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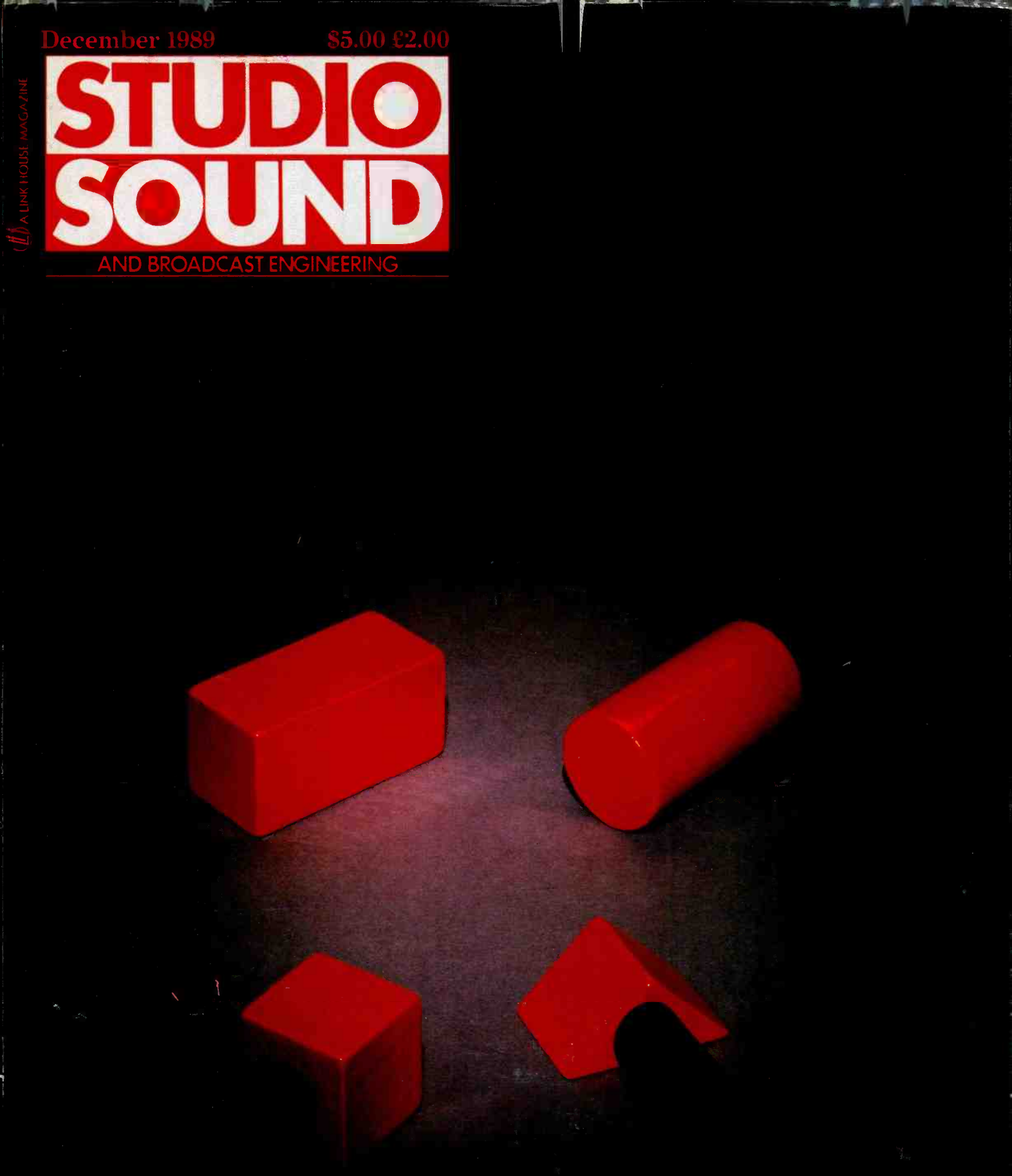
STUDIO SOUND

AND BROADCAST ENGINEERING

A LINK HOUSE MAGAZINE



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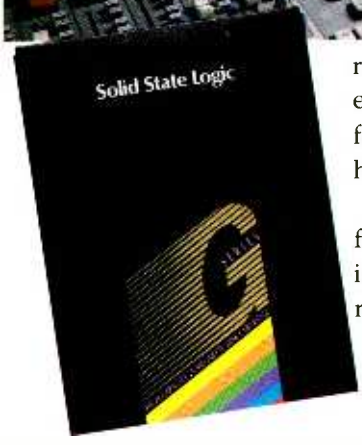
"No one will ever buy a console with a television in it"

Visitor to AES, 1977

It is hard to believe that a modest VDU could attract much attention. Today, even effects processors use visual displays. Why have they become so necessary?

Quite simply, the greater the number of functions in a system, the more flexibility there must be in showing its status. In the recording studio the central piece of creative hardware is the console. To unlock its full potential, Solid State Logic gave engineers digital control of its facilities. The VDU was a side effect of this plan.



In the intervening years our designs were enhanced, incorporating the suggestions of the many leading studios who had adopted our system. Ultimately, the G Series Master Studio System evolved.

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The '80s—polishing off the old or preparing for the new?

Slowly creeping upon us are the '90s. As I write this in mid-October there are still several weeks to go before the end of the year; the end of decade; the end of the '80s.

I find myself compelled by tradition to try to draw the ends of the decade together and neatly parcel them so that when we talk about *the '80s* it is to identify them as strongly as it is now the '60s or '70s. But try as I might, we are really still too close to see what the ninth decade of the 20th century has brought the recording industry in the way of major advances. Let us look back to see some of the developments of the '80s in technology.

How about digital recording? How about digital reverb? What about large computer controlled consoles? Whoa!—Sorry but before we go any further please note that these all first appeared in the '70s.

Then how about the full synthesis of musical tracks and the use of... Let's stop there—this was the '60s and maybe even earlier.

One can almost say 'Find me *one* good thing about the '80s for our industry' but that would be unfair. There has been much to recommend the '80s but generally as a period of development and improvement and not with major milestones. There has been a transfer of artistic control into the hands of the creative musician. At the end of the decade it is undoubtedly more certain that you will make a good recording—it is much easier. There are now so many ways of achieving the same end. We have turned the experience of numerous musicians playing together in the studio into a novelty and at one stage almost dispensed with the need for real instruments. Rather sourly I could mutter that the nearer the '80s draw to a close, the better the '70s look. Will we see the '80s as the decade when hi-tech replaced high skill? When signal manipulation replaced good sound? And the time when the creative musical focus moved from the studio to the bedroom?

None of this is really true. I would like to see the '80s as really just an energetic preparation for the '90s. The last AES Convention of the decade has kindly thrown us all the elements we need for some serious progression—ADC/DAC of 20 bits and over, true realtime DSP power, the maturity of the hard disk recording/editing system with a human interface, the recordable optical disk, fibre optic data transmission, much improved monitoring and acoustic environments, cost effective and high performance digital recording systems and an interest from console manufacturers in maximising the signal path quality. The components are ready. The domestic high quality replay medium of CD is in place. High quality film and broadcast sound is happening. CD derived video, ROM, interactive formats promise much and when combined with computer processing to draw diverse elements together the possibilities are open-ended. We can record, manipulate and produce the finished product as we desire; the quality consumer formats of software and broadcasting are ready and waiting; and our attention must turn to those who input the creative and musical drive at the front end of the industry. We must trust that their muse (and the vision of the record companies) is fully a match for the possibilities that await.

I'm looking forward to the '90s and I hope that you are too. Nothing will change on New Year's Day but in 10 years time I think that I will be able to look back over the decade with rather more positive tones than I could apply to the '80s.

May I close by wishing all our readers, wherever they are, on behalf of myself and all the staff of *Studio Sound*, a very prosperous decade.

Keith Spencer-Allen

Cover: Incompatibilities. Photography by Tony Petch

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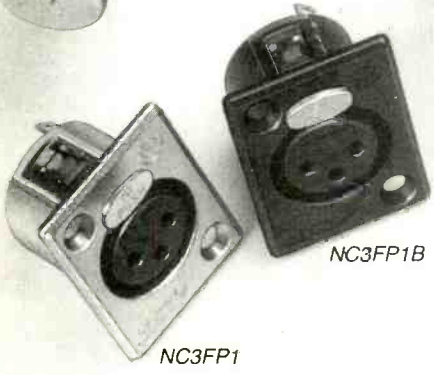

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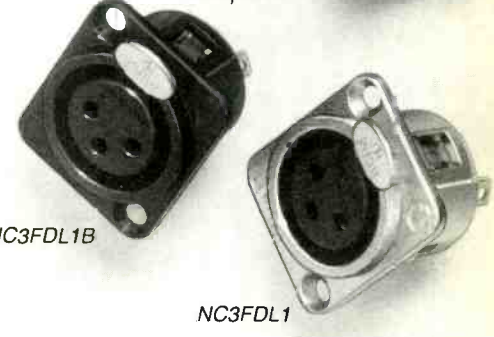

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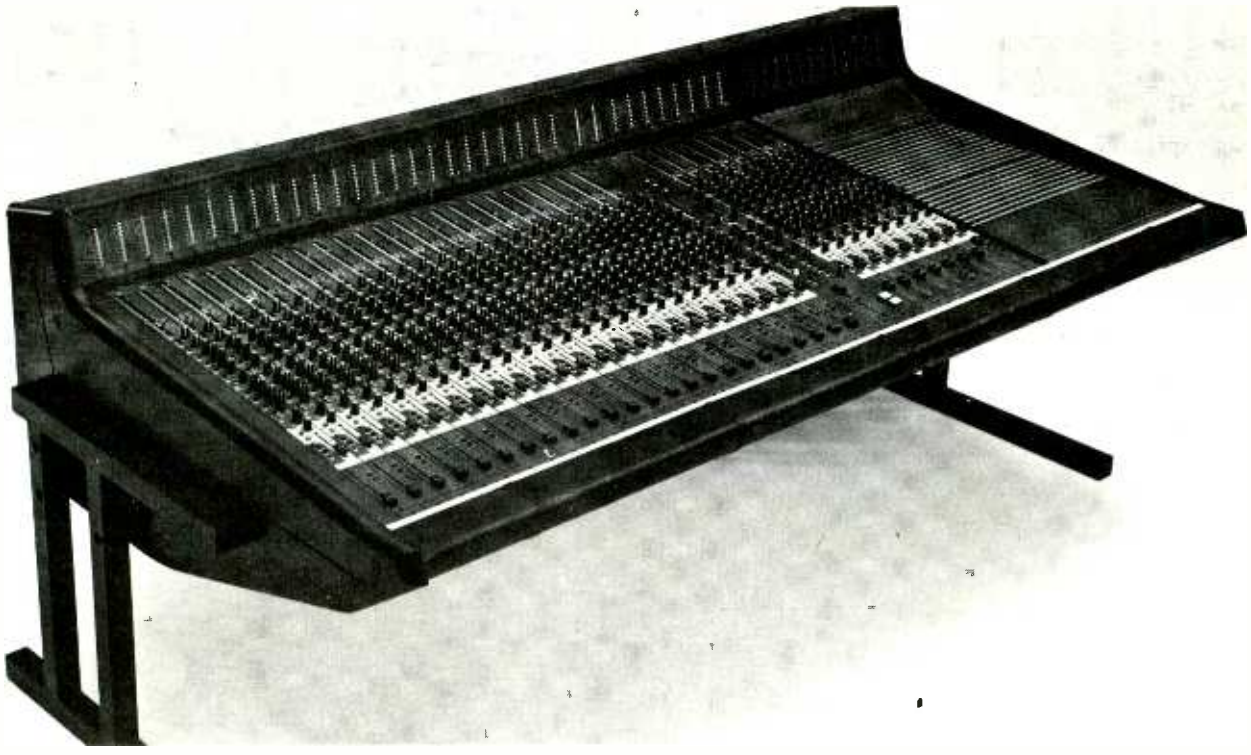

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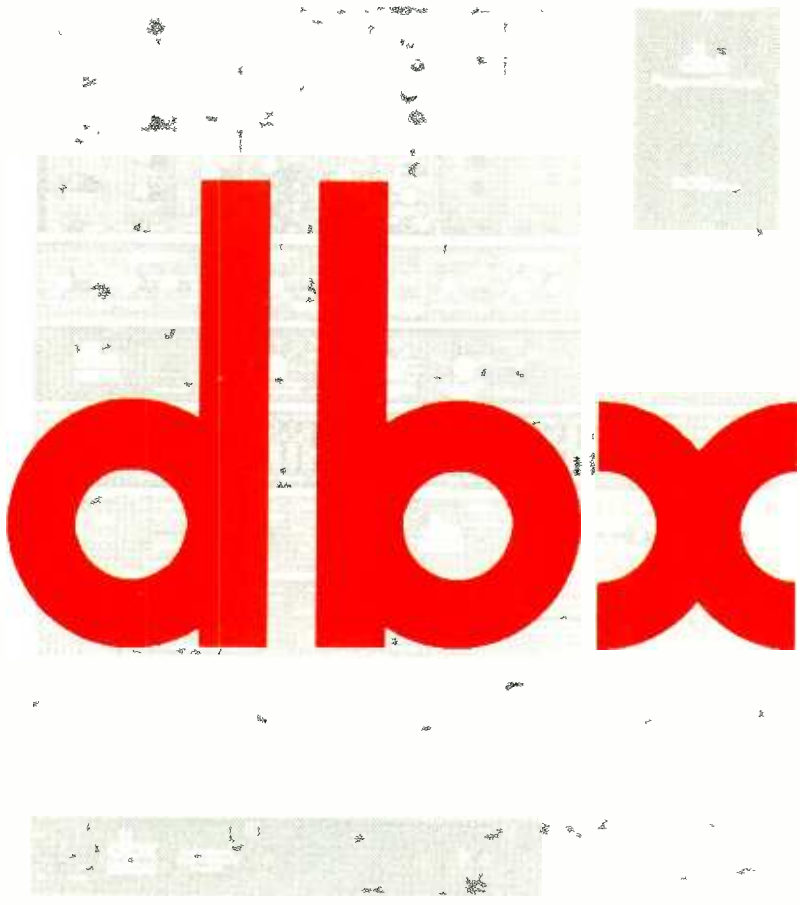

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Using the portable recorder is plain sailing in dramatic productions and audio research, where DAT picks up the smallest vibration.

Docked in a studio, the SV-360 DAT deck is equally impressive.

Like the portable, it offers all the flexibility of tape with the sound quality of C.D.

It can be used on its own or as a back up system in a recording studio.

It can also be used alongside our already well-established C.D. players and turntables which have become classics in the studio.

Out of the studio, the DAT deck is equally effective on location.

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It's 4 DAC 18 bits also delivers higher fidelity.

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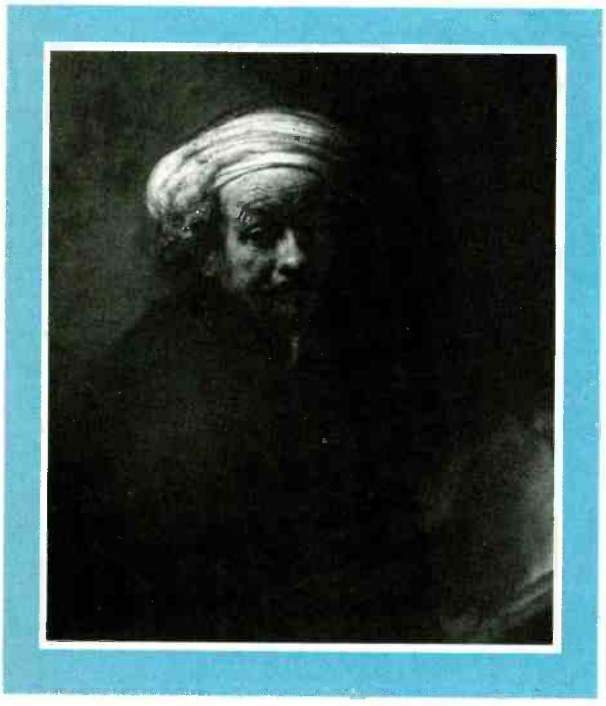

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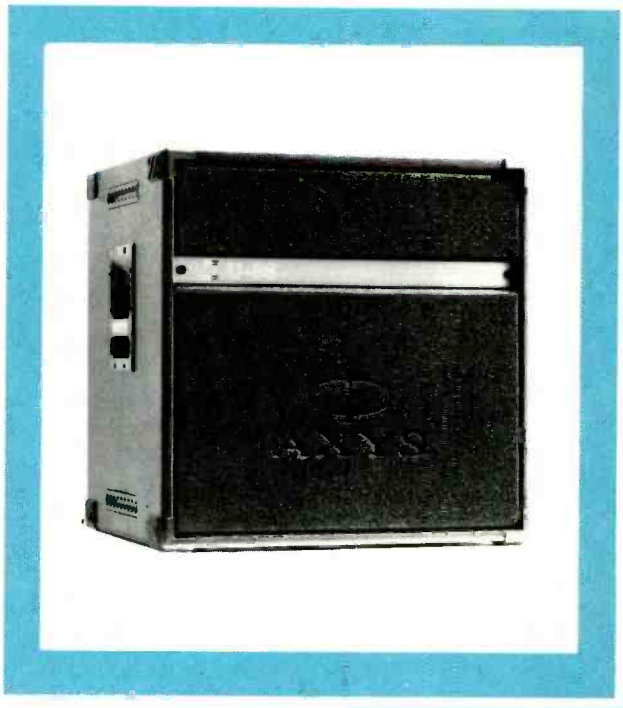

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Otari and DigiDesign develop disk recorder

Otari Corporation have announced that an agreement has been reached with DigiDesign of Menlo Park, CA, to jointly develop a professional hard disk recorder. John Carey marketing manager of Otari has stressed the need for reliability in such a venture, "DigiDesign audio editing software *SoundDesigner II* is solid and reliable. Reliability is essential for a professional application, and DigiDesign allows Otari to overcome the problems many new software-based products encounter."

Peter Gotcher, DigiDesign's president added, "We're looking forward to working with Otari to create state-of-the-art digital audio recording systems.

"The new product will be marketed by Otari and is scheduled for mid-1990 introduction."

Connector chart

AWP, suppliers of connectors and connector systems, have published a guide to help users identify and select the correct connector for a given specification. The guide was initially developed for their own use and takes the form of an A3 chart that cross references specs against product type, and for 'D' subminiature connectors, basic design part number with BS9523-F0013 reference. The major specifications are included, like MIL-C, PAN, VG and BS9000, although the listing is not intended to be inclusive.

The connector specification guide is available free on request from the sales department, AWP Electronics, Dalma House, Kings Mill Lane, South Nutfield, Redhill, Surrey RH1 5ND, UK. Tel: 0737 823421.

News from the AES

There is a broad spectrum of subjects to be covered in this year's session of British AES evening meetings and more details will be available on each nearer the time. To help future planning, the dates, speakers and titles are listed below.

1989

December 12th
Leisure and Entertainment Noise
Ken Dibble

1990

January 9th
Satellite Distribution of Audio
Chris Hibbert
February 13th
Digital Audio in the TV Studio
Paul Evans
March 20th
To be announced
April 10th
Amplifier Differences
Paul Miller
May 8th
DAT Timecode
Sony
June 12th
Active Acoustics
Philip Newell

The next lecture is on the subject of **Leisure and Entertainment Noise** by Ken Dibble of The Sound Practice. "Fifteen years ago entertainment noise was an unheard of phenomenon and the understanding of its control and regulation has grown up alongside the development of the technology which is its cause—ie audio systems engineering.

"This presentation will provide an overview of the peculiarities and particular problems associated with entertainment noise control. It will examine the reasons why entertainment noise is singled out as unique among noise sources, will consider the difficulty in application of normally adopted units of measurement, will consider the difficulties associated with legislative enforcement arising therefrom and will discuss the ways and means available for effective control.

"The presentation will be illustrated by selected actual case history data, demonstrations and some 40 slides."

This lecture will be held at the IBA, 70 Brompton Road, London SW1 starting at 7.00pm with coffee at 6.30.

We are now able to supply the revised edition of John Watkinson's book *The Art of Digital Audio* priced at £39.95 and a new book *Stereo Sound for Television* by Francis Rumsey at £12.95. The proceedings of the recent AES/EBU Interface Conference are also available at £20. These can be obtained from the address below.

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

Exhibitions and conventions

November 28th to December 3rd Sound Expo/China '89 Shanghai Exhibition Centre, Shanghai, China.

1990

January 21st to 25th Midem '90, Palais des Festivals, Cannes. Contact: Handel Communications on 01-627 8083, or Peter Rhodes.
March 13th to 16th. AES 88th Convention, Centre de Congres, Montreux, Switzerland. Contact: AES Exhibition Director, Herman A O Wilms, Zevenbunderslaan 142/9 – B-1190 Brussels, Belgium. Tel: (2) 345 7971. Fax: (2) 345 3419.
March 30th to April 3rd NAB, Atlanta, GA, USA.
April 22nd to 25th Vision and Audio International, Earls Court Exhibition Centre, UK. Tel: 01-776 0709.
June 6th to 8th APRS 90, Olympia 2, London, UK. Contact: APRS Secretariat. Tel: 0923 772907.
September 21st to 25th International Broadcasting Convention, Metropole Conference Centre, Brighton, UK. Contact: IEE Secretariat. Tel: 01-240 1871.

Agencies

• **B&B Systems** Inc, CA, USA, have announced that A/V Technology International Inc, MA, have been appointed exclusive distributor of B&B Systems' products both in the US and worldwide. A/V Technology International, PO Box 275, Newton Centre, MA 02159, USA. Tel: (617) 965-5656. Fax: (617) 965-1865.

• **Odetics Broadcast**, Anaheim, CA, have opened a subsidiary in the UK to serve as a sales and service office to enable Odetics US to better serve the UK and European markets. Odetics UK Ltd, 23 Prospect Street, Caversham, Reading, Berks RG4 8JB. Tel: 0734 461 488.

• Harman UK, distributors for the **Fostex** *D-20* R-DAT recorder have announced a network of agents to handle the *D-20* and other products. The agents include HHB Communications; Stirling Audio and Multitrack Hire. The appointed agents will work alongside Harman's own sales team.

• **WaveFrame** have announced the appointment of Naniwa Gakki Co Ltd to be the *AudioFrame* production system distributor in Japan.

In brief

• London, UK: **KFA Associates**, the studio design and construction company, recently delivered a complete 30 ton studio installation for new incremental station WNK Radio, only to find themselves repeating the process 24 hours later when WNK had to find new premises due to lease problems.

• Aylesbury, UK: **Bopla Ltd** have brought out a new technical catalogue that provides comprehensive details of the company's range of 19 inch rack cases, industrial enclosures and desktop cases, together with cable glands and other accessories.

• London, UK: **Nexus International Consultancy**, part of the TVS Entertainment Group, has been formed to offer everything that is needed to enable television programmes to be made and broadcast, terrestrially or by satellite. Services include design, studio construction and studio equipping.

• Hounslow, UK: **Akai** have announced their recommendation of Maxell 8 mm tape for their **DR1200** digital audio multitrack.

• Manila, Philippines: The second Annual Sights & Sounds Fair was the site of a workshop on tape machine alignment and maintenance held by **Otari**. The workshop drew professionals from the recording, broadcast and tape duplication industries.

• Sheffield, UK: **Audionics**, custom broadcast equipment manufacturer, have completed the installation at their Sheffield premises of a PCB and silk screening plant.

• Uden, The Netherlands: A new EMI compact disc manufacturing plant for continental Europe has been announced by **EMI Music Worldwide** president Jim Fifield. The new CD factory will be sited at Uden with installation beginning in early 1990 and manufacturing in mid 1990.

• Chessington, UK: *Beyond 2000*, an Australian TV programme seen in over 60 countries worldwide, will be transmitting a feature item on **Digital Audio Research's** *SoundStation II* early in the new year. The programme examines new ideas in science and technology.

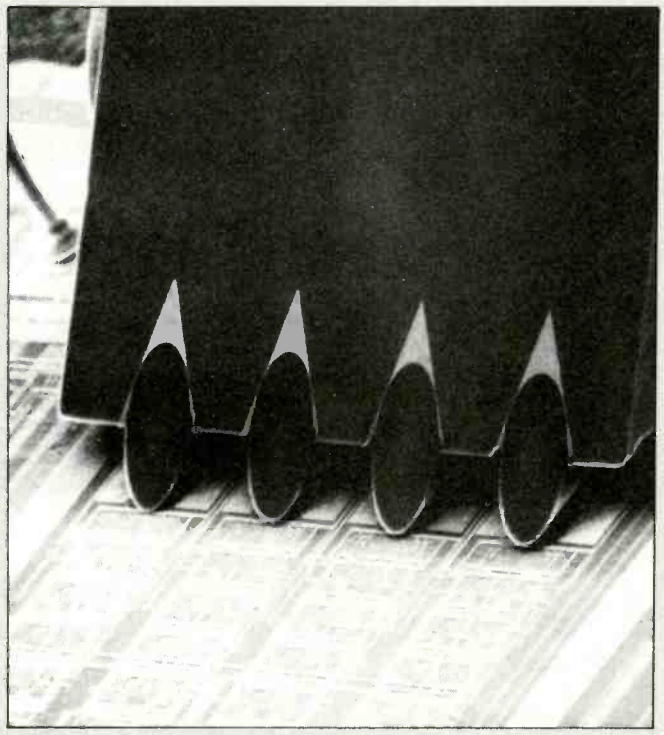
• Wendy, UK: **Thatched Cottage Audio** are opening a digital division to cater for the new generation of professional 16- and 24-track machines. Tel: 0223 207979.

AKG/Orban/dbx marketing structure

S Richard Ravich, president of AKG Acoustics, Stamford, CT, has announced the formation of a new marketing and sales team for AKG's dbx Professional Products division.

Marketing and sales for North and South America will be headed by David Roudebush, who will also be responsible for marketing and sales of AKG's Orban division throughout the same area. Sales in Europe, Asia, Africa and Oceania will be handled by Howard Mullinack, in addition to his responsibilities as marketing and sales manager for Orban products in those regions.

The dbx Professional Products Division and Orban Associates were acquired within six months of one another earlier this year by AKG Acoustics, the US-based subsidiary of AKG GmbH, of Vienna, Austria.



LIGHTFALL: Shown here magnified several hundred times is the receiver portion of IBM's experimental optoelectronic computer chip set. The four large ovals are optical fibres, scarcely larger than the average human hair, sliced at a precise angle so the light signals they carry are aimed at the photodetectors directly beneath them (unseen in dark area). The photodetectors take the light signals—arriving at one billion bits per second—and convert them to electronic signals 'understandable' to a computer. The filigree-like circuitry (in the foreground) are amplifiers. They strengthen the electrical signal so that it can be used by the chip's logic circuits.

People

• **BeyerDynamic UK** have announced the appointment of Chris Gilbert who will be responsible for broadcast and professional sales.

• David Ward and Mark Smith have joined the Broadcast Systems Divisions of **Philip Drake** as project managers. Ward comes from BBC television's planning and installations department and Smith from the projects and maintenance department at TVam.

• **Neve US** have announced the promotion of Rick Plushner to the newly created position of national sales manager for PCM products in the USA. Neve became the exclusive North American distributor of Mitsubishi Electric's professional digital audio products in April.

• **Digital Audio Research** have announced the appointment of Bob McNabb to the position of regional manager in the US. McNabb was formerly regional manager with the Mitsubishi Pro Audio Group. He will be based at DAR's LA office.

• The monitor and broadcast products manufacturer, **Croma**, have appointed Charlie Raynsford as senior sales engineer for UK and Europe.

• Philip Rambow has been appointed managing director of **Scarlett Recordings**, a division of the music based consortium the Scarlett Group of Companies. Rambow was previously A&R manager at Arista Records.

• **HHB Communications** have announced the appointment of Chris Barron as field service engineer. Barron joins from Solid State Logic, where he was a senior systems engineer.

• **Limehouse Television** have appointed John Turner as the new managing director. Turner was group resources manager for Trilion, owners of Limehouse.

Crest correction

In the 'Products' section of *Studio Sound* October we included the new additions to Crest Audio's *FA* series of amplifiers. Unfortunately we included the wrong UK distributor information. Crest Audio's sole UK distributor is First Audio Ltd, 95 Ditchling Road, Brighton, East Sussex BN1 4SB. Tel: 0273 693610. Fax: 0273 693620.

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Switzerland: (01) 3913939 · Turkey: (04) 1262919 · USA NY: (0212) 7653410 · USA CA: (0818) 7852211 · Taiwan: (02) 3214454 - 6

Fostex statement on timecode

Fostex have recently announced that should the IEC choose to adopt the NHK timecode standard for R-DAT, Fostex will provide a software and hardware retrofit to make the *D-20* DAT machines currently in use compatible with other machines.

The Fostex statement comes in recognition of the NHK proposal for subcode timecode recording, supported by Sony, Matsushita and Teac. Yuki Ikeda, Fostex international marketing manager, explained that the retrofit would only be offered in the event the IEC specifies and approves that standard, "Of course, the *D-20* will continue to support Fostex's existing implementation allowing the user to select which timecode format to decode on playback."

Contracts

• The new version of the **Drake** *8000* series studio intercom has been chosen by VCF in France for each of their two new OB vans. Inelco Electronique, Drake's French representative, recently confirmed the order. Granada Television, in preparation for stereo programme transmission have placed an order with Philip Drake for *7200* series stereo distribution modules.

• **New England Digital**, VT, USA, have supplied aircraft manufacturers the Boeing Corporation with a 16-track *PostPro* workstation for their in-house studios.

• An **Allen & Heath** *Saber* 16-track console has been used by the University of Sussex in conjunction with the Musician's Union for their courses on musicianship and audio technology.

• Music producer Steve Levine has invested in four **DCS** *900* A/D converters for his studio, Do Not Erase, in London, UK. The addition of the DCS *900* units has enabled Levine to upgrade his two Sony *3324* 24-track digital tape recorders without having to replace them with Sony's improved *3324A* versions.

• **KFA Associates**, London, have supplied a modular acoustic booth, *The Box*, to Raper and Wayman, suppliers of professional audio equipment. Raper & Wayman are to use *The Box* for demonstration purposes at their premises in Wood Green.

• **Electro Sound**, manufacturers of high-speed audio cassette duplication systems and equipment, have sold music duplicating systems to three leading companies in the People's Republic of China.

• Radio Bremen, North Germany's regional broadcasters, have installed an **SSL** audio production console in its new outside broadcast vehicle. The new mobile is part of an expansion of radio OB operations and is equipped with an *SL 5548 M* series console with 32 mono and four stereo channels plus *Instant Recall*.

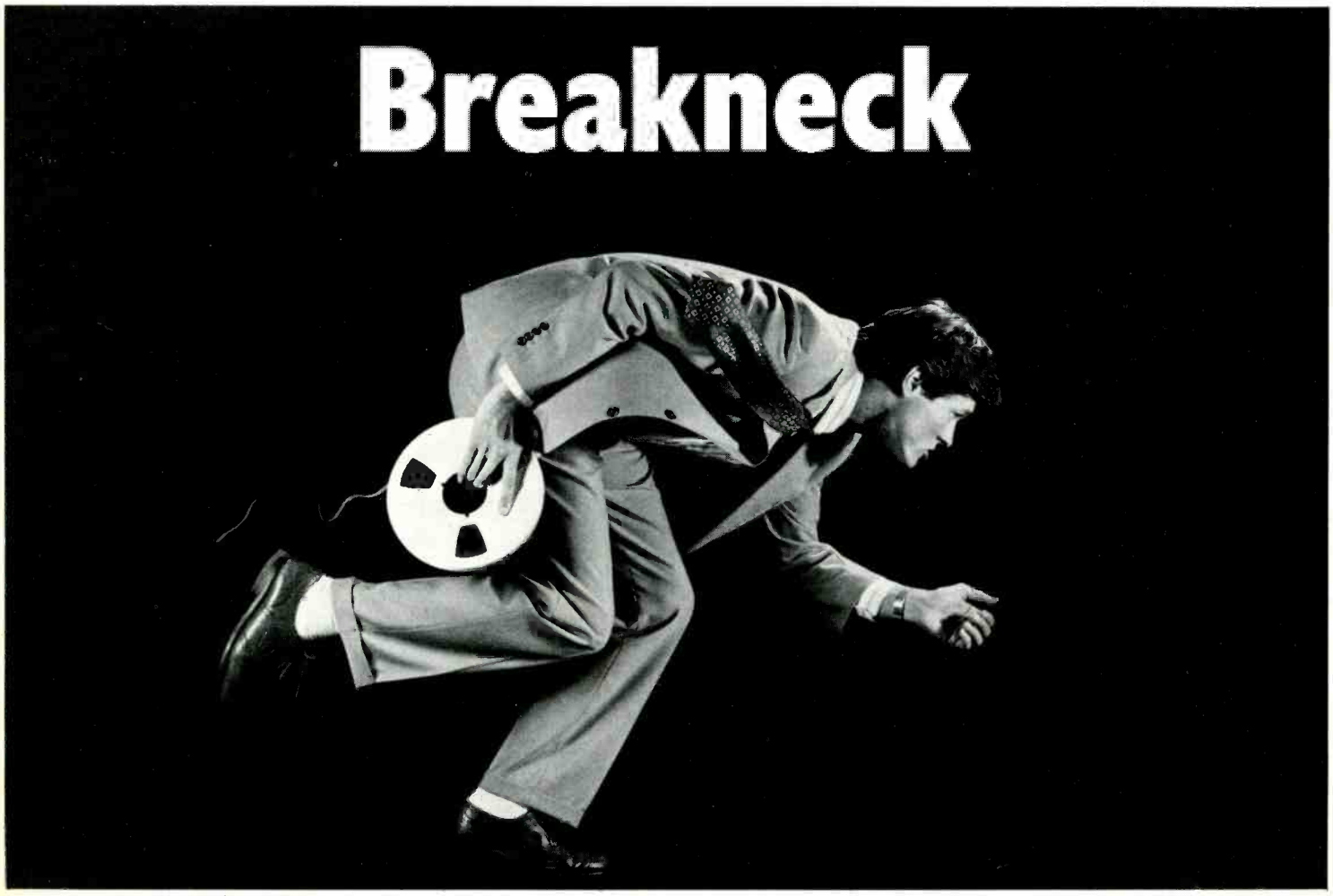
• Gerr Audio, **Allen & Heath's** Canadian distributors, have supplied a 44-input recording *Saber* console for use in a presentation and seminar room in the Vancouver offices of TMI, distributors of Akai and Fender products.

• **Solid State Logic** have installed a 64-input *SL 4064 G* series console with computer into the new Battery Studios in New York.

• A **Lyrec** *P-4400* duplicating line and four Lyrec twin slave units have recently been installed at the Nortorf operation of Teldec Press, record and cassette production company.

• **Imagination**, the London based design and communications company have opened two new 24-track studios. The studio complex was designed by Eastlake Audio and includes the SSL *6000* series console brought from the previous studio and a Harrison *SeriesTen*.

• Current **Eastlake Audio** design and construction projects include work in Africa, an orchestral studio and control room in Libya; the Far East, studio complex in Kuala Lumpar, Malaysia; and South America, a complex to house 10 radio



stations for Caracol in Columbia.

• **WaveFrame**, CO, USA, have announced the delivery of two *AudioFrame* production systems to Werner Studios in Copenhagen, Denmark; and two *AudioFrames* to Real to Reel studios in Dallas, TX, USA.

• Recent **Amek** contracts include three *BCII* consoles to Spitfire TV, London video facilities, a *BCII* console to Sync Facilities in Yorkshire; a *Classic* console to Sky TV for their Thames Valley headquarters; and a Classic console to Studio L'Equipe in Brussels, Belgium, fitted with GML moving fader automation.

• Video post-production facility Molinare SA, Madrid, Spain, have become the first operators of **Solid State Logic's** *ScreenSound* digital audio-for-vision editing suite. Recent orders have come from clients in the USA, Japan, Europe and Canada, and recent installations include Magnetic North, Canada and Pearson Roff video house, London, UK.

• The first **Solid State Logic** *01* digital production centre to leave the company's Oxford headquarters has been installed at Video Sunmall audio/video post-production facility in Tokyo, Japan. UK Film Production company Goldcrest have ordered two *SL 5000 M* series film consoles as part of their studio modernisation.

• Recent orders for the **Fostex** *D-20* DAT player have been placed by the BBC's TRU; Yorkshire TV, Granada TV; Thames TV; Marks & Spencer's in-house audio-visual department; and Network Music & Media.

• The Canadian Broadcasting Corporation have equipped three audio/video post-production studios with **Audio Kinetics** *ESbus* machine control and synchronisation products.

• Recent sales of **Soundcraft**'s video post-production console, the series *200 BVE*, include the BBC's Research Department for use in their HDTV experimental OB truck; Central TV; and Watford based Blitz Vision.

• **Prism Sound**, Cambridge, UK, is to supply VSD, a new Video Status-Display and metering system, for the Neve *DSP* series digital sound mixing consoles at WDR (West Deutscher Rundfunk, the West German Broadcasting Authority). The first VSD systems will be installed at Cologne in two live transmission studios where performance will be mixed for digital recording and broadcast. The order has been placed by Siemens, Neve's parent company and distributor for West Germany.

• Recent sales of the **ASC** *CDK006* pro audio *RS232* interface have included exports to Hong Kong, Germany, Canada, Italy and the USA. The ASC *CDK006* Pro Audio *RS232* interface enables user control of the Sony *CDK006* juke box by simple *ASCII* strings sent from any micro computer using the ASC *CDCol* serial protocol.

• **Elliott Brothers**, Oxford, UK, have won the contract for the design and installation of the complete Central Technical Area of the BBC's Queen Street Edinburgh complex.

Address changes

• **HEAD Acoustics** are now at Kaiserstrasse 100, D-5120 Herzogenrath 3, Aachen, West Germany. Tel: (49) 2407 577 30. Fax: (49) 2407 577 99.

• **Audio Design**, Pangbourne, UK, have a new telephone and fax number. Tel: 0734 844545. Fax: 0734 842604.

• **DTL Broadcast Products and Systems** have moved to DTL House, Knaves Beech Way, Loudwater, Bucks HP10 9QY, UK. Tel: 0628 819481. Fax: 0628 819369.

In brief

• London, UK: Europe's first *No-Noise* service has been based at the **Chop 'Em Out** audio facility in London, W10. Developed in California by Sonic Solutions *No-Noise* is a computer-based system that can be used to enhance poor quality or damaged recordings.

We are pleased to announce that we're ready to fit the final pieces to our new demo facility.

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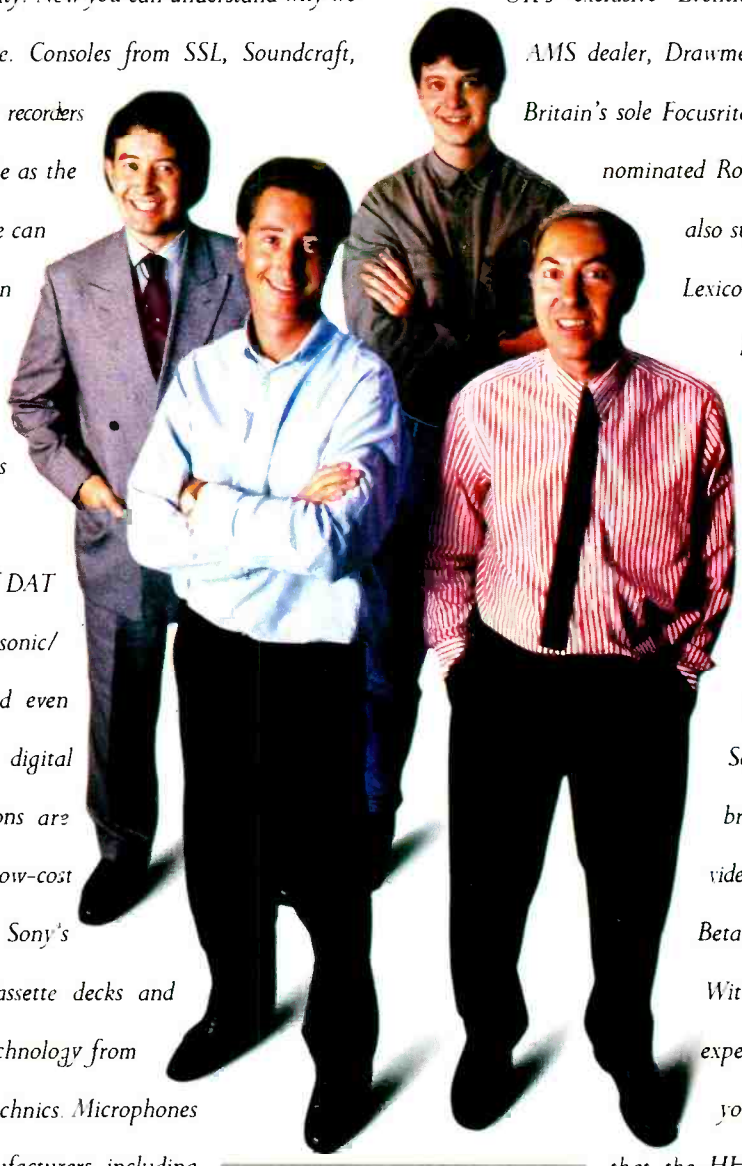
HHB is famous for its specialist digital audio expertise. We can advise on a choice of DAT recorders from Sony, Panasonic/Technics, Radio Systems and even Fostex and Casio. In the digital multi-track arena, the options are wider than ever. From Akai's low-cost 12 track device right up to Sony's PCM-3348. CD players, cassette decks and consumer audio and video technology from Aiwa, Sony and Panasonic/Technics. Microphones from a large selection of manufacturers, including Neumann, Amcron, AKG, Shure, Sennheiser, Beyer and B&K. We also support our positon as key UK distributors for Amcron and Quad amplifiers with a powerful choice of speakers – from the ever-popular Yamaha NS10M's to ATC's much-acclaimed range of studio monitors. Signal processors? We are the UK's exclusive Eventide distributor, London's single AMS dealer, Drawmer's biggest pro-audio dealer and Britain's sole Focusrite distributor. Not only is HHB a nominated Roland Systems House, but we can also supply the latest and the best from Lexicon, Tubetech, Urei, Yamaha, BSS, Klark Teknik, Akai and many, many more.



RICHARD KERSHAW, MARTIN O'DONNELL, STEVE ANGEL, MARTIN WESTWOOD.

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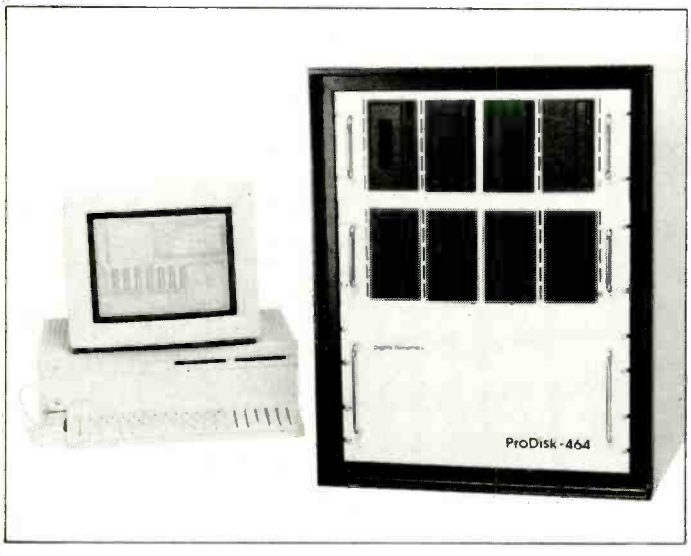
Digital Dynamics recording/ editing system

Digital Dynamics demonstrated a 24-track hard disk recording/editing system at the NY AES. The *ProDisk-464* is a 4- to 64-track recording and editing system that the manufacturers describe as suitable for all audio production areas. The system is controlled through an Apple *Macintosh* computer and displays a transport console, level meters and an editing console from multitrack reel-to-reel hardware. Editing operations include copy, cut splice, replace and align. Identical edits can be made across single or multiple tracks. Sound libraries may be compiled and auditioned in realtime.

On-screen editing facilities include crossfades, slipping, reel-rocking, auto record and individual track realignments. The system also automatically builds edit decision lists in realtime. *ProDisk* can lock to SMPTE/EBU timecode, has MIDI compatibility and accepts AES/EBU digital inputs. The minimum configuration is four tracks, and one disk provides 30 minutes per track recording time and additional drives may be added up to three track hours. The hard disks are on a SCSI bus and back-up is to 8 mm cartridge. Upgrading is available in 4-track sections.

Digital Dynamics Inc, 270-02 East Pulaski Road, Greenlawn, NY 11740, USA. Tel: (516) 271-5600. Fax: (516) 271-5607.



Agfa XT restoration process

Agfa have introduced an advanced tape restoration process that restores deteriorating tapes for playback, something that is particularly important at the present time with much remastering of back catalogue for CD release. *Agfa-XT* allows badly shedding and flaking tapes of all brands that would normally be considered unplayable to be played back on a standard tape machine without damaging the heads. The recordings must then be transferred onto a new tape reel for futher work or storage.

XT is a five-step process developed by Agfa Technical Support Services in conjunction with the Agfa Technical Centre in Munich and is being offered as a service through the Agfa US company headquarters. The archives of two major record labels are currently being processed with *XT*. Charging for the service is on a time basis and is not connected to the tape width.

Agfa-Gevaert AG, D-509 Leverkusen, West Germany.
USA: Agfa Corporation, 100 Challenger Road, Ridgefield Park, NJ 07660. Tel: (201) 440-2500.

Lexicon Opus/e and 480L software

Lexicon have introduced a development from the *Opus* Random Access system. The *Opus/e* is largely the editing and disk-based recording system of the full *Opus* but without the mixing and signal processing capabilities. It will function as a standalone editor or can be used in conjunction with an existing console. *Opus/e* can be upgraded to a full *Opus* as well as operate as a satellite accessing the system from other rooms in the facility. All the multitrack edit functions including the edit audition facilities of the full *Opus* remain. Analogue and digital inputs and outputs are simultaneously available as are the electronic patching I/O capabilities. The *Opus/e* consists of the edit controller, a colour coded keyboard and a monitor with the ability to rackmount the controller and keyboard or install directly into a console.

Enhancements were also announced during the AES Convention for other Lexicon products. New software for the *Opus* allows recall of up to 10 complete groupings and console status configurations including channel fader, mutes, pans and sends. There are also possibilities for different relations between the group master and the slave such as relative, direct, inverse and no connect. EQ grouping is also possible. There have also been changes to the recorder interface to allow manual record in/out as well as auto record against timecode values.

For the *480L* effects processor, Lexicon have introduced new software enhancements in cartridge format, Program Cartridge 1.0 features four new algorithms: Ambience, designed to control room ambience by positioning the direct signal within the ambience and tailoring the reverb of the space; Random Hall, enhanced reverb programs based on the Hall programs in the *480L* but with the addition of spread and wander; Panorama, giving the ability to 'spread out' reverb and effects before they are mixed with the main signal; and a Stereo Digital Compressor/Expander with full parameter control.

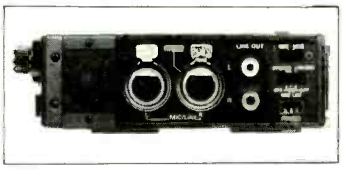
Lexicon Inc, 100 Beaver Street, Waltham, MA 02154, USA. Tel: (617) 891-6790. Fax: (617) 891-0340.
UK: For Opus products, FWO Bauch Ltd. Tel: 01-953 0091. For other Lexicon products, Stirling Audio. Tel: 01-624 6000.

QSC MX 700

QSC have added a new model to the *MX* range based on the same design philosophy. The *MX 700* is a fan-cooled 2U chassis with front panel gain controls, clip indicators and power on/off. Rear connections include electronically balanced ¼ inch tip/ring sleeve and barrier strip inputs and 5-way binding post connectors. Output rating is 150 W/channel into 8 Ω; 25 W/channel into 4 Ω; and 350 W/channel into 2 Ω. Circuit protection is for open circuit, short circuit and mismatched loads.

QSC Audio Products, 1926 Placentia Avenue, Costa Mesa, CA 92627, USA. Tel: (714) 645-2540.
UK: Music Lab Sales, 72-74 Eversholt Street, London NW1 1BY. Tel: 01-388 5392.

Panasonic SV-255 DAT



At the recent AES show in New York, Panasonic introduced an updated version of the *SV-250* portable DAT machine designated as the *SV-255*. There are several areas of update including the newly designed mic amps offering lower distortion and a quoted S/N ration of 128 dB. There is also now a dual-channel mono recording mode where the right channel input is recorded at full level on that channel and at 15 dB lower on the left channel creating a useful backup should an unexpected overload clip the normal channel. Power consumption has also been improved over the previous model.

NB: Matsushita, parent company of both Panasonic and Technics, currently has a policy of different model designations for Europe and the US which creates a very confusing situation for international coverage. As far as we can ascertain the Panasonic *SV-250* is the same as the Technics *SV-260*, and the Panasonic *SV-3500* is the same as the Technics *SV-360*. Undoubtedly there will be small differences between models such as voltage, etc. In Europe the *SV-260* was introduced after the *SV-250* and incorporated some of the changes present in the *SV-255* and now the current models of the *SV-260* (*SV-260A*) have the same specification as the *SV-255*.

UK: Panasonic/Technics, Panasonic House, Willoughby Road, Bracknell, Berks RD12 4FP. Tel: 0344 853176.

USA: Panasonic Communications & Systems Company, 6550 Katella Avenus, Cypress, CA 90630. Tel: (714) 373-7277. Fax: (714) 373-7242.

JRF CTTC for Ampex

Otari *TC-50* centre-track timecode can now be linked to the Ampex *ATR-100* with a retrofit conversion from JRF Magnetic Sciences. Retrofits are already available for the MCI/Sony *JH110*, Studer *A80* and the Otari *MX5050* for which the *TC-50* was designed.

The process includes relapping of the record and play heads with optical realignment and the installation of a fourth head (for timecode read, write and erase) between the record and playback heads. The existing erase head is replaced with a new one that has a timecode/FM read track.

The JRF retrofit is authorised by Otari and includes the rackmount *TC-50* timecode processor calibrated to the timing requirements of the tape machine manufacture. Also included are cable interconnects, timecode alignment tape, operation/maintenance manual and all the required hardware.

JRF Magnetic Sciences, 249 Kennedy Road, PO Box 121, Greendell, NJ 07839, USA. Tel: (201) 579-5773.

DAR SoundStation II upgrades

Digital Audio Research have announced enhancements for the *SoundStation II* disk-based audio editing/record system. The system is now available with 16 channels of simultaneous input and output and record as either an option or an upgrade. Crossfades can now be recorded to disk guaranteeing playback on all channels without the crossfade restrictions present on some systems, meaning that *SoundStation* can replay 32 disk segments simultaneously with a 16-channel system. Also disk-to-disk copy to enable playback of any audio segment wherever and whenever required. Data can be bounced from one bank to another and this runs as a background task that does not interfere with working.

A completely new feature is *WordFit*. This an automatic dialogue synchronisation and replacement software that will automatically edit a replacement dialogue track to synchronise exactly with another (guide or location recorded) track. *WordFit* was originally a standalone product from DAR but is now fully incorporated within the *SoundStation II* system as an option.

Digital Audio Research, 2 Silverglade Business Park, Leatherhead Road, Chessington, Surrey KT9 2QL. Tel: 03727 42848. Fax: 03727 43532.

USA: Digital Audio Research, 6363 Sunset Boulevard, Suite 802, Hollywood, CA 90028. Tel: (213) 466-9151. Fax: (213) 466-8973.

Studer D820-48 48-track DASH

Studer unveiled a pre-production version of their long-heralded digital multitrack at the New York AES. The *D820-48* is a DASH format machine fully compatible with other DASH multitrack machines, both 24- and 48-track, with tracks 1 to 24 in the standard normal density DASH format and 25 to 48 in double density format (exactly as the Sony 48-track). The transport is based on the *820* series with 14 inch reel capacity. For remote deck control there is the standard *820* parallel interface for synchroniser control as well as a serial interface with ESbus or binary protocol. There are two thin film record heads for all 52 tracks and one ferrite play head developed to match the 4x oversampling D/A converters as fitted to the 2-channel *D820X* machine. An additional erase head is provided for the four aux tracks to maintain full compatibility with 24-track DASH tapes. There is provision for an 'advanced output mode' to compensate for delays in external processing equipment. Each channel is equipped with high speed DSP for crossfading, switching, error correction and concealment. A penthouse over the deck has a 30-LED ppm meter for each channel with facilities including bouncing, group programming, individual emphasis settings, programmable user functions, master safe, auto mute and auto input. It can also store four complete setups of the panel. Different digital I/O formats are selectable with the machine supporting AES/EBU, MADI and SDIF multichannel with 2-channel AES/EBU format assignable to any two tracks.

There is a test mode with an internal DSP generating test signals that can be injected into signal paths for troubleshooting. An audio memory system is in development, which is described as 'offering new and powerful editing features'.

Delivery of production machines is being planned for mid 1990.

Studer International AG, Althardstrasse 150, C-8105, Regensdorf, Switzerland. Tel: 01 840.29.60. Fax: 01 840.41.71.

UK: FWO Bauch Ltd, 49 Theobald Street, Borehamwood, Herts WD6 4RZ. Tel: 01-953 0091.

USA: Studer Revox America Inc, 1425 Elm Hill Pike, Nashville, TN 37210. Tel: (615) 254-565.

Voyetra Sequencer Plus Version 3.0

This popular IBM sequencer package for the IBM *PC* and compatibles has been significantly upgraded with the release of this new version. Voyetra are now referring to it as an 'integrated MIDI control program' and have reduced the retail price by nearly 50%.

Version 3.0 integrates an enhanced version of the original Sequencer Plus with Patch Master Plus—their Universal Librarian. MIDI Data Analyser program—and Conversion Plus—a Sequencer Plus to Personal Composer Songfile Converter—to form a very powerful combination.

On a typical recording session the program aims to enhance your productivity in the studio by providing a uniform environment within which you can create songs, manage your MIDI instrument sounds and patches, and diagnose MIDI network problems. These tasks were previously accomplished by switching between multiple programs, interrupting the creative process.

The Universal Librarian functions are predefined for nearly 100 MIDI instruments from over 20 manufacturers, replacing expensive RAM cartridges and inconvenient cassette tapes with the *PC*'s disk logging power. It allows patches to be auditioned, named and grouped into banks in any order, making it easy to create an organised inventory of sounds for every supported instrument.

The Network Organiser allows the *PC* to download patches from disk to 32 MIDI instruments with a single keystroke. Multiple instrument settings may be saved as a Setup specific to the song, assuring that the correct patches will always be loaded in each device for every composition performed with the sequencer.

The MIDI data analyser allows MIDI data streams to be viewed, captured, analysed, saved to disk and retransmitted. In addition to these diagnostic functions, multiple MIDI data strings may be defined and transmitted from the *PC* keyboard for directly controlling MIDI devices.

Sequencer enhancements include Trackscan, which allows the number of sequencer tracks to be defined when the program boots. Over 3,000 tracks can be used and the program defaults to 64 when Trackscan is not used.

Comment: The Voyetra *Sequencer Plus* has long been established as the best sequencer available on IBM-compatible machines. These new features include just about everything you need to allow you to have complete control over your entire MIDI setup in the studio. And the price has dropped substantially as well.

Voyetra Technologies, 333 Fifth Avenue, Pelham, NY 10803, USA. Tel: (914) 738-4500.

UK: ADT, 5-7 Buck Street, London NW1 8ND. Tel: 01-482 5224.

Studio Sound's Music News is compiled by Mike Collins.

Sweetwater Sound SW800 soundblocks for Kurzweil K250 expander/RMX units

The *SW800* soundblock works with the Kurzweil *K250, Expander,* and *RMX* units. Featured instruments include the Yamaha *DX7* electric piano, wurlitzer electric piano, orchestral hits, orchestral glockenspiel, harmonica, Fairlight-type and Roland *LA*-type sounds, drum sets and basses, and more. The *SW800* adds 55 new keyboard setups to existing sounds.

This new soundblock consists of 10 IC chips that install just like the factory blocks A, B, C and D, and can be installed in any machine regardless of age or current options. The *SW800* becomes resident and can be edited and combined like all other Kurzweil blocks.

The *SW800* is the first non-factory soundblock to be developed by any company and more custom blocks are planned.

Comment: Sweetwater have chosen a 'workhorse'-type selection of sounds for their first soundblock for the Kurzweil *K250*. The Kurzweil units are popular with professional users because of their sound quality and existing users will almost certainly wish to obtain one of these new soundblocks to extend their range of available sounds.

Sweetwater Sound Inc, 4821 Bass Road, Fort Wayne, IN 46808, USA. Tel: (212) 432-8176.

E-mu Systems

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Editor/Librarian software packages have been developed by Opcode Systems for the *Macintosh* and Atari *ST* computers, and other manufacturers are working on versions for the IBM and *Amiga*

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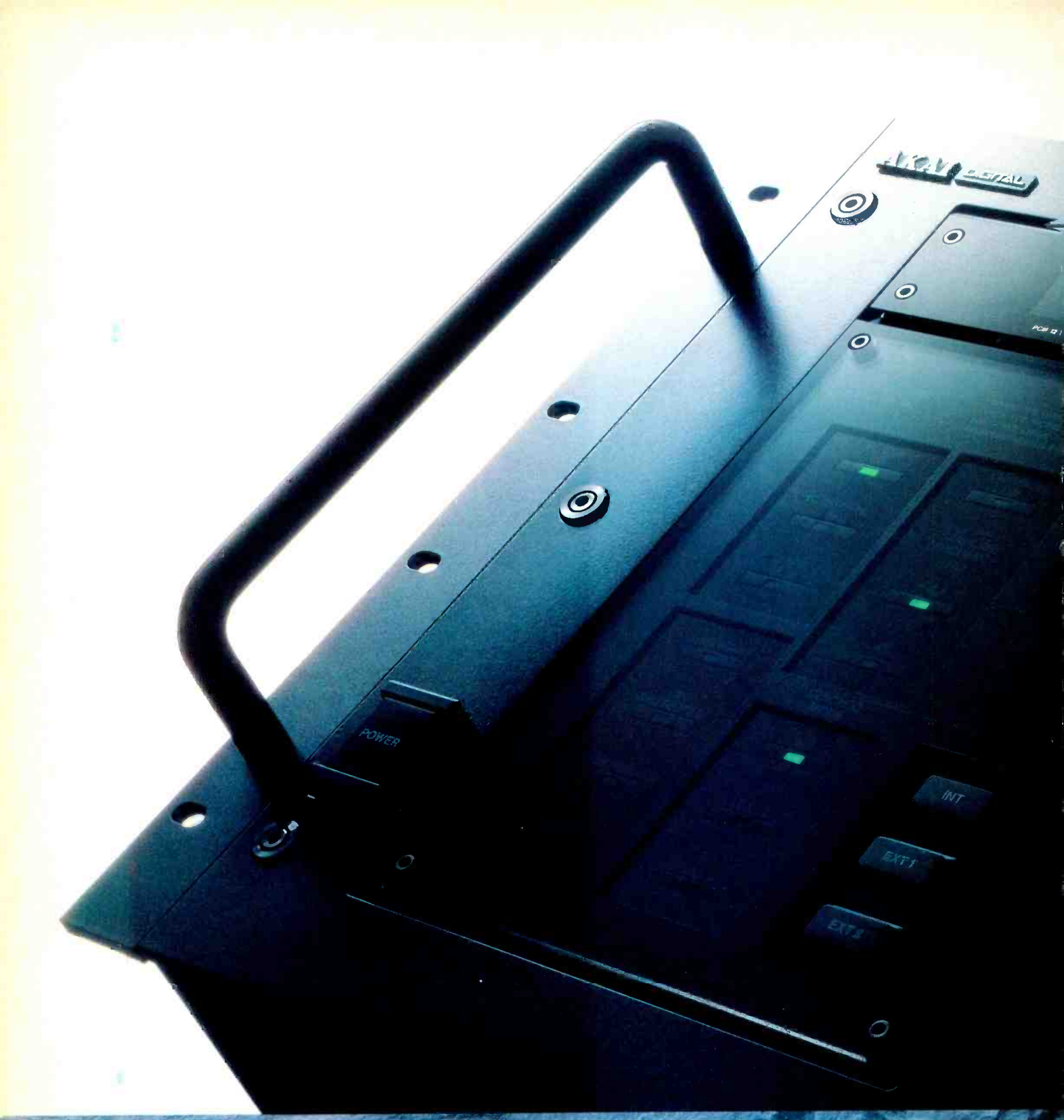

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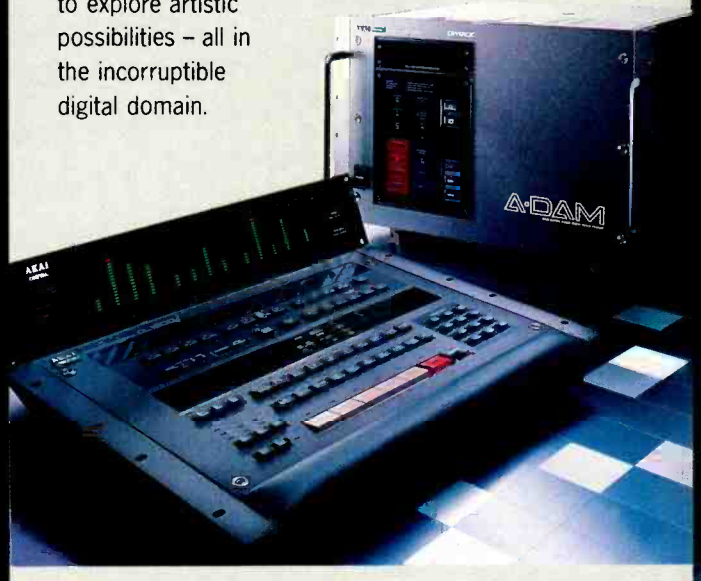

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AES/EBU INTERFACE COMES OF AGE?

Francis Rumsey looks at some of the aspects of this digital interface standard and its complexities

It would now be reasonable to accept that the AES/EBU digital interface has become the standardised means for transferring 2-channel digital audio over medium distances. Individual manufacturers may retain their generic interfaces but these may be gradually superseded as time passes. The AES/EBU 2-channel interface has been standardised by the IEC as IEC 958 (1989). A number of interesting issues now surround the interface, some of which will be discussed here, firstly concerning the relationship between the consumer and professional versions, secondly concerning the problems of decoding and synchronising AES/EBU data and thirdly concerning various multichannel offshoots based around MADI.

AES/EBU revised

To recap, for those who are not familiar with the interface, the AES/EBU format allows for two channels of digital audio to be transferred serially over one balanced interface, using *RS422* drivers and receivers. In addition to up to 20 bits of audio data, the AES/EBU subframe (**Fig 1**) contains a synchronising pattern at the start of each subframe (to identify the data as A- or B-channel, as well as delineating the channel status block boundary), four bits of auxiliary data (which may either be used for additional audio resolution or for other purposes such as low quality speech), a validity bit, a user bit, a channel status bit and a parity bit, making 32 bits per subframe (one subframe per audio channel) and 64 bits per frame.

One frame is transmitted in the time period of one audio sample, and thus the data rate varies with the sampling rate (this is different from MADI, see below). Channel status bits, carried one in each subframe, are aggregated at the receiver to form a 24 byte word every 192 frames, and each bit of this word has a specific function relating to interface operation. Examples of bit usage in this word are the signalling of sampling rate and pre-emphasis, as well as the carrying of a sample address 'timecode' and labelling of source and destination. A number of uses have been suggested for the user bit, including the possibility of a high level packet system for labelling audio data.

The interface allows two channels of audio to be transferred over distances up to 100 metres but longer distances may be achieved using combinations of appropriate cabling, equalisation and termination. It terminates in standard *XLR-3* connectors, which will look for all the world like analogue audio connectors, except that, on equipment, they may be labelled DI (Digital In) and DO (Digital Out).

The consumer/professional relationship

When the IEC standardised the 2-channel interface, two requirements existed: one for 'consumer use' and one for 'broadcasting or similar purposes'. There may be considerable advantages in the fact that a single IEC standard has resulted with only subtle differences between consumer and professional implementation but the very fact that the differences are subtle has led to some confusion over which pieces of domestic equipment may interface to which professional devices and *vice-versa*.

The consumer format interface is sometimes known as the SPDIF (or Sony/Philips Digital Interface) and can be found on many items of semi-pro or consumer digital audio equipment, such as CD players and DAT machines. It usually terminates in an RCA phono connector. As stated above, the professional interface terminates in *XLR* connectors. In many ways, the data format of the two interfaces is the same, the incongruities lying mainly in the electrical differences between them but it is necessary to understand how the channel status implementation differs to see where incompatibilities may arise when trying to connect digital equipment together.

Firstly, it is important to itemise the differences between the electrical interfaces. The professional interface is shown in **Fig 2a**, whereas the consumer interface is shown in **Fig 2b**. It can be seen that the professional interface is a fully balanced circuit, whereas the consumer interface is an unbalanced circuit. The pro interface involves a transformer at both ends (although there is a small amount of disagreement about the need for this) whereas the consumer interface has a transformer only at one end. Interestingly, the consumer interface is considered by many to form a better transmission line than the pro interface, because it specifies the need for 75 Ω termination and 75 Ω cable, whereas the professional interface has to accommodate a wide range of cable types and is required to use *XLR* ▷

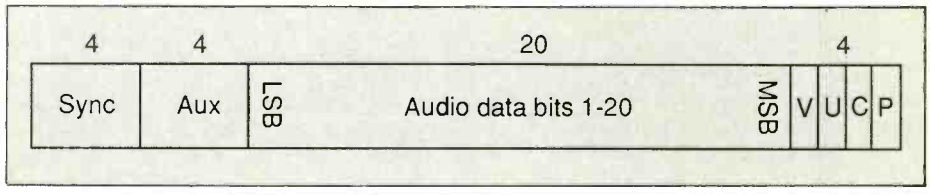

Fig 1: AES/EBU subframe format

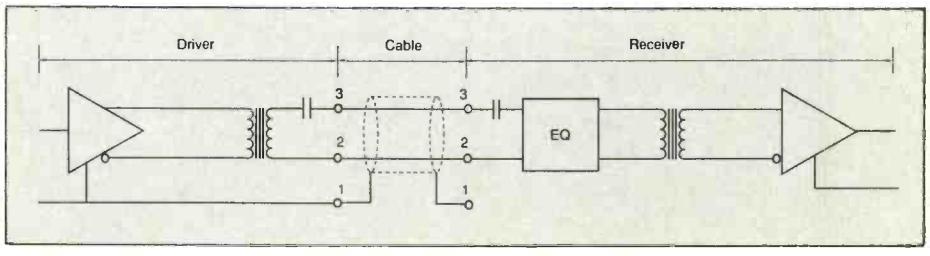

Fig 2a: Professional format electrical interface

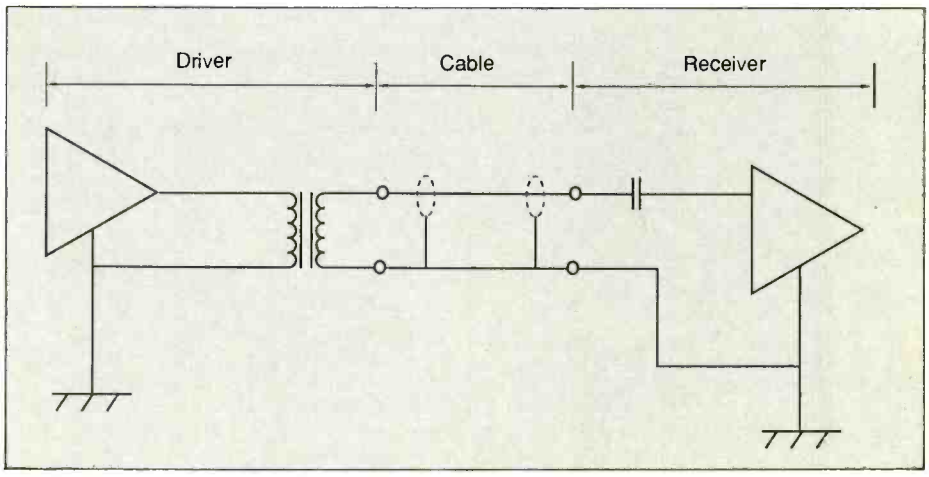

Fig 2b: Consumer format electrical interface

connectors (preventing the specification of a standard characteristic impedance, although a nominal 110 Ω is recommended).

Concerning the possibility for consumer/professional interconnection using the AES/EBU interface, one should note that the consumer version operates at peak-to-peak voltages of around 0.5 V, whereas the professional version operates around TTL levels (5 V). Nonetheless, the professional receiver must accommodate a minimum eye-height of 200 mV, thus most professional receivers will be able to accept signals from consumer interfaces, provided cable and termination losses are not too severe. The real problems are not in getting electrical interconnection to work but in ensuring some degree of compatibility in the data format.

Looking back to **Fig 1**, it will be seen that the subframe format is the same for consumer and professional interfaces, in that both formats carry data grouped into this 32 bit pattern. Whether or not devices will communicate correctly depends on a number of factors, assuming that the electrical interfacing question has been addressed. The main stumbling blocks come in the question of what the receiving device expects to see in terms of the status of bits both in the body of the subframe and in the channel status information.

In theory, consumer and professional uses of the channel status block are different, with the first bit of the channel status block in the professional format being set to a '1', whereas the consumer format is set to a '0', followed by five control bits. Devices intending to accept both consumer and professional data should be able to recognise this bit and alter their interpretation of channel status accordingly, otherwise strange things may happen. In practice, devices vary as to how comprehensively they have implemented either mode and may ignore many of the bits in the channel status block, and it should also be borne in mind that some devices may simply refuse to do anything if they see the first bit of channel status set to the opposite format from their own.

The second byte of channel status in the consumer interface has been set aside for the indication of 'category codes', these being set to define the type of consumer use. Currently defined categories are (00000000) for the general category, (10000000) for compact disc, (01000000) for a PCM adaptor (such as a Sony *PCM 601*) and (11000000) for a DAT machine. Once the category has been defined, the receiver is intended to interpret certain bits of the channel status word in different ways, depending on the category. For example, in CD use, the four control bits from the CD's 'Q' channel subcode are inserted into the first four control bits of the channel status block (bits 1-4).

Historically, a number of bits in the format have been left intentionally undefined in order that their use could be defined later. Furthermore, new uses for bits have been defined since the original specification and some equipment may not recognise these. Problems arise when a receiver is expecting certain bits of the channel status world to be set in a particular pattern, locking out any data that does not conform to this exact pattern. Conversely, other systems may be very loose in their AES/EBU implementation and will accept virtually anything, provided that it looks like AES/EBU data. There is a danger in the latter approach that audio data will be transferred with something set wrongly (pre-emphasis for example), whereas a more rigorous implementation would ensure that transfer only took place if every bit were set in the correct manner.

Depending on the application it might be suggested that there are cases where the user will want devices to talk, no matter what the niceties of channel status (in order simply to assess the viability of communication), whereas in other applications the rigorous interpretation of channel status will be necessary. Some have suggested that there would be considerable value in the AES/EBU equivalent of an *RS232* 'break-out' box, which could be inserted into a linkage to allow various bits to be manually set or reset until satisfactory communication was established (in other words, to con the receiver into accepting data it would otherwise have rejected).

A final, and rather worrying point to be made is that there have been instances in which manufacturers have taken a consumer interface chip (a consumer serial output is understood to be provided on a particular Philips oversampling D/A converter incorporated in some systems) and balanced the output, terminating it in an *XLR* connector, making an interface that masquerades as a professional interface but with consumer channel status. From the previous paragraphs it should be clear that the adoption of such a method is likely to end in tears.

Decoding and synchronisation

If a number of digital devices are all to be connected together using the AES/EBU interface, then the question of synchronisation will arise, since all devices must be operating at the same sampling rate, with very small tolerances. Additionally, if devices are to be able to receive data from remote locations (say from another room in the same building), the problems of decoding AES/EBU signals that have been degraded during transmission must be addressed.

The concept of synchronisation is not really inherent in the nature of analogue audio, because analogue waveforms are essentially time-continuous, but in digital systems where information is time-discrete the subject becomes more important. Unless two devices that are to communicate are locked to the same sampling rate clock, they will drift in relation to each other. In the simple instance where you are copying a digital tape from one machine to another synchronisation is usually achieved by the recording device deriving its sample clock from the replay device, either using a separate word sync connection or by deriving the clock from the incoming data.

In the AES/EBU interface, a clock is encoded with the data, such that a transition from high to low or *vice-versa* occurs once every clock period (biphase mark coding). A receiving device should be able to extract clock information from incoming data and this may need to be 'cleaned up' to provide a stable clock signal to the rest of the system, since there may be considerable 'jitter' on the decoded clock. Jitter, in fact, is a major evil in digital audio interfacing, as it blurs the edge of bit cells making it difficult to determine where one ends and the next begins and, if passed on to the conversion circuitry, it will add noise to the eventual output.

It is intended by the AES that two grades of sync reference should be adopted, the sync reference supplying a centrally distributed signal that all digital devices in a system will be locked to, having a very fine tolerance in its frequency. In this way it could be ensured that all sample clocks might be aligned, thus the only form of ▷

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◁ synchronisation required at receivers would be the need to perform alignment of samples whose position had changed due to timing delays in propagation (since 'synchronous' signals are to be defined as those whose sync preambles lie within a 25%, or ¼ sample period, window of the reference). Drift would not be a problem. The two grades of timing reference intended are: Grade One, intended for the timing of a large studio complex and operating to an accuracy of ±1 Part Per Million (PPM), and Grade Two, intended for synchronising the operation of a single studio with an accuracy of ±10 PPM.

The synchronising signal would take the form of an ordinary AES/EBU data stream, carrying audio or not, distributed to all operational areas, much as video sync pulses are distributed in TV operations. Thames Television, at its Euston site in London, is installing a complete AES/EBU network for the distribution of stereo audio around the complex, and such synchronising signals are used in this instance to ensure common timing of all sources.

It may be of interest to note that the BBC has designed an integrated single-chip device for handling AES/EBU interfacing, known as AESIC, which is to be commercially available and is expected to retail at around £26 ($40) in quantities of 1,000 off. This chip takes into account many of the decoding difficulties often encountered with AES/EBU signals and allows decoding of signals with any incoming phase relationship to a master clock, which is likely to be useful in large installations with timing problems.

Multichannel interfacing

Although not finalised as an international standard at the time of writing, a number of manufacturers have combined forces to propose a multichannel interface standard known as MADI (the Multichannel Audio Digital Interface), which is based on the 2-channel AES/EBU interface and is designed to be transparent to such data. MADI, for the record, is nothing whatsoever to do with MIDI, which is a control interface for musical instruments.

MADI has much in common with the AES/EBU standard described above, as the information carried for each channel is in a very similar format, except that it naturally uses a higher data rate to carry the increased amount of information. The format has been submitted to the AES and EBU for consideration as a multichannel standard. MADI allows for 56-channels of audio to be carried serially over one 75 Ω co-axial cable, such that one sample for each of the 56 channels is transferred within one audio sample period. Each subframe within the 56-channel MADI frame carries a 32 bit word, of which all but the first four correspond to the 2-channel format (**Fig 3**).

Unlike the 2-channel interface, the transmission data rate of MADI is fixed at 125 Mbit/s, irrespective of the sampling rate or number of channels, and the actual data transfer rate is 100 Mbit/s due to the use of a 4 to 5 bit encoding scheme. In this channel code each 32 bit subframe is broken up into 4 bit words, which are then encoded to 5 bit words according to a look-up table, the reason being to maintain a low DC content to the code. A synchronisation symbol is inserted at least once per frame, and in cases where the full bandwidth of the link is not being

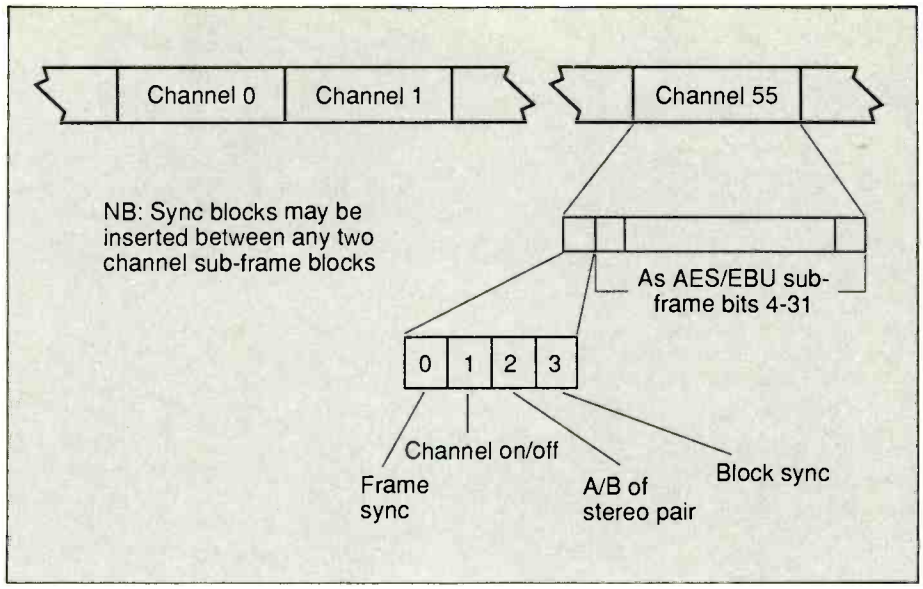

Fig 3: MADI frame construction

used, additional sync symbols are inserted to take up the capacity of the bus. Communication is handled entirely by TAXI chips (Transparent Asynchronous Xmitter/Receiver Interface), which take care of the insertion of sync symbols.

Unlike the 2-channel format, it is intended that the link will be asynchronous, accompanied by a separate synchronising clock, as it is not intended that the data be self clocking. BNC connectors of 75 Ω are to be used and the maximum cable length is intended to be no more than 50 metres. The modulation method used is to be NRZI, in which a transition from high-to-low or low-to-high represents a binary 'one', and no transition represents a zero. This differs from the bi-phase mark coding method, used in the 2-channel interface, which is self-clocking, having a transition at every bit-cell boundary, as well as a transition within the bit cell in the case of a binary 'one'.

Because of the asynchronous nature of the interface, buffers are required at both ends of the link so that data can be re-clocked from the buffer at the right rate (**Fig 4**). At the receiving end, the data is clocked out under control of the synchronising signal that may accompany the MADI audio data.

As digital mixers and multichannel tapeless systems grow in popularity, the need for a multichannel digital interface will also grow. A format such as this has wide applications for interface between digital tape machines or tapeless workstations and digital mixers, and 56 channels is likely to be an adequate capacity for all but the most demanding systems, in which case a second interface could be employed.

As a development that goes even further than MADI, the BBC is working on an experimental distribution system for 224 channels of digital audio, using optical fibres to accommodate the wide bandwidth required. In this system, four MADI multiplexes will be multiplexed together over a single link operating at 500 Mbit/s (0.5 Gbit/s). The multiplex will be distributed to a number of locations within Broadcasting House using passive optical splitters and rackmounted terminators will be provided with individual AES/EBU format inputs and outputs. Selection of signals is then a matter of demultiplexing the desired channel(s) from the 224-channel multiplex, rather than the electrical switching of 'hard' channels.

Conclusion

A number of aspects of AES/EBU interfacing have been described and from this it should be clear that not only is the interface a comprehensive and standardised means of communication but also that considerable care must be taken in its implementation. Manufacturers must be aware of the wide range of possibilities inherent in the interface and make equipment that is able to accommodate data originating from a wide range of sources. As applications become more complex, the need for rigorous interpretation of the standard will become important, as will the need for very clear definitions of the different modes.

If anyone is interested in designing the 'break-out' box mentioned above, they could well find themselves a considerable market!□

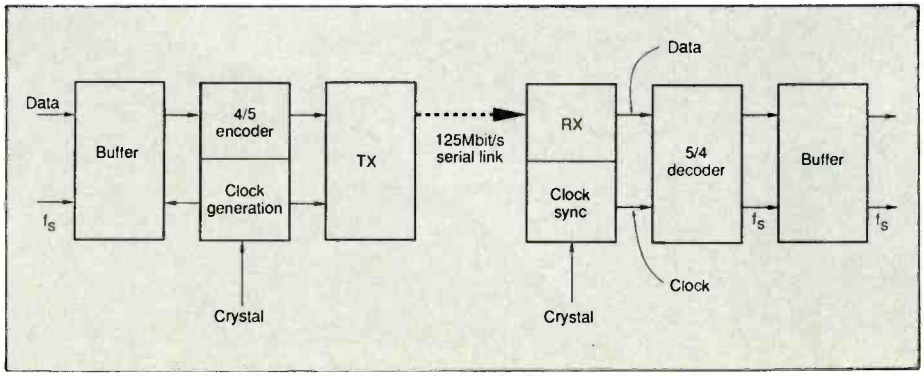

Fig 4: MADI buffering arrangement

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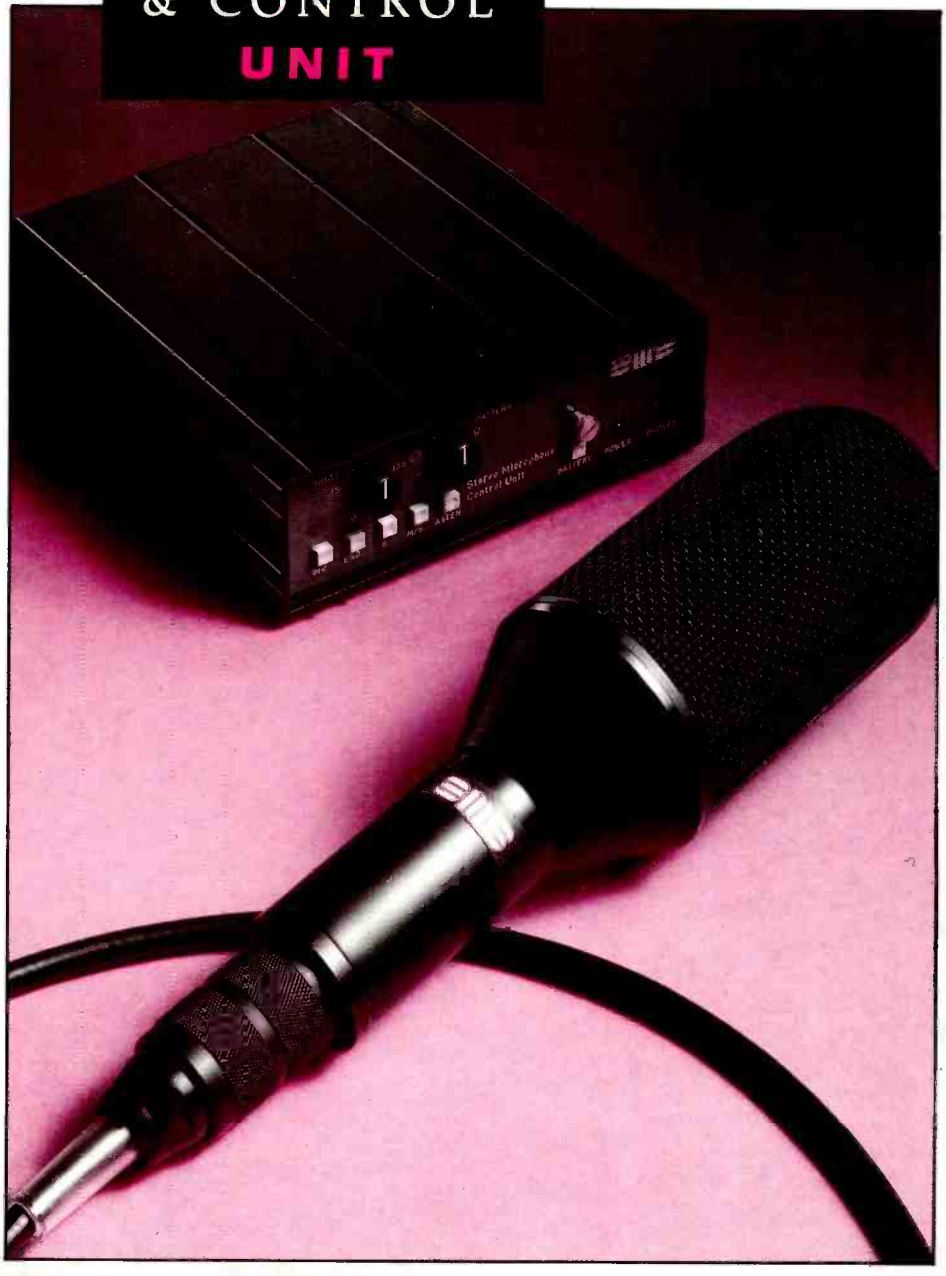

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MADI INTERFACE AN OVERVIEW

Stella Plumbridge and Yasmin Hashmi interviewed Alan Jubb and Paul Lidbetter of Neve to discover how the MADI standard has developed and to review its technical specifications

Currently, a great deal of material that ends up as a digital master will have passed through the analogue domain at some point in the production process. The most common area that still generally requires analogue audio is the mixing process. However, the continuing increase in the use of digital recording will inevitably lead to greater demands being made on sonic fidelity and will highlight the basic requirement to keep audio in the digital domain from source, through editing and mixing, to master.

Stereo interfaces such as AES/EBU and SDIF-2 have been around for some time and are only now being used widely as a means of transferring digital audio. However, until recently there was no standard interface available for multichannel digital audio and this has been a cause for concern among users and manufacturers.

Is it possible then, for a group of pro-audio manufacturers to get together, agree on a set of common objectives and provide working hardware inside a year—without the motivation of direct financial benefit? The answer is 'yes' and this is exactly what the MADI group have achieved. MADI stands for Multichannel Audio Digital Interface and the MADI design team are from these UK-based companies:

- Mitsubishi Electric UK
- Neve Electronics International
- Solid State Logic
- Sony Broadcast & Communications Ltd

History

The MADI story began because of difficulties in interfacing DSP equipment with digital multitracks. This prompted Neve and SSL to approach Mitsubishi and Sony with the idea of a co-operative venture to help solve these problems. The response was so favourable that the group was able to have its first formal meeting only a month later. Their objectives were to design a multichannel digital audio serial interface that would:

- have a high probability of being accepted as an industry standard
- be based upon readily available components—not on designs owned by any of the MADI group members
- not require a long development cycle
- be economical to implement
- be transparent to the AES/EBU twin-channel interface
- be easy to understand
- accommodate at least 32 channels

It is important to appreciate that the group set out only to provide a solution for transferring multichannel digital audio and not make a profit from it. To this end the interface design is made up of readily available components. However, the group hopes to benefit from the fact that providing such a solution helps in the general promotion of the technology.

Naturally since the group have invested time and effort in developing the standard, they want it to be as widely used as possible. Hence the criteria for economic implementation and again, a design based on readily available components. There was also little point in developing an interface in isolation so MADI was designed to be compatible with the AES/EBU twin-channel format.

Following their initial meeting, the group met monthly and were ready to make a presentation to the European AES Technical Sub-Group of the Digital I/O Interface Working Group only 5 months later. MADI was then accepted for standardisation at a meeting of the full Technical Working Group prior to the Paris AES, the Paris AES show itself included a stand (jointly funded by the MADI group) with working hardware based on prototype chips. The need for such an interface was confirmed by over 40 manufacturers, with equipment ranging from multitracks to effects units, registering an interest in including the interface in their products. Following the Paris AES, MADI documentation was updated by the AES to conform to the formal requirements of a standards document in order that it could be presented to ANSI (American National Standards Institute). The working hardware is now based on production chips with the full operating speed of 125 Mbit/s and has undergone EMC testing (tests against radio frequency interference).

The group are now working on a user guide, which will include hints and tips on MADI—their work is then complete. The MADI format should have been published in the *Journal of the AES* by the end of 1989. The speed with which the group have been able to work owes a lot to the close proximity of the participating companies, as well as the fact that the group is engineering led and not hindered by commercial differences.

MADI features

1 Based on industry standard silicon (FDDI)
2 56 channels of audio
3 32 kHz to 48 kHz ±12.5% (for varispeed)
4 Audio sample up to 24 bit
5 Compatible with AES/EBU twin-channel format
6 Serial point to point connection
7 Interconnection to be standard quality video co-axial cable with BNC connectors
8 50 metre cable length
9 Fibre-optic cable option for longer distances
10 Synchronisation of interconnected equipment by means of common master sample rate clock
11 Link transfer rate 125 Mbit/s, data rate 100 Mbit/s

MADI group milestones

June '87 Neve and SSL write to Mitsubishi and Sony
August '87 The MADI group have their first meeting
January '88 A private presentation is made to the chairman and European members of the Technical Sub-Group of the AES Digital I/O Interface Working Group
February '88 MADI is accepted for standardisation at a formal presentation to the full AES Technical Working Group
March '88 The MADI group jointly fund a stand at the Paris AES with a working demonstration of the interface
November '88 AES Sub-Group approve the documentation for presentation to ANSI
May '89 EMC testing completed

Technical overview

In order to keep costs down, the most attractive option was to go for a serial link—provided the link could offer the bandwidth required for the data rates involved. A serial link would also simplify connection between equipment. The MADI group researched several possibilities and found industry standard FDDI (Fibre Distributed Digital Interface) chips, which would convert data from parallel to serial and *vice-versa*, at the desired rates and with co-axial cable as the transmission medium.

Frame format

In keeping with the AES/EBU interface (which transmits in frames consisting of channel A followed by channel B), the multichannel audio information is transmitted serially via the co-axial cable link in frames. Each frame consists of 56 channels, starting with channel 0 and ending with channel 55. All 56 channels are transmitted whether or not they are being used. The frame format is shown in **Fig 1**. ▷

Channel format

The AES/EBU channel format consists of 32 bits—the first four bits are used as preamble, the following 24 bits for audio data and the last four bits for validity, user, channel status and parity bits. Given that most digital recorders currently use 16 bits maximum 24 bits may seem a little excessive, however, some systems are offering 18 bit conversion and the first 20 bit recording was recently reported using the new *DCS 900* A/D converter in conjunction with a Mitsubishi *X-86* digital tape recorder.

The MADI channel format retains this 32 bit channel structure but uses the first four bits for sync information as shown in **Fig 2**.

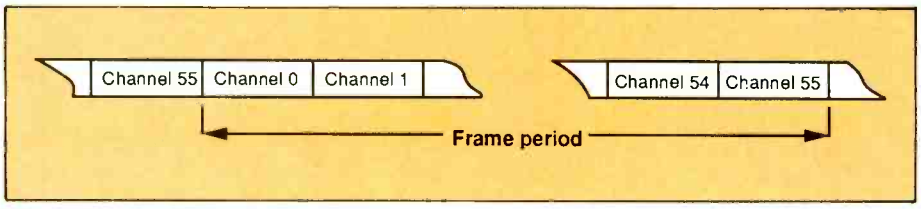

Fig 1: Frame format

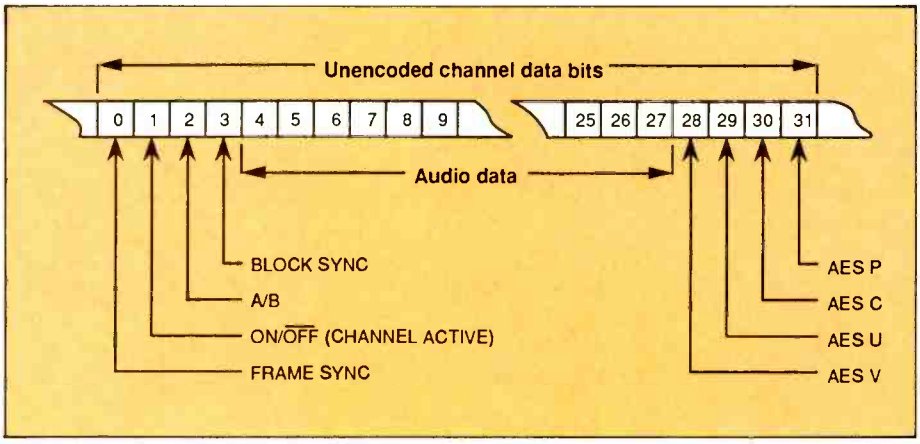

Fig 2: Channel data format

Data rates

The maximum data rate required by MADI in order to meet the specification can be calculated using the maximum allowable sample rate of 48 kHz+12.5%=54 kHz. The bit rate would therefore be 54,000×56 channels×32 bits per channel=96.768 Mbit/s. The specified data rate of 100 Mbit/s therefore easily accommodates the maximum rate actually required.

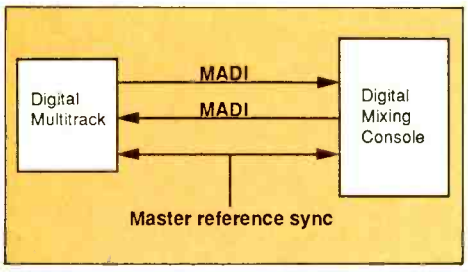

Fig 3: Example of MADI setup

Encoding

The difference between the AES/EBU and MADI formats is how the digital information is transmitted. With AES/EBU the data is bi-phase encoded, which basically means that double the number of bits are used to provide a clock that the receiver can recover and use. This results in the required bandwidth being doubled. Because MADI must transmit 56 channels rather than just two, the bandwidth required by bi-phase encoding would not be practical, so another method of encoding is employed.

This method, called 4B/5B encoding is achieved with a standard chip and involves breaking the 32 bit channel word up into eight lots of four bits. Each four bit nibble is then converted into a five bit word.

This ratio of 4/5 means that the 32 bit channel word becomes a 40 bit encoded word and the link transmission rate therefore only needs to be increased by the same ratio of 4/5, ie it is 125 Mbit/s. The transmission rate is kept constant irrespective of the sample rate.

Link synchronisation

The operation of the link is asynchronous so it is necessary for the equipment transmitting MADI data to include timing information that the receiving equipment can extract and use for synchronisation. At least one synchronising code must be sent per frame; this consists of two consecutive five bit words, which are not used in the 4B/5B encoding scheme. As already stated, the specified data rate is higher than the actual maximum rate needed—the difference between these rates allows enough space to include synchronisation codes within each frame. It should be noted that all these synchronising and coding operations are automatically handled by the FDDI chip set.

System synchronisation

Since the link is asynchronous, it is also necessary to provide a separate sample rate reference to which all interconnected systems can be synchronised. The AES working group on digital audio synchronisation favours the use of a master sample rate reference, which is fed to all pieces of equipment and allows audio data to be retimed and synchronised within specified tolerances. This standard, although 6 months behind MADI in the standardisation process, has been adopted by the MADI group and means that in addition to the MADI connection, interconnected equipment must also be connected to a master sample rate reference. A typical studio setup using MADI is shown in **Fig 3**.

Conclusion

With MADI having jumped through all the hoops required of a new interface standard, we now have to wait for the products to appear before it can be used. If comparable with the AES/EBU twin-channel format, the widespread use of MADI could take a couple of years. Neve are already developing products that include MADI and, as you would expect, they intend to incorporate MADI into any product where it is appropriate.

The lack of standards has been cited as a reason for people being either discouraged or unsure of investing in digital audio equipment. Inevitably the industry will have to standardise if it is to progress—the development of MADI is certainly a step in the right direction. It is encouraging that four major competitors can work together to provide a design which could benefit the digital audio industry as a whole.□

Bit description of channel format

Bit	Name	Description
0	MADI Channel 0	Frame sync bit—only true for channel 0, receiver detects this and resets to receive following frame
1	MADI Active	Channel active bit—all active channels must be consecutive starting with channel 0
2	MADI B/A	AES stereo B/A leg—indicates whether channel is A or B leg of a stereo signal
3	MADI Sync	Channel block sync—a block consists of 192 frames*
4-27		Audio data bits (bit 27=MSB)—can also be used for non-audio data
28	AES V	Validity bit—indicates whether audio data is valid
29	AES U	User data bit—recommended for labelling purposes
30	AES C	Status data bit—indicates which sampling rate is being used, whether emphasis is being used, whether data is audio or non audio, etc
31	AES P	Parity bit (excludes bits 0-3)

*The scheme of signalling a block of 192 frames is in keeping with the AES/EBU format where the AES C bits can be received as a data stream of 192 bits repeating every block period. The AES U bits can be received in the same way

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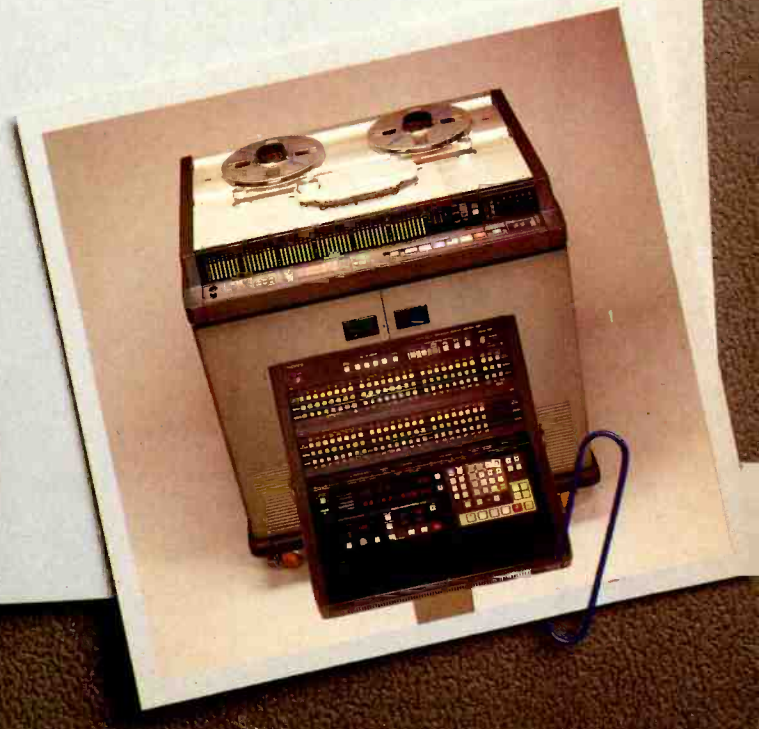

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THE SOUNDHOUSE

Dave Foister visits a London facility specialising in audio/visual soundtracks

Many recording studios have their own area of specialisation, and Paul Deeley and Phil Horne's chosen niche is audio/visual soundtracks and video sound dubbing. Their West London studio is quite clearly designed and equipped entirely with this market in mind.

The Soundhouse has two studios, sharing many common features. The first indication that this is not a music studio is the size of the recording areas, both extremely small (in Studio Two it is only 55 ft²!) with the bulk of the space given over to the control rooms. The consoles are a Soundcraft *1600* in Studio One and a TAC *Matchless* in Studio Two, both fitted with Audio Kinetics *Reflex* automation. The studios incorporate Otari 24-track machines with Dolby *A* and AK synchronisers, a choice of stereo formats, a small range of familiar outboard gear and video formats including low-band U-matic, VHS, S-VHS and Betamax. There are plenty of links between the rooms to allow 48-track work and many transfer permutations.

Deeley and Horne are proud of the specialist services they offer. Horne says, "Everybody thinks the whole world of sound recording studios centres around doing singles, albums and classical recordings, and no-one really takes any notice of the other people, like us, out there.

"We're one of a handful of studios in London that specialise in doing A/V and video soundtrack work. We specialised more in the A/V side with our client base being in A/V to start with, doing audio-visual soundtracks for car launches and corporate programmes. Then the industry developed onto doing corporate videos so we needed to be able to handle that too."

Deeley: "People have been accustomed in the past to going into a sound facility somewhere, stripping up a piece of music behind a voice and going away with what they call a soundtrack but I don't think that's the way we see it at all. Fortunately a lot of our clients are very aware of what production services we and people like us can offer, and are prepared to put time and effort into making soundtracks."

This kind of work depends heavily on the use of library music, which hasn't always had a very good reputation. However, Horne says, "Library music now is really good. Everybody's getting up to date with it, basically because the demand has been created by clients like ours who want more than was available before. Now they're digging out some good stuff and utilising it. They know more about the effects, they know what an *SPX-90* does, they know you can do these things and make things happen; they're not so ignorant about sound any more. That's why we now need to have 24-track, we need to have all the facilities that top music studios have."

Deeley agrees: "That's the overlap; we're borrowing a lot of things that traditionally you'd find in a music studio but using them in a different way, a sort of compilation way, which is basically the majority of what we're doing."

How heavily do they rely on library music? "Very; 80%. The rest is either originated here or commercially available."

In-house music recording nearly always centres on keyboards and samplers, and in fact the studio owns very few microphones, a Neumann *U87* and a recently-acquired *U47*, two Sony *C48*s, an AKG *C414*, a Reslo ribbon and a pair of Tandy PZMs.

There is very little space for music recording anyway; clearly they have designed the layout on the assumption that they will not be doing very much of that kind of work. The recording areas are on the dead side for music work, although they can be livened up by removing some absorber panels. Horne says: "It's a compromise for doing mainly voiceovers, drama with actors standing round a mic *à la Beeb*, and doing a little bit of music."

On the design side, Deeley elaborates: "We called in a consultant in the early stages to see what sort of ideas they would come up with but we had our own ideas of the way we wanted to do it anyway. The consultant who came in was frightening the life out of us with the sort of figures he was coming back with and we had to tailor it to suit the budget eventually. Practically it works fine."

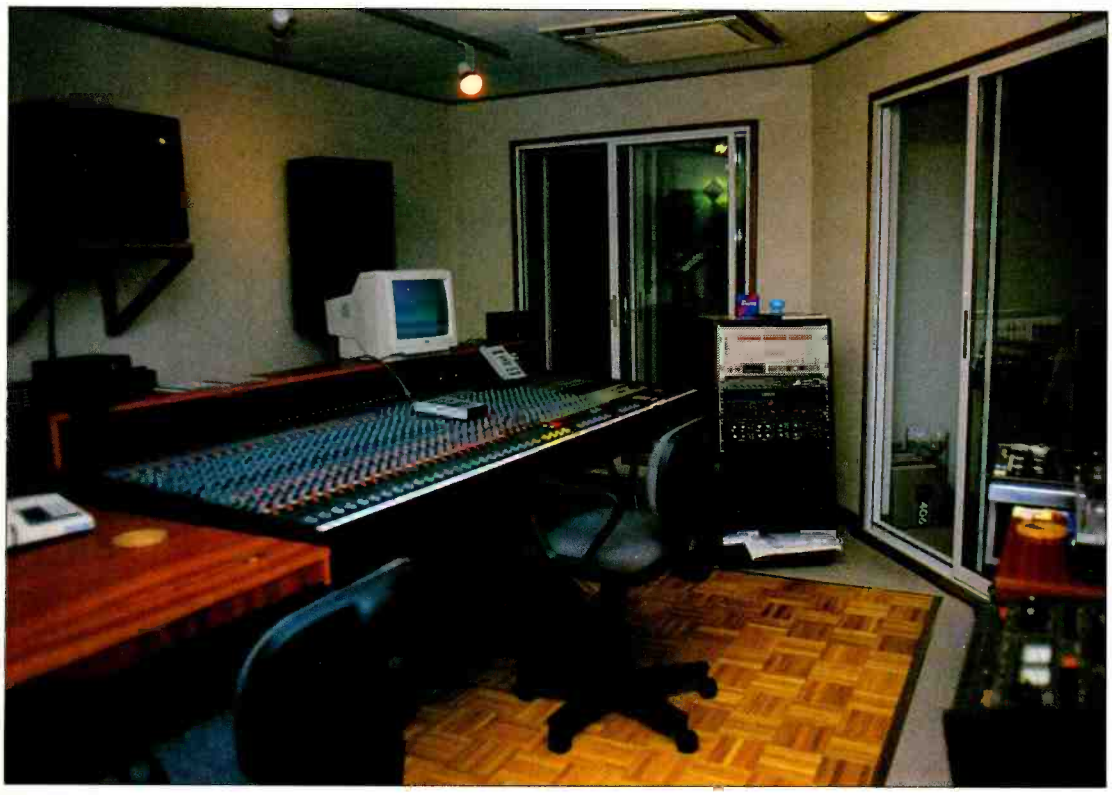
Horne: "It's a floating room concept, with no contact between studio and control room. We don't monitor at ridiculously high levels anyway."

Monitoring incidentally is on Tannoys in both rooms, chosen after extensive listening to the competition.

So, what is the nature of their work. Horne: "Most of our soundtracks are compiled off CD, which is why we've got the two Technics *SL-P1200*s. That machine is the mainstay of our work; without that we're stuffed. That to us is like a word processor to a journalist. That and a microphone is all we really need with a multitrack machine.

"The quality of soundtracks is improving because of CD. None of the libraries are producing vinyl any more—they've all gone to CD—and the old idea of 'you'll never notice when it's on VHS' is all going out the window now as people are realising that you do notice."

The Soundhouse's library of music and effects is extremely large—around 50 effects CDs, the same again on vinyl, and many more again of library music in both formats— ▷



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◁ and obviously a lot of the speed with which a job gets done is going to depend on how well they know those libraries.

Deeley: "That's a very good point. The producers tend to lean fairly heavily on our knowledge of the libraries—they're not working with them day-to-day whereas we are."

Deeley and Horne met while working at Sarner Audio-Visual. Eventually they both went freelance, separately, but found they were working in very much the same area and passing clients back and forth. This finally led to them setting up their own facility. As Deeley says, "Individually and together we were spending a lot of money dry-hiring studio time around town simply to service our clients, which was the reason for setting up our own thing. The clients expect it after a while, they like to see some substance."

There was never any doubt as to what the market would be. Horne says "We already had the market, we were already getting work. When we went freelance clients went with us, so we'd already found our niche in the market. We enjoyed doing the A/V stuff and people were working with us so we must have been doing it right."

The specialisation led to some untypical equipment priorities, such as the video and synchronisation facilities, and even though music recording was not planned, multitrack was held to be essential.

Horne explains: "When we set up, the budget was very low but 16-track was the minimum we would get away with. We needed to have the tracks because some of the jobs we do have multi-language versions, with a common soundtrack and then eight different languages, plus synchronisation codes and clock codes for the different A/V control systems. It's not that rare to have a 24-track with four tracks of different codes."

They often also provide soundtracks for conference and trade show presentations with multichannel audio on site, for which their own experience of the live situation is obviously useful. As Horne says, "On the bigger presentations we work alongside the live sound guys—they often come down for the mix, for pre-production meetings and discussions."

These contacts enable them to be more imaginative and flexible; for example, says Deeley, "Something we're doing from time to time for the better-organised show sound people is putting the voice on a separate track; we give them premixed music and effects, so they can actually mix the voice in."

To an extent The Soundhouse's choice of equipment is dictated by the need for standardisation with other similar facilities. This influenced the choice of synchronisers, particularly since the studio is often dry-hired to other engineers who like something familiar to work with. It also meant 24-track was essential from an early stage. Horne explains, "We don't have a 1 inch video machine here; we synchronise to low band U-matic—basically because we can't afford to spend £60,000 on a machine that's going to sit here for 90% of its life doing nothing—so we get our video layoffs and laybacks done at various video houses. They will layoff any audio that we need to use off a 1 inch master straight onto 24-track with timecode; we do all we need to do then mix it onto a couple of tracks on the 24-track and send it back."

For the same reasons they use ¼ inch centre-track timecode machines, rapidly becoming a standard in the industry.

"The logical next step is to go for DAT with timecode. We're waiting to see exactly what happens with the Fostex machine and who else is going to come out with one—we're not going to make any rash moves yet."

Another area of change to which they may have to conform is noise reduction. "Dolby *SR* is coming into our business now. The old standard for A/V soundtracks was a 4-track Teac with stereo audio on tracks 1 and 2 and two tracks of programming clocks, with dbx. Then people started looking at Dolby *SR* and it's becoming a bit of a trend now for the higher-quality shows."

They don't yet have their own *SR*—it was on the budget for Studio Two until they won their second *Reflex* computer at this year's APRS and had to buy the rest of the system—but they aim to have it so that they can offer what a lot of other studios don't.

What about hard disk? "For what we do it's a lot slower to work that way. It's a lot of number-crunching and you end up losing the creative flow. We're not just engineers in our studio

by any means; people come to us because we construct a soundtrack with them. With hard disk and things like this you're becoming like a lot of video editors, just pushing buttons."

Of course, much of the gear gets used rather differently from its application in a music studio. A good example of this is reverb and effects, where the need is to match the acoustic of a dub to the pictures. For this reason small natural rooms are used far more often than the larger more musical ones.

The work is more varied than might be imagined. The schedule might include dubbing work for a broadcast company (Yorkshire TV are recent clients), complete sound build-up to mute pictures, and conventional voiceover dubbing, often in other languages. At the time of this visit, Deeley had done "two or three sessions of that this week. Yesterday we did a Hebrew commercial for a vegetarian schnitzel dubbed into English. It's interesting; you get to meet lots of different approaches, lots of different production expertise."

Like any area of recording it requires its own specialist areas of knowledge in order to communicate effectively with the clients. As Horne points out "We have to understand their storyboards; we have to understand their language when they say we've got a three-stage zoom here or whatever.

Although the view of sound as the poor relation to the visuals is happily less prevalent than it once was, The Soundhouse often finds it is still having to battle to get it taken seriously as an equal contribution to a project, and as a specialist area requiring specialist facilities. Deeley says, "One of the heartbreaking things is when a track comes in which we have to start again, which has been made in a video edit suite by people who are picture specialists, not sound specialists. What we're trying to do is to persuade people to come to a sound facility and do a proper soundtrack."

Horne: "Go to a sound studio for your soundtrack and a video studio to do your video editing. You go to the baker's to get your bread and the cobblers to get your shoes mended."

Deeley points out that, "It's difficult to persuade people that it's worth coming into a sound facility and spending, say £85 an hour to get a really nice soundtrack which works well with the pictures, as opposed to going into a video edit suite where they will spend days at several hundred pounds an hour. They don't bat an eyelid at the cost of a video edit like that but when it comes to an extra hour of studio time at £85 they go 'Whoa, my music budget's gone way over.' "

Directors Paul Deeley and Phil Horne in Studio One

Fortunately there is a healthy and growing number of exceptions, according to Horne: "A lot of our clients do care about the audio quality and they realise that a good soundtrack makes or breaks a programme."

The fact remains that The Soundhouse are clearly doing well by concentrating on their chosen field and helping to develop their market's awareness of the possibilities for the future, shunning the often illusory glamour of the music business and avoiding the temptation to be all things to all men.□

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UNCOVERING SENTINEL

Granite buildings provide a great recording environment but can cause one or two problems when rebuilding. David Hastilow was involved in one such project in Cornwall which he relates here

Six months ago Sentinel Records was acquired by Admix AV with a deal that included the entire Sentinel record catalogue and the recording facilities situated in a former school in the picturesque hamlet of Paul near Penzance, Cornwall. Admix, together with their music publishing company Interspear are producers of original film and library music for major advertising clients.

The feature of Sentinel Studios that attracted its purchasers was the large ambient recording area that was formerly the school hall. Brian Smith, co-director, musician and producer for Admix explains: "We now have seven writers inputting music into our catalogue, most of them using the Atari/Steinberg combination, but increasingly our clientele have been asking for live orchestral music for which we had no facilities at our original facility in Hampton. Normally we'd just use sampling to originate string and orchestral sounds."

The original owner of Sentinel, Job Morris, had built the record label from a tiny studio in the fishing village of Newlyn. When Morris moved from Newlyn to the present studio he literally hard-wired many of his mic placements, using mainly STC *4038* mics, into the hall. Acoustically he left the hall completely untouched as it comprised hardwood floors and plaster walls to a height of 10 metres. Smith continues, "The hall divides itself perfectly into a strings and brass setup. Brass on the gallery and strings below in the hall. Even when it's not sunny in Cornwall the light here has a brilliant quality. I can understand why artists come here. In the control room I'm installing an Allen & Heath *Sigma* console, with automation and all of the latest FX and keyboard gear. Upstairs I'll probably install a pre-production programming suite. Job's duplicating setup I'll probably keep too.

"When I first decided at Christmas 1987 to go and have a look at Sentinel, I had travelled around 200 miles in a howling gale to get there. But when I arrived and saw the place I was totally enchanted."

As with just about everything else in that part of the world, Sentinel Studios, being an old school house, was built of good, solid Cornish granite. The perfect insulator and isolator. Hence the initial problem of isolating the musicians from the sound of snoring from the nearby 'sleepy hamlet' was overcome without much ado.

Granite, and any other 'rock' in studios, can be a bit of a hassle if one is planning to alter the building structurally in any way but in Sentinel's case the original layout of the building worked to their advantage.

Firstly it was discovered with the first swing of the builder's 4 kg lump hammer that the partition wall between the recording hall (the former classroom/gymnasium/assembly hall—one enormous room to serve every purpose) and the control room (the former headmaster's study) was in fact almost 1 metre thick and already had three quarters of a hole knocked in it large enough for the sliding glass partition doors.

The second swing of the hammer revealed that behind the plaster on which used to hang the headmaster's accessories—mortar board, cloak, canes, etc—was another granite wall. The builder's labourer was so excited with the find that he'd cleaned it, polished it, stabilised it and, well, finalised just about every decision regarding the control room acoustics before the rest knew what was happening. (Builders' labourers have an incredible knack of finalising decisions, in my experience of studio refurbishment anyway.)

So, 'the wall' is now the central feature of the control room and very beautiful it is too.

The acoustics 'plot' was simply to trap the lows, deflect the mids and scatter the highs, the latter scatter being already accomplished by our aforementioned friend. But first, the village had to be isolated from the studio because, well, they didn't want anyone to be disturbed from their sleep.

Co-owners Brian Smith and Brian Millar at the A&H Sigma console and Atari controlled keyboards

A laminated floor, two layers of plasterboard and two layers of *Revac*, and five gallons of *Evostik* later, the village was permanently isolated (and so was the builder almost). Once the floor was down, all the inner shell walls were built on to it leaving an appreciable air gap, deadened with Rockwool, between it and the outer building shell.

The inner walls were bevelled in accordance with angles derived from the initial room acoustic analysis.

And although one may use absorption coefficient tables, spectrum analysis and other calculations for decisions regarding use of different materials, woods, etc, a balance has to be struck between creating an atmosphere people want to work in and the theoretical ideal—and the unknown, ie what will happen to the sound when the room is full of equipment, full of people, full of people with people, and not forgetting the directivity factor, Q, placing coherence, timing alignment of monitor speakers that have only been listened to twice and that was in someone else's studio, showroom, or at a show.

Suffice it to say, the acoustic designer's lot is sometimes a lonely one but there is always the escape route, Yamaha *NS 10s*.

Joking aside, there is no substitute for practical experience and an ability to implement acoustical and psychoacoustical science to give the client what he wants... the impossible!

Natural hardwoods were used throughout to continue the rustic theme. The rear of the control room incorporates Rettinger-type traps which also, through the nature of their construction, gives the illusion that the room is higher than it actually is. The illusion also counteracts the fairly low slung false ceiling and trap, which was provided to overcome any comb filter effects and reinforce the phantom images and to enable an even sound throughout the entire control room rather than just in one area of the console. A possibility with directional near and mid field monitors.

From beneath the console to the rear wall the floor area was left hard, to create a diffuse sound field along with the battening on the front surface of the traps and the 'pyramid' tiles on the ceiling above and to the rear of the ceiling trap.

Bearing in mind all the time that the RT60 had to be short enough so that the reverb time of the recording hall could be heard well, during orchestral recording, and at the same time, long enough for comfortable monitoring of electronic music.

Once the basic constructional work was completed, equipment was moved in to help in the final trimming of the acoustics.

The granite wall, lit with directional coloured lighting, blue hessian surfaces, teak finish and a grey carpet now provide the perfect backdrop to the Allen & Heath *Sigma* console, Atari *ST1040*, numerous keyboards and ATC midfield monitors.

On the opening day, studio owners Brian Smith and Brian Millar gave local people a demonstration of how they work, bringing together astounding original compositions in just a matter of minutes. Much to the amazement of the audience.

Electronic and orchestral music is not the prime production role of the studio as the two Brians intend to continue the furtherance of wholly Cornish music, such as choirs, brass bands and local musicians through the existing Sentinel Records label.□

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RUPERT HINE

Rupert Hine talks to Ralph Denyer about songwriting, performing and his particular approach to record production

"All the artists I produce," Rupert Hine told me during conversations at Farmyard Recording Studio, "are motivated in a similar way to myself. They see music as a method of communicating ideas not simply as an entertainment format. That is the most crucial thing to establish when I talk to someone about producing them."

Including his own solo and band projects Rupert has produced over 70 albums in a 16 year period. He's also written songs for himself and many of the acts he has produced including for one of his own bands Quantum Jump. He has produced albums for Saga, Kevin Ayres, The Fixx, Chris De Burgh, Thompson Twins, Howard Jones, Bob Geldof and, more recently, Stevie Nicks and Rush. He produced and co-wrote tracks for Tina Turner's *Private Dancer, Break Every Rule* and *Foreign Affair* albums. Hine's own album *Immunity* attracted considerable attention including that of Robert Palmer, with whom he co-wrote the singer's *Pride* album, he also played keyboards and performed with Palmer as part of The Power Station. He wrote and produced the soundtrack for Skolimowski's *The Shout*.

"Most days," Hine explained, while talking about his production values, "I feel that it really is a fight against an industry that would love to have music mean little more than a lightbulb. Simply something you switch on when you go into a room and it affects the room and not the occupants. And for anyone who lived through the most powerful part of the mid and late '60s and saw—in terms of what you can record—the kind of really evocative and completely limitless vision pop music can produce, that has to be a constant reminder that it can happen again at any time.

"My initial drive was to be a record maker. As a kid, the records that I most liked were the ones that were most evocative on a sound level. Ideally they had to be great songs but the records which gave me a real spine tingle always had a very evocative and emotional sound. If I had to name one album that most convinced me I had to be a record producer, then it would be the Beach Boys *Pet Sounds*.

"A different style yet equally evocative to me—would be the Phil Spector productions, particularly *River Deep, Mountain High* (Ike & Tina Turner) and *You've Lost That Loving Feeling* (The Righteous Brothers). I think both of them have become Top 10 classic singles of all time and quite rightly. Those records and the Brian Wilson productions for me were the first records on which it was quite impossible to tell what was being played; and it was really quite irrelevant. No longer were you listening to three or four musicians and a singer. You were just listening to a sound; a very evocative picture the same as when listening to the best symphonic writing when you can't tell what parts of the orchestra are playing and it's rather irrelevant anyway. It just adds up to a wonderfully evocative whole."

Hine feels that the word genius must be applied to the 'production minds' of Wilson and Spector at their peaks during the '60s, particularly when considered in relation to the low level technology of the period, which they managed to overcome "usually in comparatively easy ways such as clever doubling-up of instruments and startling use of dimension and space. The only kind of tricks at the time were pretty much just compression and limiting as far as electronic processing is concerned. But those three major elements put together—incredible! Even now stunning records could be made with just those three ingredients. It's not particularly fashionable to go that route although I think more and more people probably will in an effort to redress the balance of instant technology."

During his teens Hine was highly motivated to be singer and songwriter. He was in various R 'n' B groups and at one time was in a duo which had a few singles released. During the early '70s Deep Purple bassist Roger Glover produced Hine's first solo album *Pick Up A Bone* for Purple Records. During 1973 Glover's time was entirely taken up with Deep Purple and he suggested that Rupert should produce himself. Glover had been impressed by Rupert's home demo tapes, which showed that he had become as involved with the recording process as he was with songwriting. So Rupert recorded his second solo album *Unfinished Picture* with the assistance of Simon Jeffes.

Singer Yvonne Elliman was also signed to Purple Records at this time. Through the record company connection Hine and lyric writing partner Jeannette Obstoj were approached to write songs for Elliman's *Food Of Love* album but she also decided that she wanted him to produce the album. Since that time he has pursued parallel careers as a producer and a solo artist with his songwriting playing a major part in both.

In 1984 Rupert worked with Tina Turner for the first time producing *I Might Have Been Queen* and *Better Be Good To Me* for the *Private Dancer* album. He co-wrote *I Might Have Been Queen* with Jeannette Obstoj. Subsequently during 1986 he produced *Break Every Rule* and *I'll Be Thunder* for the follow-up *Break Every Rule* album, both songs co-written with Obstoj. Apart from some guitar from James West-Oram of The Fixx and some percussion overdubs from Trevor Morais, Rupert played everything else on the backing tracks.

Various songwriting and production teams had essentially been asked if they could help with any spare songs or time in which to work on tracks for the album. Subsequently different production teams were working on tracks in four or five separate studios all at the same time.

Hine was asked if he could produce four or possibly five tracks. His only problem was time. The Fixx, on appreciating Hine's feelings about working with Turner, agreed to break their sessions at the Farmyard for a week. Hine agreed to write, arrange and produce one track which turned out to be *I Might Have Been Queen*.

"In the end Roger pursuaded me to do the cover of *Better Be Good To Me* which was done in its entirety in two days, the other five days being spent writing and recording *I Might Have Been Queen*." ▷

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◁ The first surprise for Turner was that Hine never writes songs off-the-shelf, preferring to work on what he jokingly calls a "bespoke songwriting" basis. Obstoj carried out the major part of the first step of the process. This is an almost patient-psychiatrist-like session in which several hours are spent discussing the singer's past life, experiences, beliefs, hopes, fears—you name it. Thus armed with a psychological profile of the artist, Obstoj and Hine set about writing the kind of song they feel that the artist would be likely to write for themselves.

"I started the whole thing with drum machines and sequencers but when using them I like to try to make them as invisible as possible. So a great deal of effort goes into making the drums sound as organic as possible at the end of the day. I usually overdub some real drums or percussion and again, go to great lengths to make it crashy, wild and thumpy—not neat and tidy. The basslines were all keyboard, which I play. The only major extra person was Jamie on guitar."

Hine in musical mood

Hine wanted a strong guitar influence on the song so involved the guitarist right from the songwriting stage, quickly switching lines from keyboards to guitar. "So it was all done in a week. I love it when it's that fast. It's so exciting."

So boredom is his biggest enemy in the studio? "Yes, but I would put it another way and say excitement and energy are quintessential to making records. The only way you'll get that is by keeping things moving along at a pace as one idea leads into another so that something is always up and happening. Keep the energy continuous because as soon as you stop and there is a long and deliberated point it's not just that one point and its resolution that will suffer. The next *x* number of things you have to do will all suffer from the slowing down of mental energies. It's better to keep the ideas flowing and re-do something later if it is proven wrong. Some people are going back to recording these albums which take months and it's dreadful. When people ask me to produce them if they've got that attitude I can tell at the first meeting and I'm straight off like a streak of lightning.

"*I'll Be Thunder* represents an effort from me both as a writer and producer to try and give Tina a track of a size and scope that she'd not attempted on record since *River Deep, Mountain High*. The lyric was once again written as the result of a conversation with Jeannette. It was a monumental love song.

"In a way if there are any influences on the track, they go right back to the beginning of our conversation. To me there are bits of the Brian Wilson and Phil Spector feeling of what I like about record making on that track. Nothing specific, just a feeling. It's a big sound with very few tangible parts. Clarity was not what I went for on that track. It's much more impressionistic. You don't want the listener to be sidetracked into detail. You want people to be hit by a force and a vocal. That was really the aim on that track. It did take some time, of course, because the arrangement was quite complicated with a lot of orchestral types of sounds carefully interwoven with drums, bass and some guitar.

"I have my home recording setup. I do virtually all my writing at home on piano and then into machines and a 12-track tape recorder. Certainly Tina heard what I would regard as a Phil Spectorised demo, albeit just recorded by me alone at home. So Tina would sing over these early recordings. She 'owns' a song before she arrives at the studio. She'd already be in charge of a song. Tina only arrives at the studio when she's ready to perform your song as if she is on stage in front of 100,000 people. That's her attitude. They've all paid their money and they want to see a great performance."

It was hoped that Rupert would be able to produce three or four tracks for the more recent *Foreign Affair* sessions but conflicting schedules restricted his input to two tracks. It seems the sessions were scheduled at short notice and by the time Hine was free of his commitment to a Stevie Nicks project, Turner's album was virtually complete and there was more than a tinge of disappointment in his voice as he related the tale. He did, however, produce the Sandy Stewart and David Munday song *Falling Like Rain*, and *Bold And Reckless*, which was co-written with Jeannette Obstoj and appears on the CD single of *The Best*.

"I did the two tracks back here in London in about 8 days which included writing *Bold And Reckless* and Tina and I enjoyed them both. I recorded as before, all on my machines on my own. I did some percussion overdubs and Tina did her vocals in the inevitable 10 minutes.

"Most of the rest of the album was done with a completely live setup. Dan Hartman did the Tony Joe White songs with a live band in New York, which again was different for Tina. I was going to put live players on my tracks but everyone thought they sounded so good that we didn't risk it."

Rupert had already heard the demo of *Falling Like Rain* as the song had been offered to Stevie Nicks. "It wasn't right for Stevie, it was too kind of funky but I said 'Oh yeah, great track for Tina.'

"They said they'd already had a go at recording the song with another producer who shall remain nameless, spent a fortune and the final result had absolutely no feel and the whole point of the song was that it had an almost 'Prince-like' feel. I said 'Yes, I'd love to have a go.'"

Rupert decided to transfer the 12-track demo on to the studio multitrack and then gradually replace parts. "That way you keep the atmosphere of the demo intact and you've always got a frame of reference right there in your ears all the time. I like working like that. It only applies if you've got one of those magical demo tapes and this certainly was that." Hine actually retained some of the original demo backing vocals, which he tidied up via sampling and then mixed together with his own voice.

He was pleased when Capitol Records in Los Angeles told him that *Falling Like Rain* was the first track from any of her recent albums that they felt could get airplay in the States on the black stations. Apparently she is usually only played on AOR rock stations.

Earlier Hine had become involved with another 'phenomenon'. At one point he was in America having meetings with three of the "biggest acts you could be offered" but was not excited by the prospects.

Flattered by the approaches he thought he should at least meet the acts in question. It soon became clear that all three were on a financial treadmill, having to maintain a high level of sales to meet commitments from previous albums and past successes. They were forced to consider money first and foremost, otherwise they were in serious financial trouble. He found the meetings depressing. He also found that he was actually getting refreshment and relief by listening to a demo cassette of a young, unsigned, unknown act he'd brought with him from England. That was Howard Jones.

"I was so excited by what Howard was doing that it became no contest." On getting back to England he went to one of the singer's famous early gigs in local Aylesbury.

"All I can say is I walked in and it was like a mini version of the Beatles at Shea Stadium. And this man hadn't even put a record out." On top of all this Howard Jones was of course making the music alone, just him and his machines. "That sealed it. I said: 'Bloody Hell! This is it!'

"The fact that a man using machines—with all the things that have been described as uncommunicative to an audience, and I still believe that generally machines are—but in his case he completely transcended the limitations and liabilities. Not only that, he made a feature of them and rose above it. When the machines were playing on their own he was in the audience really involving them and getting them to sing the songs. That was such an exciting element to add to the songs that I'd already heard on the demo cassette. If I could only catch the liveness in his voice and personality on tape, not get it all studio sounding and avoid all methodical recording, just keep it alive; which was what we did and Howard responded so well to that."

Hine was signed to produce the *Human's Lib* album, recorded at the Farmyard in true Hine style, in just five weeks. Colin Thurston had produced Jones' first hit *New Song* before Hine became involved so Warner Brothers were politely requesting a follow-up single as soon as was practically possible.

"I suppose Howard's motivation for making records was exactly like my own, music is a handy and evocative vehicle through which to communicate ideas to as many people as possible. A means of communication that is not an end in ▷

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◁ itself. That was a chance for me to work very closely with someone on a one-to-one basis who was naturally commercial. The kind of music he most liked to write and that came naturally from the ends of his fingers and through his voice was, as luck would have it, right for the time. So he had that going for him and yet his motivation for making music was identical to my own. In other words my music would come out with angst-ridden darker sides and again by definition was hardly Radio One fodder. But exactly the same attitude in Howard would produce absolutely perfect Radio One material. So that was a very exciting starting point for me. We were not going to make superficial pop records but committed ideological records as far as the motivation went but they would just be so much more accessible and palatable."

Rupert recalls that WEA Records were extremely supportive of Howard Jones' musical direction and that they did not try to force any artistic compromise. But he also feels that there are more than enough producers meeting the more common record company demands for instantly commercial and disposable music.

Rupert Hine with Steve Taylor and percussionist Trevor Morais

"The follow up album *Dream Into Action* could have been a disappointment but wasn't. There was an album which Howard, unlike a lot of artists, took firm control of in terms of expanding what he was trying to do. There's some pretty chunky stuff there in terms of him taking on fairly serious subjects that he needed to sing about. I enjoyed that album equally as much as *Human's Lib* and it sold equally as well on a worldwide basis.

"He's become and stayed a very good friend and it was a difficult decision not to carry on working together after *Dream Into Action* in that both of us would have liked to continue but we agreed not to. Part of Howard's thing is to keep changing and looking for new areas. He is such a positive person to work with. His whole attitude is: Be positive and positive things will happen to you. And he generates such positiveness from people around him. It's so contagious that you do your best every minute of the day. Working with Howard you wouldn't possibly think of doing otherwise. I don't want to sound semi-religious about it but I think his strength is his personality and it goes beyond anything you can really talk about."

A couple of years ago Rupert produced The Rebel Heels debut album for Atlantic Records. Atlantic's president Doug Morris was very impressed by Rupert's work and thought of him when Stevie Nicks came to record *The Other Side Of The Mirror*. Both Morris and Stevie Nicks' management thought Rupert's strong songwriting base would be a desirable asset to the project.

"Stevie and I hit it off pretty quickly. She's got a little studio in her house and I went over there and did about 5 weeks pre-production with her. She had so many parts of songs that needed finishing and rather than finishing some unnecessarily I was trying to work through the ones that had most promise. She had lots of fully completed songs as well. Pre-production with her was crucial."

Rupert was working with Nicks on uncompleted songs while working up arrangements on his keyboard rig.

"First and foremost Stevie is very much a poet and lyricist. That is her main drive and she has co-written with various musicians in the past. I co-wrote four or five of the songs on the album which was great fun and very enjoyable. She writes poetry *all* the time. At the end of every day where most people might write a diary, she'll write down her feelings in a combination of prose and poetry. That gives her a colossal bank of ideas for songs."

Rupert decided to hire an Otari 24-track and start recording his basic instrumental work on to 2 inch tape in Stevie's studio. "Rather than uproot Stevie and say: 'Now we're going to start recording the album for real,' we just stayed in her house and upgraded the equipment a little bit. So the seam between pre-production and actually making the record was invisible and we ended up doing about 6 weeks recording, just in her house. I did all the programming of my stuff first, which meant she was able to hear it in a kind of natural surrounding without feeling that it had suddenly got serious. That way she could hear the arrangements and warm to them or criticise them. For the most part the tracks grew naturally and we started to add a few players there, like Mike Campbell from Tom Petty and the Heartbreakers."

Eventually the main reason for moving to a commercial studio was that Stevie and some friends who were staying with her could not get away from the noise: "It was fine when Stevie was in the room working with me but a lot of the time I was in there pottering away on my own and I like to work through the night a bit."

Rupert started working with Steve Taylor in various studios around Los Angeles adding top calibre musicians including Tony Levin and Vail Johnson on bass, Jerry Marotta on drums and Kenny G on sax. "Some great players. They started to replace my parts on tape. I kept some of my playing but for the most part the guitar, bass and drums parts were all replaced. Back in England we added Jamie West-Oram, guitarist with the Fixx and Geoff Dougmore played some drums.

"Stevie's a very natural singer. Recording Stevie means recording her at the right time when she's in the right frame of mind. It's just timing. A classic case where the psychology of music is much more important than anything inherently more musical. It's a bit like going fishing. Fishing for the right moment to reel in a good vocal. She's one of those singers who will sing a song all the way through maybe once or twice and there's not much point in doing any more because it is just the wrong time of day. It's nothing to do with working harder. That's pointless. Stevie hates 'dropping-in' and so do I. A vocal is supposed to be something you sing from top to toe and get it right, and that's got nothing to do with getting it right technically. If the actual spirit of performance is absolutely there, then use technology to take care of little inaccuracies rather than start dropping-in and lowering the spirit of the performer. That is certainly the way I feel."

Hine seems to work in a series of organised stages. Firstly there is a distinct songwriting stage, then an arranging or pre-production stage, then the actual recording and mixing. Consequently his songs have structure not based on the finer points of the final record.

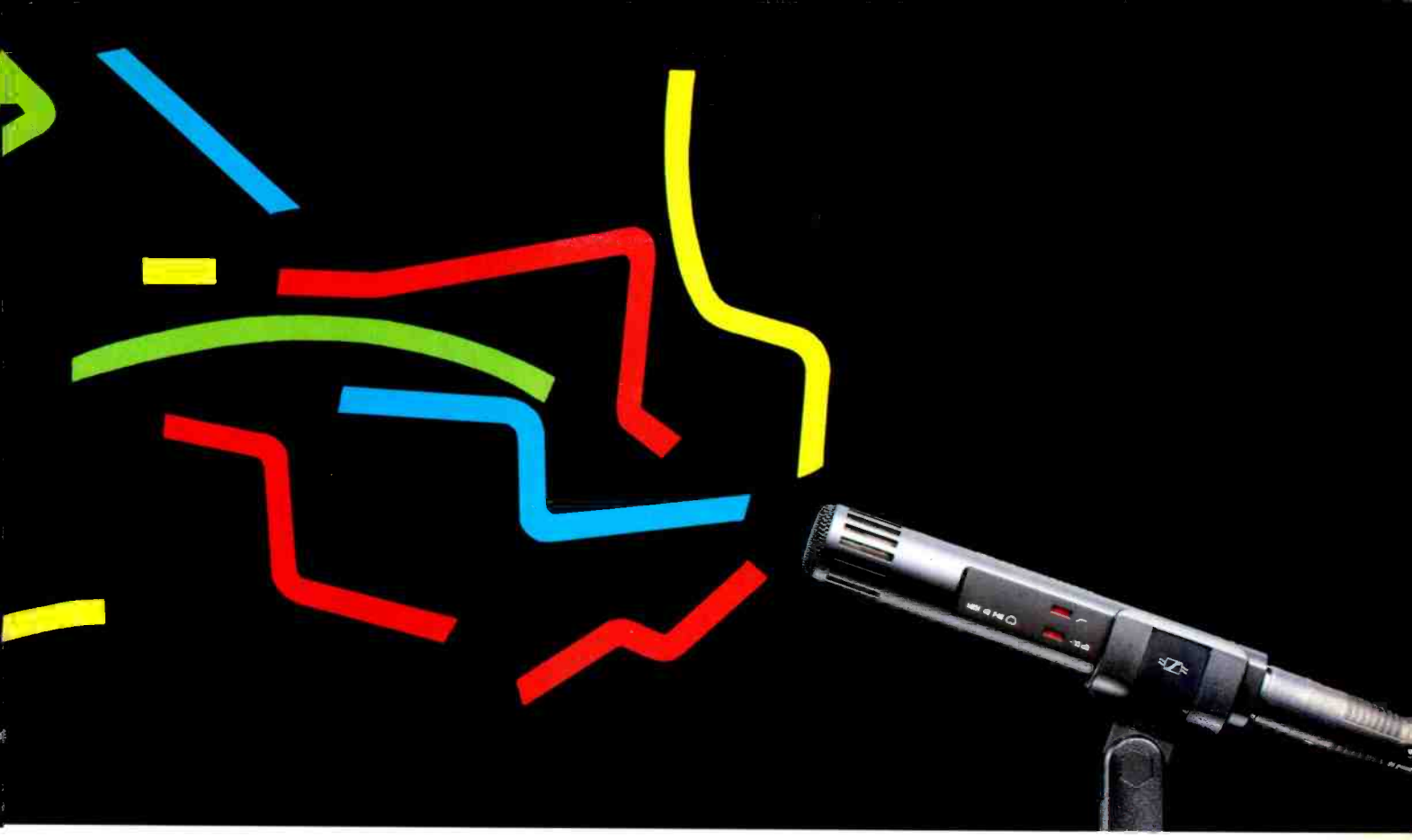
"One of the biggest criticisms I have of the disposable side of pop music is that it is arranging nothing. It's starting with an arrangement and trying to make a song out of it. It almost always has some very intangible beginning which is just a couple of riffs or arrangement ideas of even just a couple of sounds that people think would make a great sound for a track. They start to work with that and hope, with their fingers crossed, that they end up with a song. That has never been my outlook. I've got too much of a songwriter background.

"Pre-production to me is sitting down with the artist or the band and trying to get the song to do everything it should do, and to be a satisfying song, to have the right emotional possibilities arrangement-wise, and for the song itself to be of a very satisfying shape so that you can do it in a very simple way. You don't need loads of equipment to do that. Obviously if you are working with a solo artist it is a lot trickier. If they are self-accompanying it's not so much of a problem but if they aren't it's a question of whether I would do that work or try to get them involved with someone else. For the most part I do that work with the artist because I prefer to.

"I feel a real responsibility in what I try to do, trying to redress the balance wherever possible to remind people that music—just as much as film—is one of the most powerful creative mediums, especially when you utilise the sung word as well. So you've got all the emotional and pictorial abilities inherent in music coupled with the very much more literal communication via a lyric. It's such a powerful medium but the industry itself doesn't really encourage it to be used in that way.

"I react to so much of what I hear on the radio and dislike. A lot of my production work lately has been a reaction to the current obsession with superficiality. I'm always trying to produce the records *I* would like to hear.

"My feelings towards production are always on the change and always on the move. That's part and parcel of being a good producer. I don't just have one way of making records." ▷



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◁ The pairing of Rupert with Canadian rock band Rush for an album in September was not an obvious one. The band had approached him a few years previously with a view to working on their first album without Terry Brown. Rupert had gone over to Canada to meet the band.

"When they asked me *this* time I felt absolutely ready to do an album with a *real* self-contained band of three very fine instrumentalists. They don't rely on umpteen sequencers and keyboards although in my opinion they had got a little too much into keyboards on their last two albums. I'd reached a point at which I would just love to hear guitar, bass and drums again. I'd done so much work in the electronic area. And if I was going to work with a guitar, bass and drums group it would be even more fun to work with extremely talented musicians, which Rush most certainly are. 'Progressive rock'—a label I haven't used to describe anything for about 12 years—actually seems quite fresh again. Everything is cyclical.

"To hear people work out very intricate arrangements and very carefully crafted songs with lots of depth and interest—all of which they handle completely themselves without studio trickery—was extremely refreshing. That was absolutely the appeal for me; to have a refresher course in production and recording, and I must say it has been exactly that. Rush are not only talented, they are also extremely together. Both in the studio and pre-production, everything is finely crafted."

Hine worked on the basis of recording the band very much as a three-piece, restricting keyboards to a secondary role. He encouraged the group to return to their 'powerbase' roots, while retaining their development in melodic songwriting.

The Rush album was the first Hine has recorded completely digitally. A couple of years ago when Studer first introduced the *A820* with Dolby *SR* Hine bought the machine for Farmyard Studios, a move many studios soon followed. At the time Rupert felt that the digital multitracks on offer had yet to be proven, particularly as Dolby *SR* had given analogue recording an extended lease of life. He heard A/B tests and went for the *A820*. He felt that even when a digital multitrack arrived he was confident about investing in, he and the studio would benefit from having the very best of analogue machines for work on back catalogue. Up until the last year or so most of Rupert's own productions were recorded at Farmyard and thus Rush was his first completely digital project.

"For me analogue and digital are fine as long as they are good machines. Digital tends to be fine almost regardless of the machine whereas you need a good analogue machine. So accepting that as the basis, in recent years I've recorded a lot of projects Dolby *SR*, both multitrack and mixdown. Invariably I've mixed down to a Sony *1630* and before that a *1610*. For some territories I've made straight copies to *1630*, for vinyl sometimes I've taken a Dolby *SR* copy. But I'd never recorded an album on multitrack digital and Rush hadn't either."

Hine and the band decided to set up a direct Dolby *SR* and Mitsubishi digital comparison on Neil Peart's drum kit: "Neil has a vast '70s type of kit with loads of cymbals and great-sounding drums. He's a loud, strong, intricate and fast player. None of us took more than about 2 minutes to go for the digital recording, mainly because of the way it kept the impact on the front of the transients. To me, there was not a great difference in the nature of the sound, it was just the way it was handling transients, particularly on the bass drum. We certainly didn't think we were losing anything and decided to do the whole album using Mitsubishi multitrack and never felt the need for more than the 32 tracks."

Hine decided to record the album exclusively on SSL *G* series consoles in preference to *E* series as he finds *G* series quieter. He also prefers the *G* series EQ, computer and stereo channels.

Hine experimented with various mixdown media on the Rush project: "We're mixing on the Technics *SV360*, which is the sweetest sounding DAT machine I've ever heard. It's the same as the Panasonic machine in America (*SV-3500*)." (The Panasonic equivalent had been used by Bob Ludwig in the USA who was mastering engineer for many previous Rush albums.)

After a period of A/B testing "like a bunch of loonies", Hine says they finally decided that the semi-professional Technics *SV360* sounded at least as good, if not better, than professional digital systems costing 15 times as much: "You could kid yourself that you'd figured out what the difference was but if someone switched over and didn't tell you you'd be lost again."

Rupert first worked with Bob Geldof in 1986/87 on his first solo album *Deep In The Heart Of Nowhere*. During October he was working with Geldof on a second solo album.

"The new album is worth a mention because it illustrates another totally different way of working. We decided to try to record an album which was much less to do with the machine approach and much more to do with players. This time we had seven players, which included accordionists and violinists as well as guitar and bass. We had a seven-piece rhythm track line-up and in 5 days we recorded 26 songs. All of them had a nice loose edge, which was in a way why we were able to record so many songs in such a short time. We wanted a very spirited performance rather than getting everything finely crafted. I had obviously done a lot of pre-production with Bob so that the songs were as together as they needed to be.

"In the studio we just kept throwing songs at the players and seeing how they reacted, capturing takes at those times when they were still, in a way, getting off on the songs; before they reached the point where they'd 'got it together'. It's a strange mixture of Celtic and Cajun with elements of World Party or Waterboys or even Elvis Costello. The only thing we cheated on was using a very basic drum machine as a click track, which I played controlling it from the studio, varying it and keeping it lively. We'll add real drums or percussion to the songs that need them at a later stage."

Again to keep the track spontaneous, multi-instrumentalist Geoffrey Richardson—who plays viola, guitar, mandolin, sax and flute—was set up in a corner of the studio with a couple of general purpose mics and encouraged to play just whatever instrument he felt was appropriate.

Hine has always been a fan of Van Morrison's *Astral Weeks* and he and Geldof had listened to it before the sessions.

"Hearing *Astral Weeks* fresh on CD and hearing how out of tune and approximate things were and how brilliant the whole record is, made us want to respond to those kind of feelings now, when tightness is the easiest thing to achieve. Looseness is the hardest thing to achieve and for years, of course, it was the other way around. If you want to get looseness now, it is to do with people playing together, responding to each other. For me the whole fascination with tight grooves and pockets has long since gone. It became so easy it became the norm. It's strange but understandable. Everything is so cyclical."□

THE LOUDSPEAKER CONNECTOR DEBATE A FINALÉ AT LAST

Ken Dibble has spent a number of years calling for a standard in loudspeaker connection. It is possible he has fulfilled his quest?

The debate over speaker connectors has been covered in *Studio Sound* since 1982, and elsewhere long before that. Several proposals and counter proposals have come and gone while the industry continues to fit the *XLR/3* male as the output connector on an amplifier and the *XLR/3* female as the input connector to a loudspeaker system.

From the USA there are signs that the industry is at last about to go for change—and maybe, at last, there is a product of sufficient merit to warrant a commitment.

The story to date

Recognising that the standard jack plug was not the answer, the Association of British Theatre Technicians (ABTT) issued their own recommended standard as long ago as 1970 in which the Cannon *XLR/4* was used with the female component carrying the source signal and the male on the loudspeaker enclosure. The pin configuration was as follows:

Contact #1: Low impedance −ve
Contact #2: Low impedance +ve
Contact #3: 100 V line −ve
Contact #4: 100 V line +ve

Recognising that the *XLR/4* connector did not comply with the Electrical Safety Regulations for use at 100 V line level, that the current per contact is limited to 10 A and because of the conflict with the APAE Standard, the ABTT Standard was withdrawn in May 1979.

Next came the Association of Public Address Engineers (APAE), who, in 1974, issued their own standard, again using the *XLR/4* but with a different contact configuration:

Contacts #1 and #2 commoned = −ve
Contacts #3 and #4 commoned = +ve

This overcame the current limitation problem but did not address noncompliance with the safety standards. If a connector wired to the APAE standard were mated with one wired to the ABTT standard, you shorted both the low impedance and 100 V line circuits to ground! Those early solid state amplifiers did not have the sophisticated protection circuits of today's designs.

Meanwhile, because of widespread practice by established systems houses and sound hire companies at the time, most equipment manufacturers in the studio and live sound market sector were fitting the *XLR/3*-type male connector. Despite efforts of the ABTT and APAE to come up with an acceptable alternative, the problem of interchangeability between low level signal and loudspeaker circuits, and the blatant flaunting of the Electrical Safety Regulations, this caught on and became standard practice throughout the industry in a very short space of time.

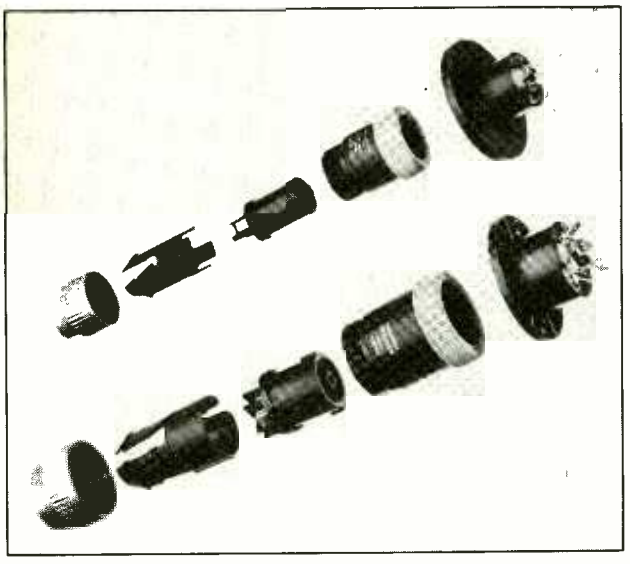
Because this practice contravenes the first principle of any connector concept, ie that 'volts should come out of holes' the BBC issued a directive in November 1981 in which the amplifier output connector is the *XLR/3* female wired contact #2 +ve, #3 −ve and the input connector on the loudspeaker cabinet is the *XLR/3* male, thus reversing the system so that volts do indeed, come out of holes. However, in order to overcome confusion between the input and output connectors, this recommendation requires that the input connector to the amplifier be male, ie the exact opposite of the present arrangement and in complete contradiction to *BS6840*, *IEC-268* and ANSI *RS-221A*. Clearly this makes no contribution to the present standardisation endeavour.

Then Robert Grunberg and Mike Dixon came onto the scene with their hermaphroditic connector, which, after being hawked around the world's connector manufacturers for a few years, finally saw the light of day as the Utilux *UX*. For various reasons, however, the Utilux never took off as a serious contender and now seems to have faded from the field.

Meanwhile, ITT-Cannon came up with the *AXR-PDN*, a two-pole plus earth system based on, but not compatible with, their already established *AXR-LNE* mains connector. Again, this has not really taken off even though it is featured as a loudspeaker connector in the RS Components catalogue, is carried in stock by a number of ITT-Cannon distributors and has since been adopted by the ABTT as a new theatre standard.

Similarly prompted, Switchcraft looked at the loudspeaker connector market during 1984/5 and after carrying out their own market survey based on proposed *20A* and *30A* hermaphroditic designs not dissimilar to the Grunberg-Dixon proposal, concluded that the market was not sufficiently large to justify ▷

Neutrik L series Speakon



◁ the tooling costs for a new connector and that in any event, prospective users were unable to agree on the form such a connector should take. The survey did, however, show that there was sufficient interest within the music industry to warrant the development of a new heavy duty two-pole 6.3 mm jack plug and socket. It is the author's view that the industry needs another jack plug like it needs a hole in the head but that was the outcome of Switchcraft's market survey and this is the way they have gone.

Following the failure of negotiations with Grunberg and Dixon, Neutrik looked at the market and, unlike Switchcraft, concluded that there was an unfulfilled need and proceeded to develop an altogether different approach to the problem. After several years of consultation and prototype development the Neutrik *Speakon* has emerged and it is this which seems, at last, to have captured the imagination of the market—in the USA at least with signs that one or two UK manufacturers may be about to follow suit.

The requirements

On the face of it, it may seem odd that a sophisticated, hi-tech industry such as ours should be unable to come up with an acceptable connector suitable for use as an amplifier/loudspeaker interconnect. Upon further investigation, however, the problems become apparent. Let's consider some of the often conflicting requirements.

Electrical safety: The UK Electrical Safety Regulations were fully discussed in 'Electrical Safety Requirements' (*Studio Sound*, October 1983). In essence, these lay down basic requirements for the safety of electrical apparatus. There are many ways in which the regulations can be met including compliance with *BS415* in Britain and *IEC65* in Europe. There is very little difference between the two although *BS415* is a little more stringent in certain areas. Section 9, part 1, clause 1 of *BS415* simply states: 'Accessible parts shall not be live.' And goes on to qualify that a part or terminal is not considered live if: '...the current measured through a non-inductive resistance of 50,000 Ω does not exceed 0.7 mA (peak) AC or 2 mA DC and...for voltages between 34 V and 450 V (peak) the capacitance does not exceed 0.1 μF... For frequencies above 1 kHz the limit of 0.7 mA (peak) is multiplied by the value of the frequency in kHz, but shall not exceed 70 mA (peak).'

By simple calculation, an amplifier rated at 150 W RMS into 4 Ω will develop a peak voltage of 34.6 V and a peak current of 0.69 mA. Therefore, any loudspeaker circuit rated at or above 150 W/4 Ω should be terminated so that the output terminals are not 'accessible'. This means that the live parts must not be within reach of a 'standard test finger', the dimensional details of which are defined within the Standard. The *XLR/3* flagrantly breaches this requirement so the majority of amplifiers and loudspeaker systems currently manufactured do not comply with the Regulations.

The first requirement of any new connector must be compliance with *BS415* and *IEC65* for voltages up to, and possibly beyond, 120 VRMS AC.

Current rating: High current carrying capability—say 20 to 30 A AC at 100 Hz.

Contact resistance: Low contact resistance, <5 m Ω, needs to be maintained after several thousand connects and disconnects.

Hermaphroditic design: Grunberg/Dixon, Utilux and Switchcraft all seem to prefer a hermaphroditic design. While this has much to commend it, it also has the disadvantage that the output of one amplifier could readily be connected to that of another, as was pointed out by the late Hugh Ford in 'A Speaker Connector—Some Response' (*Studio Sound*, March 1987) and Paul Garrity of Artec Consultants of New York in private correspondence. So maybe it is not such a good idea after all.

XLR format: Several correspondents consider we should retain the *XLR* format because of industry acceptance and to facilitate retrofitting to existing equipment, or to allow a changeover without the need for retooling of presses, etc. Others argue that the *XLR* shell is too small to allow anything greater than 2.5 mm cable, whereas any new connector should be capable of accommodating at least 4 mm cables, if not 6 mm. Also, some prefer a screw terminal option for ease of field servicing, which would not be feasible inside an *XLR*-style housing.

Durability: Robust construction, resistance to corrosion and durability are obvious requirements.

'Twistlock' engagement system: Some, especially Artec Consultants, express a preference for a 'Twistlock' design, while others are less certain whether a latching device is a good idea, some being clearly against any form of latching whatsoever.

Four-pole contact system: Some consider a four-contact device to be preferable to allow bi-amp wiring in the same connector.

EP format: Because of its rugged design and ability to accept larger CSA (Cross Sectional Area) cables, many suggest adopting a particular version of the Cannon *EP* connector. Unfortunately, the *EP* is already in use in various formats and with various wiring configurations; it would therefore be difficult to convert existing users to a different system. Also, the *EP* range is very expensive and not readily available.

Airtight receptacle: The component intended to be mounted on the loudspeaker enclosure should be airtight for obvious reasons.

Cable clamp: The cable clamp should be effective over a wide range of cable sizes. A size range of 6 mm to 15 mm is often advocated.

Universal point of agreement: Without exception, everyone agrees that a new connector is needed, that it should be a brand new dedicated design, adopted for use as a loudspeaker connector with the wiring convention standardised at the outset.

TABLE 1

Contact number	Low Z full range	Low Z bi-amp	Low Z stereo	100 V line full range
1−	Grnd	LF grnd	LH grnd	Safety grnd
1+	Signal	LF signal	LH signal	NC
2−	NC	HF grnd	RH grnd	0 V
2+	NC	HF signal	RH signal	+100 V

The Neutrik Speakon

In developing the *Speakon*, Neutrik have obviously listened long and hard to the industry. They have blended the views expressed with their own engineering expertise and design resources and come up with a product clearly intended as a serious market contender—probably the best to date, even though on first sight it looks more like a garden hose coupler. It would appear to address all but one of the design requirements earlier discussed:

- Tough, durable, all moulded, virtually indestructible, electrically non-conducting body shell
- Accepts 4 mm stranded or 6 mm solid cable
- Collet-type cable clamp effective up to 15 mm OD
- Dual purpose solder/screw terminals
- Rated at 250 V and 30 ARMS AC
- Complies with *BS415/IEC65*
- Low contact resistance: <3 mΩ after 5,000 operations
- Twistlock-style bayonet latch with locking ring
- Airtight chassis component available with small square flange to fit existing *XLR* cut-out or large circular flange to fit existing *EP* cut-out. The latter is especially suited for fixing to timber cabinets
- Four contacts to facilitate bi-amp operation
- Larger version, soon to be introduced in *EP* size format only, has eight contacts to accommodate multichannel systems
- Dedicated connector—no previous usage

While not entirely hermaphroditic, there are only two component parts, the cable mounted or 'free' component, and the chassis-mount receptacle, neither of which can really be described as male or female, or plug and socket. However, as there is only one chassis receptacle and one cable component, cables cannot be connected end-to-end without an in-line adaptor and the system does indeed make it possible for the output of one amplifier to be connected to the output of another. But these are the only drawbacks apparent from a fairly close examination of the design.

Composition

The basic design very cleverly combines the apparently contradictory requirements that the chassis-mount component should fit a standard ▷

◁ *XLR* panel cut-out while the cable component should be larger than the present *XLR* shell in order to accept larger CSA cables and allow for easier termination. The connector is essentially co-axial having one pair of diametrically opposed contacts on the outer annulus and another pair, rotated by 180°, on an inner spigot. The inner pair are designated 1+ and 2+, the outer pair 1– and 2–. The cable component is inserted into the chassis receptacle and twisted approximately 20° in a bayonet cap motion to lock, wiping the contacts in the process to ensure minimum contact resistance. A locking ring is also provided, which requires a 120° turn to lock the bayonet spigot firmly in place and prevent any further movement.

The cable component comprises only four parts, is very simple to assemble and requires a small screwdriver only to tighten the cable termination screws. The four parts comprise the contact barrel sub-assembly, a three-jaw cable clamp collet, an outer shell sub-assembly and a gland ring. All are reinforced high impact plastic mouldings, the only metal parts being the contacts themselves. In order to overcome the problem of in-line connection, a back-to-back version of the chassis receptacle is available as an in-line adaptor.

The arrangement is very cleverly thought out and, if at first appearing unnecessarily complex, has been simply executed to produce a serviceable connector. The available variants are as follows:

Part No. NL4FC	Cable mount connector
Part No. NL4FRC	Right angle cable connector but with reduced cable size capacity
Part No. NL4MP	Chassis receptacle, *XLR*-style
Part No. NL4MPR	Chassis receptacle, *EP*-style
Part No. NL4MM	In-line adaptor

Wiring convention

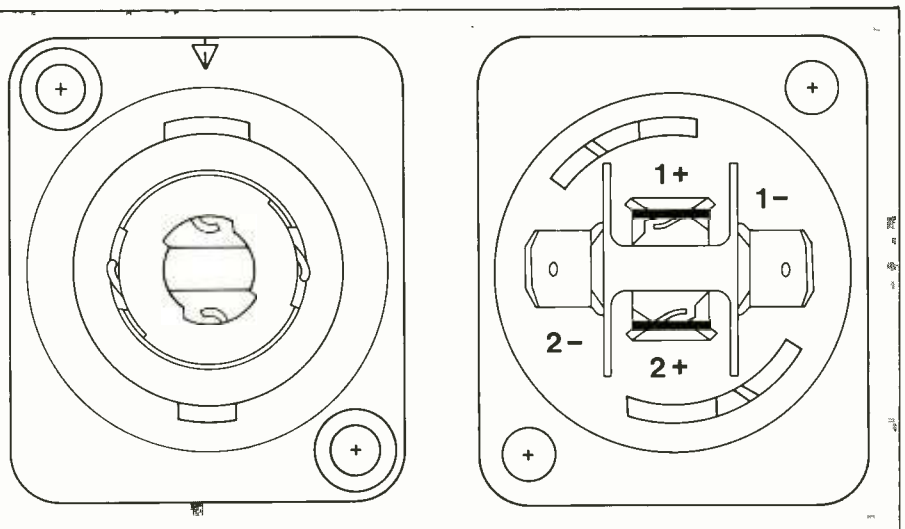
The manufacturers have suggested four different wiring conventions for single-channel low impedance applications, single-channel constant voltage line systems, low impedance bi-amp and low impedance stereo as shown in **Table 1.**

Canford Audio, a major Neutrik distributor in the UK, have published this information in their March 1989 catalogue as a 'proposed convention' and have invited comments for onward transmission to Neutrik.

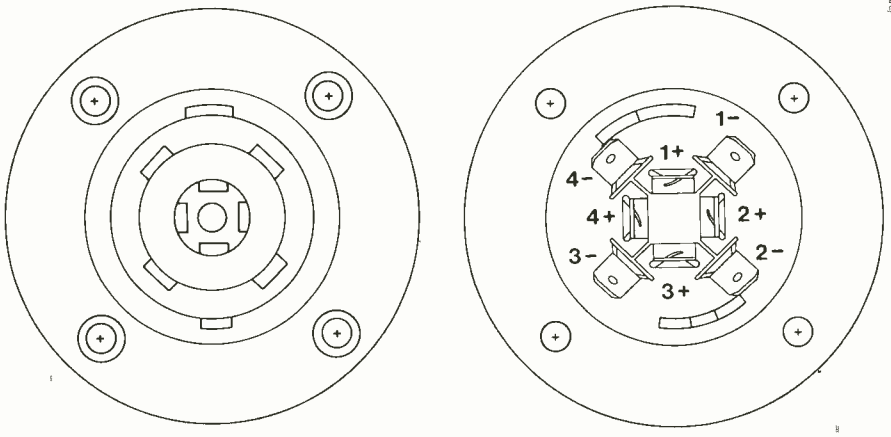
The following is a suggested reaction:

Low Z full range: Adopt
Low Z bi-amp: Adopt
Low Z stereo: Disaster area 1!
This would cause full range signal to be applied across the HF driver with fatal consequences if inadvertently connected to a receptacle wired for bi-amp operation
100 V line: Disaster area 2!
This would cause full range, high voltage signal to be applied across the HF driver in a connector wired for bi-amp or across the RH channel in a connector wired for stereo. This is a totally different application that would be better served by a connector not related to the *Speakon* range, perhaps Cannon *AXR-PDN*

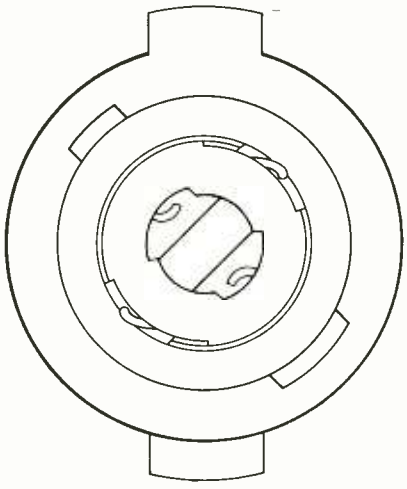
The four-pole Speakon should therefore be restricted to full range and bi-amped low impedance systems only. Stereo systems should use two separate connectors as with the XLR/3 at present. Having got this far, the adoption of any other arrangement would be nothing short of a disaster and would be likely to prevent eventual IEC recognition of the system.



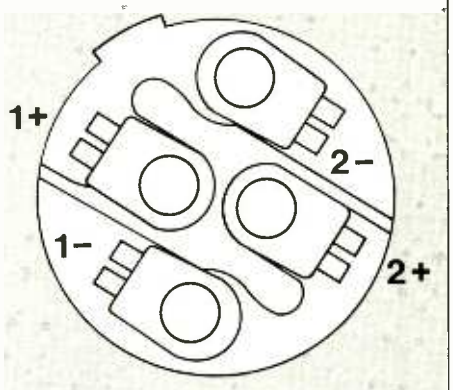
4-pole panel mount socket and wiring configuration

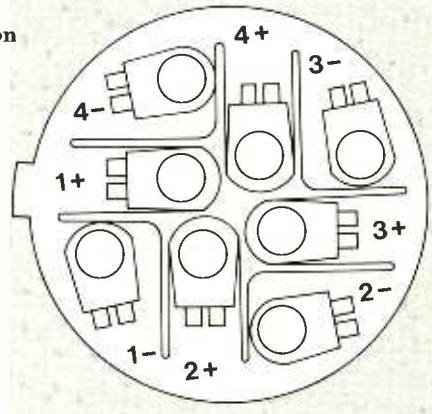


8-pole panel mount socket and wiring configuration



4-pole connector and wiring configuration





8-pole wiring configuration

Neutrik's technical designs for the new Speakon (L series) 4-pole and 8-pole connectors

During the 1988 Paris AES, at a technical forum on loudspeaker processors and connector/wiring conventions, an eminent panel of experts, including Dave Martin, John Meyer, Garry Margolis (JBL) and Eric Vincenot (Nexo) agreed that the *Speakon* was the way forward and that the proposed full range and bi-amp wiring convention be agreed. No mention was made of a stereo or 100 V line alternative.

One further point: the contact designation places the first contact pair at 0° and 270° and the second at 180° and 90°. Given the proposed wiring convention, it would seem more logical if the first contact pair were at 0° and 180° with the second at 90° and 270° so the cable strain could be more evenly distributed over the connector housing, as well as more logically placing both contacts of circuit 1 within the inner spigot and circuit 2 in the outer annulus. As things stand all the cable strain is on one quadrant of the connector when used in the proposed full range mode—which will doubtless account for most use—with the +ve contact located on the inner spigot and the −ve on the outer annulus. It is, however, clearly too late for this to be addressed now.

The pioneers

The industry lead is coming from the USA on this issue with the following major manufacturers having changed over, or being poised to adopt the *Speakon* system in the very near future: Apogee Sound Inc, USA; Community Light & Sound Inc, USA; BGW, USA; Dynacord, West Germany; EAW, USA; Electro-Voice, USA; JBL, USA: Matsushita, Japan; Montarbo, Canada; Peavey Corp, USA; Philips, Holland; Quad, UK; Renkus-Heinz, USA; Stage Accompany, Holland.

So the wind of change is very much in the air with others set to follow in the very near future.

Some reaction

Allen Mornington-West at Quad has carried out extensive testing of the *Speakon* prior to its adoption including contact resistance tests, durability and corrosion tests, jumping and trampling on mated pairs and experimental termination with a wide range of cable sizes, and has confirmed its acceptability from an engineering aspect. Quad's only reservation is over the small size of the screw terminations used to secure the conductors and the lack of any wraparound provision in the cable mount component but they acknowledge that by removing the grub screw the terminals do double as solder buckets. Consultation with customers has also produced a positive response.

Quad are confident over the change and believe that irrespective of which way the rest of the market may go someone has to make the move away from the *XLR/3* male, and changed over their entire production from September 1989.

Apogee Sound are already in full production with the four-pole *Speakon* as the standard option connector on their entire product range with the eight-pole version on their 3×3 concert system.

JBL are set to introduce an updated version of their *Sound Power* series, late '89 or early '90, which will include the standard four-pole and the eight-pole *Speakon*. The precise timing is dependent upon the availability of the eight-pole version in production quantities. The *MI* range will follow soon after and a spare *NLAFC* will be supplied in case of difficulty in obtaining the *Speakon* from a high street music store during the changeover period. The *Concert Series* will also be brought into line during the year to coincide with other design changes, so 1990 should see all relevant JBL product leaving the factory with the *Speakon* as standard equipment. Garry Margolis, marketing director at JBL International added, "We like the connector very much and are pleased to see a rational connector philosophy emerging," and went on to promise full support for a rational standardisation of use and wiring.

John Wiggins, vice-president of sales and marketing at Community Light & Sound, had this to say:

"The *RS880* is fitted with *Speakon* and others of our enclosed systems will follow. We talked to various tour companies and different cable manufacturers and it seemed pretty unanimous that they liked the design features. And the cable companies should know—they have to assemble them. It seems to have the voltage and current ratings and I guess we'll find out if it's rugged or not. We did the sound for the Special Olympics using *RS880*s in three different outdoor venues and we had some pretty bad rain conditions. They all held up and although it's not a waterproof connector, we had no shorting problems. So it appears that it probably is reliable."

Conversely, despite being included in Neutrik's *Speakon* users' list, Turbosound's Richard Frankson gave a categorical assurance that Turbosound were not about to change over to *Speakon* in the foreseeable future. They had looked at samples but did not like the hard latch feature, and in fact preferred the Cannon *AXR-PDN* if a change became necessary. If market ▷

◁ forces dictated a change they would consider it but, so far, no one has asked for *Speakon*. The *EP-6* was now so widely established on their product worldwide that they would not entertain changing to the eight-pole *Speakon* for that application.

Electro-Voice are also set to change over during the coming months and Mark Burgin of UK agents Shuttlesound welcomed the change saying that provided the market were generally happy with the new connector and that the industry all goes the same way, he was very pleased to see a move away from the *XLR/3*. He does not see the *AXR-PDN* as a valid alternative and considers the *EP* too expensive, difficult to obtain and complains that the chassis-mount version is not airtight. He would, however, be happier if Neutrik were to complete the *Speakon* range by tooling up for all four versions, ie male and female in cablemount and chassis formats. He is not happy at the prospect of using an in-line adaptor to extend cables and sees this as the only significant drawback. This is a view echoed by many others.

Renkus-Heinz, with immediate effect, are offering the *Speakon* as an alternative to their established use of the Hubble connector—again subject to the availability of the eight-pole version—but Paul Trew at Smart Acoustics, RH's UK distributors, was very positive that as far as the UK is concerned, RH product will be coming in with the *Speakon* option fitted as standard, adding that the imminent Renkus-Heinz UK hire operation would be entirely *Speakon*.

Peavey are known to be strong supporters of the *Speakon* and a number of products are already on line with the *Speakon* as standard equipment.

The list goes on.

Neutrik and marketing

At the 1984 AES Paris Convention Neutrik assured that they had no intention of addressing the loudspeaker connector requirement as, like Switchcraft, they saw insufficient market demand to warrant a commitment.

Following an article on the Utilux *UX* and Cannon *AXR-PDN* connectors ('A Speaker Connector—3 Years On', *Studio Sound*, December 1986) I was castigated by Bernhard Weingartner, president of Neutrik, for not including Neutrik in the feature ('A Speaker Connector—Some Response' *Studio Sound*, March 1987).

During the 1987 APRS exhibition, where a prototype was on show I was promised early samples of a *Speakon* to evaluate for a follow-up feature. This would have widened the debate and increased public awareness of this latest contender but the samples never materialised.

Meanwhile, with the *Speakon* in limited production, Neutrik have been quietly probing the market, encouraging users and manufacturers to consider it as a valid alternative to the *XLR/3* and, in the USA in particular, have clearly been successful in this endeavour. There was considerable confidence and a positive reaction to the *Speakon* at the 1989 NSCA in Nashville but Eardley Electronics (Neutrik's UK importers) were unable to supply samples and full details of the range at the time of writing. (*Eardley were however in a position to supply samples and diagrams by the time the article was in production—Ed.*) Canford Audio have been keen supporters of the *Speakon* project throughout and have been a most helpful source of information.

Now it seems, we have another softly, softly approach with the eight-pole version, with major international manufacturers waiting for this to come on-line before adopting the system, while Neutrik continue to dip a toe into the water.

Now Neutrik are asking existing and prospective users whether it is absolutely essential that the chassis-mount version be airtight as apparently the sealing of this component is causing a production slowdown.

Conclusions

Although there are a few minor criticisms here, it seems we may at last have a connector that meets most of the criteria and would appear to be acceptable to most users. In particular, it complies with the safety standards, it is durable, it accepts larger CSA cables, it is quick and easy to assemble, it is available to fit the *XLR* and *EP* panel cut-outs, the standard version will accommodate bi-amp working and a big brother is (or soon will be) available for larger systems—and it is affordable.

The time is right; the market is ready. All it needs is for Neutrik to make the *Speakon* available in production quantities, back it up with a positive international marketing effort and encourage its endorsement by the IEC and ANSI standards authorities.

But if we are to adopt the *Speakon* let's not try to make this one product all things to all people. Let's agree on a realistic level of application and adopt a standard wiring convention right from the start. That means dropping the proposed stereo and 100 V line application areas. For stereo, use two separate connectors, for 100 V line use *AXR-PDN*.

Adopt the present Neutrik/AES wiring proposal for full range applications: first contact pair *only* should be used; the second contact pair should *not* be wired in parallel to increase current rating in this mode.

Adopt the present Neutrik/AES wiring proposal for bi-amp applications: first contact pair LF, second pair HF.

Reject the use of this connector for any other purpose or with any other wiring configuration. □

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MONITOR SYSTEMS - CROSSOVERS

PART FIVE

Phil Newell continues his series with a personal view of crossover options available, the design problems and subjective audibility

In the words of one of the owners of a well-known studio: "I never realised that a different crossover could change the sound so much!" Granted, it had just taken 3 years to develop the new crossover, but nonetheless, the crossover it had replaced was a first class unit. With conventional crossovers, be they active or passive, what you put in is not what you get out. The outputs just will not add up to replicate the input. The only exception to this is the first order, 6 dB/octave Butterworth, but unfortunately 6 dB/octave is far too shallow a slope for most conventional requirements.

Crossover design and technology is enormously complex. The effect a crossover has on the audible characteristics of a monitor system, is usually grossly underestimated. Indeed it is so mind-bending, that many people prefer to use a stock unit and hope somebody else has done their homework.

To obtain the best results, the amplifier/loudspeaker/crossover/room combination must really be seen as one unit. This philosophy is brought home by the fact that most specialist monitor manufacturers now use highly dedicated, specific crossovers for their systems, as opposed to using stock items from specialist manufacturers of crossovers; now largely aiming their products at a sound reinforcement market. Not that this is less demanding than the studio market but the physical distribution of the many reinforcement systems changes from night to night. Together with the large physical displacement between the individual drivers, this renders accurate crossing-over an impossibility given current knowledge and technology.

There is a very large range of parameters to choose from in crossover design. Active or passive; high level or low level; slope—6, 12, 18, 24 dB/octave or others; slope shapes on initial turnover; power handling (high level)/output voltage (active). All these things are crossover variables. Different people, with different philosophies have pursued, and continue to pursue different paths.

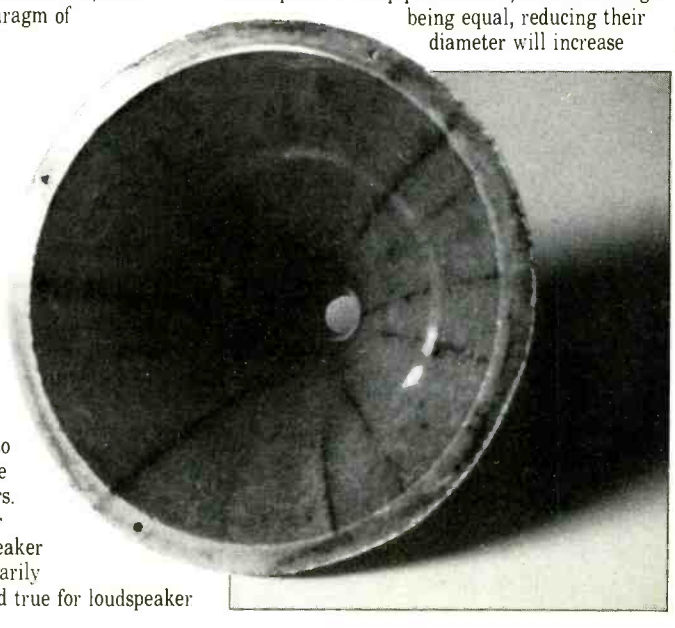
In 1984, I began developing a family of monitor systems, ranging from 2½-way nearfield systems to large, 4-way systems. The full design philosophy of the latter systems, was described in great detail in 'Studio Monitoring Design' (*Studio Sound*, December 1986). The article reflected the state of play in early '86 when it was originally written. The significant changes to date would be the standardisation on the Emilar mid-range driver, the more or less exclusive use of Crown amplifiers and the development of a new family of crossovers. Initially, the first two changes, or rather standardisation, would not seem to be relevant to a feature on crossovers. In reality, however, they are! The vast majority of mid-range compression drivers, are designed and developed for sound reinforcement/public address systems. This market is very much larger worldwide, than the studio monitoring market, so the manufacturers tend to produce units to fulfil the requirement of the larger market.

It was for their very high output capability and general indestructibility that I opted for compression horns in the first instance. Given the widely disparate systems with which they may be used, it is necessary for the manufacturers to maintain the electrical parameters as closely as possible to an idealised norm. This very much helps a system designer, who would expect an 8 Ω driver to be near to 8 Ω over the widest possible range of frequency and voice coil temperature. Diaphragm loading, resonances and coil inductances do all they can to thwart the former, whereas input power, efficiency and effective cooling affect the latter. The voice coil of a badly cooled, inefficient driver, when sustaining high input levels, will get hot. The temperature rise in the voice coil will increase the impedance, as will the voice coil inductance as the frequency rises. Inductance will also increase due to resonances on poorly damped drivers.

Manufacturers must keep these things in check if the drivers are to work well with impedance-dependent, passive, high level crossovers. Design parameters are a compromise. Companies producing drivers to the sole dictate of the greatest sonic accuracy from a very specific system, would soon go out of business. To my own ears, in this specific system, the Emilar is the driver that allows the smoothest overall performance from the system. I also think that the aluminium diaphragm of the Emilar is a smoother match to the pulp cone of the low-mid drivers, and to the aluminium ring diaphragm of the high frequency unit. The titanium diaphragm of the JBL and the beryllium diaphragm of the TAD were more difficult to blend with the remainder of the system. This in turn, as we shall see later, has a great bearing on the selection of the shape of the initial crossover slopes. This is not to say that the Emilar is a better driver than the JBL, the TAD, or the Coral but I believe it to be the most suitable match to this specific system. Likewise the choice of power amplifiers. If amplifier 'A' sounds better than amplifier 'B' on loudspeaker system 'X' it does not necessarily apply that the same will hold true for loudspeaker 'Y'. Somewhat ridiculously, in my opinion, amplifiers have usually been tested and rated into resistive loads.

Ah, for truly resistive loudspeakers—I have certainly never seen one! Whenever there is inductance and/or capacitance in a loudspeaker, there is consequently a reactive element where the voltage and current are not in-phase with each other. This produces the classic 'wattless power' so well-known to engineers of heavy electrical machinery. Factories using heavily inductive machinery must install 'power factor correction' capacitors to restore the phase relationship; otherwise the electricity board's wattmeter would not measure the total electricity used, and the subsequent bills would be lower than normal. A power factor of unity would be resistive. A reactive element resulting in a power factor of 0.7 would, if uncorrected, result in electricity bills of only 70% of the actual, generated power consumed. The electricity generating people are not too keen on that state of affairs, hence their insistence upon the installation of the correcting capacitors.

Why does this occur? Well, it is probably easiest to explain by an analogy with a familiar piece of everyday plumbing—a toilet. Electrical engineering and plumbing really are remarkably similar in their fundamentals. Pipes are wires, water is electricity, water pressure is voltage, water flow rate, in gallons per minute, is current in amperes. With pipes or wires, all other things being equal, reducing their diameter will increase ▷



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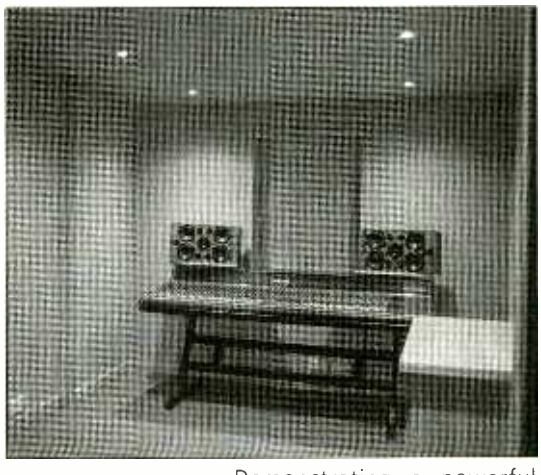

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◁ their resistance to the flow of either water or electricity. Capacitors have a capacity (the original term for capacitance), and toilet cisterns have a capacity. A resistor/capacitor series circuit has a 'time-constant'—the length of time it will take the capacitor to fully charge through that resistance. The time constant is determined by the multiplication of the resistance in Ω, by the capacitance in Farads (F). A 10,000 μF capacitor (.01 F) fed through a 100 Ω resistor, will take 1 second to 'fully' charge. That is, 100 Ω×0.01 F (10,000 μF); 100×0.01=1 (second). A toilet cistern of 1 gallon capacity, when fed through a pipe allowing a flow rate of 1 gallon per minute, will fill to capacity in 1 minute. Thus, the time constant of the toilet plumbing would be 1 minute. A plumbing system with a closed loop, for example an electric pump coupled to a turbine, with the outflow from the turbine coupled back to the pump, will provide a closed loop-resistive loading. The pressure in the pipes will remain constant for as long as the pump speed is maintained, and the back pressure on the turbine remains constant. This is analogous to a battery, connected via a wire to one side of a bulb, the other side of the bulb being connected to the other side of the battery (see **Fig 1**). Current will flow dependent upon the battery voltage and circuit resistance. Voltage and current are directly proportional, a higher voltage producing a higher current and *vice-versa*. In the plumbing analogy, a higher motor speed will produce a higher water pressure, which will in turn produce a higher rate of flow. Other things again being equal, water pressure and flow rate are in direct proportion. The bulb resistance is the analogy of the loading on the turbine.

So, back into the toilet! In this instance, the plumbing system is not a closed loop (ignoring the sea, evaporation, the clouds and the rain, which ultimately link the reservoirs to the sewers—a reactive coupling if ever there was one). When the toilet is flushed, the water drains from the cistern and enters the pan. As the cistern empties, the ballcock falls, opening a valve that allows water to flow into the cistern. Prior to flushing, the valve seals the end of the inflow pipe. The water pressure is at a maximum (the voltage) while the rate of water flow (the current) is at a minimum. As the falling ballcock opens the valve, the blocked end of the pipe is opened, allowing a flow of water. At that point in time, the pressure in the pipe (the voltage) is reduced as the valve is opened. The current (water flow) is consequently at its highest when the voltage (the water pressure) is at a minimum. As the cistern fills up, the valve begins to close. As this occurs, the back pressure in the water pipe (the voltage) increases as the water flow (the current), is reduced. When the cistern is full, the water flow ceases (no current flows) and the voltage (the water pressure in the pipe) rises once again to a maximum. It can surely be seen that the terminology of current/water flow and voltage/water pressure are largely interchangeable. It can also be seen that in a capacitor, as in the toilet plumbing, the maxima and minima of the current and voltage are not in harmony. When the water flow is at a maximum, the pressure is at a minimum and *vice-versa* (see **Fig 2**).

When a capacitor charges through a resistor, initially current flow is at a maximum as the voltage across the plates is at a minimum. The voltage across the plates eventually rises to the peak value of the charging voltage and the current ceases to flow. The current flows first, then the voltage rises. The current is thus considered to have a 'phase lead' over the voltage. In the case of an inductor, a voltage must be present across the terminals before a current can flow. Hence the current in this instance is said to have a 'phase lag' compared with the voltage. Back to the resistor again; current and voltage are always in direct proportion and are considered to be in phase. Just as with the plumbing system, electrical analogies can be made with the moving system of loudspeakers. We discussed in parts three and four (*Studio Sound*, October and November) *vis-à-vis* direct radiators, that the inefficiency of direct radiators is due to their poor loading and coupling to the surrounding air. The loading is reactive hence the 'power factor' is low. The transfer from electrical to acoustic energy is poor.

It has long been my opinion that the accuracy of the impulse response of a system is the most important criterion in any system's performance. In order to create an accurate impulse response, the phase response/phase slope of the system must be as coherent as possible. Only this, together with a smooth 'frequency response' can re-create an accurate impulse. Given the impossible complexity of interaction between the reactive elements of the loudspeaker drivers and the high level, passive crossover components, the inductors and capacitors, I cannot seriously consider such crossovers as contenders in the search for more phase accurate systems. Compensation can sometimes be incorporated in order to present a relatively constant frequency/impedance characteristic at the amplifier terminals but strange things can still be going on in the crossover/driver interface. All in all, some very strange loads can be presented to amplifiers, when coupled to loudspeakers. The 'wattless power' of phase leads and phase lags can demand truly enormous current surges from the amplifiers. This cannot be computed on a simple,

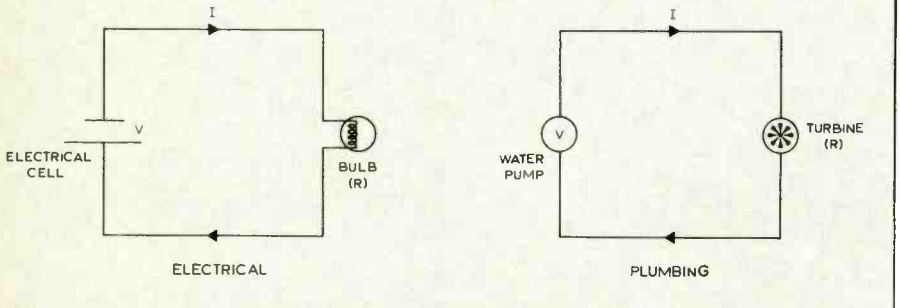
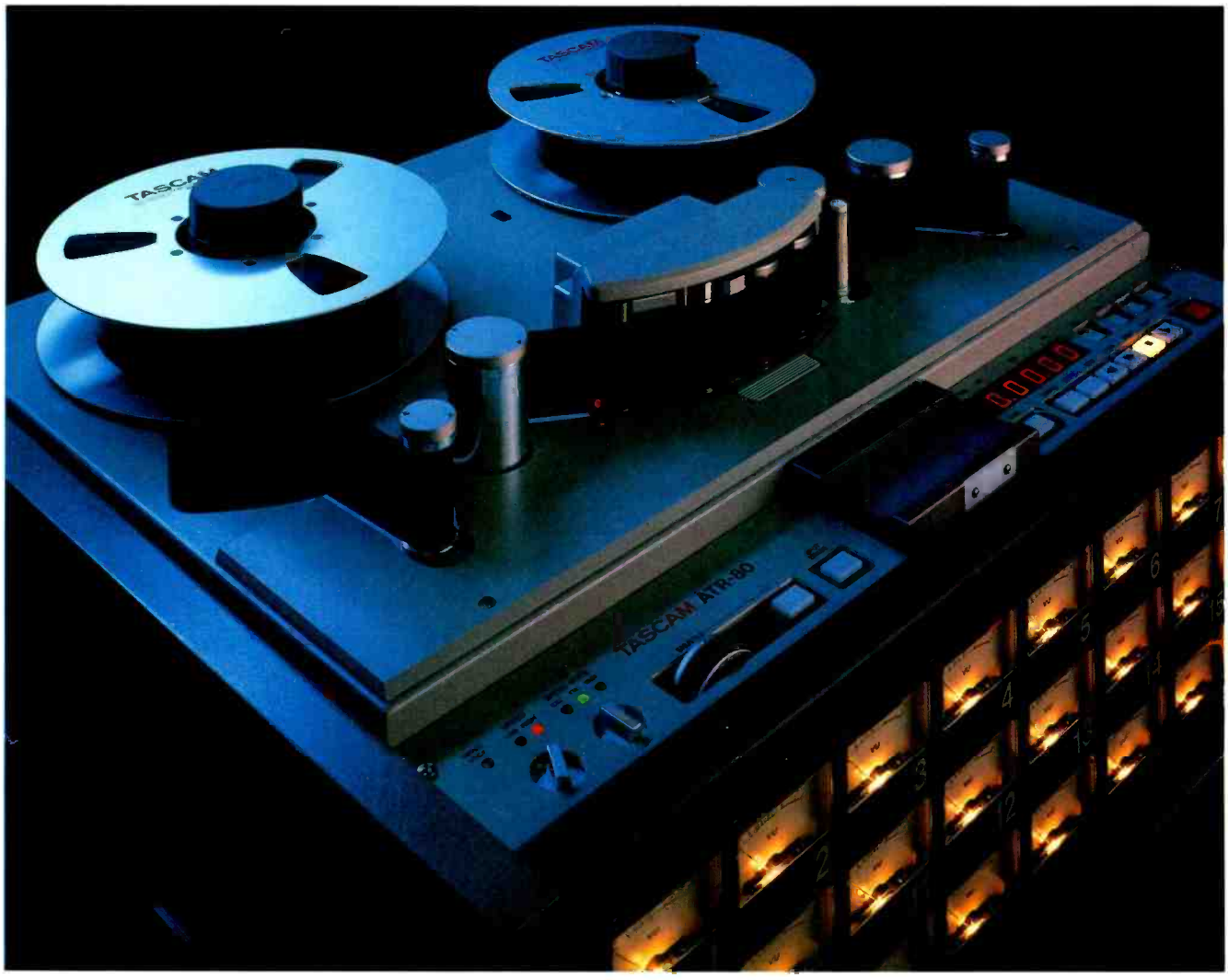


Fig 1: In both cases, increasing the voltage/pump pressure would increase bulb brightness/turbine speed which would always remain in proportion: Increase for increase, decrease for decrease. These are closed loop systems with voltage/current, brightness/speed, in-phase, and are principally resistively loaded.



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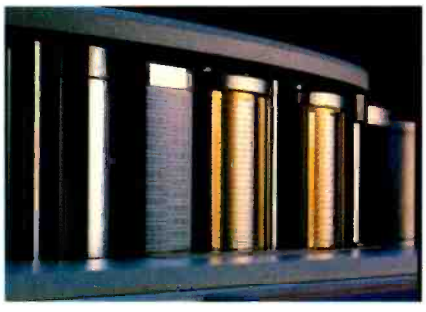

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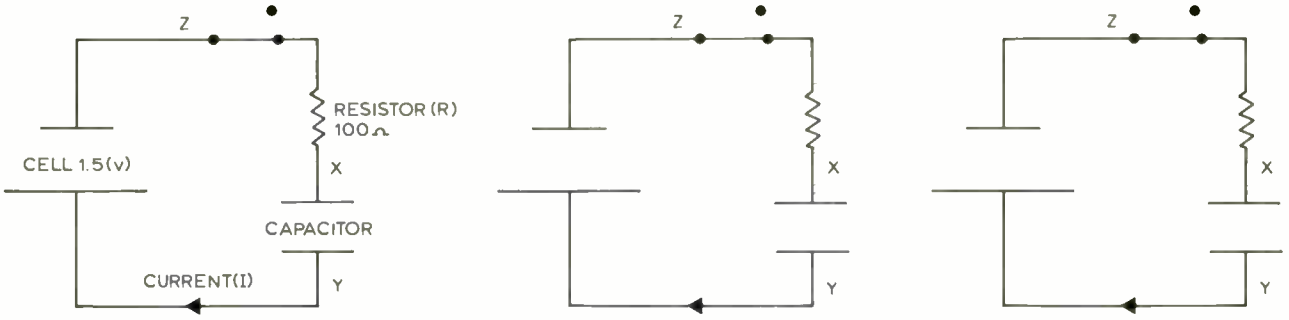


Fig 2:
Reactive load
Ohms law V = IR ($I=\frac{V}{R}$)

T=0
Instant of switch being closed
Capacitor begins to charge
Effectively short circuit
Current (I)=$\frac{1.5\ (V)}{100\ (R)}$=15 mA
Voltage ZY=1.5 V
Voltage XY=0 V
(short circuit)

Voltage across:
Resistor=1.5 V (max) (ZX)
Capacitor=0 (min) (XY)
Current through:
Resistor=15 mA (max)
Capacitor=15 mA (max)
∴where V=voltage
and I=current:
Resistor V/I max/max
Capacitor V/I min/max

T+5
Switch recently closed
Capacity charging but not yet fully charged

Current=7.5 mA
Voltage ZY=1.5 V
Voltage XY=0.75 V

Voltage across:
Resistor=0.75 V (falling) (ZX)
Capacitor=0.75 V (rising) (XY)
Current through:
Resistor=7.5 mA (falling)
Capacitor=7.5mA (falling)

Resistor V/I falling/falling
Capacitor V/I rising/falling

T+10
Some time after closing of switch
Capacitor has charged to full voltage of cell so no further current can flow

Current=0
Voltage ZY=1.5 V
Voltage XY=1.5 V
(effective open circuit)

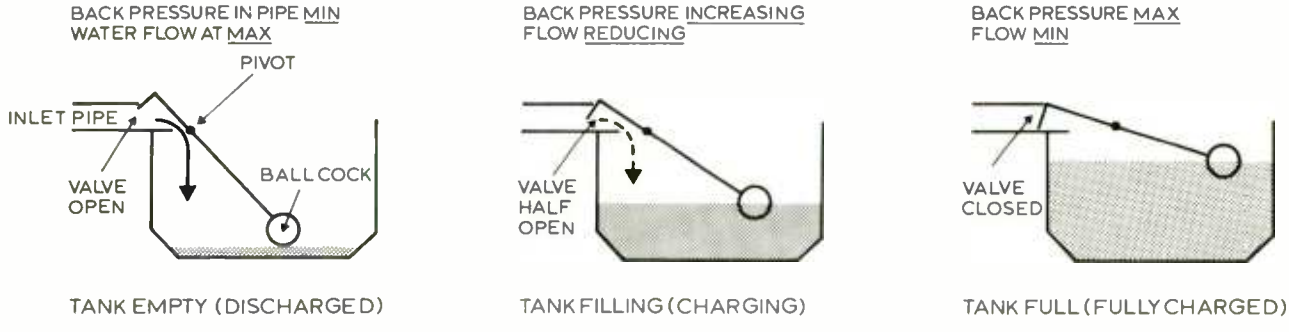
Voltage across:
Resistor=0 V (min) (ZX)
Capacitor: 1.5 V (max) (XY)
Current through:
Resistor=0 mA (min)
Capacitor=0 mA (min)

Resistor V/I min/min
Capacitor V/I max/min

NOTE: At any given time, the voltage and current for the resistor are either minimum, falling, or maximum together, hence in-phase. The voltage and current for the capacitor are always out of step. The current flows before the voltage rises, so is said to have a phase lead. The voltage has a corresponding phase lag.

The in-phase current and voltage in the resistor multiply together to produce watts—the resistor heats up—the power is dissipated.

The out-of-phase current and voltage in the capacitor multiply together to produce volt/amps or wattless power, the capacitor does not heat up, no power is dissipated.



Plumbing analogy of capacity: Flow (I) and back pressure (V) are always out of step at valve.

◁ conventional calculation of power being equal to the resistance multiplied by the square of the current, yet time and again, I hear people saying, "100 W into 8 Ω is about 3.5 A, so I'll use 5 A cable." The truth is, with reactive loading 100 W into 8 Ω may demand 20 A under certain loading and drive conditions. This obviously has a bearing on the current carrying/saturation ratings of any inductors used in passive, high level crossover circuitry. It also makes great demands of an amplifier in order to drive these absurd loads.

Back again to the choice of the Crown amplifiers, which work well with the systems I use them in. People will say one amplifier is better than another because they have made a series of careful listening tests. These listening tests usually are carefully controlled in one room,

◁ with the same programme material and obviously, through one set of loudspeakers as a reference standard. They draw up a table in their order of preference and then proclaim to all and sundry that Amplifier 'F' is definitely superior. It is a total farce. It may simply be that amplifier 'F' was more suited to matching the complex loading of the loudspeaker used. If the test were repeated 10 times using 10 different loudspeakers and the same series of amplifiers, if amplifier 'F' was more suited to matching the then maybe we could be getting close to some more rational generalisation. Even so, it is still quite possible that with a given complex impedance of yet another loudspeaker, Amplifier 'F' may run into trouble and Amplifier 'D' may prove superior. Rash generalisations that one amplifier, one loudspeaker driver, or one crossover is universally 'the best' is totally out of order.

At one time UREI refused to guarantee their loudspeakers if used with a MOSFET amplifier, common in the USA. The popular amplifier was prone to high frequency instability when driving the UREI *Time Align* crossover. UREI opted for a passive, high level crossover to the 'Long' design. They circumvented many of the impulse/phase caveats but were still left with the other problem of such crossovers, components in circuit between the amplifier and bass drivers. If the series inductors, in the crossover filters feeding the bass drivers have any resistance, which they must have, then the amplifier's damping capability is reduced. One Ω of resistance in series with an 8 Ω driver will limit any possible damping factor to around 8:1. This is the case whether the amplifier has a damping factor of 10, 100 or even 1,000. The bass cannot be truly 'punchy' unless the power amplifier has total authoritative control over the motor system of the loudspeaker. The prime requirement to achieve this is a damping factor somewhere in excess of 40 and a minimum cable/crossover resistance between the amplifier and the bass drivers. UREI made use of a sensing circuit, terminating on a separate connector on the back of the loudspeaker cabinet, which I believe took a feed from the bass loudspeaker, intending that feed to return to a separate control input on their specially designed amplifier. The above system helped to overcome the inherent problems with high level crossovers but was only effective with UREI's own amplifier.

Certainly in the UK, I rarely see these amplifiers in use. Too many people still think that an amplifier is an amplifier no mistake! They will not pay the price for the expensive, dedicated UREI amplifier but you get what you pay for. The UREI amplifier/crossover/loudspeaker is an integrated system. Replace or substitute any part of that system and you will not achieve the optimum performance. UREIs sound best with the UREI amplifier systems. So they should; they were designed to work together!

MOSFETs are inherently higher in internal resistance than comparable bipolar transistors. Consequently, when these seemingly ridiculously disproportionate currents are demanded by certain loudspeaker systems, MOSFETs cannot always deliver the goods. Obviously, this is most apparent on low frequency, high power applications. Also MOSFETs have inherently higher distortion than their bipolar counterparts. While this is often compensated for on steady-state signals by increased negative feedback, under certain drive conditions transients can pass through the system before the feedback systems can respond. There can be time delays in the application of feedback, due to the phase lead, phase lag properties, as already discussed, in any reactive components such as capacitors or inductors that may be in the feedback circuitry.

As a result of this, transients can sometimes cause the amplifier to momentarily run wild, with results such as instability and transient intermodulation distortion. I believe this was at the root of the problem with the amplifier that so disliked being used with the UREI loudspeakers. BGW have recently opted for a bipolar design, using discrete transistors rather than operational amplifiers, to minimise the distortion of the basic circuit and hence to require a much reduced overall amount of negative feedback. I totally agree with this principle.

My own preference in amplifiers has been for DC coupled devices. In these, there are no coupling capacitors and they operate as servo amplifiers. The feedback occurs at the speed of light, without any significant phase lead or lag over the range from say DC to 100 kHz. With my own design philosophies, they appear to provide the best overall performance, hence my continuing use of Crown, though they are typical of an entire *genre* of amplifier. They provide the performance I need from a specific philosophy of monitor system design. A Porsche gearbox may well be 'better' than a Ford Cortina's but don't even think about buying one to 'improve' your Ford Cortina. The ratios are probably all wrong and anyhow, it won't fit. The system must be considered as a whole.

Notwithstanding UREI's valiant efforts at system design around a passive, high level crossover, I still consider active, low level crossovers and multiple drive amplifiers, to be the only truly viable approach to high level, high quality monitoring. There are many reasons for this that have been discussed elsewhere, but four particular reasons relate specifically to crossover

design. They are component tolerance and ageing, power loss and component design problems at high power levels, response slope shaping particularly in high order (high slope) networks, and ability to select drive components for audible compatibility, free from restrictions imposed by impedance mismatches or sensitivity discrepancies. On the first point, component tolerance and ageing, electrolytic capacitors are the major culprits. The non-electrolytic option is not always open, especially when low frequency, low impedance and high power handling are the requirements. These necessitate large capacitances and high working voltages. Non-electrolytic capacitors may become so large that their own inherent inductance may become a hindrance to correct circuit design. These inductances can create undesirable elements in the circuit design and cause many hidden problems. It must not be forgotten that the voice coils of the drivers themselves are electro-mechanical components of the crossover circuits, and that drivers age. While this can usually be dealt with quite simply in active, multi-amplifier systems, their delicate and complex inter-relationship with passive, high level crossover components, may produce performance drifts, which are very difficult to remedy.

The second point was power loss and component design at high power levels. Especially with the higher order 18 and 24 dB/octave networks, the increased number of components can produce significant power losses. At high power levels, these losses can produce considerable heating of the components. The heating changes the value of the components, and hence the carefully chosen parameters can be subject to level-dependent drift. Heating in the voice coils will also change the driver impedances and once again, we move further away from our design criteria. We have already discussed the problem of finding suitable high value, high voltage capacitors but inductors can also be a problem. The difficult phase relationship of current and voltage can produce remarkably large transient current surges at high power levels. It can be difficult to design inductors that will not saturate at these high current levels and to keep them sufficiently small to avoid stray, leakage inductances, which once again upset the intended choice of component values. Furthermore, when the inductors are large, it becomes increasingly difficult to site them such that their stray magnetic fields do not interact one with another, causing even further complexities to an already very complex system.

The third restriction is on response shapes. These problems apply to passive crossovers, be they high level or low level. Passive crossovers can, of course, be used ahead of the amplifiers as with active crossovers but are only infrequently used in this way. The advantage of active crossovers, utilising the gain of an amplifier stage, either simply or cascaded, is to provide precise control over the filter shapes. While there is gain available in the amplifier stage, feedback can be used to provide contours that could not be achieved within the conventional circuitry of passive components.

The fourth drawback of the passive, high level approach is in the choice of drive units. Should it be decided on the grounds of subjective, audible, sound characteristics that mid-range unit 'A' with a sensitivity of 92 dB for 1 W at 1 metre, matched very well with bass driver 'B' of 97 dB sensitivity, a problem arises. Accepting the undesirability of introducing components between the bass driver and the power amplifier, and especially considering the far from constant impedance of the voice coil, the introduction of an accurate 5 dB pad into the circuit of the bass driver would be neither desirable nor realisable. The only option would be to transformer-couple the mid-range driver with a 5 dB step up. Once again, at high power levels, such a transformer, quite apart from weight, size and expense, could introduce even more phase shifts and inaccuracies into the system. It would not be desirable and would be difficult to put into practice.

The use of individual power amplifiers provides an almost ideal, low impedance, constant voltage signal source, which damp resonances, are tolerant of widely varying impedance irregularities, and provide a stable drive that can be altered easily should the need arise. They also accommodate the use of drivers of very differing sensitivities, compensation being made with simple adjustment of a gain control. Once again, back to the UREIs. They overcome the power level restraint of problem two with the use of very high sensitivity drive units but this must restrain the choice of problem four. I have been in studios that have replaced the original Eminence bass unit with Gauss and JBL drivers, blissfully unaware of the consequence of a sensitivity loss of at least 3 or 4 dB, and even up to 7 dB at the lower frequencies. With the intended amplifier of their own design, the UREI system incorporating separate feedback from the bass drivers, can probably really only be considered as a quasi-passive design. The most important thing is that it is a system and should not be disturbed. Truly passive, high level systems are not, in my opinion, suited to the wider frequency range, higher dynamic range, faster transients and more accurate phase/impulse response of modern digital recording; especially of computer-generated signals.□ *(continued next month)*

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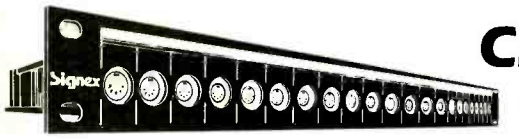

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CPX16

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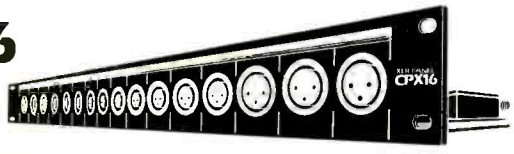

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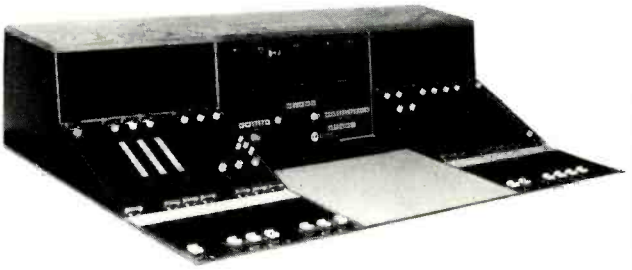

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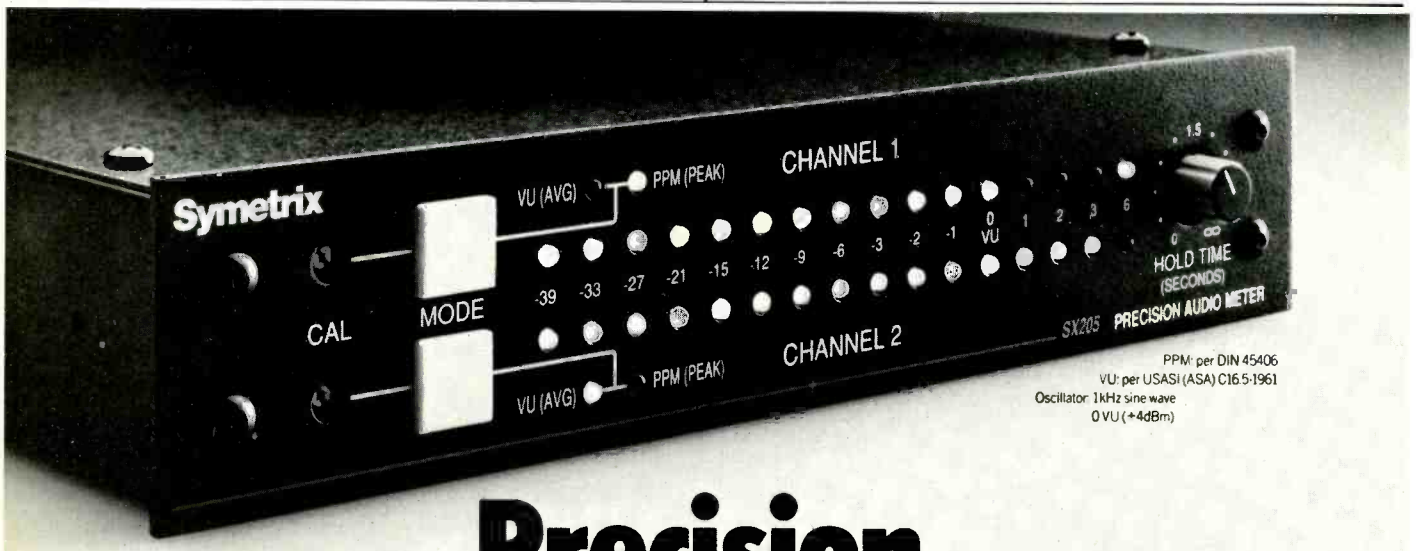

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DIGITAL AUDIO METERING

Rod Duggan Totalsystems Ltd calls for new standards in metering to meet the demands of digital signal level control

Metering for audio signals has evolved as an interface between engineer and equipment with developments that satisfy the needs of both. Specifications are summarised in several international standards documents and are readily available. However, changes in equipment due to digital signal processing and the associated engineering practices demand new international standards for digital audio metering.

VUs and PPMs

The vu meter is well liked in the recording industry, cheap and universally used. It is a moving coil meter with a full wave rectifier and a series resistor. This resistor (3.6 kΩ) means that the movement is not fed from a low impedance source, and hence almost no electrical damping is available. The response time is specified at 300 ms, and the decay time is the same. Short peaks of signal are not indicated but if you are feeding equipment that can cope easily with these—such as an analogue tape machine or a limiter—they may not be a problem. Bargraph versions of vu meters should have all these qualities reproduced with electronics and hence do not need further analysis.

The more sophisticated ppm shows programme peaks. Electronic circuitry puts the signal into a form easily shown on a moving coil meter or bargraph. Signal peaks are shown with fast response times to capture the shorter peaks, however, if this response time were repeated on the meter decay it would produce a very fast, frantic display, which would be unusable. Short troughs or 'negative peaks' (quietness) are not as relevant so the decay time is slowed considerably to ignore them. The result is the fast attack, slow decay movement that characterises ppms. The slow decay does omit information from the display but it shows those high level short peaks very well in contrast to the vu meter.

Typical differences between vu and ppm readings depending on programme material are quoted at between 8 and 20 dB. A common line-up level for digital equipment fed from vu metered sources is −15 dB (relative max bits) and this is marked on the metering scales of the *PCM-F1*, etc. The *PCM-1610* meter has its line-up reference at −20 dB, and is well known for needing high levels to record with and then bending vu meter needles on playback. A popular modification is to increase record sensitivity by 6 dB and reduce playback sensitivity by 6 dB, which gives us a new line-up level of −14 dB.

It is interesting to note that ppms are designed to gradually ignore signal peaks of less than a certain duration. The BBC, who developed early ppms, listened to material with short duration peaks causing clipping and discovered that very short clipped peaks were not audible. Anyone who has a power amplifier with clipping indicators, or has over-recorded on a digital recorder, must have noticed this occasionally. An integration time is therefore defined with various times and level differences to lower the sensitivity of the ppm for very short peaks, eg 10 ms reads 2.5 dB lower than steady state, 5 ms reads 4 dB lower, etc. The various ppm standards have slightly different values for this integration time and, more recently, analogue ppms intended for digital recording have been produced with much shorter integration times.

Considerations for digital metering

Equipment has headroom above a nominal signal level, where the nominal signal level is a level that has optimum performance, and the headroom is the amount this can be exceeded before clipping. In a tape machine we can set the nominal level at a certain nWb/m figure to give acceptable distortion, with tape saturation occurring perhaps 10 dB higher. We then have headroom of 10 dB. On a mixer we can choose a voltage to give levels with an acceptable S/N ratio, with clipping of signals occurring perhaps 18 dB higher. We then have headroom of 18 dB, enough for sudden large signals to pass without distorting. On digital equipment we have a noise floor that does not decrease with signal attenuation. (For noise floor in digital, read 'quantisation residue'. If there is no signal in digital, there is theoretically no noise.) We also have distortion figures that improve with increasing level right up to clipping. To make the most of the digital domain we need to work with levels as close to clipping as possible, so (a) where's the nominal signal level? and (b) what's the headroom? The answers are maximum bits and none.

This absence of headroom has serious implications for digital metering. With analogue, scales have been developed with part of the headroom above the nominal signal level indicated in red or shown brighter, so it is clear when signals enter the zone. For digital signals, this zone doesn't exist, so 'zero' on a digital meter should be right at the top, maximum bits.

Sony DMU-30 digital meter unit

We thus have to record as close to the maximum as possible, without going over it, and all peaks must be monitored. It makes sense therefore that the ppm is the more appropriate device to adapt for digital audio.

Direct digital metering

Analogue metering measures the electrical analogue of a signal, and as such is usually indirect, ie it does not directly measure the end quantity represented. Your vu/ppm meter is measuring a voltage that corresponds to a pre-aligned flux density on your analogue tape recorder. The console vu/ppm is reading only its applied voltage, with the end quantity dependent on some conversion figure that has been arrived ▷

Totalsystems DBM-1 digital bargraph meter

◁ at by measurement or theory, and included in the alignment. In the case of say, a transmitter, the peak deviation is made to displayed by a ppm because someone has carefully worked out and aligned the volts/kHz calibration of the signal chain. This indirect metering can be summed up by 'what you measure is not what you indicate– but the analogy has been carefully measured and calibrated'. Problems and the necessity for re-alignments can arise with programme exchanges when different organisations use different calibrations, eg different tape flux densities, different transmitter levels, general miscalibration, etc.

Bargraph displays are a line of segments, each addressable to illuminate the level to be shown. They can be numerically controlled, with each segment having its own unique address code. Digital audio is in a similar form, with each sample having its own numerical code. We can therefore avoid analogue conversion by correlating each bargraph segment with a predetermined digital audio sample code. This is done with a digital circuitry and the result is a very precise audio bargraph.

Digital metering, therefore, is derived from the bitstream directly, and so measures the recording medium directly with no conversion errors. A digital signal level is indicated that will remain the same throughout copying or transfer (assuming no digital 'processing' takes place). If we make sure always to clock words on the MSB (Most Significant Bit), different length words (16/18/20 bits) will always read the same level and ensure future meter compatibility. Direct metering means, in reality, that when you see the meter peaking to full scale you know you are using all the bits available and you needn't worry about meter calibration. It means you can check all the mastering levels on your CDs. You are monitoring the actual recording medium itself. It's rather like having a nWb/m meter reading off tape on your analogue machine.

This direct metering helps the exchange of signals between equipment or organisations as the digital levels are the same and thus predictable and precise.

Overloads in digital

In the digital domain, overloading occurs as hard clipping at maximum bits, ie 7FFF for positive peaks and 8000 for negative peaks. It is, therefore, impossible for a meter to show the amount by which a signal has overloaded the bitstream. We can detect this and then measure the length of time that the signal stays at maximum bits, ie measure the width of the continuously maximum portion of the signal and assume this suggests clipping has taken place. If the signal is at maximum for one sample, then it is safe to assume we have reached maximum level but no clipping has occurred. If for two samples then a small period of clipping has occurred, and so on. This clipping detection is used to activate an overload indicator for that particular channel.

If you play a test CD with a pure sinewave recorded at maximum level at 1 kHz, you will find maximum bits are reached for one sample only on signal peaks. If you play the 20 Hz maximum level track, however, you will find maximum bits are held for longer. In the case of the Technics *SH-CD001* this is seven samples, whereas on the Sony *YEDS-7* this is only five

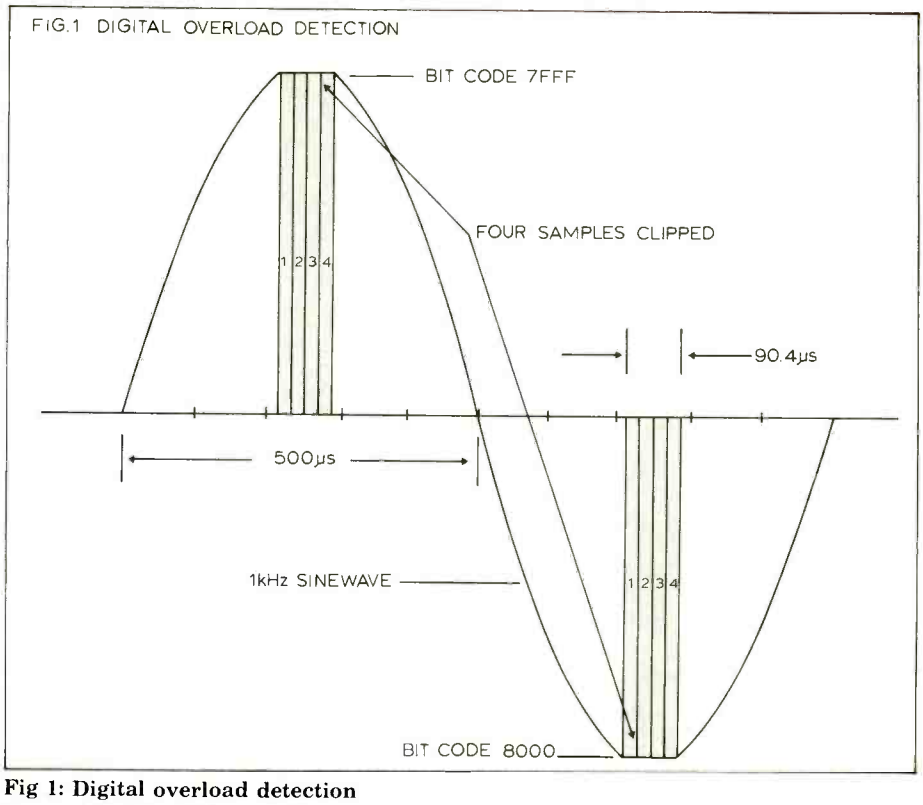


Fig 1: Digital overload detection

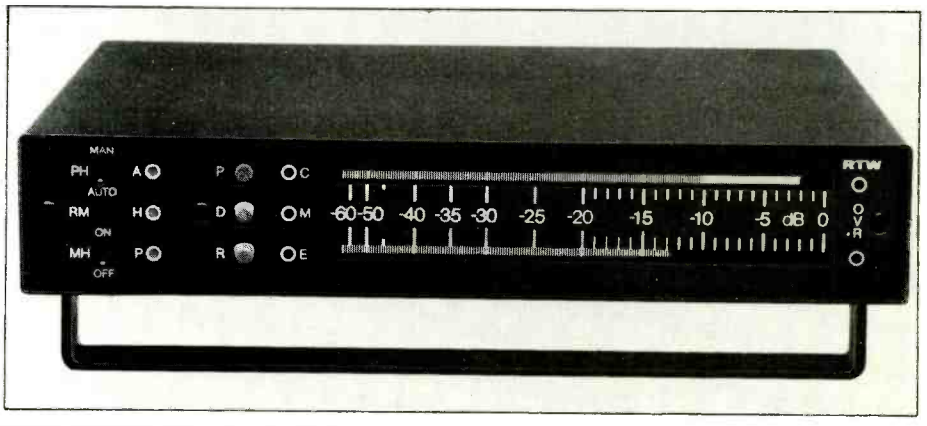
samples. A half cycle of 20 Hz contains about 1100 samples and at the peak of the sinewave, where the rate of change is zero, it is not surprising that these few samples are given the same numerical value. This does show a shortcoming of the digital overload detection, for if we have specified four samples as our threshold then the 20 Hz pure sinewave would have caused our master tape to be rejected although, to be fair, it is not a common occurrence in programme material.

We discussed the integration time of analogue meters earlier and the way ppms were designed with a specific integration time to reduce sensitivity to small signal peaks that caused inaudible distortion. Well the digital domain can cope with small overloads without giving rise to audible distortion. The system converts squarewaves (samples) to sinewaves (signals) as part of its normal operation, so a one sample 'overload' is invisible and inaudible, two samples gets well rounded off by the anti-aliasing filter (and oversampling if present), and the popularly accepted threshold figure for an overload to be indicated is four samples, ie clipping for 90 μs.

Fig 1 shows a 1 kHz sinewave suffering from this amount of clipping on signal peaks, which incidentally gives a THD figure of about 3%. This figure of four samples may be considered too low by some, however, its existence may be traced to the widely used Sony *DTA-2000* analyser, whose threshold settings are only available at two, four, eight or 16 samples, narrowing the choice to either four or eight samples in this region.

An illustration of all this is the Kate Bush CD *Hounds of Love*, on track 1 there are overloads of six and five samples each on the drums, left and right, at 4 min 01 s to 4 min 02 s. Also on track 5 at 4 min 47 s there is a five- and seven-sample overload, and on track 10 at 1 min 36 s a six- and 22-sample overload left and right. Theorists may say these are faults but can you hear them?

Digital equipment with its own overload detection often has the threshold set by the user by means of jumpers, switches, etc. The *PCM-F1* had it set at four samples (incidentally the *F1* metering was analogue, which is why 'over' could be indicated before maximum level were reached if meter calibration was slightly out). Some users are blissfully unaware of their threshold setting and confusion runs riot when separate units handle the same signal and have different overload settings. Perhaps it is time to set an agreed standard for this parameter. ▷



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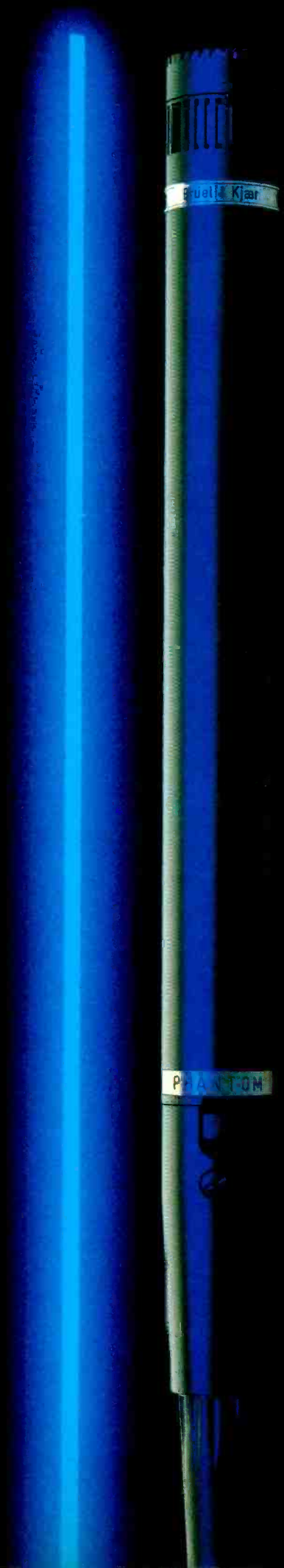

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Advantages of digital metering

With direct digital metering, the engineer has instantly gained access to the internal capabilities of his or her digital equipment. The A/D and D/A converters no longer act as barriers to signal metering. Any digital processing such as level changes or emphasis/equalisation are shown clearly on the meters. Once digital metering has been tried, it is generally accepted as invaluable and a more relevant indicator to digital recording than its analogue counterpart.

This means that with digital recording and mastering becoming more widely used, greater numbers of engineers will recognise the benefits of direct digital metering and thus promote its rapid growth and general acceptance throughout the professional audio industry. Therefore, we should look towards standardisation now.

At the moment, metering on digital audio equipment shows a variety of specifications, depending on the manufacturer's design. Examples of some of the scales commonly used, normalised to the same length for comparison, are shown in **Fig 2.**

What should be changed?

Some analogue meter specifications seem unnecessary or need to be altered when applied to digital metering:

Sensitivity has a different meaning as no analogue volts are involved. It needs only to mean maximum bits reads maximum scale.

Scale range of analogue ppms is from 24 dB (EBU) to 55 dB (DIN). The average digital scale is about 60 dB.

Integration time defined by the response to 5 kHz tone bursts, is not needed with digital metering as this only applies to the masking of overload signals, so perhaps even a one-sample peak should be indicated by the digital meter. (Integration is applied in the overload detection facility and this needs standardising.)

Decay time of 2 to 3 secs for 0 to −20 dB has been reduced on most digital metering and a figure of about 2 secs for 0 to 60 dB appears to be popular. This may be due to engineers previously accustomed to vu meters and the faster decay time that gives more detail and similar apparent ballistics.

Scale linearity appears to have been generally modified in digital, with expansion to the top 15 dB as compared to the approximately linear dB scale for analogue meters.

Delay time to be quoted as less than 300 or 150 ms in several analogue standards, may be viewed as being too long for modern digital recording.

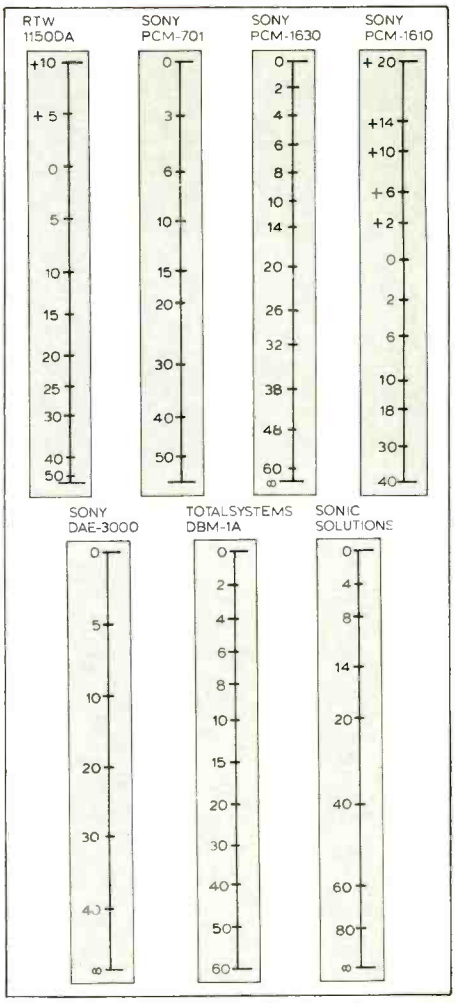

Fig 2: A comparison of several current digital audio meter scales

Conclusion

With such changes appearing in popular commercial equipment and the growth of digital audio, especially its advantages in transmission and copying, it appears that the old standards for analogue metering cannot easily be translated for the digital domain and the result is that manufacturers are producing their own 'standards'. Perhaps it is time the international organisations set about developing a digital metering standard specification to help conformity of recording practices and ensure the best possible use of the digital medium. □

Acknowledgement: Thanks to Finesplice Ltd for help in compiling this article.

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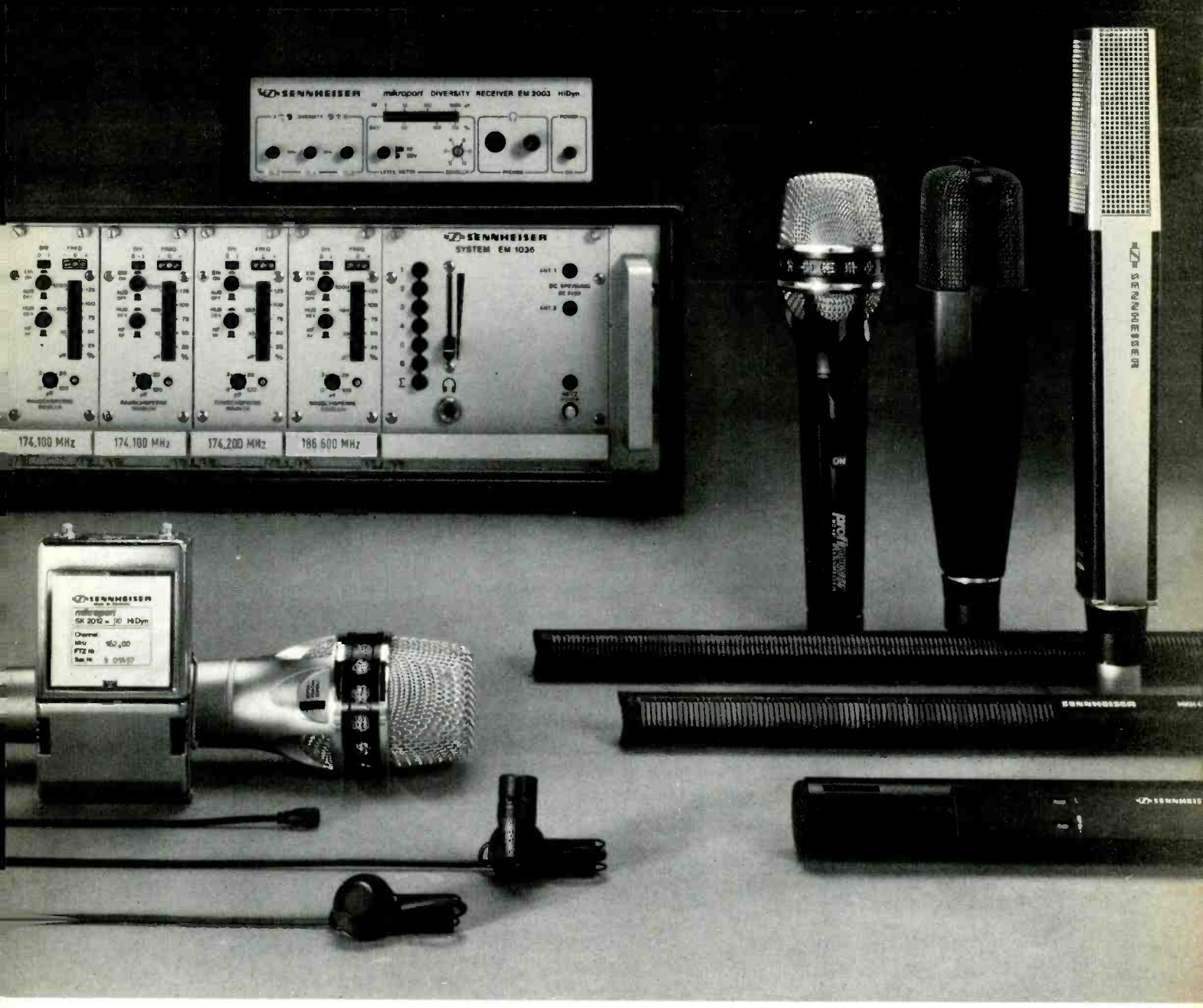

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Barry Fox

Stereo TV fiasco, live sound reinforcement and some advice to an editor

In mid September I got a call from a firm of market analysts employed by the British government's DTI (Department of Trade and Industry) to prepare a report on the market prospects for new television and sound technologies. The caller wanted to know where to get information on sales of professional studio equipment in the UK. I was my usual ratty self. Why is the DTI paying good money to a market research company to advise on a subject they know so little about that they have to phone a journalist to get a telephone number for information which is central to the report they are compiling?

But that's not the point of the story. It happens all the time and is probably why some sectors of UK industry are in such a mess.

The point of the story is that the researcher referred to stereo TV sound broadcasting as being 'in the future'. A week after the official start of Nicam stereo broadcasting in the UK, by Channel 4, Thames and LWT in London, and Yorkshire in the North, the DTI's advisers still didn't know it had happened.

Total publicity put out by the IBA amounted to one dull press release a month ahead of the launch. The ITV companies' publicity was one press release from Channel 4, which I never saw, and, judging from the lack of press coverage, no-one else saw either. What it needed was a press conference to explain what the event meant.

It becomes clearer every day, that the ITV companies are more worried about their franchise renewals in 1992, than technical niceties like stereo sound. Under the new government policy, the franchises will go to the highest bidders.

The ITV companies' lack of enthusiasm and the IBA's lack of publicity has turned off the advertising agencies who originally liked the idea of making commercials in stereo.

And even after the official launch there were still engineering hiccups, with the London transmitters still mistakenly sending out silence or test tones on the Nicam channels instead of programme sound.

In short, the launch of Nicam seems to have been a very well kept secret, even to the people running the stereo stations. The event will go down in history as a sodden wet squib. The real losers are those members of the public who have recently bought 'stereo' TV sets and VCRs and will soon find out that they are incapable of receiving Nicam.

In London the *Evening Standard* ran an advert from Dixons on September 21st—10 days after the start of Nicam—that featured a £450 non-Nicam TV set with 'four speaker stereo sound (from a stereo source)'. What is Nicam if not a stereo source?

Either the public doesn't bother to phone the IBA for advice anymore, or the IBA won't admit to the confusion it has caused over Nicam. But the BBC freely admits it has recently been flooded with calls from viewers asking how they go about converting stereo receivers for Nicam. "Sorry, you can't," is the BBC's stock response. The best viewers can hope for is an add-on black box, which strips out and decodes the Nicam signal ahead of the aerial socket.

Although the IBA has provided stereo links to transmitters, the studios have done little or nothing to install stereo switching. There is still confusion over the stereo sound-in-syncs system that is to be used. The BBC has a pile of SIS encoders ordered 3 years ago from Pye Varian. These work at 676 kbit/s, because the BBC feared that 728 kbit/s SIS signals would be too easily corrupted. The IBA opted for 728 encoders from RE Electronics in Copenhagen.

As previously reported, the first models tested by the IBA gave problems, which the IBA only admitted to when confronted with pages from their own internal report.

The IBA now says the problems are solved, with reframers to repair discontinuities in the data and mute briefly if errors cannot be corrected. The IBA also says that since the August Bank Holiday, 1989, Channel 4 has been using these encoders without any audible nasties. But the BBC needs convincing before it switches the order from Varian to RE.

By chance I was in Denmark recently with some BBC and ITV sound engineers. There are two TV stations in Denmark: TV1, which, like the BBC, has no commercials, and TV2, which is partially supported by commercials and partially financed by the very heavy licence fees that viewers pay (almost twice the British fee of £66). Both are committed to Nicam.

TV1 converted their main transmitter in Copenhagen to Nicam. This serves one-third of the population and although there has been no formal announcement of a stereo service, anything available in stereo is broadcast in Nicam.

TV2, which started broadcasting in October 1988, was equipped for Nicam from day one and serves the whole country. But it is also still an unofficial service.

Not surprisingly, Danish television is buying its sound-in-sync stereo encoders from RE. But TV1 admitted to us that its stereo sound was still going to the transmitter by analogue FM subcarrier. Why? Because, says TV1's sound boss, there are still problems with the clicks, splats and pops encountered by the IBA during tests with ITN. (News people like the idea of digital stereo because they can keep clean sound effects and clean commentary separate.) Some of the TV2 links use the SIS encoder, others use 140 Mbit/s microwave links.

BBC, ITV and press ears flapped at this revelation.

The next day we all attended a meeting

arranged by the Danish Pro Audio Group, of which RE is a member. A representative of RE was scheduled to attend but he failed to appear. Obviously embarrassed, the very professional Pro Audio Group organiser later asked why.

RE had *forgotten* the meeting with the BBC, ITV and press!

This is probably the first time in history, that the consumer electronics industry has been ready with a full range of the domestic hardware needed to receive a new service, before the service suppliers are ready with hardware to deliver it.

You can be sure that the consumer electronics industry will remember this, next time the broadcast industry asks them to support another new service launch. The next time will be in 1992, when the Barcelona Olympics are broadcast in high definition TV. I wonder whether the market analysts preparing that report for the DTI will have recognised this unpleasant home truth.

More on 'live' sound. For 25 years the BBC has kept an in-house Radio Orchestra that tackles all types of popular music, from classical hits, through sweeping strings to big band jazz. Musically the standard has always been very high.

The Radio Orchestra gave a concert recently to celebrate its 25th anniversary. The BBC booked the Queen Elizabeth Hall on London's South Bank and gave tickets to selected listeners. The augmented orchestra has around 65 musicians. The Queen Elizabeth Hall is quite small, and has a live acoustic.

So what happens? All the musicians were close miked, for recording for future broadcast. But was it really necessary to pipe the live sound through a couple of small stacks at the side front of stage, with the level wound up so high that even towards the rear of the hall, the sound of the drummer's high hat was quite clearly coming from the speakers, rather than the orchestra?

Do 65 musicians really need electronic assistance in a small, live hall? In a situation like this surely the sound reinforcement should simply be used to give soloists a little lift.

The sad truth is that in 25 years the public has grown to expect a reproduced sound, even from live orchestras that need only subtle assistance. A generation has grown up so used to listening to sound through loudspeakers, that live sound no longer sounds right.

Mike Thorne used to edit *Studio Sound* and sister publication *Hi-Fi News*. He went on to become a highly successful record producer, with hits from Soft Cell and the Communards.

I sat with Thorne in a pub recently. We talked about some musician's obsession with playing too many notes. Or the juke box there was a nice R'n'B track with a simple trombone solo—just a few well-chosen and well-placed notes.

"Everyone benefits from a bit of editing," he remarked. "Someone to say, hold on, you don't have to prove anything. In fact, being a record producer is not much different from editing *Studio Sound*."□

Martin Polon

Our US columnist takes his hat off to a brave new world and looks back in angst

Entering the 1990s is a lot like going out with a girl on a blind date who has a mysterious secret: you're going to do it but you really aren't sure whether you are going to enjoy it or not. For better or worse, we are on the threshold of the last decade of the 20th century. It would seem that we look forward with much more trepidation than our forbears did 100 years ago. To them, the end of the 19th century and the coming of the 20th promised to embrace the coming of the world of technology. Now, technology has come and come and come. We are not altogether sure that we want it to continue coming at quite the pace we have accepted over the last few decades.

Before we look forward, it might be interesting to look back. One could clearly identify the beginning of the 'audio era' with the 1920s and the commercial advent of amplification. The pioneering work of companies such as ATT and their Bell Labs, EMI, General Electric, RCA and Telefunken made amplification a commercial reality. The radio and recording studios of the 1930s used by the BBC, CBS and NBC, RCA records and others helped to produce programmes and recordings that entered millions of homes around the world. The dulcet tones of Al Jolson poured from the movie screen at the same time that ATT was experimenting with stereo sound. It was at this time that the concept of electronic entertainment actually became a reality. Also a harsh reality was the direct access to news and newsmakers given us by the live radio reports of those whose names became household words (Murrow, Kaltenborn, Shirer) while covering other household names not so pleasantly connotated (Hitler, Mussolini, Tojo). By 1939, television had become a reality for the BBC at the Crystal Palace and for RCA at the 1939 New York World Fair.

The '40s were a difficult time, at least for the first half of the decade. It became obvious that there was a downside to the miracle of audio amplification. Through it and radio, and with the aid of the newest innovation of magnetic tape recording via Magnetophone, Adolf Hitler and others like him were able to rouse the masses into the suicidal horrors of World War II. Without his rallies and broadcasts, Hitler would have faded from the scene quite quickly. Fortunately, the radio also nurtured Winston Churchill and Franklin Delano Roosevelt to raise the democracies into an arsenal of victory. The Goddess of Audio also supplied the Gods of War with acoustic fuses—torpedoes, mines, huge Beachmaster sound systems to co-ordinate the Normandy landings and other developments.

The second half of the '40s was marked by the gradual absorption of the incredible quantum leaps that World War II had brought to the business of audio and entertainment technology. We had audio power tubes—both pentodes and triodes, miniature tubes, FM broadcasting, audio tape recorders, the first not-so-crude hi-fi systems and, of course, the LP record. Disney and Bell Labs had already brought multiple tracks of film sound to the big screen with *Fantasia*. It's funny how what comes around comes around again. The first hi-fi fanatics began to mount Altec, Goodmans and Wharfedale speakers in closet doors or in walls to take advantage of the infinite baffling—today's latest trend to the tee from Kef and other makers.

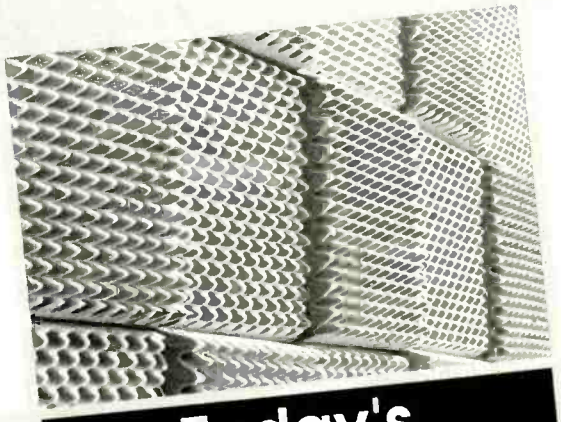
The story of the driving force behind tape recording is a good one, too. It seems that fledgling Magnetophone-inspired Ampex Corporation had trouble selling the idea of broadcast audio tape recording until Bing Crosby heard about it and realised how much extra time it would give him on the golf course. In a way, the late 1940s were a time for the birth of high fidelity audio as we know it today. The combination of high fidelity FM radio and long playing (LP) discs plus the post-war entrepreneurial push for anything electronic meant that there were quality sources of music and quality makers of speakers, amplifiers and tuners. And of course, for what it was worth, there was the new kid on the block—television.

The '50s have to bear the accusing finger of history as a time when things were 'just too perfect'. The middle classes rose to prominence, women in starched dresses and heels, with perfect hairdos and pearls raised perfect children to go off to perfect schools; even if at least one of those kids was called 'Beaver'. Men clothed successfully and synonymously in grey flannel went off to work daily with rolled newspaper and umbrella. Cars attempted to emulate the styling imagined for space flight. Everybody had more of everything than they had ever had before. Rationing in the UK and the US just seemed a bad memory. The '50s were an age of recovery, an age of national pride, an age of political stability and of suburbia. The demand for consumer goods had never been stronger and it begot the time of the universal audio system installed in every hearth in every cottage and castle from Edinburgh to Brighton and Boston to San Diego. All of these wonders started in Los Angeles so the myth went and travelled across the country and then across the waters. Britain had a painful 'due bill' to pay over World War II and achieved its triumphs about 5 years behind America. Audio continued to expand and improve as stereo debuted via dual-headed binaural record players and 2-track tape machines for the studio as well as for the home. There really was an audio industry with both a user base and equipment makers. Recording was going great guns as new titles for catalogues were being laid down. Then, near the end came Sputnik and the dream seemed to crumble.

If the '50s seemed like a daydream until Sputnik, it may have been because so much of what was sold was designed to sell rather than to last. Replacement had become a way of life. Even in the more stable audio field the consumer and the pro were being asked to discard the first generation of tube monophonic equipment for the second generation of tubed mono and then in the '60s to go for tubed stereo and again in the '60s to buy the first generation of solid state stereo. Consume, consume, consume. But it was more than that. It was a time of rebellion. Rebellion against the staid ways of the '50s, against 'the system', against the war in Vietnam or Northern Ireland, against almost anything. Against boring suburbs, trading stamps at markets, planned obsolescence, conformity and women's clothes with skirts wider than doorways. In audio, the rebellion against analogue and against monaural sound was raised. The stereo LP took off and obliterated all of the 7½ inch open-reel stereo tapes that had been struggling along as a release format. The first wave of products from Japan arrived on foreign shores; being neither reliable nor exciting but having such copious volume and low prices that an ominous chill was felt up and down the electronics industry.

The '70s were in many ways analogous to the '50s; at least for audio. It was a time of relative prosperity and super groups. The record industry set new records for sales and recording studios were the hottest properties in town. The '70s was the decade of touring and both groups and sound systems were honed to the cutting edge of state-of-the-art. The Grateful Dead carried a sound system with three times the power of a clear channel 1A AM radio station. Vacuum tubes dropped by the wayside as the transistor gave way to the IC chip with multitudes of transistors. Auto sound became a legitimate category and FM stereo radio had become a routine feature. Products from Japan had become reliable if not too exciting and certainly in copious supply at reasonable prices. Woody Allen made *Sleeper* and movies became big business as successful single picture revenues worldwide moved into the nine-digit category. The general population found the '70s an era of doubt. Doubt over Vietnam or Northern Ireland questioned our national goals, politicians' influence, as did messy scandals. We experienced an energy crisis and a recession. Some would say that the digital audio era was established at this time in labs and classrooms as the mathematics of digitisation were consummated. And oh yes, the '70s saw the introduction of the portable personal stereo which was certainly a shot in the arm to the audio business at all levels and for decades to come.

The '80s have been the decade of zero or one, yes or no, open or closed. Binary. Digital. Computer. The personal computer, based on an engineered specification from IBM swept the world. The decade was also marked by the advent of digital compact discs, digital synthesis, MIDI buses and equipment, digital DAT recorders, digital multitrack audio recording, digital consoles and digital audio editors and workstations. Not to forget home studio digital mixers, digital 2-track and multitrack recorders, digital signal processors, digital reverb, digital equalisers, etc. ▷



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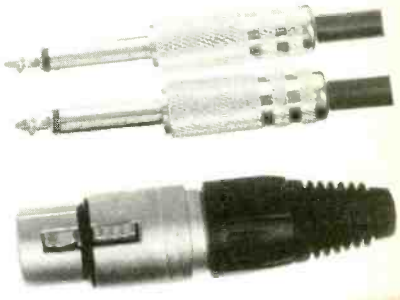

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Products from Japan that were and are reliable, exciting, innovative and relatively affordable. The LP was given its death sentence. For many, the '80s will be remembered as the era of information—the audio information age if you will. Stereo television was born only to lapse into a creative coma as it reached its 3rd birthday.

The '90s. Where will the '90s take us? To new heights or to maturation and/or stagnation? Good questions deserving good answers. One thing is clear. The level of success to be achieved by the professional audio marketplace in the 1990s is solely dependent upon the amount of consumer business to be generated during the same time frame. High levels of consumer record buying, movie attendance and TV watching yield large volumes of recording studio activity, post-production work, concerts and tours, upgrading and new installation of sound systems for performance centres, *ad infinitum*. New professional equipment sales would similarly boom. So the strong interconnection of the professional and consumer sectors in audio means that both areas must succeed together. Listed below, are three issues that could severely impact the further development of audio in the '90s.

Copyright protection of recorded materials is perhaps the most important technical issue to be faced in the 1990s, since it can exert the most constraint against any new technology developed. As long as the copyright holders, in this case the record companies, persist in their claim that there is no established precedent of fair use for any new digital technology, the options for the free and open introduction of new products diminishes in mathematical lock step with the record companies' complaints. We have already seen this happen during the end of the 1980s with the campaign of intimidation against DAT importation and the barely veiled threats against the introduction of recording optical disk and/or recording CD. None of this is to say that there is not a valid position for the record companies in trying to protect their profitability. It is just that this position is diametrically opposed to open use of any new digital recording format.

In the '90s we have every reason to expect to find large optical disk drives replacing large hard disk drives in next generation personal computers. Every such machine will be able to record and playback digital audio, without flaw. The record companies will want a multiple recording 'spoiler' introduced to prevent unrestricted copying. They will also want large sums of money for tribute in the form of royalties on both equipment and software. So if a restrictive atmosphere is created over this issue, the likely result will be to poison the waters for further development and still the market for current developments. Certainly an issue worth watching in terms of the impact on product development and product developers.

Consumer interest in audio and audio-related hardware and software products could continue to be weak. Weak consumer demand could consequently soften demand within the professional marketplace. The issue is to keep the creation of new product foremost in an endless loop. Keep the music, film and television software pipeline filled from the top to the bottom. That means business to the audio professional. Unfortunately, the most important question of 1990 is also the most important question of 1989 and the jury is still out: 'Do the entertainment software providers, the movie and record companies, recognise that the mass and affluent marketplace is in the hands of the over 35s?'

As the baby boomers who fuelled the record and movie boom of the late 1970s marry, have their own 'baby boomers baby boom', and buy houses as well as grow older, their tastes and discretionary spending patterns change. Beyond this grouping, who will all enter their 40th year during the '90s, there exists another majority of 50 and older. For the '90s, the population thrust is toward the over 40s who have the affluence to sustain any trend that catches their fancy. It is not that the youth market so favoured by record and film promoters currently has to be abandoned, rather the entertainment software companies have to offer records and tapes and films desirable to the all too numerous greying citizens of the western world as well.

Do the movie and record companies recognise where the mass and affluent marketplace is?

With the LP record, there was always a catalogue of 10,000 titles and at least half of that number were relatively easily accessible in one way or another. Now, both record companies and record stores prefer to stock in a minimalist sense and availability is discouraged. No quantum jump in future growth in the audio marketplace can take place without the diversity demanded by the 'old folks'. It is curious that the car companies point at least half their offerings towards the over-35s while the record companies merely throw classical music as a sop to the so-called 'older' buyer. Looking forward into 1990, it does not appear that rock groups Midnight Oil, The Sugarcubes, and Living Colour will capture many sales of the over 35s. Neither will the 12 major feature films pointed at romantic love among the yuppies turn the trick. "I want Indiana Jones—not Indiana Harvest Moon, after a day with the office, my mortgage and two kids," cooed one stressed-out super-Dad recently.

Even the act of trying to buy music is frequently offputting to anyone over the age of 20. Most suburban record stores are in shopping malls, business districts or on high streets. Except for megastores in national and regional centres, the catalogue stock of most stores excludes most titles that are favoured by 'older' consumers. The cupboard is frequently bare if you want Elvis Presley instead of Elvis Costello or (BJ) Billy Joel instead of (BJ) Bon Jovi. The other problem is the presence in virtually all suburban and exurban record palaces of a 6 foot tall, air headed, motorcycle leather clad down to his shorts behemoth who is usually found trying to 'chat up' the 16-year-old female store clerk. This beast in his spike-clad leather 'sans deodorant', usually called 'Spud', 'Spike' or 'Killer', does not encourage a 55-year-old woman to enter the store even though she might spend $100 on gifts if she felt safe enough to come in and shop. The beast will not spend $100 in the store in a year. Bottom line—the youth market is no longer the mass market.

The third conundrum clouding the technological future of audio in the 1990s is similar to the first; in this case, the issue of 'look and feel' in software. Another copyright issue, in this case inherited directly from the computer industry, 'look and feel' is being smilingly labelled the 'copyright lawyer's full employment act of 1990'. With the personal computer likely to replace much of the dedicated audio hardware used or developed in the '70s or the '80s for studio work, the importance of function-defining software cannot be too heavily overstated. It is already expected to replace dedicated digital equalisers, reverbs and signal processors since these units are basically PC-type microprocessors with changeable analogue circuit interfaces driven by the computer and its associated software. As we venture into the world of PCs with hundreds of thousands of Mbytes of dynamic RAM and optical or hard drives with several Gbytes of storage, the ability to record, play, edit, manipulate and process in realtime becomes just another feature with the appropriate software. Several analysts of a companion industry to audio, that of video editing and post-production, expect much of the dedicated machinery of video plus audio editing and audio-for-video post-production to be completely replaced during the '90s by super PCs running editing software. Software will become the audio 'power' issue of the last decade of the century. It will define form and function for the PC or PC-type hardware.

The threat of 'look and feel' is to reduce competition among audio program developers. The first piece of computer software dedicated to turning a PC into a digital audio workstation could be the only piece of software with a legal franchise if 'sounds like' is added to 'look and feel'. More important, in audio many functions and features are common by necessity. It would be very difficult to survive a 'sounds like' or a 'look and feel' challenge if the principle of 'look and feel' is permanently validated by the courts. The threat of 'look and feel' is to eliminate competition and improvement from the business of software development. Since so many dedicated pieces of audio hardware today are dependent upon complex operating software, the dangers of a 'look and feel' suit by a competitor are very real.

It is clear to all observers that the '90s have the potential to be the most exciting era ever for electronic entertainment. There is no doubt that improvements in digital audio editing, recording, mixing and playback will happen. There is no doubt that the technology will continue to march forwards. The doubt is that the infrastructure of law, demand and management will allow technology the freedom of movement of past decades. Let's fervently hope that the '90s are not characterised in a similar piece 50 years hence as 'not with a bang but a whimper'.□

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