is adjusted to be flat, so that the program material designed to be heard over conventional direct-field loudspeakers in relatively small rooms, such as for television monitoring, can be accommodated. ■

Apogee Sound cabinets and *MPTS-1* processor unit

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Tapeless Technology in Radio Applications - the Users Point of View

SYPHA has published the results of an extensive survey into the use of tapeless technology for cart replacement, station automation and production applications. Over 500 radio stations in the United Kingdom and United States took part and the issues examined included:

- **Awareness and opinion of systems and technology**
- **Reasons for system selection and purchase**
- **Expectations of system performance**
- **Operational and technical support**
- **Applications and features required**
- **Investment decisions and future plans**
- **Sources of information and advice**

PSN Europe and Studio Sound magazines, as well as AMS Neve, BASYS, Broadcast Electronics, Computer Concepts, Harris Allied, Korg, RCS, Sony and Studer Digitec provided the sponsorship necessary to conduct the survey. However, the method and results were managed independently by SYPHA.

SYPHA specialises in providing consultancy and research services on the use of random access technologies for audio and picture recording, editing and replay. Other publications available from SYPHA include:

- **The Nonlinear Buyers Guide** - a buyers guide to random access video systems covering nonlinear and mixed mode editors, digitising cards and software, video disks, video servers and RAM stores;
- **The Tapeless Directory** - a buyers guide to digital audio workstations covering production, post production, cart replacement and station automation systems.

A summary of the survey can be obtained by sending a stamped addressed envelope to SYPHA. The price of the full report, entitled ***Tapeless Technology in Radio Applications - the Users Point of View***, is £225 or US$380 and can be ordered from:

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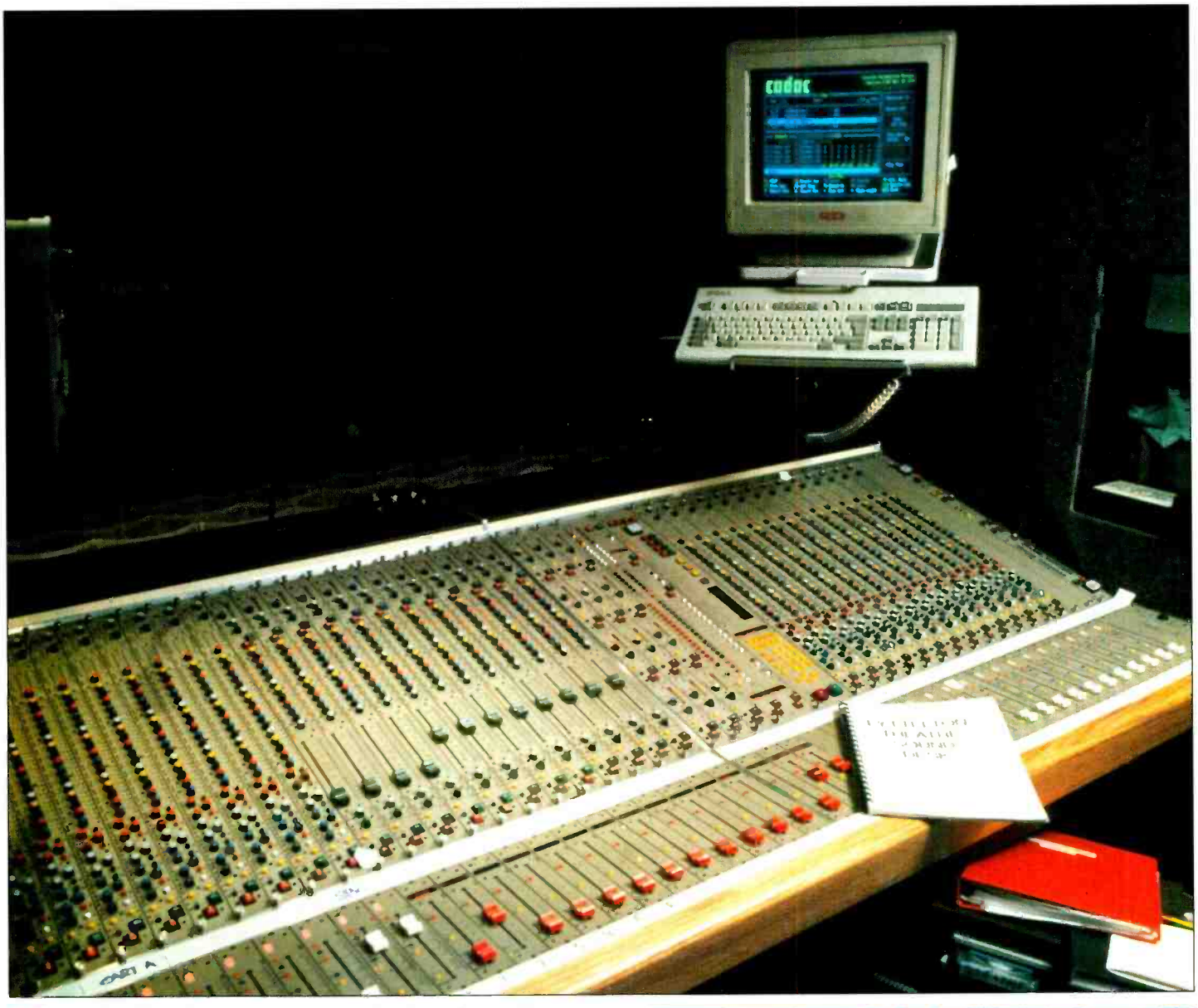
CADAC CONCERT



Having played a part in the conception of the *Concert*, the Royal National theatre offered the ideal opportunity to assess Cadac's latest console. Exclusive review by Patrick Stapley

London's Royal National theatre is deceptive: to the outsider walking into the large booking hall with its calm, relaxed exterior, it's hard to visualise the frenetic atmosphere that exists behind the scenes. But with three separate auditoriums —The Lyttelton, The Olivier and The Cottesloe—plus additional touring commitments, the National is the UK's biggest and busiest repertory house.

Being a rep theatre means that each of the three auditoriums will be running a cycle of three of four productions with changeovers occurring every few days—this could total as many as 12 different plays in a week. There is never a 'dark night' in any of the theatres, which will often be putting on matinees as well as catering for rehearsals—one begins to get an idea of the kind of demands that are put on the NT's sound department.

Head of Sound is Rob Barnard, who apart from looking after a team of seven operations, is responsible for equipment purchases—the latest of which is a Cadac *Concert* console which has been installed in the Lyttleton Theatre.

'Everything we buy here is driven ▶

MODULE FEATURES

INPUT CHANNELS

- Dual mixable mic-line inputs
- Routeing to 12 subs and 12 matrix outputs
- 16 aux sends each selectable Pre-Post On-Off
- Direct output pre-post selectable
- 3-band fully parametric EQs each with In-Out and Bell-Shelf for HF and LF
- Variable frequency HPF with separate In-Out
- Post-EQ-Pre-fader insert point with bypass
- PFL and Mute
- Manual, VCA or motorised fader

OUTPUT CHANNELS

Subgroups

- 12 subgroups each with PFL and Mute
- Insert point with bypass
- Separate input to subgroup mix amplifier with level control and switch
- Separate input mixed to insert return with level control and switch
- Routeing to 12 matrix groups with individual level controls and routeing
- 100mm Fader controlled output plus 20-segment LED metering (57dB range)

Matrix Groups

- 12 matrix groups each with PFL and Mute
- Insert point with bypass
- Separate input to group mix amplifier with level control and switch
- Separate input mixed to insert return with level control and switch
- Output level controlled by pot plus 20 segment LED metering (57dB range)

CENTRAL MODULES

- Check mode enabling channel mutes to be used as solo-in-place switches
- 11 frequency oscillator/pink noise generator with routeing
- Versatile communications system
- Aux group with 20-segment bar graphs, line adjustable IPs, and OP insert points
- Individual master mute switches for chans, subs, matrix and aux masters
- 4 Stop-Start switches for external machines
- RAM card slot for future use
- Display and controls for Cadac automation system
- Back-up computer switchover
- Assignable control of switching for channels, group modules and aux masters
- Up to 15 DC VCA group master faders—motorised or nonmotorised

Rob Barnard: 'I can confidently put a less experienced operator in charge of a show'

by the art,' states Barnard. 'What this means is that we're not buying pieces of equipment just because we fancy them, but because they serve a real need and allow us to perform our job better and offer our clients more.'

'We already had long-standing experience of Cadac having installed the unique *D-Type* console in The Olivier nearly ten years ago. That console was built to our specification, and in fact my predecessor, Tony Waldron, who now works for Cadac (and played an important part in designing *Concert*), was responsible for specifying the design with Cadac's Clive Green. *Concert* really grew out of that desk and includes the best parts such as central assignability and the automation which is very specialist.'

Concert has been developed over a 3-year period, and is Cadac's most highly-specified live mixing console to date, incorporating a full recall system and sophisticated automation package. As far as Barnard was concerned there was no contest in choosing a new desk for the NT.

'There was no point going to competitive tender because no one else was making the kind of desk we wanted. *Concert* is specifically tailored to our requirements in a variety of ways—for example we're expected to changeover plays very, very quickly—so having a console that can be reset more of less by the press of a button is fantastic. also during the rehearsal period for a new play, we need to be able to react as quickly as possible so we don't hold things up—directors want to be able to go back over complex sequences immediately without waiting around while we reconfigure everything and lose the mood in the process—here again *Concert* allows us to do that and we don't have to say no as much as we used to. Before we installed the console we used to start changing shows over at 10am without fail, but now we generally begin around midday—that gives you an ideal of how it's improved our efficiency, as well as our confidence in the system.'

Automation

Looked at simply, *Concert*'s automation system involves a series of scene changes (Cues) which are stored sequentially as a Show on hard disk—the operator then steps through them during the show recalling changes at the appropriate moment. Cues contain both console information and external data, allowing complex change to be reset effortlessly and precisely by a single operator.

The desk has two levels of internal reset: automatic and manual nulling. All switches and functions are stored in a Cue and will be instantly reset as the Cue is activated. Also stored in a Cue are fader and knob settings, and depending on whether faders are automated (VCA or proprietary moving fader) or manual they will either by automatically reset along with the ▶

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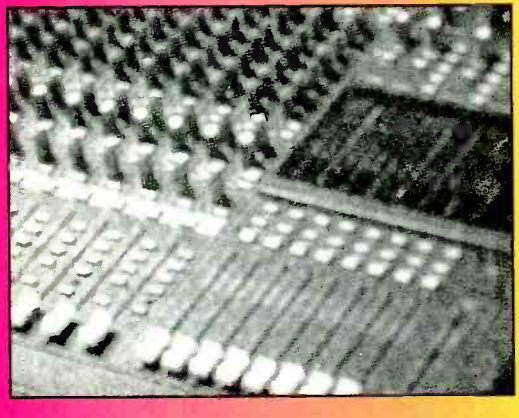

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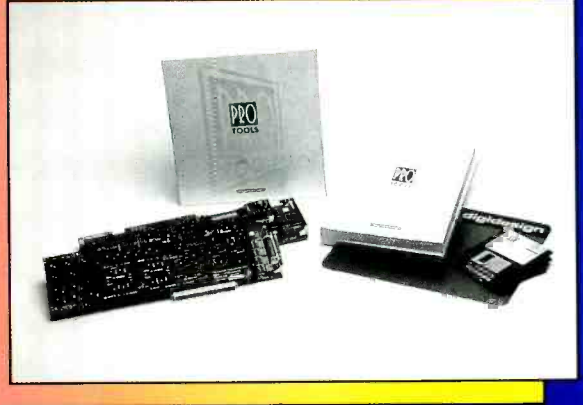

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switches, or can be manually nulled together with the knobs. All knobs and faders include two local nulling LEDs which show at a glance (especially in dimly lit conditions) which direction a control must be moved to match the Cue setting. An additional overall Recall LED, provided on each channel-section, will light if any pot is not at its recorded position.

'The automation systems cuts out a lot of time traditionally taken for an operator to "learn" the show,' states Barnard. 'It allows me to make better use of my staff, as I can confidently put a less experienced operator in charge of a show knowing that the automation system will guarantee repeatability and accuracy from performance to performance. It means that we can have as many people operating the shows as possible. Another benefit is that the operator can relax a bit more and devote more attention to the sound rather than having to worry about manually performing complicated resets.'

The automation does not stop at desk functions, and cues will also switch MIDI and events—changing over programs on outboard processors and triggering tape machines and carts. Cadac are also putting the finishing touches to sequencer software which will add yet another dimension to the automation process. As a stop-gap, the National are currently using Autograph Sound's proprietary sequencing software to control an Akai sampler.

'We use two PCs—one to control the *Concert* automation, and another to control sequencing,' says Barnard. 'Both step together as the operator changes Cues from the Central Control Module (CCM), so that cue data and sequencing occur simultaneously. The Akai sampler has become a very important part of theatre sound and has been adopted as something of a standard. You, of course, have the advantage of it being random access and digital, but it has the terrific advantage of being used in the Studio (the NT have two dedicated postproduction rooms where sound effects are created from CD libraries and their own 1/4-inch library that goes back 20 years), in the rehearsal and in the theatre. It allows us to modify things on the spot, whereas before when everything was on NAB Cart, we'd be continually scurrying backwards and forwards to the studio to make changes.'

Features

Another important consideration for the Lyttelton's console was that it should offer plenty of outputs.

'We're not usually dealing with huge numbers of inputs; typically there might be eight outputs from a sampler, another eight from an *ADAT*, a number of NAB Carts and perhaps a small orchestra. However, we do have to deal with a lot of outputs because we need to distribute sound in a very controlled way all around the theatre. So that's another reason why we can't buy a live sound console off-the-shelf simply because of output limitations.'

The National's *Concert* has 30 inputs, 15 DC VCA masters, a 12 x 12 matrix, 12 subgroups and 12 auxiliary buses 916 are standard). On occasions where 30 inputs are not sufficient, the *Concert*'s dual-input facility is used: each input channel has two inputs that may be accessed independently or combined allowing pairs of mics to be paralleled through one channel strip. A common use of this facility is in combining stage rifle mics which will feed a digital reverb to create different stage ambience-effects.

The physical size of the console also had to be taken into account due to limited space in the control room. With the large number of facilities being offered by *Concert*, it's surprising just how compact the desk is—an achievement that has largely been made possible by incorporating central assignment.

The desk features a double-width Central Assignment Module (CAM) which contains all the switches for the input channels, group modules and the auxiliary masters. By controlling the majority of console switching from one central location, a large number of local switches are removed leaving just the indicators; the desk thus becomes more compact (a plus point in most theatre environments), as well as leaving space for local nulling indicators, and more finger room around pots.

The CAM also allows switch settings to be copied from one area to another which can greatly speed-up initial setup time. Taken a stage further, the digital control element of the console makes it possible to perform console setup off-line from a PC—a facility that Rob Barnard finds potentially very exciting.

'Although we've yet to explore the off-line aspect of the console, it offers some very interesting possibilities. For example, if you're setting up a show say a week in advance, you can sit at a PC and design how the console's going to be laid out—using a lap top you can even sit in the theatre during rehearsals putting together Cue information. The next stage for us will be to get some of the console remoted, because ideally we'd like to work in the auditorium rather than being cooped-up in a small and rather isolated control room. There are plans afoot to take the Central Assignment section and a handful of faders and stick them on remote module out front which would make the sound designers job a lot easier.'

'Having said all this though, we have actually installed the console so that it can be easily taken out and operated in the auditorium which is essential for musicals. The problem, however, is that it takes up seats and of course my bosses ▶

don't like that; also it only really works if you've got a long-running production like *Carousel* which was here for three months—in normal repertoire circumstances, moving the console in an out every few days would be ridiculous.'

Cadac have an enviable reputation for sonic quality, and apart from the console's 'very true' signal path and 'superb' EQ, a fundamental issue for Barnard was noise.

'We're dealing with an enormous variation in dynamics which requires a very powerful loudspeaker system with amplifiers fully open. This means that the console electronics must be fantastically quiet, and from our experience with the *D-Series* we already knew that Cadac offered a very low noise floor. People's expectation of theatre sound is going up all the time and this requires us to deliver the very best in terms of sound quality.

'Over the last five to six years theatre sound has become much more high profile, and the sound designer, who really didn't exist until quite recently, is looked upon by the director as a key member of his creative team.'

'All our operators are multiskilled and can wear three hats—they can be sound designers, operators, or technicians and will swap over these rolls from production to production. Everything is done in-house, and we only call on freelancers when our people are out on the road; so far no director has ever come here and insisted that they bring their own sound designer which I think we can be very proud of. On the other hand our operators do go outside, for instance one of my team, Paul Groothuis, has recently been involved in codesigning the gigantic sound system for *Oliver* and Mike Walker (also ex-NT) at The London Palladium which features the biggest Cadac *J-Type* console yet built.'

Progress

With the rapid advances in theatre audio, how are directors reacting to the new technology and making use of it?

'There's generally a much greater awareness among people of what can be achieved now,' observes Barnard. 'Playwright Alan Ayckbourn, for example is a fanatical sound man he has his own small studio at home and gets very involved in production sound and the equipment. We put on a children's play of his *Mr A's Amazing Maze Plays* which is all about a child and a dog who go into this house which is like a maze. The set is quite surrealistic with steps, ramps and trap doors, and every time they pass through a different area an associated sound effect is played from the Akai—the only problem is that the audience dictate which direction they go in, so the operator is really kept on his toes. This kind of reaction time is something that would have been very difficult if not impossible some years ago.

'Some directors, such as out MD Richard Eyre, put enormous demands on sound, and use it almost in a filmic sense, continually underscoring the action. We're also seeing directors from all sorts of different disciplines including TV and films where they're used to different kinds of technology and are bringing new ideas with them. Things are moving fast in theatre sound, and it's true to say that sound is no longer the poor relation in theatre production—gone are the days where a play would have the odd dog bark and a carriage departing on gravel.'

Things do certainly seem to have come on a long way in a relatively short period of time, and there's every indication that rather like film sound, theatre sound will continue to go from strength to strength. Credit for that must go to the forward thinking of theatres such as the National, but also to companies like Cadac who have played a key roll in making it all possible. ■

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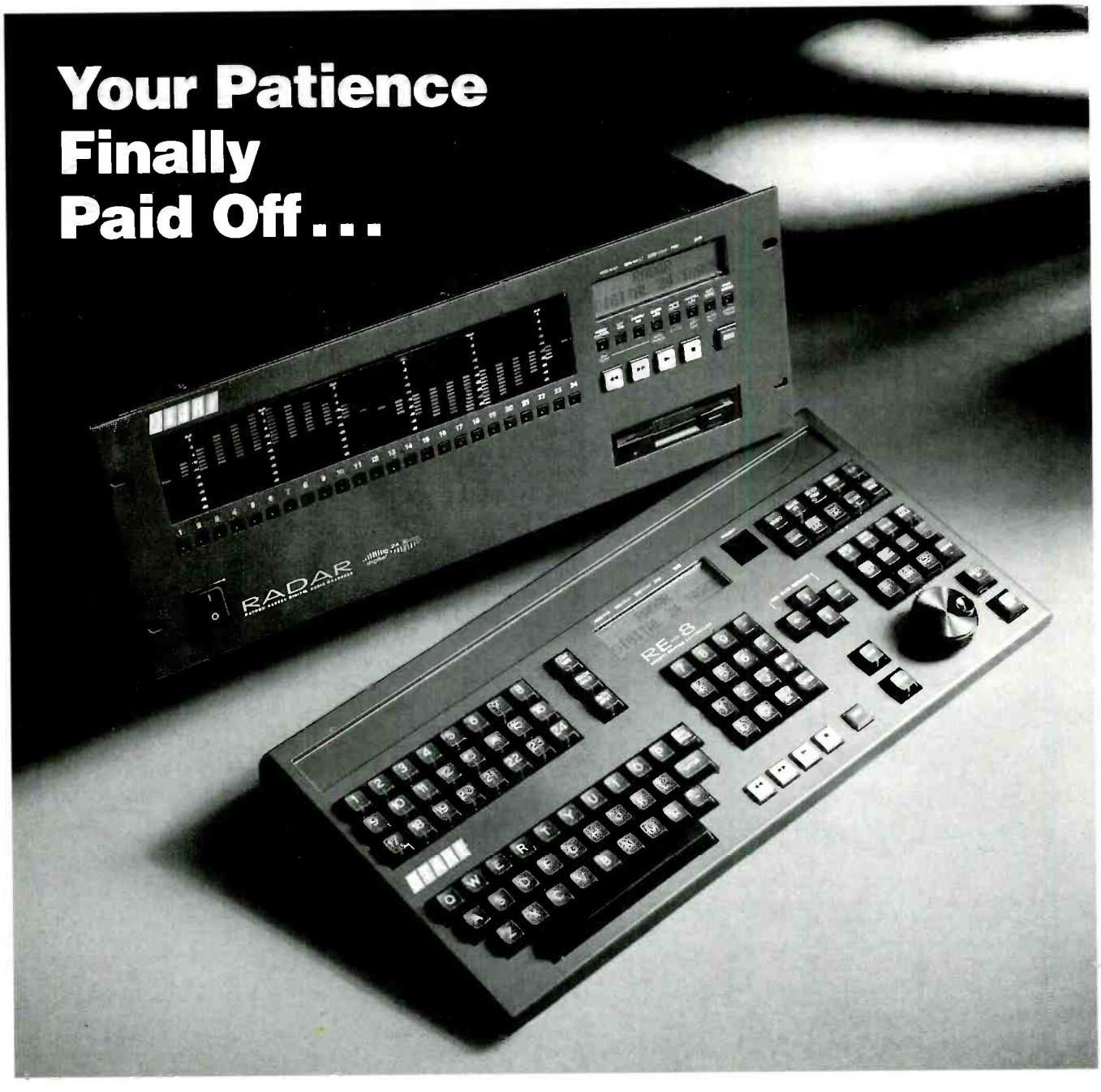

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OR

2. In file manager, select 'RUN' from 'File' menu, and enter 'A: install', then click on 'OK'.

The appropriate files will be installed on the 'C:\drive'. To run the program, double click on the 'PDI EXE' file in the PDI directory from file manager.

LETTERS

EMC: the backlash

Dear sir, EMC has been a well-aired topic for a number of years and there has been a number of sources of widely available information even in the limited PAVI, for those who were open to receiving it. There are a number of points which could be made, many of which I suspect will be echoed by others who work in the EMC field in some capacity.

UK trade bodies such as AORS, SCIF, IABM and others, such as AES, have all contributed to the effort of increasing awareness and inviting participation. Small companies do not have the resource to put aside the effort to make a large contribution to the standards process. Many have written to those notionally elected to be our obedient servants in Parliament (but in practice seemingly more overtly wedded to the party whips and perks of the trade) in an attempt to secure some understanding from government neurones that there were serious issues at foot. Some credit then that the combined efforts helped our DTI representatives to secure a four-year period before which the Directive would be imposed. At the time we are led to understand that Germany was pressing for a period of only two years. So the rate of introduction is not that fast—it's actually twice as slow as Ben indicates. With awareness that should be time enough to evaluate the net present value (NPV) if maintaining each product into the EMC era—it's usually a natural part of product management process.

The OJEC of 25th March 1991 carried a reply to the question of the damage to small firms and to the (then exceedingly short timescale) impending implementation date. M Bangeman said that the commission was aware that the maintenance of the TCF and its necessary third-party certification would involve substantial costs in particular for small and medium companies. The Commission 'asked the European standards bodies to prepare as soon as possible the standards required to enable manufacturers to make full use of the certification procedure for a declaration of conformity to be issued without the intervention of a third party'. There were other comments but if 'despite all of these steps major difficulties are encountered in the application of the Directive, the commission will propose... appropriate measures to guarantee a flexible changeover from the rules currently in force...while ensuring the free movement of the products concerned'. This does indicate that there was a little sensitivity to the problems which were going to be caused throughout the electronics industry?

Some of those taking part in the BSI WP represented the technical interests of UK PAVI companies and although they were rarely the source of any reports on supportive test work they were repeatedly able to reply that their companies could meet the proposed standards though, in some places, work would be needed. It is only recently that an article in which the audio apparatus review featured EMC testing has appeared. The FBA carried out some testing in 1991 using EN55013 and EN55020 as basic standards. It is a matter of interpretation as to whether a hi-fi company should choose to use the Generic route (using EN 50 081-1 and EN 50 08201) when a product-specific family standard apparently exists. The principle problem for hi-fi systems appears to exist in the immunity measurements and provisions exist to justify variation of the relevant test. However, a hi-fi company would be well advised to heed the advice of a competent body as their justification for the choice of whatever standard or test method will satisfy the essential requirements for each of their equipments. This advice would contribute to the potential defence of due diligence in the event of any dispute.

There is nothing new about the LVD. It is an Old Approach Directive. What is new is that the CE mark will indicate that apparatus complies with the relevant standard which, as a recent statement from CENELEC makes clear, will mean that all professional apparatus will conform to IEC65 (BS415:1994 and EN 60 065).

Ben refers to industrial robots being affected by RF fields, so it should not be surprising to learn that proposals are afoot to test motorised wheel chairs for disabled persons in fields as high as 300V/m, for example. The issues of interference are real and the industry which gave us DECT and GSM will also require us to adjust existing EMC standards proposals in order to accommodate such toys. So there will never be a stationary standard and this has been the case all through the drafting of the PAVI standard. Throughout this exercise various informative provisions with the Generic standard have been published as final standards. to the extent that this is possible the PAVI standard has adopted these. You would still have to test these standards even if they are only present in an informative annex in the 1991 version of the Generic standard because they exist now. And there are changes...even now in the EN60 555-2 limit for exclusion from the requirements to meet the mains current waveform distortion have been lowered from 75W to 50W. For conventionally powered audio power amplifiers this implies a limit of around 100W into 4Ω per channel.

CENELEC's five-year phasing-in period permits apparatus declared as confirming to the existence of relevant standards—whether or not they are mentioned in the Generic standards—to remain validly on the market it does mean that new apparatus must confirm to the relevant standards—such as the proposed PAVI family standard—in force at the time of introduction.

Other than an hysterical outburst there is no indication that XLRs will become useless. The issue of audio (and video) cabling practices has been receiving attention for some time in AES standards committees, for example. There is no reason why EMC standards should result in reduced audio or video quality, nor in increased prices, nor in reduced choice nor in radical changes to equipment practice. What is very likely to happen is that audio —and to a lesser extent video—design is likely to improve as designers understand that audio systems have a bandwidth which extends from 0Hz beyond 1GHz. It may take the faddism out of design but will not dent the spirit of a true engineer.

Since the EU wishes to achieve a removal of technical barriers to trade it may have to reckon with the continuing *de facto* imposition of standards in some member states. If you are already trading

with one of these countries then you may already be meeting this standard. Indeed many companies marketing into Germany have been meeting the VDE0871 requirements for radiated and conducted emissions for years as a condition of entry into the market and they will have benefited much from this experience. In principle it becomes an inapplicable standard unless, being stricter than the proposed limits in prEN 55 103 or EN 50 08-1 or EN 55 013, it is imposed as a condition of a specific purchase contract.

The EC Directive 89/336/EEC does not use the word 'reasonable' and, of course, its very terseness is an invitation to bar-room interpretation. However we might role Art 7.3 and Art 10.2 because these appear to state that where a manufacturer applies only part of a standard the attestation of conformity will require a TCF. We might also note that although much reference is given to communications equipment the provisions of Art 4 and Annex III should he noted. The UK DTI have published a number of texts which are intended to clarify these issues. Note, however that the DTI's accessibility in this regard is being changed. Tony Bond, who has held the remit to assist industry with the EMC Directive, is rumoured to be passing on to assist with a new Directive (apparently covering Pleasure Boats) and that his department will be closed in march 1995.

The Directive makes it quite clear that there is not historical right to presumption of conformity and the UK SI 2372 reflects just this in Part III. It is easy to find examples where previous existence is not an indication of compliance—think of PCs in the EMC field and unshrouded valve amplifiers in the safety field. In general SI 2372 makes reference to equipment operating undisturbed and undisturbing in its intended environment and it is incumbent on the manufacturer to state the intended environments for its use.

The only obvious use of the term 'reasonable' in SI 2372 is in Part VII paper 75 under powers of search wherein it is provided that 'A duly authorised officer of an enforcement authority may at any reasonable hour... enter any premises other than premises occupied only as person's residence...' There is a number of provisions in this part which are probably similar to provisions enjoyed by the IRS, Customs and Excise (VAT), and bailiffs. There is the defence of due diligence in which 'it shall be a defence for a person to show that he took all reasonable steps and exercised all due diligence to avoid committing the offence'. From a legal viewpoint the matter is probably cut and dried and it is in the sentencing that any mitigating circumstances would be considered.

There is a hint at a waiver in SI 2372 Part 1 para 3.6 wherein it is suggested that an apparatus shall not be regarded as being taken into service if it is used for demonstration purposes. How long does a dem take?

One of the problems facing test houses is that it is important for their long term business that their testing procedures are uniform. It will not be the test house which will decide to vary the provisions in a given EMC standard though...that decision remains with the person seeking to declare conformity. If you are going to have measurements carried out one important thing to look for in a test house is one which understands the process of making measurements on your kind of equipment.

The UK IEE have been approached with respect to providing advice on professional indemnity insurance, principally for Chartered Engineers. Since the signature on the declaration of conformity is a personal one it might be held that a personal indemnity would be a sage provision and you should really consult the IEE's legal department. As I understand it the indemnity is a general professional one with no special provisions for particular risks. However, you might find it difficult to obtain professional indemnity insurance without the professional qualification.

Allen Mornington-West.
Quad Electroacoustics, UK

Dear sir, Ben Duncan's article on the EMC Directive ('Green New World', November 1994) was well written, although I feel that the 'whinge' level was a bit on the high side.

The EMC writing has been on the wall now for seven years—so why are people only starting to complain now? The fact that trade regulators are taking an interest in the electronics industry is a measure of how mature that industry is becoming, and this can be turned to advantage by those companies who are professional, responsible, and keen to be successful in the future. Who, for example, would wish to turn the essential protection requirements of the EMC Directive on their head and say that it is okay for a product to interfere with other equipment, and be unreliable due to its susceptibility to interference, in its normal operating environment?

It would, of course, be helpful if all colleges and universities prepared their engineering and science students in any way at all for their individual legal obligations under EMC (and Safety) laws, ready for when they leave their ivory towers and to earn a living.

There is an awful lot of misunderstanding about EMC, partly caused by the fact that the most pragmatic course for each manufacturer to declare compliance and affix the CE mark depends very much on the particular circumstances of his products, company and marketplace. What the DTI will judge as a 'reasonable' approach to EMC compliance differs from a multinational making standard products in volume to a small company building custom systems.

What Ben's article does not show is that it is quite possible for every pro-audio manufacturer or systems builder to CE mark their products for a reasonable cost that they can well afford. Stating that EMC testing at a test house can cost from £5,000 to £100,000 without balancing this with the fact that one can also test products oneself for next to nothing, and without pointing out that the law does not actually require any testing to be performed at all, is, I am afraid, scaremongering —even if it is standard journalistic practice.

Ben's article does not also point out the plus side of EMC. It has been well established for many years by electronic engineering professionals that good EMC design actually reduces the development ▶

costs and time-to-market of new products (especially those that mix analogue with microprocessors), reduces installation time and costs, and improves quality and reliability in operation—thereby reducing the costs of field service and warranty claims whilst gaining a better reputation in the marketplace which is subsequently reflected in higher levels of turnover with reduced costs of sales.

For readers who are thinking that this is only possible if you have spare cash in the bank, I would point out that the nontechnical accountant's view of all this is that the 'break-even time' for a new product is reduced dramatically by the use of good EMC practices, so that the costs of finance are reduced.

It is also erroneous to assume that good EMC design inevitably leads to more expensive unit manufacturing costs. Although there are additional costs (such as groundplane PCBs and filters), good EMC design practice usually results in making savings elsewhere. I have seen examples of mixed analogue-digital products which actually cost less to manufacture once they had been redesigned for EMC compliance and CE marking.

One of the problems of EMC compliance, which is also met in other professional industries, is the large numbers of companies providing specialist products and services. Most of the information, seminars, exhibitions, articles, and advertising, are aimed at the majority of the electronics industry (mass produced products), consequently the needs of small companies and those providing custom services are not addressed. I suggest that Trade Associations, or *ad-hoc* groupings of small companies with similar needs, work with carefully-chosen experts in their field to provide 'EMC workshops' tailored to their specific requirements. This could make it possible for such companies to learn which specific low-cost EMC compliance methods will work best for them for a cost of less than £100 per head.

EMC is a new thing for many people. We all hate change, and it is true that there is a learning curve to be overcome. The careful use of EMC consultants who have experience in the pro-audio industry, and who have a healthily pragmatic approach, is the obvious low-cost way to make that first most challenging step. In a year or two we can all be achieving EMC automatically, gaining the financial benefits, and wondering that all the fuss was about. The penalties are real, but the route to compliance is well understood and affordable and the rewards are there for the taking.

Eur Ing Keith Armstrong CEng MIEE, Cherry Clough Consultants, UK

Dear sir, with reference to the article 'Green New World', why is it that the UK press have to emphasise the doom and gloom associated with the European EMC Directive 89/336/EMC and scaremonger, rather than simply give the facts and offer sound advice to the readership. I believe that the article contained a number of inaccuracies and made sweeping statements which painted a picture of punitive legislation which is virtually impossible to comply with both from a technical and a cost aspect.

EMC is not new. BS800 now BS EN55014, the emissions standard for domestic type products, was first published in 1937. The EMC directive was published in 1989 and came into force in 1992 with a four-year transitional period. The DTI have issued a number of documents for public comment, some as far back as seven years ago, culminating in the WS Atkins report in April 1989, so why is it necessary to start kicking up a fuss now? The writing has been indelibly on the wall for a long time.

The author makes the arguments that if the draft PAVI standards, which will be numbered EN55103-1 and EN55103-2 are not implemented in time then the generic standards would need to be used and that these standards are not suitable. The actual methods of testing for any of the above is very similar with references being made to the same, or similar basic testing standards using the same pass and fail criteria. Even so in the absence of the PAVI standards then EN 55013 would be the appropriate standard to apply since this covers amplifiers, magnetic recording and playback equipments, which logically can be extended to cover mixing consoles and outboard equipments under the 'associated equipment' umbrella. Since this is a product specific standard it must be used in preference to a generic standard.

Figures of up to £100,000 for testing a large console and up to £9,000 for a technical construction file are quoted. The only thing that I can do here is to increase my charges to £1,000 per hour. Nice work if you can get it! The directive and the UK law does not require testing to a harmonised standard but merely that a standard is applied. This may be a technical assessment against the relevant standard or some simple testing, with a higher allowance against the required limits to assess compliance. This does give a higher risk element but with lower cost and it is up to the individual manufacturer to assess how much risk they can afford to take. Even when a product is fully tested against a standard at a NAMAS accredited laboratory the results are published with a confidence factor of 95%, so there is still a small amount of risk involved even with full testing at a higher cost. In addition to the Article 7 of the directive says that 'member states shall presume compliance with the basic protection requirements where equipment meets the requirements of harmonised standards'. So again a product could meet a standard but still not meet the requirements of Article 4. Chris Marshman sums it up in the article when he cites clause 88 of the EMC regulation SI2372 which gives the defence of due diligence.

To suggest that manufacturers should put the money away for some serious EMC trouble shooting as when required is in my opinion a ludicrous approach to comply with the directive. Surely it would be better to get as high up the EMC learning curve as quickly as possible and start to build EMC into the product. This from experience will reduce development times and get the product into the market place earlier, increasing the products commercial viability and giving earlier returns on the initial investment. Any company financial director I'm sure would not argue against this approach.

A statement is made in the article that all existing XLRs will become useless because of poor screening. This kind of statement causes far more damage than good. An XLR is a balanced connector and in certain applications will operate quite happily without the screen connected and still meet the requirements.

EMC legislation is daunting, but it is not insurmountable, I've tested equipment which failed miserably, went through a single design iteration and subsequently passed with a lower unit build cost that the original noncomplaint version. How about further articles offering sound pragmatic advice covering legislation, design techniques, low cost in-house test techniques instead of all of this doom and gloom.

Ian Ball, EMC Test Centre Manager, Manchester, UK

Ben Duncan replies

The pragmatic and optimistic opinions of the consultants are most welcome but if EMC is so wonderful, why can it not be a voluntary, prestige matter, like BS 5750 'Quality' assurance? The keynote made by Soundcraft's Martin Reynolds that appears to be being missed is that for manufacturers of purely analogue audio (which covers about 99% of mixing consoles and power amplifiers, and 50% of audio processors in use), EMI (Electro-Magnetic Interference—what EMC aims to prevent) has been for the most part a nonproblem. Being lumped in with computers (surely the most prolific cause of radiated RF hash in civilised domestic and light industrial environments) is a gross insult. The change from almost not having to think at all about EMI matters, to having to learn all about the far broader topic of EMC that is of marginal relevance, is painful to analogue audio manufacturers the world over, and doubly so when the expense and hassle is scheduled with scant regard to the world economy still being deeply in recession. Any 'recovery' stimulated by increased manufacture of EMI suppression components will be a fake.

Another issue is that 'expert' assessment of EMC requirements varies immensely, depending on whom you address. A law that's (presently) interpreted in 11 shades of grey is no way to create a 'level sales field'. It seems likely that companies manufacturing in (or importing into the EC via) the 'more relaxed' southern member states will find compliance is cheaper and easier than it is for manufacturers in the UK and Germany—the only two EC states where EMC is thus far being taken seriously. In Ireland, they haven't even put EMC into law!

For overseas readers, it's worth noting that the apparent pessimism expressed by the UK manufacturers over EMC is based on a full awareness of the standards to be met. It is also instructive to learn that five years or so ago, the Federation of British Audio approached SCIF, as a representative of UK pro-audio, with a view to cooperating on researching how to apply EMC, but in the event, the parties went their own way. Today the UK's volume hi-fi makers (Arcam, Linn, Naim and Quad) are successfully applying the generic standards, a year ahead of compulsion.

Even as I write, the UK's 16-year-old administration whose 'Iron Lady' repeatedly promised less red tape for business, is tearing itself apart over the runaway costs and regulations imposed by the EC's disconnected bureaucracy. ■

Kevin Hilton

The IBC-ITS face-off draws closer while the BBC take a stand over DAB

There was a time when people went to exhibitions to see new technologies, which they would then discuss for the rest of the year, waiting to see whether they would be implemented, and if they were, how they would affect the way business is done. Now it seems that the main reason to go to the 'big shows' is to see how much life is left in such events, or whether they're going to cave in under pressure from rivals.

The suits on the Amsterdam tram appeared satisfied enough with 1994's International Broadcasting Convention (IBC), but we all realise that the real indicators are only going to emerge after this year's double header with the International Television Symposium (ITS) in Montreux.

There is much at stake in this fight, and the ITS people have obviously realised that they need to raise their profile and spread the message—especially when many in the trade, while acknowledging that the location is a beautiful one, think that Montreux is expensive, difficult to get to, and badly laid out in comparison to Amsterdam. The IBC took advantage of their last solo year to mark out their ground, and now the ITS have begun to reply.

I have always felt that the ITS' message has had problems getting out of Switzerland during non-Symposium years. Perhaps to counter such an opinion, the organisation have hired a leading London corporate PR firm to handle the publicity build-up in the UK, which started in December with a fleeting visit from Michel Ferla and Philippe Guillemin, currently joint directors of the ITS.

This double-act—the long-serving Ferla is the fast-talking main-man, Guillemin his almost silent stooge—is to be a short-lived one. After June, Ferla will transfer to the Swiss Tourist Board in Paris, handing over full control to Guillemin, who has been drafted in after stints with both Swiss Broadcasting and Sony to guide the exhibition through these days of convergence.

Or so one would have thought. There is provision for the multimedia future in the symposium schedules, including the event's first joint programme on terrestrial and satellite formats, but it appears that the ITS organisers are working with such new areas at a slower rate than their northern rival. 'Last time we introduced some multimedia coverage, but now we understand this whole subject better,' says Ferla.

Guillemin picks up by saying that it is perhaps too early to talk about a tangible shift away from the domination of traditional broadcast companies and techniques towards a cohabitation with data, computing and telecommunications operations. 'It's happening relatively slowly,' he says. 'They [the computer companies] are coming now but they have to adapt themselves to the market.'

Among the big computing names attending this year is Hewlett-Packard, of whom Ferla says, 'They're coming by the front door this year, not the back door'. At the time of the interview, 248 companies had booked; Ferla was confident that they would reach the 1993 figure of 300 by the time the Symposium opened.

The Big Five, which includes Sony and Panasonic, have again underlined their intention not to attend IBC in September, which confirms the attitude adopted by some other companies, that they would continue to work on an alternating, biennial basis. 'It's too bad that neither organisation found a solution,' Ferla says of the stand-off between the two shows. 'But as far as the industry is concerned, major exhibitors have signed up with us and not with IBC three months later. People like coming to Montreux.'

Which is true—manufacturers, distributors and hacks alike love the place. What they're less keen on is the cost of living and the shortage of affordable hotels in the town itself, which usually means having to stay some miles out in the boonies. (However, at last year's IBC a friend remarked that he was staying in Utrecht, which prompted my response that he might as well stay in Belgium.)

Ferla and Guillemin give assurances that facilities—including catering and information—have been improved, while hotel prices have been frozen at 1994 rates. This may not overly reassure foreign visitors, but the organisers appear sincere when they say they want to avoid delegates being taken for a ride.

All of which may win this first stage of the war with the IBC, but throughout this protracted toss-up between the European pair, nobody has made the obvious comment that both could be circumvented by going straight to NAB (although not necessarily collecting £200).

Any one of this year's big shows could reveal the next turning on the never-ending road that is broadcast technology, but I'd much rather get through the heavy traffic that was created by the last batch before we start on anything new. One of the best hidden exhibits at IBC 94 was the BBC Research and Development department's display of the Eureka 147 Project for digital audio broadcasting (DAB).

The BBC representative I spoke to hedged his bets over the implementation date, first saying September 1995 before going for the more abstract 'end of the year' option. He need not have worried, because at the end of last year, the managing director of BBC Network Radio, Liz Forgan, announced September as the month when it would all start to happen (in the London area, at least).

> Initially, the old terrestrial network will be used to distribute the DAB signals

Just as with digital wide-screen television, the BBC have decided to be the first in the field to state their intentions and commit to the Eureka format. The Radio Authority, who licence and regulate commercial radio in the UK, said that they have no plans at the moment to go for DAB, adding that they knew the BBC were intending to be the first to broadcast radio programmes digitally, but did not intend to spoil the party.

Speaking at a *Voice of the Listener and Viewer* conference during November, Ms Forgan said, 'Today is a significant milestone in the BBC's progress towards a digital world which offers benefits to every listener. DAB is the next step on from the world of AM and FM, VHF, medium wave and long wave—there is clearly an element of risk for the BBC in deciding to be the first broadcaster anywhere in the world to make a firm commitment to launch a DAB Radio service.'

But Forgan is convinced of the benefits of the new technology, specifically near-CD quality, unshakeable reception, and extended services. However, those who are convinced that life in the UK is unashamedly London-centric have more evidence now, as the rest of the country (in effect 60% of the population) will not be able to benefit from these advantages until the end of 1998. Even then, this coverage will be centred around the major towns, cities and motorway routes.

Rural radio fans will supposedly have to keep their trusted steam wirelesses, although the Beeb foresees a large amount of simulcasting in the early days until the number of DAB receivers increases to an appropriate level, or until the retail cost drops to something sensible (whichever comes first). 'The pricing of DAB receivers will affect people's willingness to buy,' Forgan points out. 'Estimates vary, but the first mass production sets are likely to cost several hundred pounds. As with CD players and satellite dishes, prices will fall rapidly.'

The BBC's Engineering Information Department estimate that there will be 27 DAB transmitters in the UK: five around the capital (Crystal and Alexandra Palaces, Guildford and Reigate, and Bluebell Hill), with the rest in the major metropolitan areas and regions (Bristol, Cardiff, Belfast, Leeds, Scotland, and the North West and North East). Initially, the old terrestrial network will be used to distribute the DAB signals, but it is hoped that satellite will be brought in for the later phases to make the whole project more economic.

In this way, the Corporation intends to squirt its five national channels—Radios 1, 2, 3, 4 (stereo services) and Radio Five Live (the mono news and sport channel), plus additional sports and Parliamentary coverage—down a single frequency. This will, eventually, create more space on the existing FM frequencies, improve reception quality, and make tuning round the bands easier for those who regard the selector switch as an invention of the Devil.

Unfortunately, it is unlikely to make Radio 1 DJs sound more intelligent. But then technology can only be expected to do so much, can it not? ■

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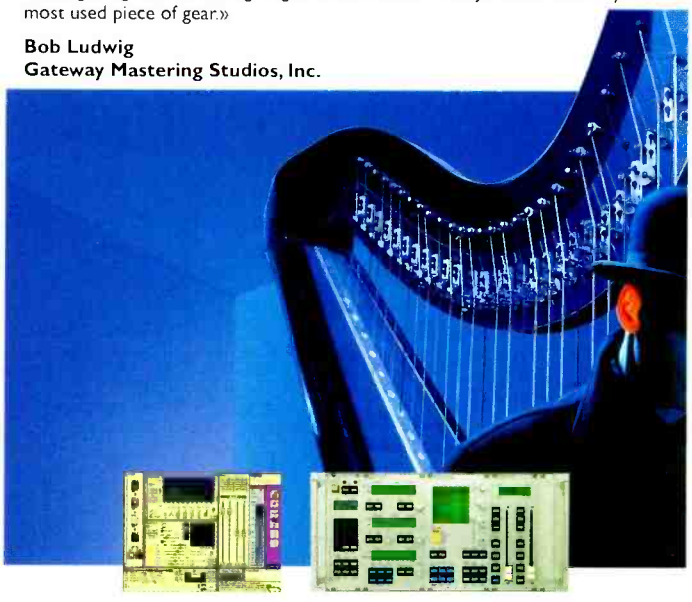

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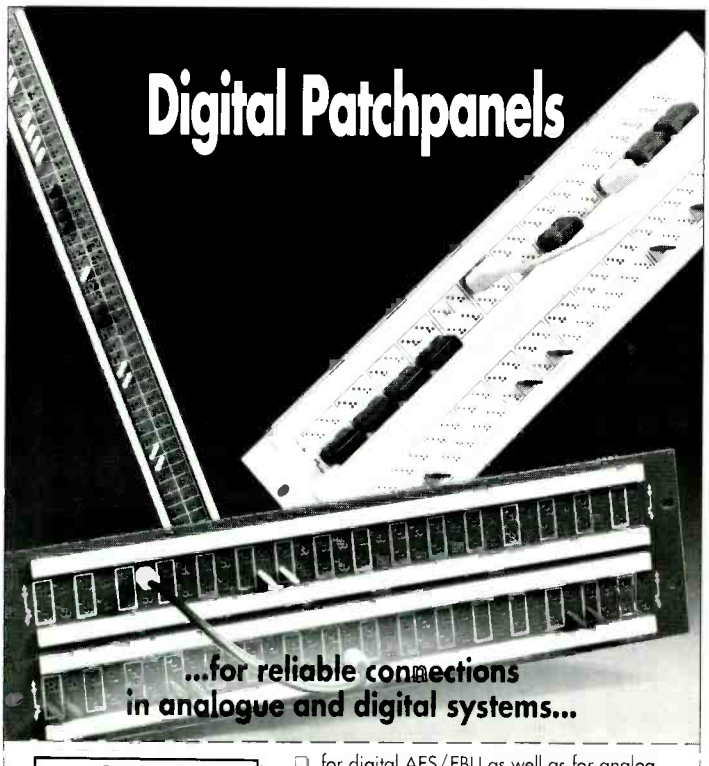

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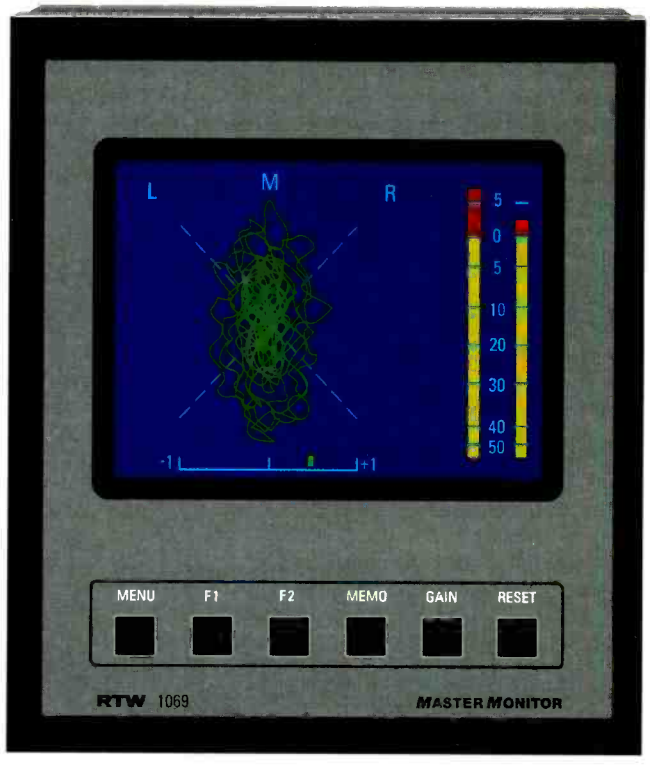

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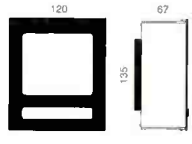

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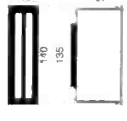

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Martin Polon

Repent, repent, repent, re: Pentium

Mainstream recording studios, postproduction facilities, sound reinforcement sites, electronic music labs, personal and project studios, broadcasters, schools and so on have come to depend upon personal computers for a wide range of applications. These might include audio recording, audio production and postproduction, audio facility management and equipment test and repair functions. Virtually, no facility does business without a computer in at least one of the above functions and many have them in all. The current trend towards the 'tapeless' studio with computers recording to hard drives shows no sign of abating. Yet there is a willingness to accept a personal computer as a flawless device when any number of different kinds of audio equipment that might be purchased will be subject to the most rigorous scrutiny.

The current flap over the reliability of computers in general and the microprocessor chip known as the Intel Pentium in particular, therefore has a large number of ramifications for the audio studio and recording industry as well as the other areas of endeavour in the audio business. The Intel Pentium processor—the fifth branch of the X86 chip family tree, that has been powering the so-called Personal Computer from the beginning—has at least two critical design flaws. The major flaw involves the absence of correct binary reference material in 0.33% of the 1,500 locations in the division look-up table. The table is consulted by the Pentium's floating point unit when the result of division is not a whole number and the appropriate binary equivalent is desired. This can reduce the accuracy of division from 16 places to less than five places. This disparity of 11 digits produces an error—either as a wrong answer or potentially as an incorrect operation or output to a specific piece of software.

The other flaw is minor by comparison and involves an occasional problem with the onboard memory cache. Generically, this is not an easy problem to solve in either case since the *raison d'ëtre* of the Pentium family is speed and power, and the application of patches to fix both flaws slows it down considerably. The greater problem remains the slavish devotion to 'chips' without fully comprehending just what the devices are—especially when served without the 'fish'.

The family of Intel microprocessors, which began with the 8086 chip in the first IBM-made machines produced at the beginning of the 1980s, continued in the mid-1980s with the 80286 and the 80386. There were bugs in these chips as well, though not of the magnitude of those found in the Pentium. The 286 flaw involved memory access problems to other chips and the 386 could not always access certain 32-bit instruction sets properly. These were promptly fixed and did not involve a large number of chips. Intel moved to the first 80486 chips that went into the more advanced PC models at the end of the 1980s. Then the 486 chip, in early yields, had a heat problem. Again, these problems were fixed promptly. In 1993, the 80586 reached maturity and was labelled by Intel as the Pentium.

Estimates of five million Pentium computers to be sold by the end of 1994 suggest that many audio businesses have these machines. In fact, if a studio or post house is determined to use *Windows* software to record and edit and manipulate audio, and-or to manage finances, the Pentium would be the platform of choice in terms of speed and computing power.

The direct threat of the Pentium's failing is that it can deliver incorrect answers where floating point math is being used and would be most disruptive to financial spreadsheets and scientific packages used for the design of amplifiers, loudspeakers, digital equalisers, other chips, and acoustic spaces. How many functions of hard drive recording through the computer are impacted by the flaw remains a mystery. And what artefacts the flaw might leave in other audio uses is a greater mystery yet.

Intel executives decided to keep the major math flaw a secret from their customer base. They did this by refusing to acknowledge it for a number of months—six months by most accounts, though some suggest for a year or longer. The company finally admitted the occasional division inaccuracy in the Fall of 1994 after being embarrassed on the Internet by a university professor who had uncovered the problem.

The company continue to claim that the mathematical division problem could impact only a tiny percentage of the Pentium's millions of customers, based on their usage patterns. Further, Intel claimed that the problem would strike the 'casual user' only once in 27,000 years or if you prefer a one in 9bn operations occurrence rate. IBM among others, violently disagreed with the Intel error-rate assessment for the Pentium and decided to cease production of all Pentium models—at least, until the problem is fixed. IBM also publicly stated that the flaw can occur for some users as often as once every 24 days. Computer industry experts in print and on the Internet have agreed with IBM's assessment. However, Intel's competitors may be using this scare to their own ends.

And what artefacts the [Pentium's] flaw might leave in other audio uses is a greater mystery yet

Intel's response to all of this was to agree, just prior to Christmas, to exchange the Pentium chip for any current or past customer who requests the exchange. This offer is to be available for the life of the Pentium-powered machine, so that the user-base will not have to make the exchange in a limited time period. In Intel's defence, the task of checking the Pentium's 3,300,000 transistors in every possible combination of software and applications is virtually impossible, with potentially 5m–6m 'sites' to check on the chip alone. Newer, more powerful chips will be even more complex and that much more difficult to purge of flaws. The time to stop chip 'bugs' is before the prototype microprocessor is setup for the 'mask'. Once the chip gets to the point of the manufacturing die, it is a lengthy and time consuming process to recast the chip.

We must also remember that the computer industry does not take itself as seriously as the outside world does. Consider the numbering system that it uses to identify its hardware and software products; this allows relatively constant adjustment of product version identification to accommodate updating to remove bugs. Suppose a company produce v1.32 for some product—if they start with v1.0, then we have moved through three relatively major fixes and two minor ones. So if one stated that bugs were to some extent endemic to the computer industry, they would not be too far off the mark.

What alternatives are there? There are chip vendors such as AMD, Cyrix and NexGen who plan to offer Pentium-style super chips of their own in 1995. There are 100MHz 486 chips in the offing from Intel and IBM, AMD, and Cyrix. Then there are the Apple *Macintosh Quadra* computers which do not use Intel chips, but run with Motorola 68040 processors, and the emerging Power PC chips from the consortium of Apple, IBM and Motorola. Computers based on the Power PC are already available from Apple Computers with IBM's models available in 1995.

What should the audio industry do about the Pentium problem? If the user is comfortable with past performance of the Pentium-powered computer, then the need for exchange of the chip remains an open-ended option. By leaving replacement of the chip up to the user, Intel took some of the pressure off of the major PC makers who did not terminate production of Pentium systems with the flaw. In fact, the supply of corrected Pentium chips may not reach the customer base until spring 1995.

If there is a message here, it is for computer users to recognise that computers are not infallible. Audio users need to be more vigilant in monitoring their computers' performance, filter positive or negative comments on the effectiveness of a chip and its associated computer system from the special interest groups on the Internet, from magazines, from newspaper columns, from computer user-groups, from word-of-mouth or whatever.

One can only hope that both the audio industry and the computer industry will learn from these problems but it just may be the price of technological progress! ■

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THE LAST BARRIER?

With all the other audio components reaching a refined state, John Watkinson argues that the last unconquered technology is the loudspeaker. Is there such a thing as the ultimate loudspeaker?

Beliefs or facts? The last few years have brought about a radical redefinition of the limits of audio quality. I remember a time when virtually every component in the audio chain was capable of contributing its own audible impairment and an essential ingredient in achieving quality was a degree of determination —and plenty of time to adjust recalcitrant hardware to the finely balanced point from which it would soon drift. The quality was never good enough and the ethic of the time was a constant struggle for improvement. The results are now all around us. Looking at a typical modern audio chain, we have a microphone feeding an A–D convertor, connected to a digital recorder, playing into a D–A convertor, driving a power amplifier connected to a loudspeaker. We could chuck in a mixing desk (analogue or digital) at the appropriate point to complete the picture. As audio is a chain, the weakest link determines the quality.

Is this the microphone? Modern microphones have a frequency response like a ruler, a frightening dynamic range and as much linearity as you want. Is it the A–D or the D–A convertor? Again, early devices were imperfect, but modern units using noise shaping and oversampling with 18 and 20-bit resolution are outperforming our ears provided some attention is given to clock jitter.

What about the digital recorder? Well, using the digital I-O—provided this does not use compression—it does not have a sound quality. Numbers coming in are the same as numbers going out, unless it is broken.

The mixer, then? Not guilty; modern digital and analogue mixers are so transparent they are basically not there.

Moving to the power amplifier, today's power amplifiers can be so good that they approach Peter Walker's 'piece of wire with gain' criterion quite well. Further developments in power amplifiers will not be so much in the area of sound quality, but in fields such as efficiency and the friendliness of the load presented to the AC supply. The obvious advantages of switched-mode power supplies are starting to be applied, but the switched-mode amplifier is also attractive because it is efficient and cannot suffer crossover distortion. An obvious application is in LF driver amplifiers.

Responsibility for the majority of impairments in today's reproduced sound has to be laid at the door of loudspeakers which have not seen the dramatic quality leaps of other components. The logical conclusion is that loudspeakers are now causing a quality bottleneck. Anywhere a bottleneck exists it is logical to search for a solution because the rewards are likely to be significant compared to the effort, whereas in more mature technologies the returns diminish as the ideal is approached.

Any designer hoping to improve on the existing art must not only understand the physics of the processes taking place but must also understand the human condition and how popular perception can be driven a long way from reality. The laws of physics cannot be changed, unless we prove conclusively that they are wrong. Fortunately, the laws of physics which are involved in audio reproduction are sufficiently well established that they are only called into question by hi-fi journalists.

I should perhaps make it quite clear that I am interested in precise sound reproduction rather than hi-fi. There was a time when the two were synonymous, but nowadays in many respects hi-fi has become a religion in which beliefs are more important than truths. The temples of hi-fi are the phenomenally expensive hardware installations and the high priests are journalists who find pseudoscientific reasons to make the believers feel comfortable with the vast sums they have spent. Iconoclasts like myself find occasional pleasure in attending their sermons and suggesting that they should try triple-blind testing.

It is impossible to make other than accidental or empirical progress without a clear picture of the processes involved and an understanding of the key criteria. Thus in attempting to determine what part of one's knowledge-base can be trusted it is necessary to remove from it all of the myths and pseudoscience and to establish what is and is not the case. It is surprising how long this takes if one is to be impartial and scientific about every spurious theory.

Psychoacoustics is one vital area of knowledge without which it is impossible to weigh the merits of differing approaches. The human hearing system is complex and highly sensitive in some areas, yet surprisingly casual in other areas. Clearly design effort has to be extended in areas of sensitivity, while shortcomings are concealed by placing them in other areas.

At the end of the day the criteria for audio system quality can only be subjective. Audio systems act as a window between the listener ▶

LOUDSPEAKER PERFORMANCE FACTORS

- **Max output SPL required:** High output precludes use of electrostatic transducer, moving coil may have heating problem if inefficient, horns more efficient thus advantage for very high SPL.
- **Frequency response:** Difficult to reproduce extreme LF. Smooth non-flat response better than flat with dips. Response affected by siting and difficult to measure. On-axis response meaningless without polar diagram.
- **Linearity-Intermodulation:** Critical quality factor. In moving coil caused by motor nonlinearity and cone breakup. Electrostatics very good here. Low distortion at LF difficult owing to pressure generated in cabinet.
- **Polar diagram-directivity:** Important for stereo imaging at least in mid range. LF tends to omni response. Difficult to maintain polar diagram at crossovers with spaced drivers. Co-axial m/c and electrostatic very good here.
- **Cabinet flexing:** Destroys clarity, causes distortion at LF. Should be designed out with rigid structure and materials. Mass is often used instead of rigidity, but speakers do not have to be heavy: it is rigidity which matters.
- **Cabinet diffraction:** Damages polar diagram and confuses stereo image with false sources. Caused by large cabinets with sharp corners.
- **Damping factor:** Critical if amplifier is to control diaphragm velocity. Easily impaired by passive crossovers. Active crossovers and one amplifier per driver for highest quality. Integral amplifiers help.

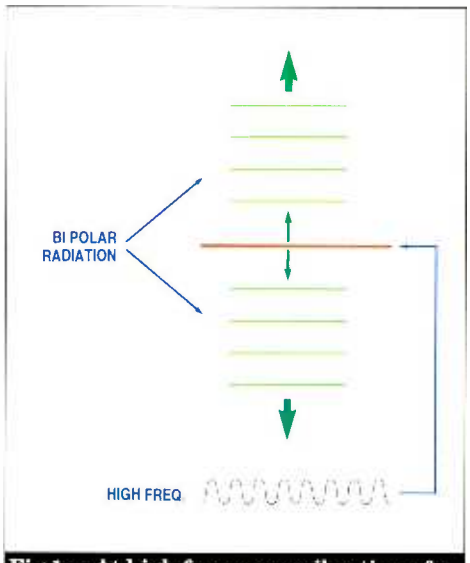

Fig.1a: At high frequency vibration of a diaphragm produces bipolar radiation, whereas low frequencies result in short circuit as in fig.1b

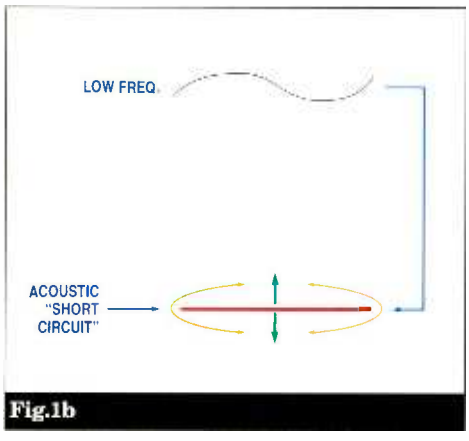

Fig.1b

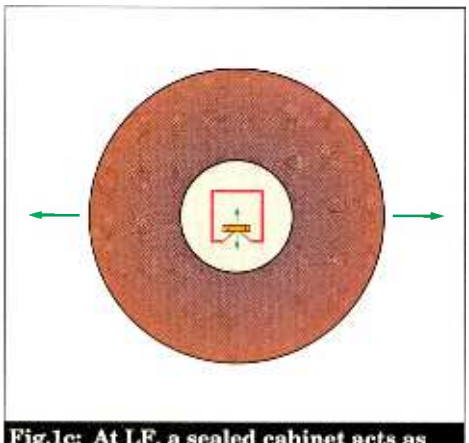

Fig.1c: At LF, a sealed cabinet acts as an omnidirectional source irrespective of the geometry. SPL is affected by surroundings; nearby walls will push up the level

and the original sound. One of the goals, then, must be to make that window larger than the sound passing through it. If human listeners are unable to detect an impairment, then the quality is sufficient and the window is big enough. Making it even bigger simply drives up the cost.

In any audio work—and particularly in loudspeaker work—listening tests are vital once all objective tests have been passed. However, in order to be significant, such tests have to be properly conducted to avoid bias. This author is capable of listening to a loudspeaker as well as anyone, but unlike many, does not consider himself competent to do so alone. This is simply because the spread of human hearing performance is so great that I cannot be truly representative. I will naturally listen to my own creations more favourably than those of competitors.

Listening tests have to be carried out with care. Neither the operator nor the subjects must not be aware of the reason for the tests, and the design of the tests must be approved by a statistician who can determine how likely it is that identical results could have been obtained by chance. My incompetence should now be clear. I can only listen to a loudspeaker of my own design to ensure that it has no obvious warts, but to compare it in any significant manner with another speaker of similar performance is beyond me.

The problem

The greatest problem of loudspeaker design is the range of frequencies involved, or more precisely, the range of wavelengths that such frequencies possess in air. These range from a few millimetres at the highest audible frequency to several metres at the lowest. There cannot be many disciplines in which mechanical motion is required over such an octave range. Wave theory is dominated by the relative sizes of the source and the wavelength. Thus in a loudspeaker at the highest frequencies the transducer is much larger than the wavelengths, whereas at the lowest frequencies it is much smaller.

Fig.1 shows some examples. When a plane diaphragm transducer is much larger than the wavelength, as shown at **(a)** it tends to produce plane waves which are directional. In the case of an unenclosed diaphragm, a bipolar response is achieved in which the front and rear radiations are identical but in antiphase. Directionality rises with frequency and the result is that the highest frequencies can only be discerned directly on axis. When the transducer is small with respect to the wavelength, the air finds it simpler to move from one side of the diaphragm to another, effectively short-circuiting the system as in **(b)**. The air in the near field may be moving quite fast, but it cannot affect the far field. This near-field effect is used to obtain LF response in supra-aural headphones.

In order to allow a diaphragm to generate low frequencies, it must be provided with an enclosure which prevents the acoustic short circuit. As **(c)** shows, provided the wavelength is larger than the enclosure, the resulting radiation will be omnidirectional and the result will be exactly the same as if a pulsating sphere had been used. This is unavoidable as it is simply not possible to produce a bipolar sound-radiation pattern at low frequencies without a diaphragm several metres across. Even if this was done, the presence of an ▶

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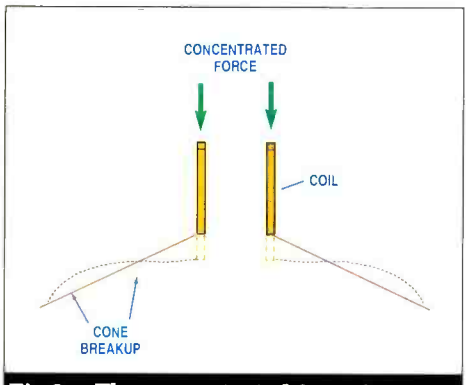


Fig.2a: The concentrated force from a moving-coil motor may distort the cone

enclosing room would destroy the bipolar characteristic. Our ability to localise sound sources is also quite poor at such frequencies.

Air is not very dense and as a result it is not possible to influence very much mass at once. Thus it is difficult to radiate energy into air with a mechanical device because the mass of the moving part of that device will eclipse the mass of air influenced. In engineering terms a diaphragm has a high mechanical impedance but the air has a low impedance, resulting in a mismatch, meaning that it is doomed to inefficiency forever. The horn loudspeaker is a kind of acoustic transformer which raises the impedance of the air adjacent to the diaphragm in order to improve the power transfer. Unfortunately, acoustic transformers are difficult to make linear and the resulting distortion is difficult to eliminate.

Thus, for high quality applications, we are left with the direct-radiating diaphragm. Efficiency can be raised by reducing the mass, but only if the stiffness can be maintained. New composite materials may help here, especially in moving-coil designs where the driving force is concentrated at the coil, but has to be spread over the air load on the whole diaphragm surface. **Fig.2a** shows that it is easy for the diaphragm to break up and fail to represent a piston.

The alternative is an approach such as the electrostatic loudspeaker shown in **Fig.2b** where the diaphragm does not need to be rigid because it is driven uniformly. As a result it can be lighter with corresponding benefits in efficiency and transient response and freedom from intermodulation distortion. The electrostatic diaphragm is supported between two driving plates and the spacing is a compromise between the amplitude of motion possible and the drive voltage needed. Thus electrostatic transducers are not capable of extremely high SPLs. They are invariably used in bipolar mode without a cabinet such that they suffer an acoustic LF roll-off. Their operation is enhanced greatly if they are driven via a high-pass filter which prevents them being driven with frequencies they cannot reproduce. Such frequencies generate large diaphragm excursions and if they are eliminated the transducer can generate a higher SPL over the range it can reproduce.

For optimal reproduction of high frequencies, a loudspeaker should be physically small in order to avoid excessive directionality. For stereophonic listening, the polar diagram of a loudspeaker is critical, probably more critical than the frequency response. In the case of the most spatially accurate stereo, using coincident microphones, the stereo illusion is obtained by amplitude differences at the loudspeakers which are converted to phase difference by the geometry of the space between the speakers and the two ears. If the polar diagrams of the loudspeakers are poor, the relative amplitudes of the sound from the two channels will only be correct on a line joining points equidistant from the speakers, resulting in a 'sweet spot' where the stereo image is alone audible.

In order to prevent the stereo image moving with frequency, the polar diagram should be relatively independent of frequency, at least over the frequency range where localisation is strongest. In a moving coil loudspeaker, the outermost sections of the cone may be mechanically decoupled so that the radiating area becomes smaller as frequency rises. In the Quad *ESL-63* electrostatic the diaphragm is formed electrically from concentric rings. A crossover network contains delays which make the concentric rings act as a phased array. This simulates a point source having an advantageous polar diagram.

With a traditional approach, the optimal reproduction of low frequencies requires a physically large loudspeaker. The mass of the diaphragm and the stiffness of the air in the enclosure behind it form a resonant system, as **Fig.3** shows. Below the resonance there is little output and so the lower the resonant frequency the better. The smaller the cabinet, the higher the stiffness of the air within, and the higher the fundamental resonance. Also the internal pressures generated rise with small cabinets, resulting in a large force on the diaphragm and an increased likelihood of breakup. The resonant frequency can be lowered by raising the diaphragm mass, but that reduces the efficiency too, causing a coil-dissipation problem. The force on the diaphragm can be reduced by using a smaller diameter, but then the throw has to be increased, increasing distortion. Thus if a good low-frequency response and low distortion are required at a reasonable SPL, the traditional loudspeaker has to be large.

This conflicts with the requirement stated above that the loudspeaker has to be small for good HF performance. It is inevitable that a quality ▶

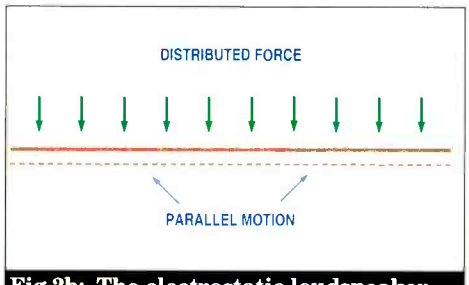
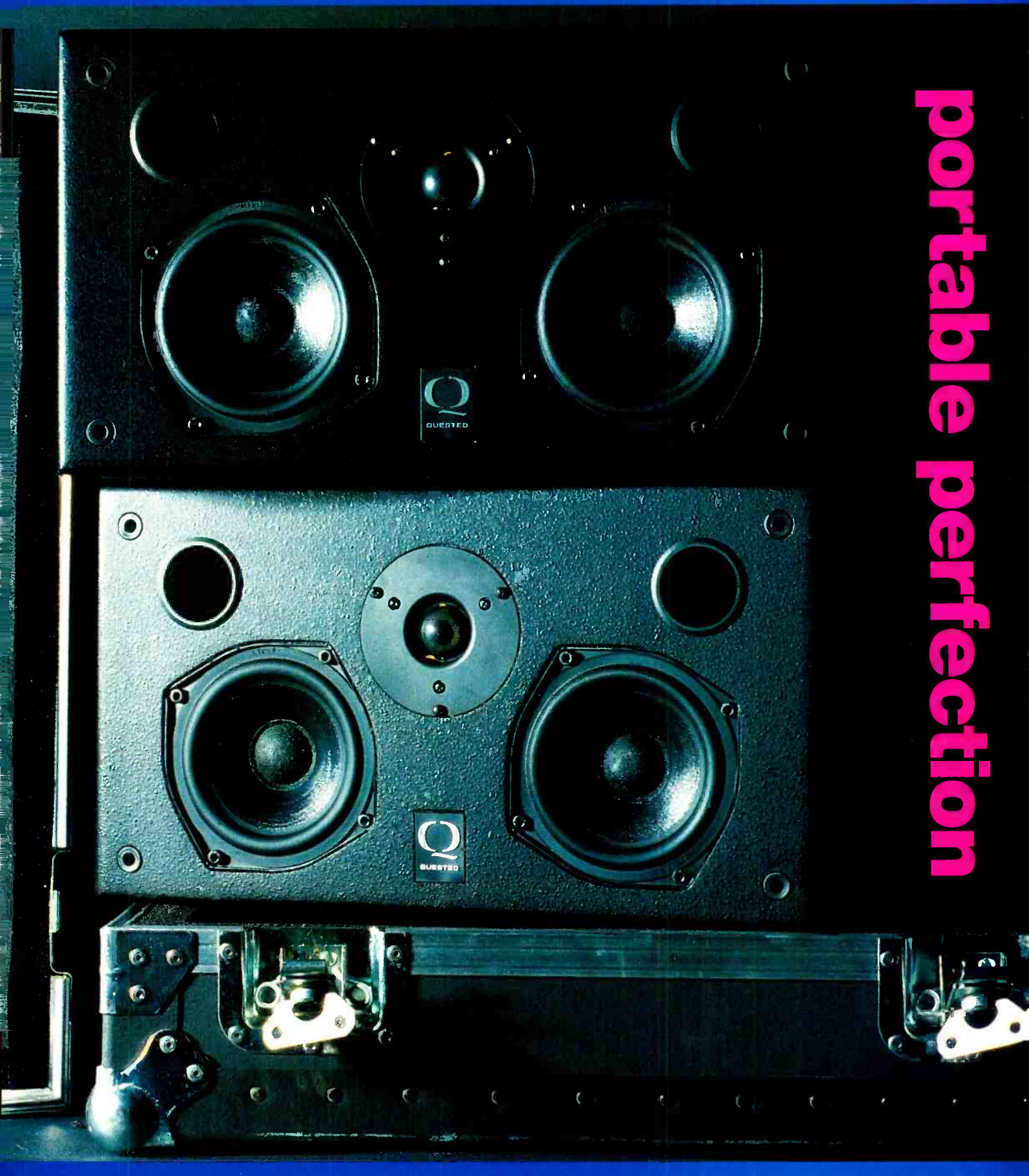


Fig.2b: The electrostatic loudspeaker produces uniform drive over the surface of the diaphragm so there is no tendency for it to break up

Q205



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loudspeaker must have at least two transducers, optimised for the different conditions at the opposite ends of the spectrum. Some crossover system is necessary to distribute the signal to the appropriate transducer.

At the crossover frequency two transducers are sharing the radiation and so their relative location is important so that a coherent wavefront can be launched. (see **Fig.4**) If several drive units are employed, they should be one above the other in a vertical line. Any other arrangement gives a poor horizontal polar diagram and consequently poor imaging. A two-way system will give better imaging than a three-way system because the single crossover frequency can be placed below the mid-range critical band and then all direction sensitive signals come from a point source. In this respect a single electrostatic driver combined with a moving coil LF unit has many attractions. The electrostatic unit is capable of acting as a point source from a few hundred hertz up and has attractive imaging characteristics. The LF driver relieves the electrostatic unit of frequencies where it is inefficient and allows the diaphragm excursion to be used to produce higher SPL. The combination of a bipolar electrostatic unit with an omnidirectional LF unit requires some care with time alignment so that at the crossover frequency the overall polar diagram makes a clean transition from bipolar through cardioid to omnidirectional.

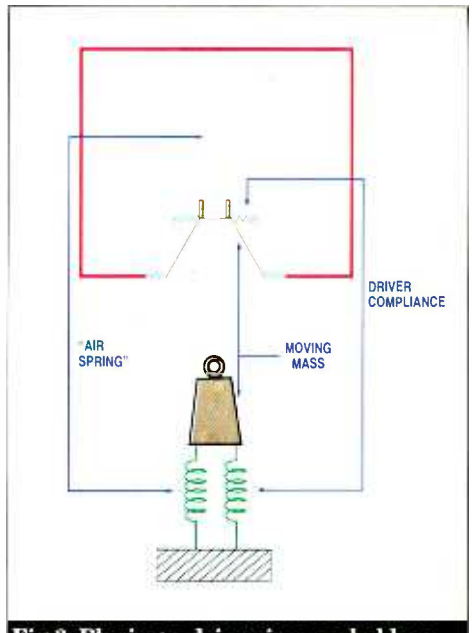


Fig.3: Placing a driver in a sealed box prevents the acoustic short circuit of Fig. 1b but the trapped air volume acts as a spring in parallel with the driver suspension. The fundamental resonance is raised. Driving the cone against this air spring can cause cone breakup in small enclosures and results in a phase lag if the damping factor is poor

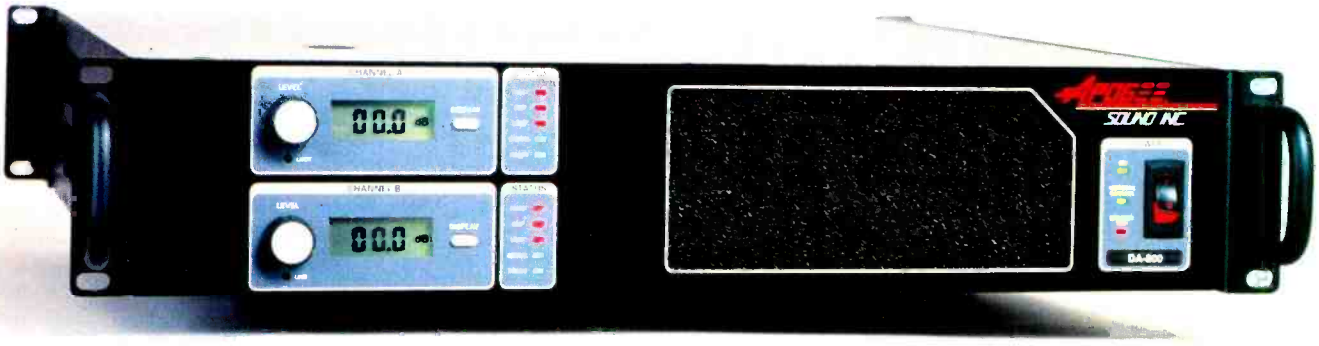
In more conventional approaches, two extreme types of loudspeaker exist, with a range of compromises in between. The first type is extremely large, powerful and with an excellent low-frequency response, but the high-frequency imaging and polar diagram are spoiled by the HF units being set in a large front face. Unless very heavy or made of exotic materials, cabinet resonance is likely to be a problem.

The second type is a compact unit with excellent imaging and polar diagram, free from cabinet resonances, but unable to produce much LF.

A good compromise is to use a tapering construction in which the low frequency volume is primarily near ground level and the HF units can occupy a slimmer cross section at the top. With a sufficiently low crossover frequency, a detached subwoofer can be used.

In strictly theoretical terms, a low frequency loudspeaker only needs to be able to displace a sufficient volume of air to achieve the required SPL, and this has nothing to do with its cabinet volume. Thus in principle at least, a small LF loudspeaker is possible, but this will not be based on the conventional approach. If it can be built, a small LF loudspeaker would solve most of our woes because the problems of cabinet stiffness and imaging are then eased. Cabinet stiffness problems are also eased by the availability of composite materials which are phenomenally stiff yet easily mouldable into complex shapes. The ease with which arbitrary shape can be produced means that cabinets can be made in which attention has been paid to edge diffraction. One development which would transform loudspeaker design would be a superconducting material ▶

Digital Power.



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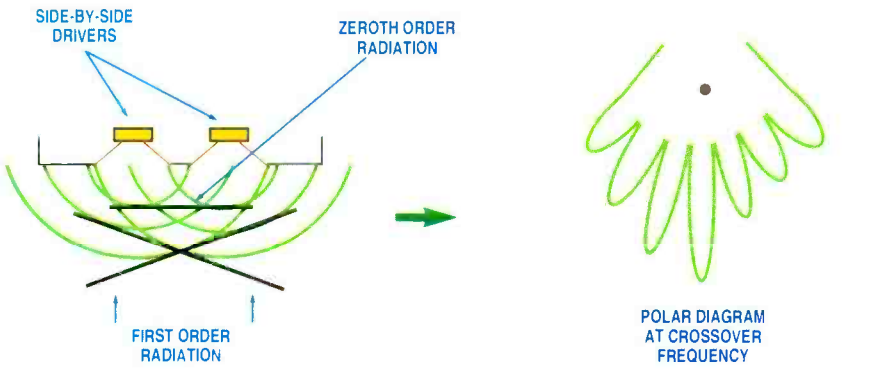


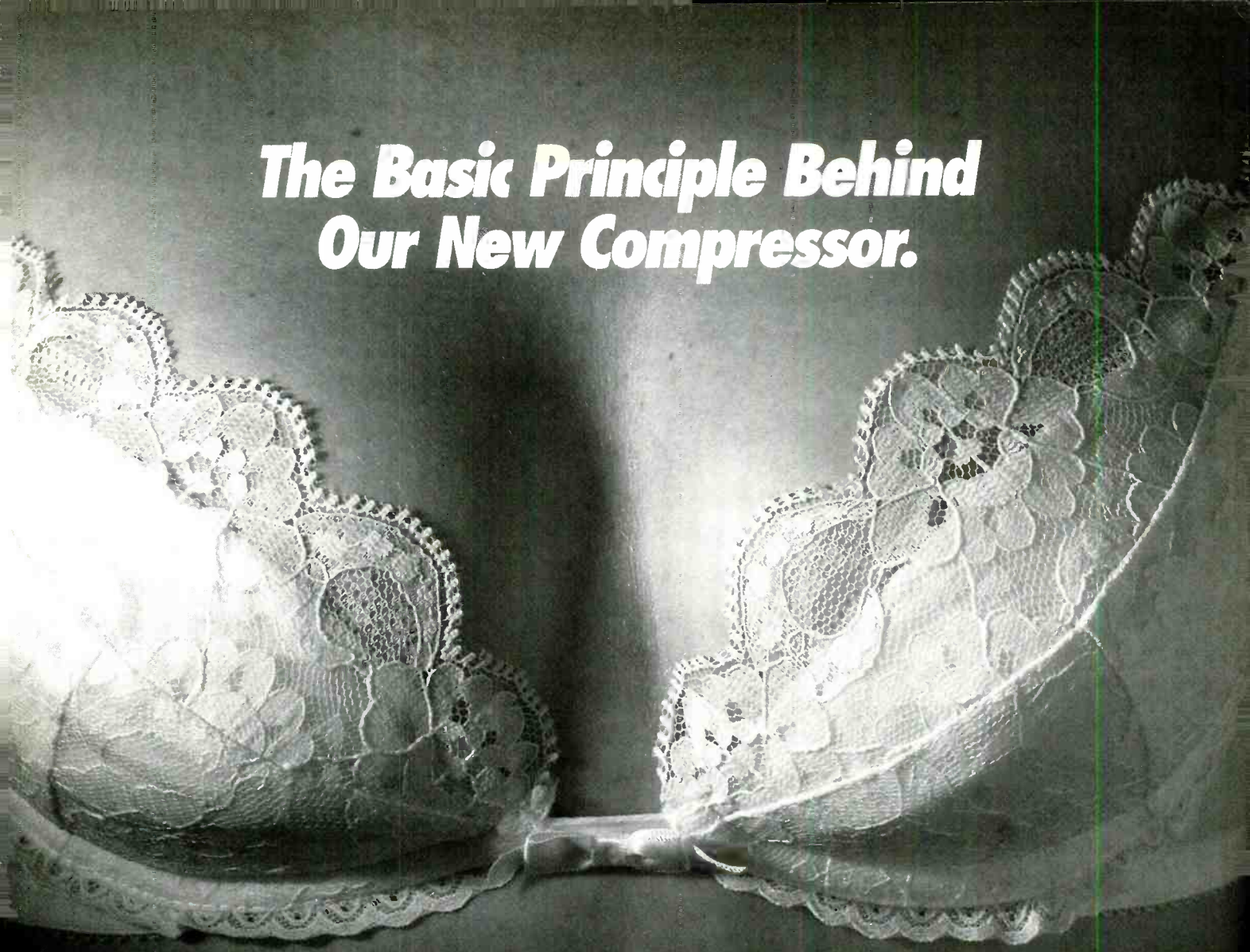
Fig.4: If drive units in a multiway system are wrongly placed side by side, at the crossover frequency when both units contribute to output there will be interference between them resulting in a spiky polar horizontal diagram which impairs the imaging

from which to make coils. This may, however, be some way off.

One approach to improving loudspeakers is to treat the amplification, crossover and transducer stages as part of a single system having an overall transfer function. When this is done, a great many new avenues open. The tradition of building general purpose amplifiers which are remotely sited from passive loudspeakers built by someone else has little to recommend it. The tradition of testing amplifiers into resistive dummy loads is still quite strong, probably because these are easier to drive than real loudspeakers!

One engineering tenet which is seldom broken with impunity is to put the power source near the load. Loudspeaker drive units should be electrically close to the amplifiers which drive them. To take the example of a moving coil unit, this is basically a linear electric motor in which the cone velocity should be proportional to the applied voltage. The difference between the back EMF, which is an indicator of actual velocity, and applied EMF determines the current in conjunction with the resistance in the circuit. In LF units there is appreciable cabinet pressure to overcome as the cone deflects, and this slows down the cone, reducing the back EMF. Unless there is tight coupling between the coil and the amplifier, there will be too much resistance for a correcting current to flow. The cone velocity and phase are no longer determined by the applied voltage. Long leads and passive crossovers, particularly the latter, prevent tight coupling. The only solution is to use one power amplifier per transducer with the crossover function performed at signal level prior to the amplifiers. Power amplifiers are so cheap today that there is little excuse for any other approach. Another advantage of integrating the amplifiers into the loudspeaker is that the mythology of loudspeaker cable audibility is neatly sidestepped.

Passive crossovers also suffer from sensitivity to the load they drive and if the load is a reactive transducer having a resonance the frequency and phase response may be quite different from the response driving a resistor. Crossovers implemented in active circuitry at signal level need not suffer this failing. ■



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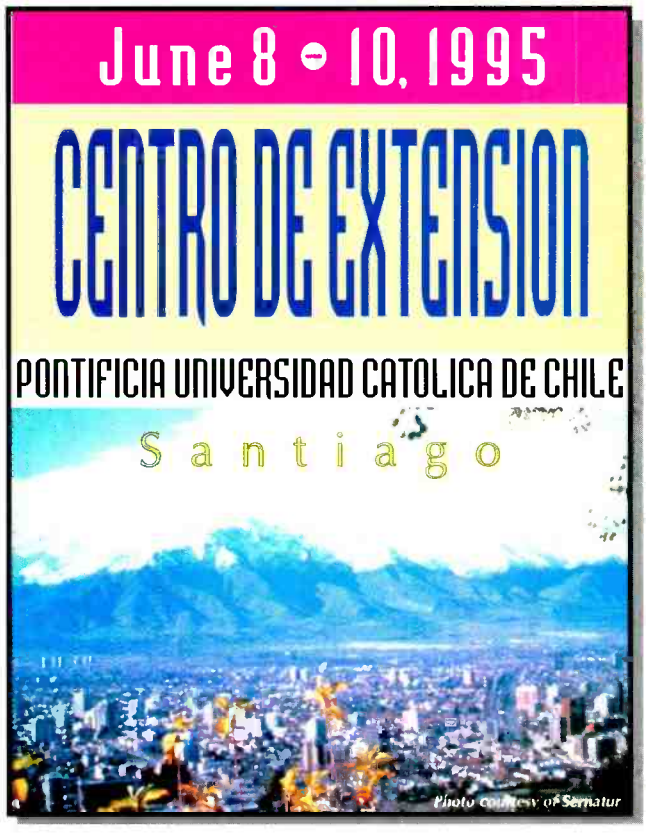

South American countries are now joining the international pro audio market. The standard was set at the First Annual South American Pro Audio Expo in June 1994 in Buenos Aires, Argentina. The next host to the South American Pro Audio Expo is appropriately, Santiago de Chile. Santiago boasts South Americas strongest and most vibrant economy. Chile is a leader in the education of sound and acoustic engineering, offering four specialized Universities. Chile, as well as other South American countries, will now continue to have a unique annual hands-on experience with the latest in sound technology. We look forward to having you and your company be a part of the growing Pro Audio Marketplace in South America.

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A NECESSARY EVIL?

The use of sampling frequency conversion is on the increase, yet little is documented about its sonic consequences. Sony Classical's David Smith goes on record

The concept of time-invariant sampled systems as opposed to analogue working, explained in the 1920s by Nyquist and later elaborated upon after the Second World War by Shannon, is a very good example of practicality trailing far behind thought. So great was the trailing distance in this case that these systems could only be created mechanically at first, and by Shannon's time, only with the extended application of valves and even then, for only short periods of time. Analogously in this discussion, we could consider the special case of the sample-rate convertor (SRC). Sample-rate conversion seminally appeared in the literature in 1968, as suffering a similar fate. Sample-rate convertors were 'laboratory curiosities' in that they and their applications were poorly understood if at all, not to mention the fact that they ran on mainframe computers and certainly not in real time. Their early use in multiplexed telecommunications combined with the advent of consumer digital audio late in the 1970s began a rethinking of their role in the larger picture of things, a role that has become directly related to certain characteristics of their implementation as we shall see.

This subset of signal processing can be accomplished via the process of continuous time digital-analogue-digital conversion. However, an argument does exist in 1995 for considering it entirely as a discrete-time digital process accomplished via elaborate digital filtration. It might be said that the discipline has attempted to make a break from its jaded past—a past that has arguably earned it a sonic dunce's cap. This break takes the form of a monolithic sample-rate convertor IC that appears to be gaining acceptance in professional and consumer circles. It is the implementation of this device that brings to light several aspects of the architectural and sonic history of these devices. This article briefly details the various types of rate convertors, and attempts to cast some light on how and why they sound as they do.

History

With respect to sound quality, two issues bear mentioning. SFCs—like many things digital including analogue convertors, dynamics sections, and equalisers—strongly defy prediction concerning their sonic behaviour. No matter how well the devices measure nor how skilled and careful the designer, they frequently disappoint the ear. Secondly, it must be remembered that sonic evaluation is a subjective animal; what you read here is not fact but opinion, based on the observations of several listeners. It is an attempt on our part to crack a small piece of a puzzle, not to write the 'gospel'.

Our first exposure to digital audio and sample-rate convertors outside of engineering school was the premier AES Conference on digital audio in 1982. Two papers were presented at that conference, one by Rabiner on the interpolate-decimate model of sample-rate conversion and one by Lagadec on a new application of that theory. Heretofore, the applications of SFCs were limited to telecommunications wherein the input-output frequency ratios were both fixed and related (as described in Rabiner's paper). However, the commercial application of digitisation to audio did away with neat fixed related ratios and threw in the monkey wrench known as digital varispeed. Lagadec and his staff presciently went about developing a convertor that could deal with these new obstacles by designing a variant of the fixed-rate convertor that was essentially 'adaptive' in real time and automatic. We must assume here to a limited degree that the reader is familiar with sample-rate conversion, as a detailed explanation of the theory surrounding it is beyond both this article and its author.

As shown in **Fig.1**, this 'asynchronous' convertor is one employing multiple digital-filter stages for interpolation for up sampling, followed by a buffer to take up any timing variations followed by multiple digital-filter stages for decimation or down sampling. The twist is twofold; parts of the filters can be reprogrammed on the fly with respect to the oversampling-decimation factor and the transition frequency, this reprogramming being controlled by the ratio of the incoming and outgoing sampling rates. This sampling-rate ratio is determined by measuring the time between like (rising or falling) edges of the input and output clocks and taking the running mean of 128 of these measurements. On the interpolate side, filter f1 interpolates ▶

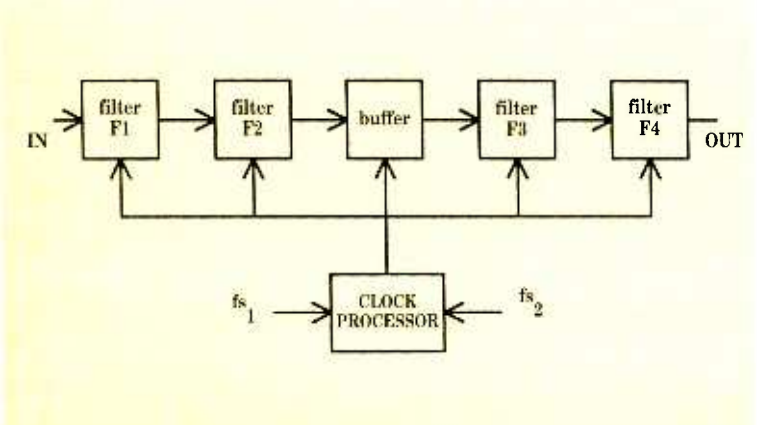

Fig 1: Asynchronous convertor schematic employing multiple digital filter stages

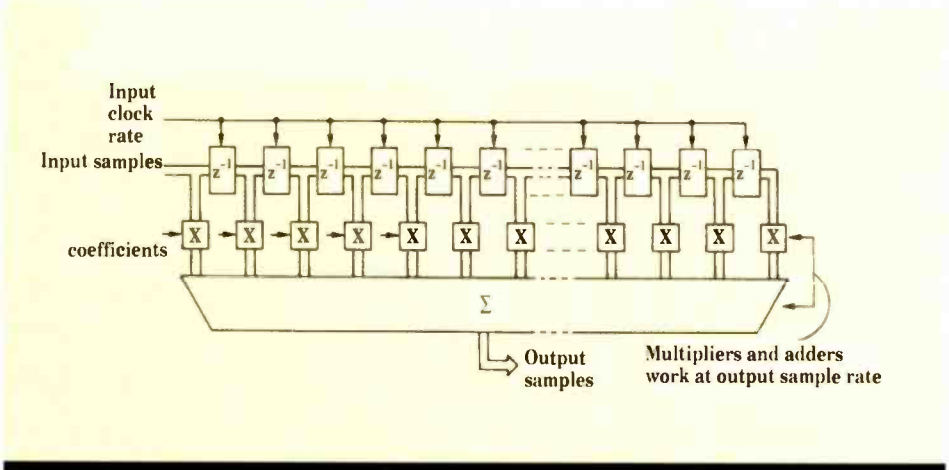

Fig.2: Schematic representation of a finite impulse-response filter

by 4 (2x2) and filter f2 can interpolate by a varying range of factors falling near 64 x 128, the factor being determined by the fs ratio. In the event of dynamic changes in sample rate, the buffer will absorb or supply extra samples, and the filters f3 and f4 accomplish the decimation function with respect to both factor and transition frequency—remembering that if the output frequency falls below the input frequency, the decimation process must also involve additional low-pass filtration to prevent aliasing.

The *SFC-16* sample-frequency convertor, manufactured by Studer, was parsecs ahead of its time and remained an industry standard for some ten years. It also spawned the *DFX-2400* which is manufactured by Sony. The Studer work remains important to this day as current technology borrows largely from it. The time-varying coefficient filter structure was proven here as well as an effective means of measuring sampling-frequency ratios, a method that embodied jitter suppression as a part of its operation. The unit, however, was extremely complex electronically and if one reads the literature surrounding its development one sees mention of a much simpler approach based around a single filter structure instead of the four used. But this is where the trail grows cold.

Processes

The initiative for this article was a formal listening session during which a very jittery DAT machine (9ns RMS) was played back into an outboard convertor with a monolithic SFC chip switched into and out of the SPDIF link between the two. The impression upon hearing the background cloudiness of a choral recording made in a cathedral vanish completely, of hearing the image specificity improve (not subtle at all), of hearing the sound stage expand, was one of both horror and amazement. Is jitter effecting this much sonic damage in our facility? We went on to listen and study extensively.

Fig.2 is a schematic representation of a finite-impulse-response filter where the z-1 boxes represent delays (serial shift registers) with pickoff points or 'taps' between them. At each tap there is an 'x' box which represents the action of a multiplier that very quickly flies down the entire length of the filter multiplying by a different coefficient (or the same one) at each tap. After it multiplies, but before going on to the next tap, it dumps the product into a long register known as an accumulator which is a large adding machine. When the multiplier reaches the end of the filter, the contents of the accumulator are output as a new filtered word, the multiplier ▶

The impression upon hearing the background cloudiness of a choral recording made in a cathedral vanish completely, and of hearing the sound stage expand was one of both horror and amazement

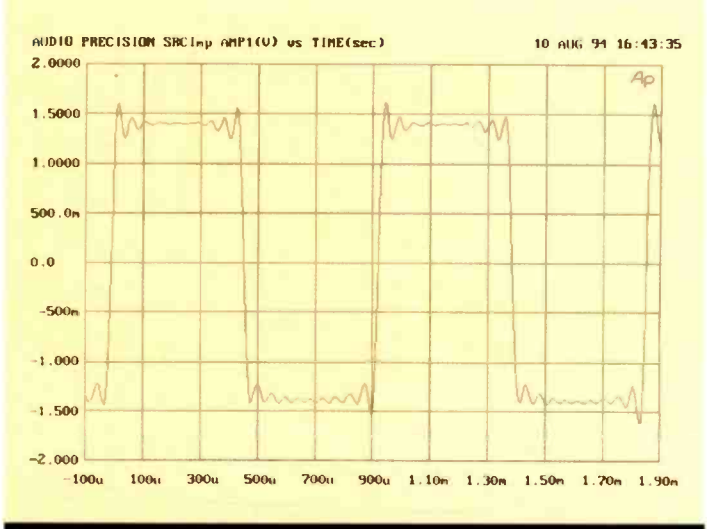

Fig.3: Square-wave response for 7% sample-rate increase through AD 1890

Fig.4: Square-wave response for 7% sample-rate increase through Motorola 55001

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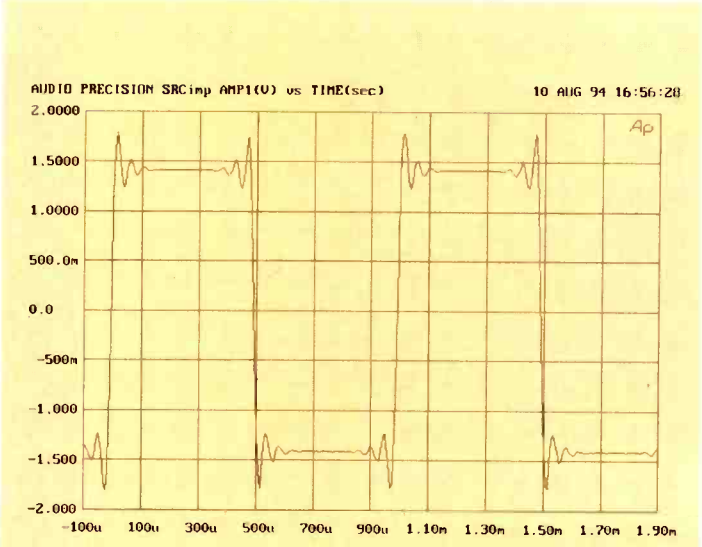

Fig.5: Square-wave response for 7% sample-rate increase through DB Technologies Model 3000

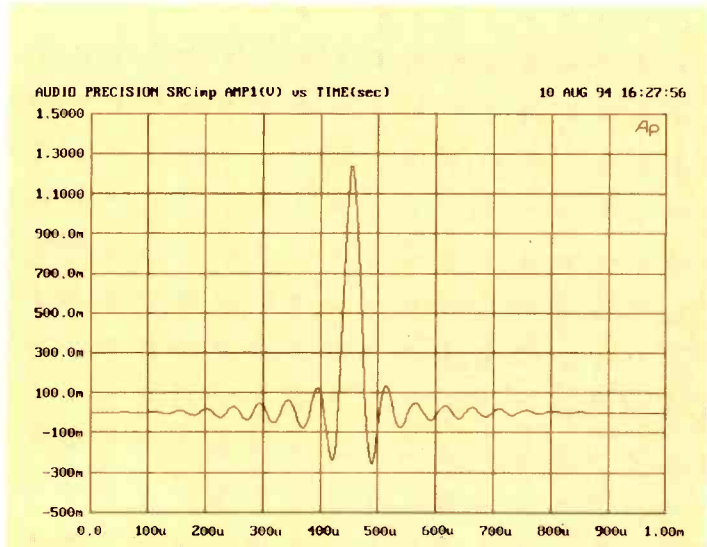

Fig.6: Impulse response for AD 1890

returns to the front, and the whole process starts again. This type of filter is inherently linear phase, exhibits flat (constant) group delay, and can be structured for band-pass, band-reject, low-pass, high-pass, decimate, and interpolate functions. Four of these filters are used in cascade in the Studer rate-convertor; however, one of these filters can, if cleverly and carefully applied, do the entire job itself and it is the belief of this author that the Studer AES papers were alluding to this fact.

As **Fig.2** is labelled, it is actually drawn as a fixed-rate SFC where the input sample-rate is the rate that the samples shift down the delay line and the output sample-rate is the rate at which the multiplier-accumulator flies down the taps in the delay line (they move at different speeds). The coefficients that the multiplier uses are determined by the ratio of sampling frequencies and if we change the coefficients in time as the ratio of the sampling frequencies changes, we have a rate convertor made from a single filter. Of course, this development did not happen overnight and there are some 'hooks' in the implementation that are interesting. The Analog Devices AD 1890 (see *Studio Sound*, August 1993) contains one digital filter which varies in length from 64 and 128 taps. This filter can be loaded on the fly with sets of coefficients stored in memory to effectively simulate 65,536 different filters—one for each of the different sampling-frequency ratios that the device can accommodate. The determination of sampling-frequency ratios is done via the method patented by Studer; however, much thought has gone into the time constants and servo loop used to control the loading of coefficients into the filter as the sampling-frequency ratio changes. A clever aspect of the 1890 concerns the fact that it only supplies an output sample when requested. Its filters take into account the fact that during the process of interpolation, many of the coefficients are zero so the multiplies need not be performed. It also understands that during the decimation process samples are discarded so it knows which samples will be discarded and does not compute them at all. Combine this filter (known as a polyphase structure) with the 'only when requested' aspect of the output and you have a smart device indeed.

Not only did the monolithic device sonically ameliorate the effects of jitter but as a sample-rate convertor it did surprisingly little sonic damage. Yet when we had the opportunity to formally take a careful listen to it another single-filter SFC was auditioned at the same time only this one sounded even better. There was, of course, a set of trade-offs involved as this device was built around a Motorola 56001 DSP chip and could deal with varying sample rates at the input only with the output remaining fixed. The fact that sonics were further improved became food for thought. The Model 3000 from DB Technologies was not only an SFC but also a dither/noise-shaping box, and oscillator, a digital meter, and a distortion test set. It differed from the Analog Devices chip in several respects, most notably the fact that it stores far more of the coefficients for its filter. Both devices interpolate extra coefficients as they function, however the accuracy of this interpolation is related to the 'density' of coefficients retrieved from RAM thereby increasing the accuracy of both the interpolation and the output sample remembering that the output rates are fixed so the filter does not have to wait for a 'request'.

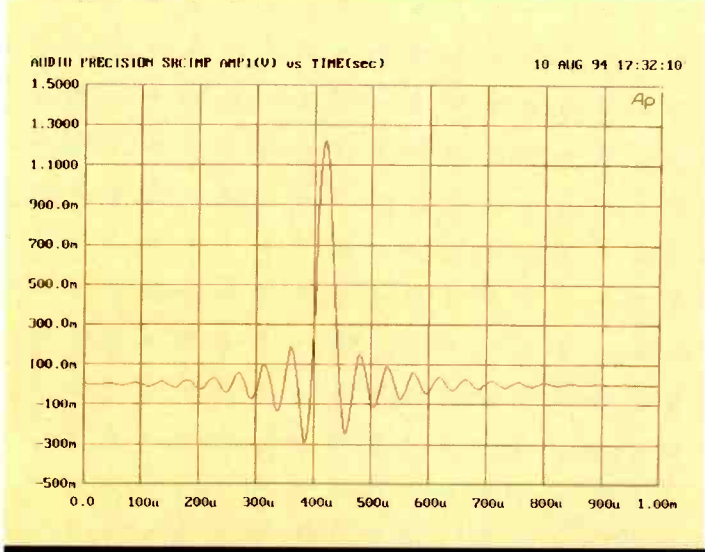

Fig.7: Impulse response for Motorola 55001

In conjecturing why these three devices sound as they do, we begin by taking square-wave responses while the respective convertors are increasing the incoming sample rate by 7%. As can be seen, **Fig.3**, **Fig.4**, and **Fig.5** are similar except for the nature of the ripple exhibited. The two single-filter devices sound very similar and different from the multiple-filter decimate-interpolate device, yet the responses do not conclusively show this. Much of the difference shown in the square waves is happening at high frequencies near the transition band and is known as Gibbs phenomenon, a consequence of both the action of a low-pass filter in the time domain and requantisation effects which suggests the first rationale for sonic behaviour. Note the fact that the Gibbs ripple on the rising edges of ▶

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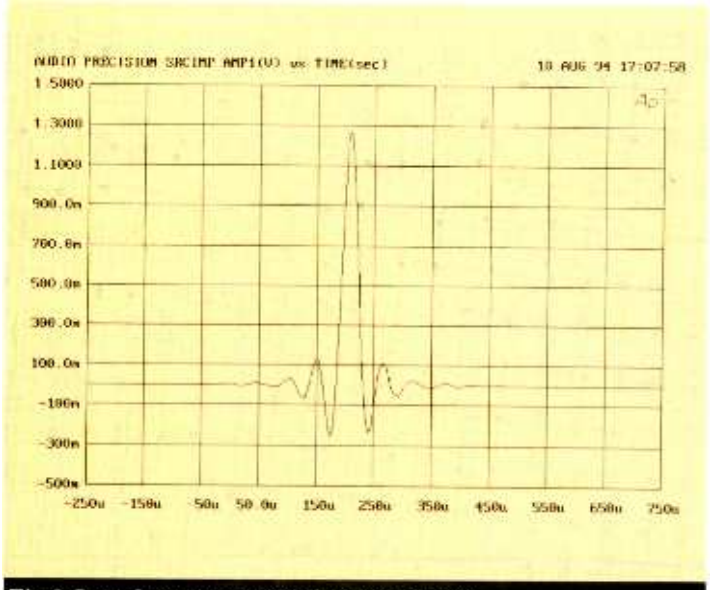

Fig.8: Impulse response for DB Model 3000

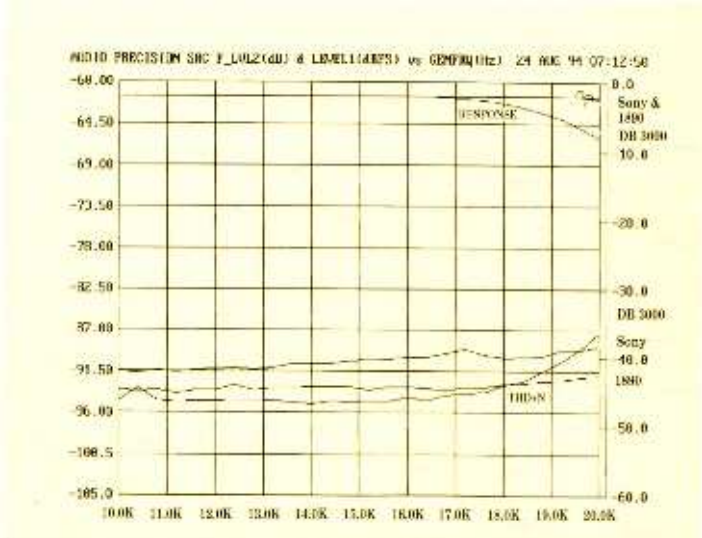

Fig.9: THD+n for AD 1890, Motorola 55001 and DB Model 3000 — 10kHz-20kHz with 16-bit word length

the 1890 square waves is not identical to the falling edge ripple. This is probably a measurement anomaly; however, if it is not it might represent a very slight deviation from linear phase.

Mathematics in digital filters must by necessity be limited in its precision (word length). For example, in the 1890 the coefficients are 22 bits long and the accumulator is 27 bits wide whereas in the 3000, the coefficients are 48 bits long and the accumulator is 48 bits wide. Generally, the longer a word is, the more precise it is—but how do we reduce these words to manageable lengths? It can be achieved by truncation, by rounding, by noise shaping (with dither or without dither—but what type of dither?) These points all affect the sonics of a device such as an SFC. While the 1890 accepts a 20-bit word and outputs a 24-bit word, presently not one implementation properly redithers or requantises the output for shorter word lengths. On the other hand, the Model 3000 inputs and outputs a 24-bit word with two types of dither and four noise-shaping curves adjustable to almost any word length.

The impulse responses are shown in **Fig.6**, **Fig.7**, and **Fig.8**. These again vary with respect to their pre and post responses. These plots show how the digital filters obey Nyquist and how they correspond to the theoretical brick wall. It was again demonstrated by Lagadec and his team at Studer that the pre and post responses (the ripple before and after the central peak) are audible and affect the sound of a digital filter.

Fig.9 shows THD+n versus frequency and frequency response for all three devices plotted from 10kHz to 20kHz with a word length of 16 bits. The 'Achilles heel' of an SFC is visible here as a rise in THD+n with frequency when a -2dBFS amplitude 20kHz sine wave is input. This rise is in large part due to aliasing caused by the finite attenuation rate and depth of the filter itself. However, the more 'relaxed' or gradual a filter is, the more benign its square wave and impulse responses are with respect to ringing and ripple (note the 3000). This easing of specification carries with it a measurement penalty but does the word 'penalty' also apply to the sonic characteristics?

Related to the concept of mathematical precision in a broader sense is the fact that the digital filters being implemented in this application, as in most others, are low-pass devices converging on the equation sin x/x also known as the sinc function. This function is defined for all time beginning in the past at the big bang and ending in the future at the big crunch. As such it is infinite for both positive and negative time and must somehow be adapted for solvability via finite mathematical means. This is accomplished by superimposing another function on it, that function being known as a window function and there are different types of window functions designed for differing mathematical purposes. These functions—which look just like the illustrated impulse response plots—differ with respect to their sharpness and their attenuation in that the faster their attenuation, the broader their central peak so there is the usual parametric trade-off. The width of a window is determined by the length of the filter remembering that a broad window favours low frequencies with respect to noise and distortion and a narrow window favours high frequencies. Some windows such as a Kaiser window have adjustable widths. Might the choice of window function and its width play a role in determining how a digital filter sounds?

As was mentioned previously, if the incoming sample rate is tracked by the SFC and the outgoing rate is locked to a crystal, these devices can very effectively suppress the effects of incoming jitter by trading it off against computational accuracy. If a severely jittered single frequency signal is applied to an SFC, the jitter will be tracked and will cause coefficient changes with time giving rise to noise sidebands appearing to either side of of the signal raising the noise floor. The property of jitter suppression in an SFC is a subtle aspect of its operation yet how the SFC tracks jitter and how the filter is reprogrammed with respect to time and jitter amplitude and frequency probably affect sonics.

Conclusion

One might be prompted to say that much of what has been presented here is conjecture, a statement that is patently true. So complex are the interactions between the few factors that have been discussed here that to draw conclusions is risky at best. All of this ignores what we are not aware of and do not understand with respect to digital recording and the ear. The only conclusive point is derived from observation and that is the fact that these devices, by virtue of technological development and experimentation, sound better than they used to. They are worth revisiting if necessary and might no longer be feared. In 1995 it is neither difficult nor expensive to find out. ■

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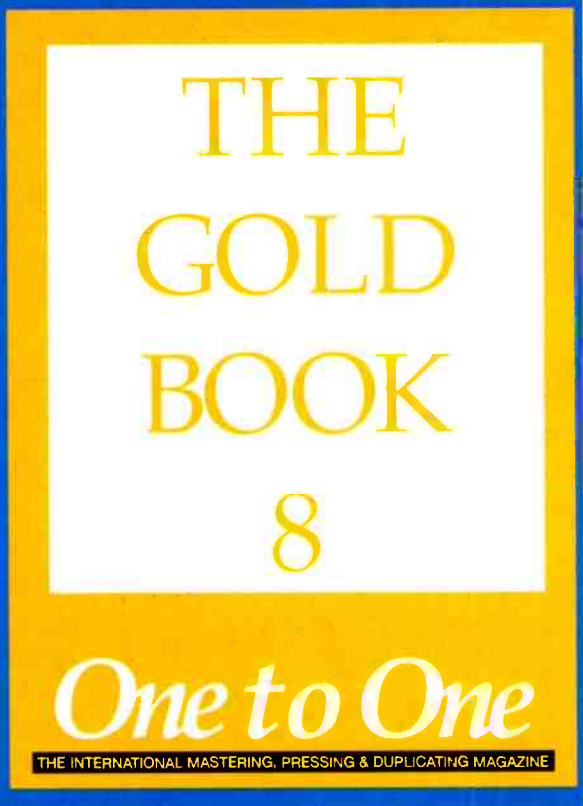

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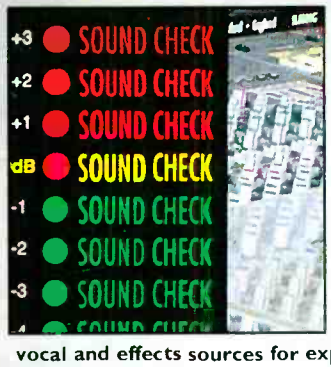

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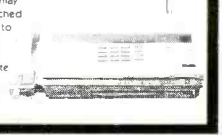

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Fresh format battles challenge the viability of future Video-CD standards

There is nothing quite like a standards battle. Such confrontations promise rich pickings for the winner and years of wasted research and development for the loser and bring out the worst in people. No sooner have we got a nice solid *White Book* standard for Video-CD (which stores an hour of digital video compressed to the MPEG-1 standard on a conventional CD), than the industry splits itself down the middle, undermining confidence in Video-CD, by developing two rival and incompatible high-density versions which each store over two hours of MPEG-2 pictures.

Although Philips have so far retained dignity, the rivalry between Sony, Philips' partner on one of the rival new systems, and Toshiba and Time-Warner, who back the other, is turning into a real humdinger. Matsushita-Panasonic has been split internally, with disagreements between hardware engineers and universal movie studios, on which systems to back.

Both sides are claiming that the other cannot press high-density Digital Video Discs (DVD), must use a caddy to protect against fingerprints and are unable to maintain full picture quality for the whole playing time of a full side. 'Specious and unverifiable' they are saying about each other's claims. There hasn't been so much bad blood around since the early days of the VHS-Beta battle, when each side was developing ever longer playing times from a single cassette.

The best scenario is that Panasonic back Sony and Philips and the battle ends before any commercial launch. The worst scenario has Panasonic siding with Toshiba, and Sony forging ahead to fight a format war in the marketplace.

While all this is going on, the market for MPEG-1 Video-CD faces a daunting challenge. Not only do Philips, the main driver behind MPEG-1 Video-CD, have to tell the buying public what a Video-CD is, Philips also have to reassure them that it is not already doomed to obsolescence by the MPEG-2 DVD system which Sony and Toshiba are shouting about.

Although everything is moving so fast that an article in the morning will be out of date by the evening, the basic technical facts remain solid. Well, so far, at least.

On all existing CDs, the spiral track of data pits has a pitch of 1.6 microns. The read-out laser emits infrared light, with wavelength of 780nm and is focussed by a lens with numerical aperture of 0.45. The single-sided disc can store around 650Mb of data which is read at a constant rate of around 1.5 million bits/second. The MPEG-1 system codes full frames, not individual fields.

The new Digital Video Disc proposed by Philips and Sony uses a red light laser which has a shorter wavelength, 635nm. The lens has a numerical aperture of 0.52. This lets it focus into a tighter spot and thus read smaller pits in a spiral with 0.84 micron pitch. The disc has a storage capacity of 3.7Gb.

The rate at which the bits are read also varies, between 1 and 10 million/second, and on average 3Mb/s, depending on whether the picture contains moving detail or stationary objects. This variable bit-stream is controlled by a buffer memory. Together, these tricks let the new disc store 135 minutes of pictures coded to the higher resolution MPEG-2 broadcast standard, which processes individual fields.

There is room for six channels of sound, either three stereo pairs (perhaps different languages, or censored and uncensored dialogue) or 6-track discrete to cope with the new digital cinema digital systems.

IBM, Apple, Compaq and Microsoft are already working with Philips and Sony to set a multimedia standard for using the new disc with a new generation of PCs.

Sony is already suggesting that DVD can in the future be used to 'activate the very high end audio market' by storing recordings sampled at 96kHz and coded in 24-bit words for 'enhanced music quality'.

Thanks to work done by 3M, the storage capacity can be doubled to 7.4Gb and 270 minutes of continuous movie playback. The double-density DVD is pressed from conventional polycarbonate plastics, with the micro-pits in one surface. This pitted surface is then coated with a thin layer of semireflective material, like a two-way mirror. This layer is then coated with a soft film of photopolymer material which is hardened by exposure to ultraviolet light while in contact with a mould which impresses a second layer of micro-pits. The result is a three layer sandwich which is then topped with a conventional aluminium reflective layer, just like any other CD.

For playback the laser focuses first on the pits in the polycarbonate, and then during a second play-through it refocuses on the pits in the photopolymer layer, while ignoring the out-of-focus polycarbonate pits.

The photopolymer moulding-system process was first used fifteen years ago, by Philips, to make Laser Discs, it was then known as 2P.

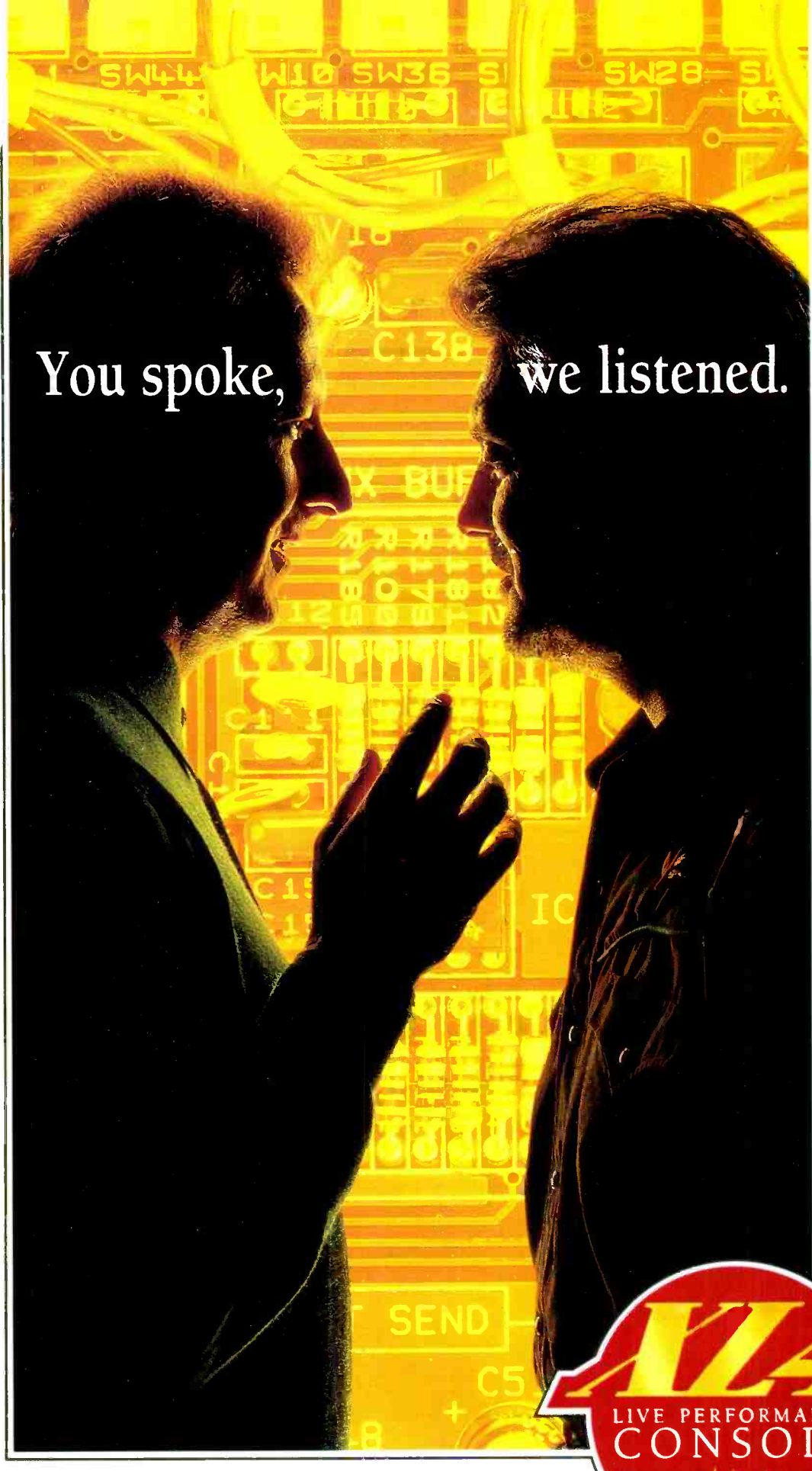
At the first open demonstration, given in January at the Winter Consumer Electronics Show in Las Vegas, Sony and Philips ran a split screen demonstration with one half of the screen showing pictures from DVD, and other half showing the same material from VHS and Laser Disc copies bought in a local video store. DVD demolished HS, and bettered Laser Disc. Another split screen demonstration showed DVD quality to be indistinguishable from the original D1 video studio master tape.

Details of Toshiba's proposed disc have been hazy, but it uses smaller pits to give a single-side capacity of 4.8Gb. Most importantly, Toshiba plan to glue two separately pressed discs together, back to back, just like an analogue Laser Disc. As in Pioneer's top end LD players, the laser reads one side of the disc first, then moves round to the other side of the disc and reads that to double overall playing time. Not surprisingly, Pioneer look likely to support the Toshiba-Time-Warner camp.

Philips and Sony say it will be difficult to mass-produce Toshiba's glued discs. As each half is made to half-thickness, even single-sided discs must be made from two pressings, or the laser in the player will not be able to focus on the disc.

Toshiba say it will be difficult to make and read the double-layer, single-sided discs used by Philips and Sony. The difficult part is getting the partly reflective layer to be of just the right reflectivity to reflect the laser beam when it is focussed to read the first top layer of pits, but let through enough of the laser light to read the second layer of pits. In practice, the reflectivity must be between 20% and 40%. If aluminium is used, the layer must be applied with a thickness tolerance of less than 5nm. 3M claim success with a new material which gives a tolerance of between 30nm and 7nm.

However fascinating all this technical detail may be, the VHS-Beta battle taught a valuable lesson. No-one ever said that the best system had to win. ■



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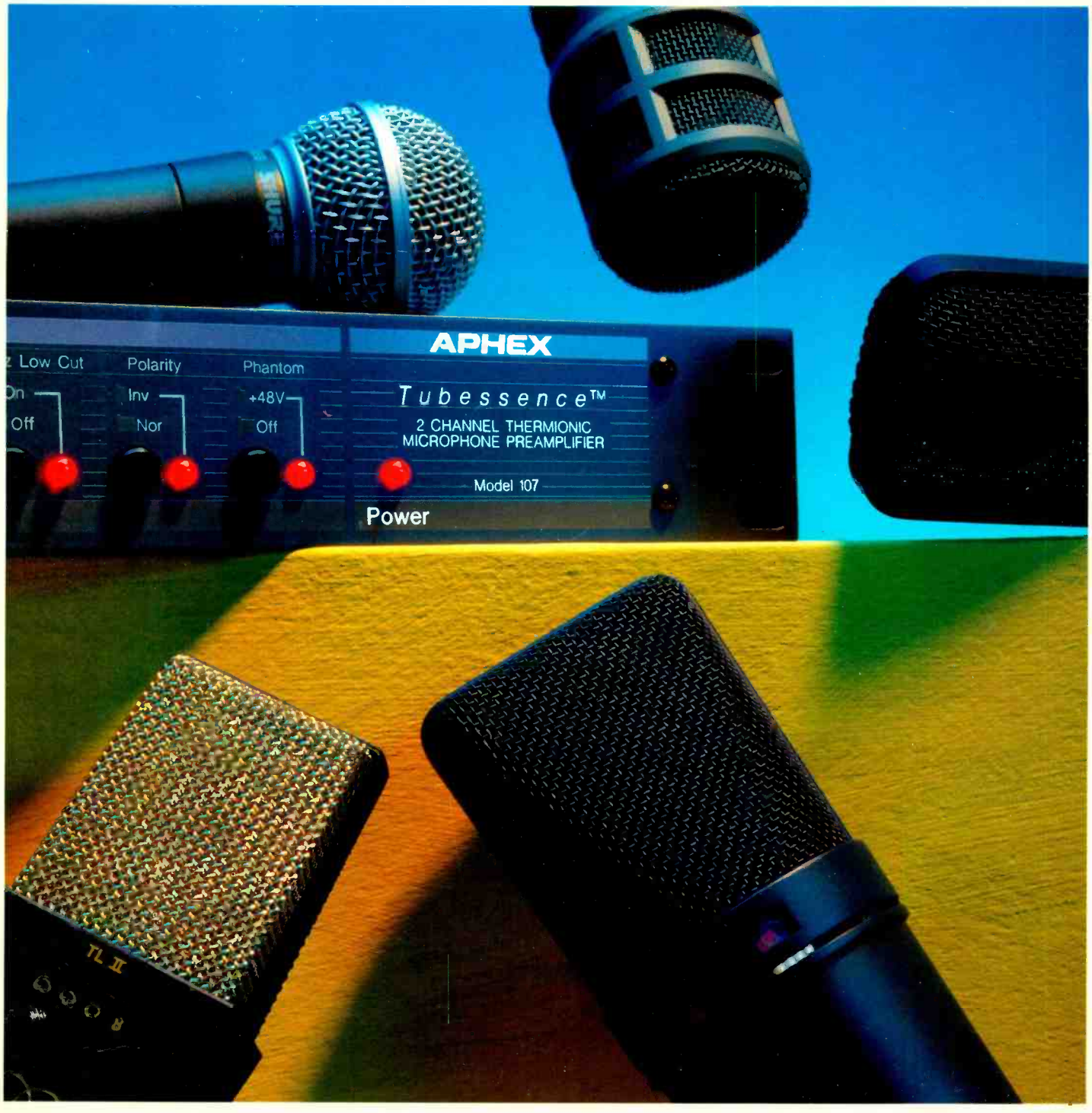

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