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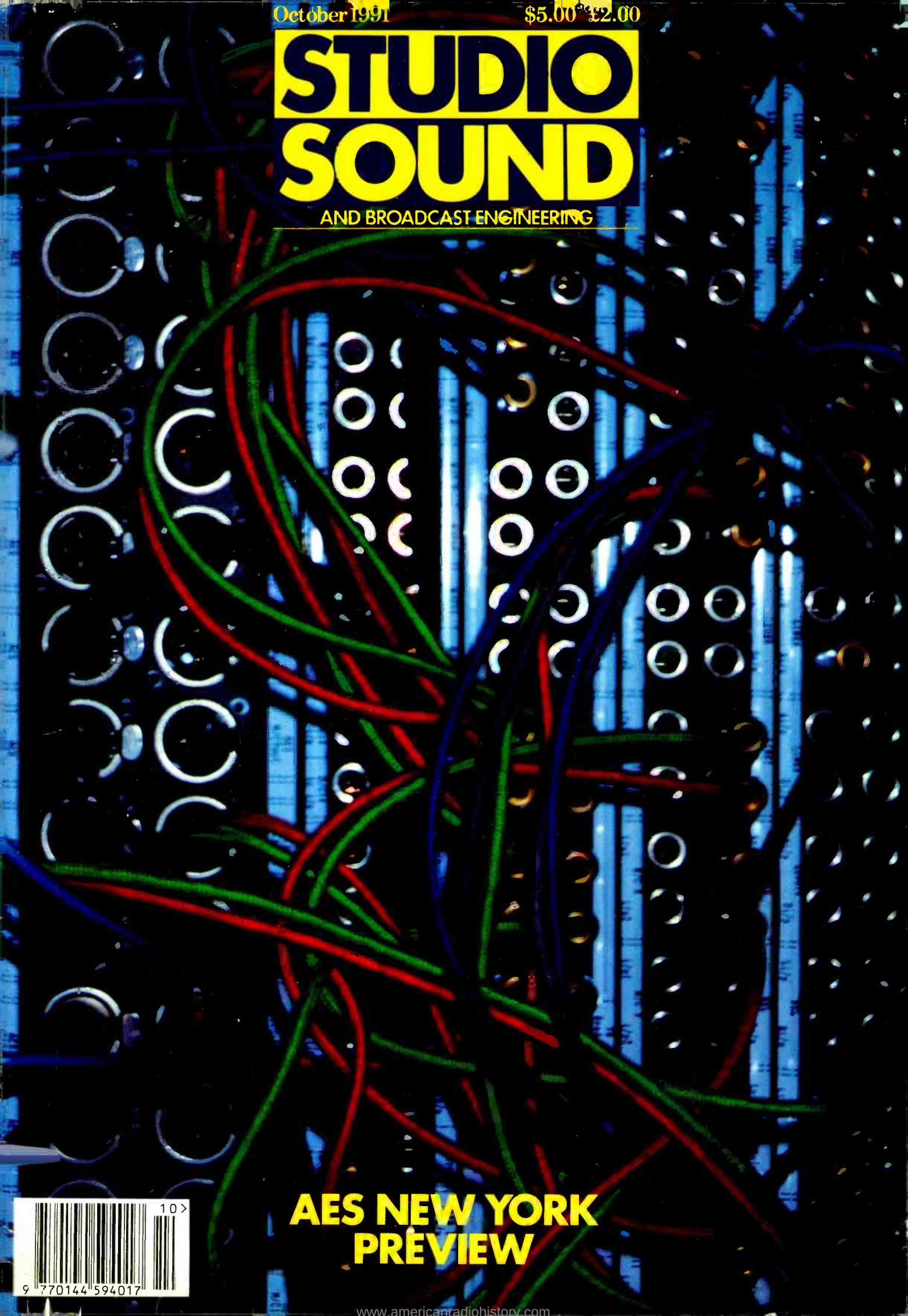
STUDIO SOUND

AND BROADCAST ENGINEERING



AES NEW YORK PREVIEW

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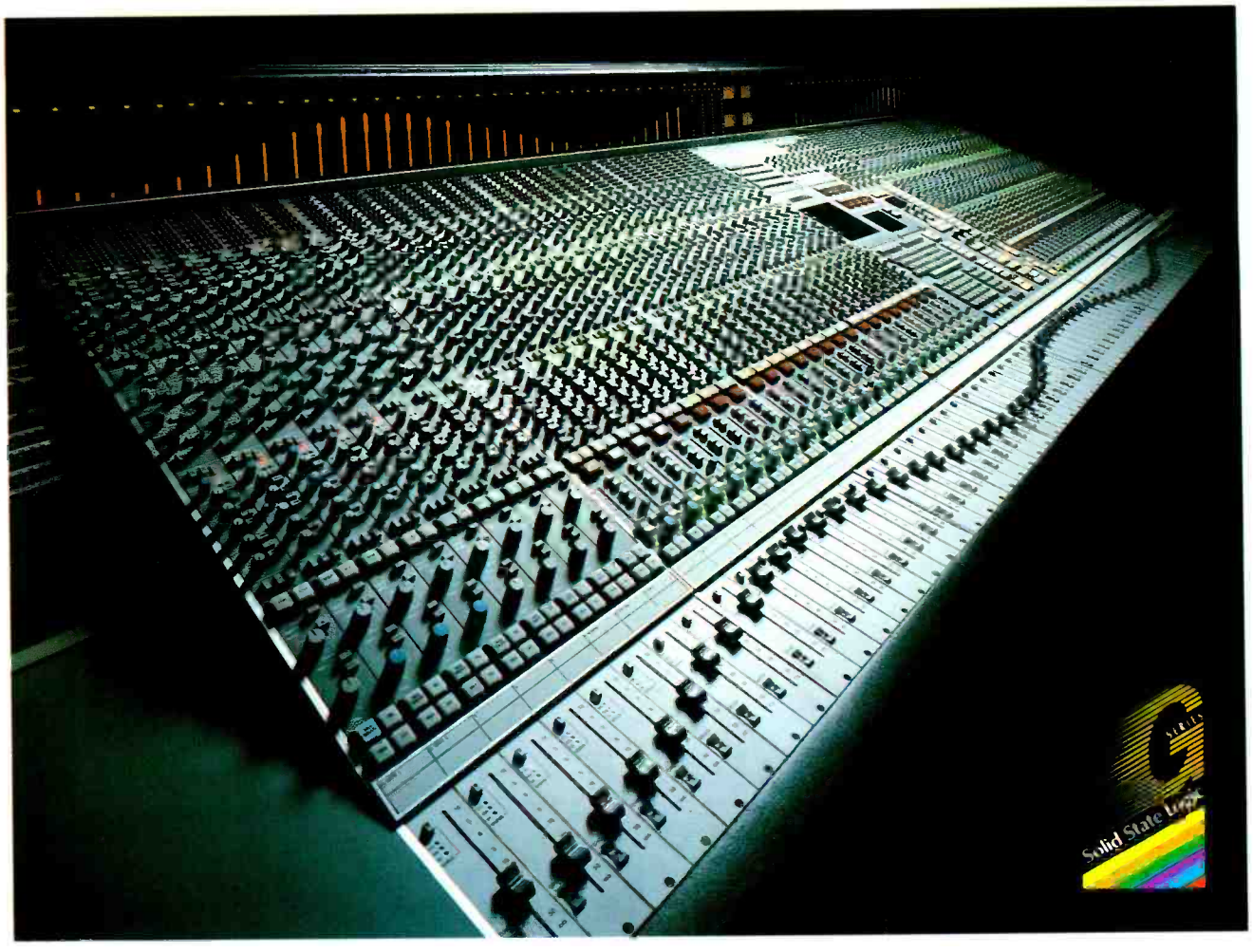
The Churchill Tapes



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STUDIO SOUND
AND BROADCAST ENGINEERING

October 1991
Volume 33 Number 10
ISSN 0144 5944

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EDITORIAL & ADVERTISING OFFICES
Spotlight Publications Ltd, 8th Floor, Ludgate House, 245 Blackfriars Road, London SE1 9UR, UK. Tel: 071-620 3636. Fax: 071-401 8036.

NEWSTRADE DISTRIBUTION (UK)
UMD, 1 Benwell Road, London N7 7AX, UK. Tel: 071-700 4600. Fax: 071-607 3352.

AMERICAN OFFICE
Publishing Services Manager: Fred Vega
Studio Sound, 2 Park Avenue, 18th Floor, New York, NY 10016, USA. Tel: (212) 779-1212. Fax: (212) 213-3484.

JAPAN & FAR EAST AGENT
Contact: Mikio Tsuchiya, Media Sales Japan, Inc, Tamuracho Bldg 3 3 14, Shimbashi, Minato-Ku, Tokyo, Japan. Tel: (03) 504-1925. Telex: J25666. Fax: (03) 595-1709.

Origination by Project Three Filmsetters Ltd, The Print Centre, 70-72 John Wilson Business Park, Thanet Way, Chestfield, Kent CT5 3QT, UK
Printed in England by Riverside Press, St Ives plc, 2 Grant Close, Gillingham Business Park, Gillingham, Kent ME8 0QB, UK.

Studio Sound and Broadcast Engineering incorporates Sound International and Beat Instrumental.

Studio Sound is published on the third Friday of the month preceding the cover date. The magazine is available on a rigidly controlled requested basis, only to qualified personnel (see classified advertisement for terms)

ABC

Total average net circulation of 19,166 per issue during 1990.
UK: 6,655. Overseas: 12,511.
(ABC audited)

Hard on software

I really wish I could remember who it was that said the best computer in the world is the human brain: designed by God, built by amateurs and runs on sugar. I don't know if that is all of the quote — I mean they didn't mention the software or the programmers.

There is so little equipment now of any type where you don't need to mention the software. However, to the end user of the equipment the software is such an intangible that it is at once infinitely versatile and infuriatingly limited, so ephemeral that it seems without limit of possibility yet so often never what you want.

Despite advancements in the techniques and abilities of software there is still an air of acceptance that the perceived role of software is a problem. It is probable that I am not viewing this from the typical position of an end user or a manufacturer and so look at me as an impartial observer. We are unfortunately not in receipt of the manufacturers mailings to registered end users so we do not know exactly what you receive. Equally, I am sure I don't have any precise idea about the degree of feedback the opposite direction. But I do hear the complaints and these cover the extremes and all points within. From "How can any manufacturer legally release software that is not free of bugs?" to "We need to get the software out there as the end user is likely to find the bugs in the system quicker than we could!" — both actual statements.

I have long looked admiringly at the lists of latest software releases that the computer magazines publish, their ability to openly review all amendments and comment on the features of the updates. This is a degree of openness that is just not found in our industry — we may be told about major upgrades but the smaller but important amendments and fixes we are generally ignorant of. I have frequently made approaches to most of the major manufacturers over this but while they may listen the reaction has been very poor.

There is a real need for a more mature approach to software from all parties. While avoiding the comment 'it's only a matter of software' there are a number of changes that need to occur. Users may have to take a realistic attitude to the costs of software, which in turn the manufacturer should reflect in being more responsible about premature release. Greater openness even about 'fix' releases would seem to be in order because surely if it fixes a problem who would complain and if it adds facilities this should then be treated as new feature. In the audio business we have a degree of uniqueness over the computer market in that it is commonly the same people that produce the hardware that deliver the software and maybe this accounts for a degree of reticence that is shown when compared to simply software companies operating on an established hardware platform from a third party.

From my position as a neutral observer it does seem that many of the prevalent attitudes to software that may be holding back mutual development are totally interlinked. So there is always the problem of who moves first. With manufacturers tending to play down new software (which in reality is a new product), it is this attitude which may make the end user suspicious and/or in turn sceptical and not viewing updates in a positive manner. And certainly being most reluctant to pay for it.

From the other way, we need the user to understand the complexities of software and the wide permutation of circumstances that are very difficult to permutate all before release.

I have no answers. I would welcome comments from all sides particularly if you feel that my view as a neutral observer is askew.

PS: Having reread this page I am not sure we didn't have exactly the same sort of problems when all our purchased wares were simply hard rather than also soft!

Keith Spencer-Allen

Cover: Patchfield decor. Photography by Patrick Stapley.

THE COVER VERSIONS

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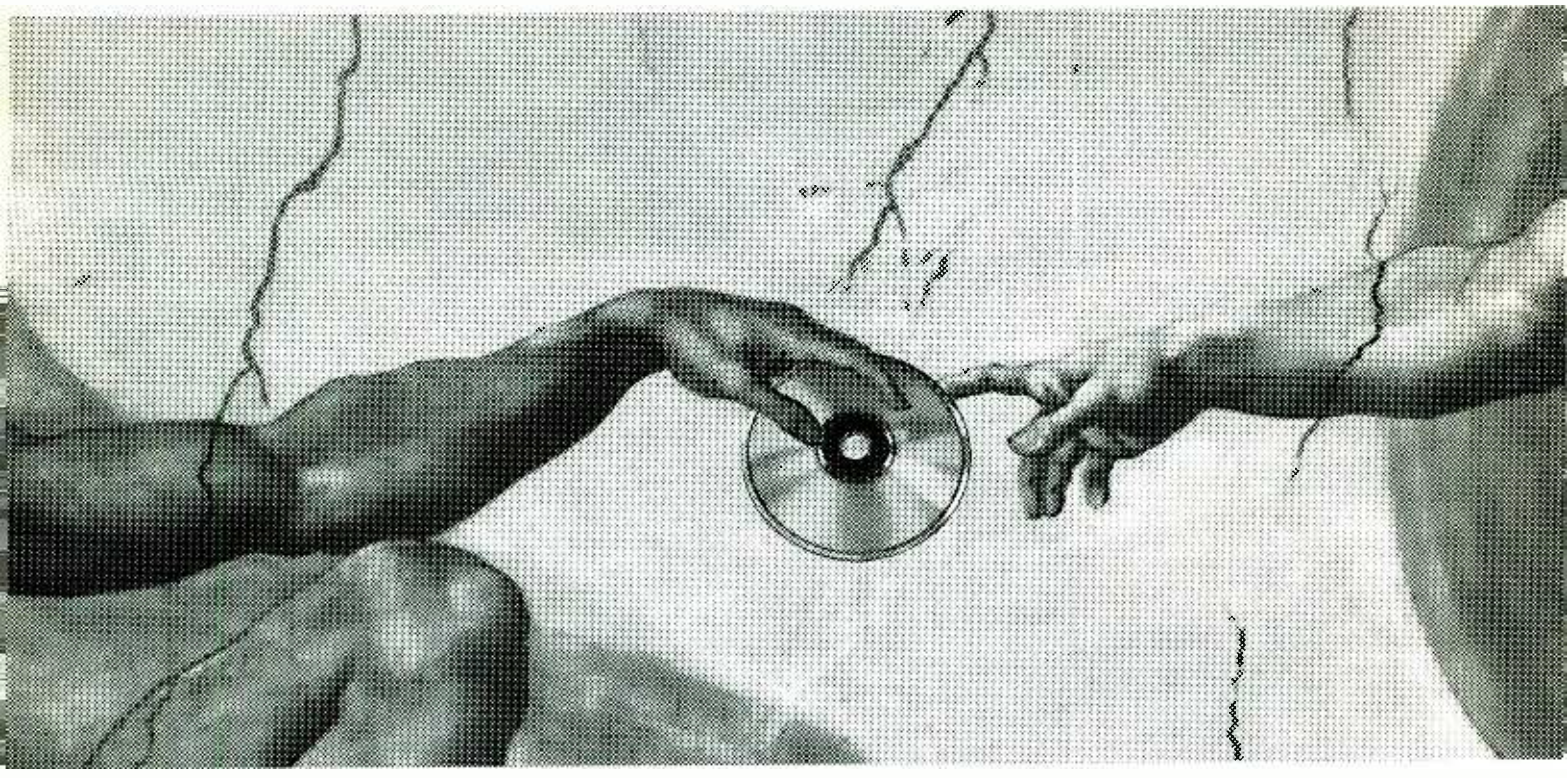

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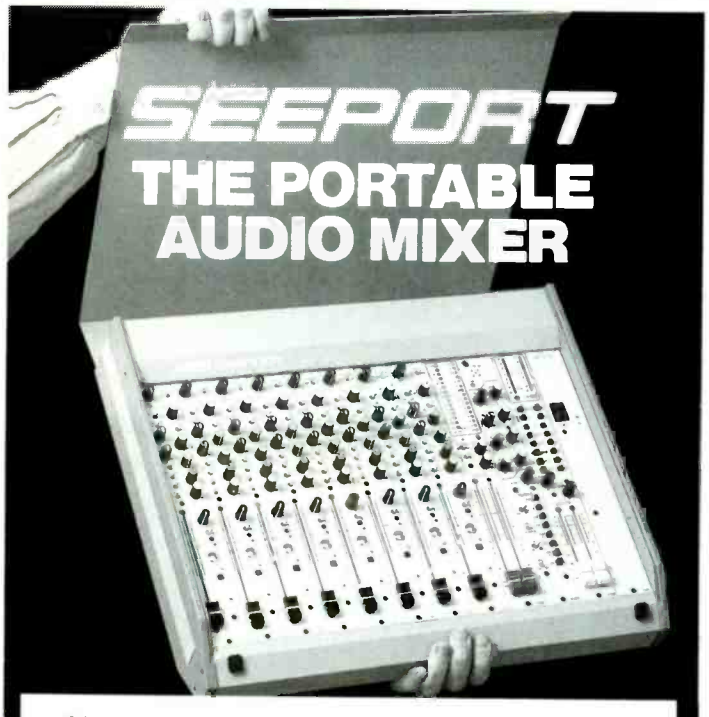

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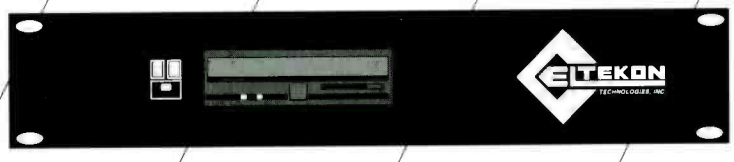

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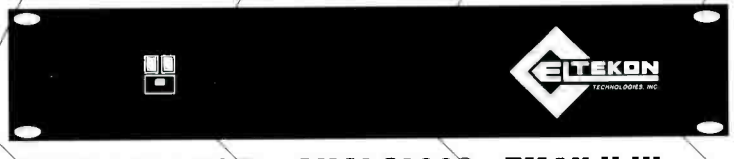

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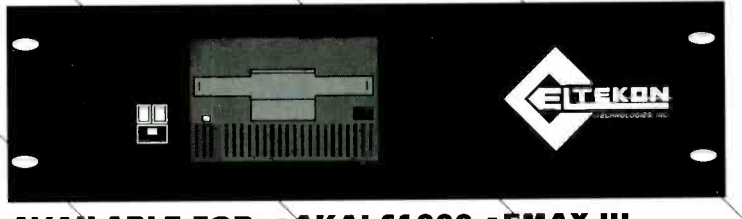

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Reworking old masters with DD1000

With its ability to record very long stereo samples the Akai *DD1000* was chosen to rework an analogue mix. The music consisted of two verses, a bridge, a verse and then an orchestral section before the fade but it was decided that the third verse should resolve after the instrumental section again and then go into a fade.

Had the music been composed using MIDI software this would have been easily accomplished by doing a pattern copy and simply sampling the vocal line. To accomplish this with an already mixed tape was a different kettle of fish, so the whole piece was sampled into the *DD1000*, edit points being 'soft marked' during recording using the front panel keys.

If editing with tape/blade one would normally cut to bass drum, or snare beats but on the mix in question this would have been impossible as the third verse vocals started slightly before the eighth beat preceding the first bar of the next section. Pressing the marker keys displayed on the *DD1000* LCD graph the amplitude waveform graph for that section, and from this and audio, and using the large cursor knob, it was possible to locate the in and out points very accurately. The section was then copied and pasted into position.

One great advantage of this over blade and tape was that at the join point the previous section did not have to be cut abruptly thus losing 'hangover' reverb, etc. Instead, it was faded into the new 'paste-up' giving a very natural effect. The degree of fade could be adjusted very finely using numerical information.

On listening to the 'final' mix it was decided to lay over some saxophone in the first orchestral section to make it different from the final faded 'copy' version later in the track.

Using Cubase software in its 'human sync' mode (to find the bpm of the original track) a brass section was created with MIDI synths and samples and then sampled into the *DD1000* and pasted onto the original mix. Also, chorus lines were 'beefed up' recording live mic into the *DD1000* thereby enabling the work to be carried out continually in the digital domain.

In fact, any number of 'live' takes could have been added and copied with no degeneration of the original signal.

With a digital console such as the *DMP7* the *DD1000* certainly offers a comparatively low cost digital recording set-up.

David Hastilow

Yet another picture of a very expensive piece of professional audio equipment hanging below a helicopter but this one is a little bit different. The land below the helicopter is the Isle of Capri, the legendary island off the coast of Naples. We are reliably informed that the flight is taking the crate to Capri Digital, a completely new recording facility on the far side of the island that is almost inaccessible except by foot — and helicopter. Due to open in October, the studio is lavishly equipped with almost all tape formats, an SSL console and a reportedly spectacular view. Full story on page 37 of this issue.

Exhibitions and conventions

October 4th to 8th 91st AES Convention, Hilton Hotel and Sheraton Centre, New York, USA.
October 16th and 17th The Playback Show 91, RDS Industries Hall, Dublin, Eire.
October 17th to 21st IBTS '91, Milan, Italy.
October 17th to 21st Mediatech 91, Milan, Fiera, Lacchiarella, Italy.
November 29th to December 1st Broadcast India 91, Bombay, India.

1992

March 24th to 27th AES 92nd Convention, Vienna, Austria.
July 8th to 10th Pro Sound & Light Asia, Singapore.
October 2nd to 5th 93rd AES Convention, San Francisco, CA, USA.

Pink Room opens

The two London based realtime tape duplication specialists Chop Em Out and The Sound Cellar have collaborated to open The Pink Room, which they describe as the world's first computer controlled all Nakamichi realtime cassette copying room. The room has 150 Nakamichi *RX505E* cassette decks all under computer-based control and subsequently requiring only a single operator. The Pink Room's control electronics were designed by the British digital processing specialists Audio Digital Technology. IBM T.DOS custom written software controls the system and incorporates self-diagnostic capabilities.

Chop Em Out claim the new room offers clients an improvement in audio quality, while savings in turnround time combine with the single operator set-up and the above average reliability of the *RX505E* machines to allow reduction of almost a third on previous prices.

Chop Em Out performed a market survey before deciding to go with the *RX505E* 3-head recorders although they anticipated the machines would be their final choice. They decided that while other professional quality machines could offer robust operation they could not at the same time match the *RX505E*'s recording quality. Also the machine's mechanism physically turns the cassette housing, which they regard as giving more consistently reliable azimuth alignment than machines that reverse the copy tape's direction. The 3-head feature also allows independent control over record and playback azimuth.

The company has an agreement with BASF to supply professional type II super-chrome exclusively in the UK.

Ralph Denyer

Contracts

● The British Library's National Archive have bought a **DAR** *SoundStation II* digital audio production system for their sound compilation and archiving operations.

● Recent **DDA** contracts include an *AMR24* recording console to the Art Music Show Theatre in Moscow; a *DCM232* console to Dutch studio Zeezicht; and an *AMR24* to CTS studios in Wembley, UK, the console has gone into their Studio Three.

● US studios Colorado Sound Recording and Crystal Clear Recorders in Dallas have both installed **Audiomation Systems** *Uptown* moving fader systems. Warner-Hollywood film studios in California have installed 96 channels of the *Uptown* moving fader console system in a Harrison *PP1* film mixing console.

● Studios Miraval and Des Dames in France have both bought *PCM-3348* 48-track recorders from **Sony.** Recording artist Vangelis and mobile operation Voyageur II have also recently bought two DASH 48-track machines.

● The BBC have chosen **The Professional Monitor Company**'s *BB5* system for their new facility at Maida Vale Studio Four in London.

● Garry Clarke, ex-Danny Wilson band member, has ordered a 44-input *Sapphyre* console from **Soundcraft** for his new private UK studio.

● Recent purchasers of the **TimeLine** *Lynx* timecode modules include Paul McCartney, Sarm West studios in London who have ordered six units, and Granada TV.

● **Elliott Bros** have designed and installed a stage management console and SR system in London's Old Vic Theatre, providing a new production and communications centre.

● CTS studios in Wembley, UK, have just installed the **AD Systeme** *Optifile* 3D automation system on a DDA mixing console.

● The Telecine-Cell group have ordered a second **Raindirk** *Symphony* console for their audio/visual studios in West London.

● **Fairlight-ESP** have delivered an *MFX* digital audio production system to Albert's Studio in Sydney, Australia. The first Canadian *MFX* system has gone to Digital Music Inc in Toronto.

● Recent **Amek/TAC** contracts include a number of consoles to Radio Television Malaysia. The consoles include a *Classic, BCII* and TAC *Bullet;* an Amek *Classic* has been installed at Singapore's broadcasting complex. *Mozart* consoles recently supplied include those to broadcast facility CDIS and two to Singapore Broadcasting Corp (SBC).

● Joe's Garage in South London have opted for the Andy Munro designed **DynaudioAcoustics** *M1* reference loudspeakers.

● Norsk Film Studio A/S, in Norway, have bought a **DAR** *SoundStation II* digital audio production system.

● The Reflections studio, Nashville, USA, recently added a second **Studer** *A827* recorder; Future Entertainment, a new facility in Virginia, US, have bought two Studer *A827*s along with an *A820* 2-track machine.

Address changes

● The address for the newly formed **Revox** UK is 1 Berkshire Business Centre, Enterprise Way, Thatcham, Berks RG13 4NA. Tel: 0635 76969. Fax: 0635 72556.

Agencies

● **Fairlight-ESP** have announced the appointment of CD Videosuono Spa as exclusive Italian distributors and Sycom Japan as exclusive Japanese distributors of the Fairlight digital audio and video product range.

In brief

● London, UK: **Stirling provides free training on Euphonix console.** Stirling Audio systems are to provide freelance engineers with free training on the operation of the Crescendo *Euphonix* console. The scheme is a direct result of the interest the console generated at June's APRS show in London.

● Slough, UK: **C-Audio wins THX approval.** The *RA 2000* power amplifier manufactured by C-Audio has been added to the Lucasfilm list of equipment approved for use in *THX* cinema sound systems. The *RA 2000* is a stereo unit capable of delivering 450 W into a 4 Ω load.

● Los Angeles, USA: **Record Plant sells mobile.** Design FX Audio have bought Mobile Unit 3 from the Record Plant. The Design FX Remote is currently equipped with an API console, and a choice of single or dual Otari *MTR-90*, Mitsubishi *X-880* or Sony *PCM-3348* multitracks.

● London, UK: **Poolside drum samples.** Over a 5 hour recording session at London's White City swimming baths, hits from more than 70 percussion instruments were committed to DAT via a customised Neve console, resulting in more than 700 hits. The *Poolside Drums* sampling CD is now available from London Sample Workshop Ltd.

● Huntingdon Beach, CA, USA: **BBE Sound honoured.** President and CEO of BBE Sound John McLaren has been honoured by both Pepperdine University and Orange County High School of the Arts.

● New York, USA: **HGA in The Big Apple.** Following a significant increase in overseas business Harris Grant Associates have opened an office in New York c/o John R Menz & Richard Cook Architects, 8th Floor, 106 East 19th Street, New York, NY 10003. Tel: (212) 477-0287. Fax: (212) 477-4521.

● Africa: **Wildlife cameraman relies on JVC.** Joe Phillips used JVC *Professional S* cameras and JVC mics to shoot and record his latest programmes in Africa.

Chop Em Out install more Sonic Solutions editing

Chop Em Out have acquired their second Sonic Solutions hard disk editing system and now operate a total of five CD mastering rooms. The two Sonic Solutions rooms are complemented by two others with Sony *DAE-1100/PCM-1630* equipment, while the fifth room is equipped around a Sony *DAE-3000* system.

The company say the speed and versatility of the Sonic Solutions system combined with their two years experience with their first system and software updates, has greatly enhanced the range of services they now perform for record companies. This includes the capability to perform complex 4-track editing work for laying voiceovers and SFX to stereo for pop promos and commercials efficiently and economically. Also, increased familiarity and operator speed has allowed them to reduce the costs to clients using the *NoNoise* de-clicking and de-noising software to remove unwanted noise from masters. Chop Em Out suggest that a growing consumer awareness of the benefits of the system as a result of use of the *NoNoise* logo on end product gives an added incentive to record companies to use the service. **Ralph Denyer**

JBL ES52000 loudspeaker controller

The *ES52000* is a new digital loudspeaker controller from JBL. The system combines the functions of 2-way dual-channel or 2-, 3- or 4-way single-channel active frequency division, system equalisation, time delay and transducer protection limiters (in single-channel configurations). The crossover filters are FIR (Finite Impulse Response) designs with 50 to 100 dB/octave slopes, which include pre-programmed signal alignment correction to compensate for the different acoustical centres of the various transducers used in JBL systems.

The *ES52000* incorporates 18 bit 64 × oversampling ADCs, 18 bit 8 × oversampling DACs, has a sampling rate of 44.1 kHz and 24 bit digital signal processing.

JBL International, 8500 Balboa Boulevard, Northridge, CA 91329, USA. Tel: (818) 893-8411.
UK: Harman Audio (UK) Ltd, Mill Street, Slough, Berks SL2 5DD. Tel: 0753 576911. Fax: 0753 535306.

Monster Musician cable

Monster Cable, the high performance cable manufacturers, have announced the launch of their *Musician* series cable. The company state, "Each cable has been developed to enhance specific aspects of the instrument's musical output, supporting the musician's sonic needs in terms of clarity, definition, pressure, attack, tone and feel."

There are eight cables in the new series — *Monster Rock, Monster Jazz, Monster Bass, Monster Club, Monster Studio Pro, Monster Keyboard, Digi-Link MIDI* and *Monster Power,* which is designed for amplifier-to-speaker connections.

Monster Cable, 274 Wattis Way, South San Francisco, CA 94080-6761, USA. Tel: (415) 871-6000. Fax: (415) 871-6555.
UK: Harman Audio (UK) Ltd, Mill Street, Slough, Berks SL2 5DD. Tel: 0753 576911. Fax: 0753 535306.

Cadac Concert live console

Cadac have launched a new live sound console called the *Concert*, which they say optimises sound quality and fast reset. The desk avoids solid state switching, using subminiature switching to improve noise and distortion figures. All switch functions in the channels and matrix groups with the exception of local mute and PFL, are controllable from centrally assigned switching. The *Concert* incorporates Cadac's system for complete status recall, encompassing all centrally assigned switching, aux sends, EQ settings, 12 × 12 matrix and output faders. All potentiometer positions are reset with the use of nulling LEDs.

The input channel features dual, mixable mic/line inputs, 16 aux sends, 4-band fully parametric EQ and high- and lowpass filters. There are 12 sub groups and 12 matrix groups. A 32-channel board provides a total of 112 inputs and 40 outputs. Each *Concert* will be built to order, allowing for individual frame layout and size requirements.

Clive Green and Co Ltd, 1 New Street, Luton, Beds LU1 5DX, UK. Tel: 0582 404202. Fax: 0582 412799.

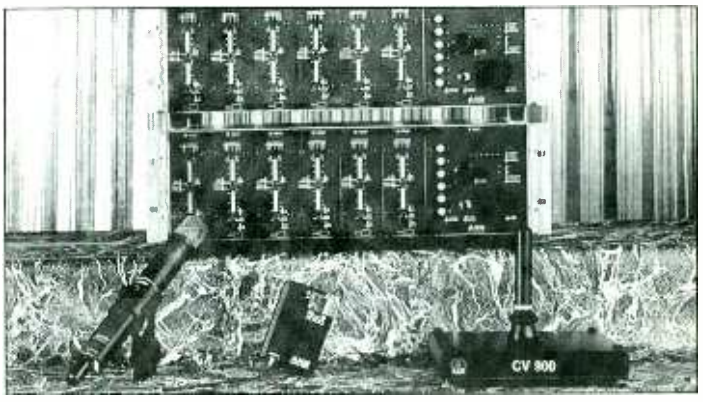
Audio Kinetics Masterlink software

Masterlink is a PC-based software package, available to *Reflex* or *MasterMix 2* users, that integrates console automation, machine control and MIDI interfacing. Automation features include, centralised data storage allowing hard disk retention of the entire mix history, an increase in mix and mute data accuracy to ¼ frame, realtime editing of mixes allowing Compare and Execute facilities, unlimited group update in realtime, off-line editing of all parameters, and single command cueing of mix data and machines. A *SwitchMix* interface is available for certain consoles, including the Amek *Mozart*, that allows automation of most channel strip switches. A *Microscope* event editor will time-slip data against timecode, and Keyboard Macros are now available.

A new low cost synchroniser module — the *MasterLink MS1* — has been developed, which is controlled from the PC. Up to four of these modules may be interfaced with the system and most popular machines are supported, including direct control of video transport when used as system master.

Alterations to audio, controller, note on/off, sequencer, start/stop can all be set up, edited and controlled against a timecode event. MIDI set-up data from peripheral devices may also be stored on the mix disk.

Audio Kinetics UK Ltd, Kinetic Centre, Theobald Street, Borehamwood, Herts WD6 4PJ, UK. Tel: 081-953-8118. Fax: 081-953 1118.

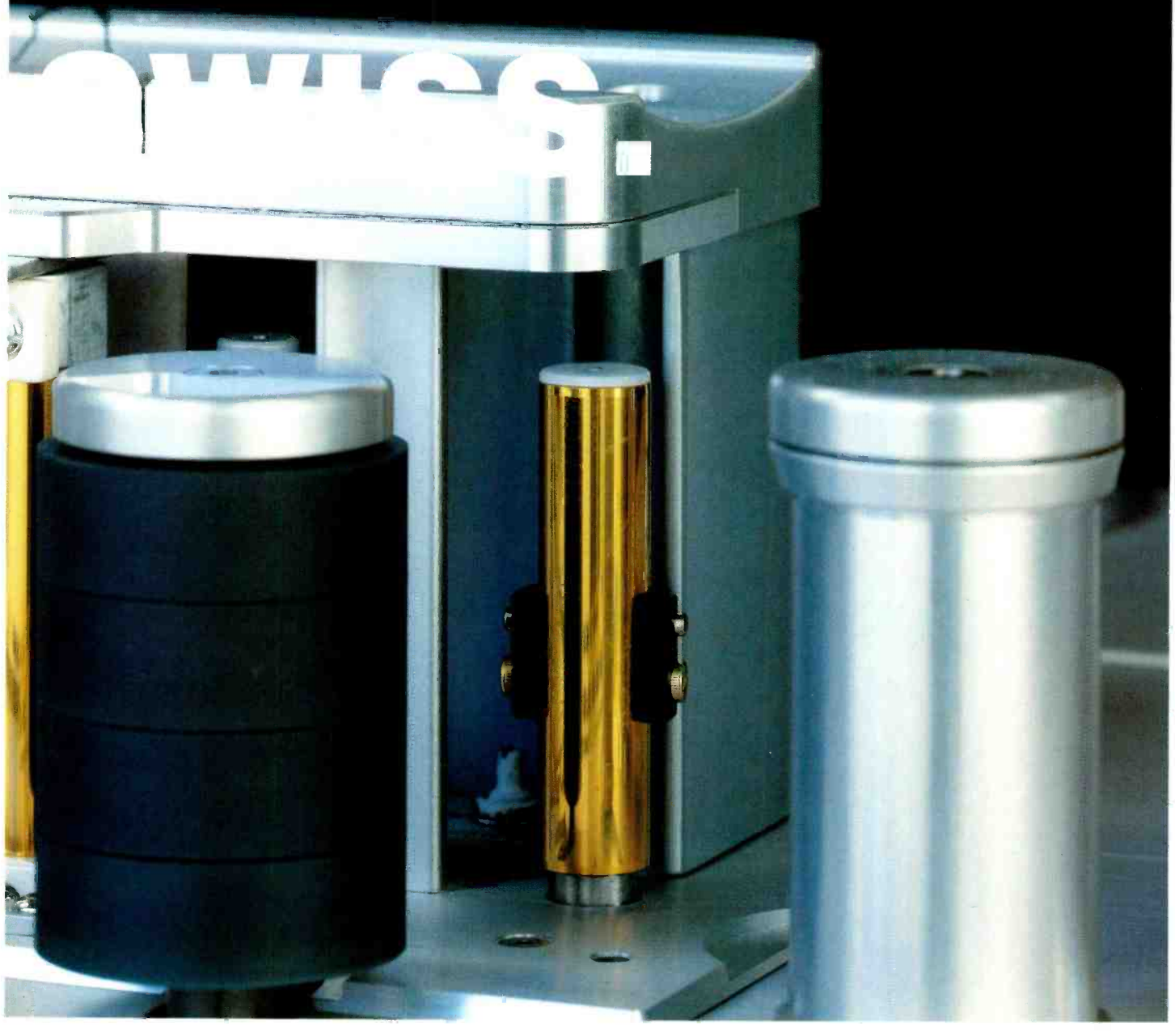


AKG WMS 900 radio mic

AKG have launched the *WMS 900* wireless microphone system. Operating in the UHF range from 470 to 970 MHz (TV channels 21-68), up to 12 microphones can be operated in one 8 MHz TV channel. The transmitters are either a handheld design incorporating the *C 1000 S* microphone element, or contained in a corrosion resistant bodypack/beltpack — both operate with the same interchangable plug-in transmitter modules. Two lavalier mics are available: the *C407* (special theatre version) and the *CK 67*. The receiver uses an antenna combined with a converter to allow standard antenna cable runs of 100 metres. The converter converts the UHF carrier to a VHF carrier in the 70 MHz band. The receiver is a true diversity design and up to six can be installed into the 19 inch rackmount unit, which also contains power supply and headphone amplifier. Front panel metering on each receiver includes audio and RF levels, and output level. Each receiver can be separately muted, and for 12-channel systems, two racks are linked together.

AKG GmbH, Brunhildengasse 1, A-1150 Vienna, Austria. Tel: (43 222) 95.65.17-0.
UK: AKG Acoustics Ltd, Vienna Court, Lammas Road, Godalming, Surrey GU7 1JG. Tel: 04868 25702. Fax: 04868 28967.
USA: AKG Acoustics Inc, 1525 Alvarado Street, San Leandro, CA 94577. Tel: (415) 351-3500.

PRECISE.



When confronted with the decision of cutting costs or maintaining high quality standards we always decide in favour of quality. This certainly is the case for our professional **A827** multichannel tape recorder where we have spared no effort. This is demonstrated, for example, by our unique tape compensation lever for protecting the tape edges and for balancing the various tape tensions. An ingeniously simple solution for which we immediately filed a patent application. But that's not all. Our conception of precision and quality are evident in each element of the **A827.** Our world-renowned precision manufacturing and our unique conceptions of analog tape recordings are in perfect harmony in the **A827.** Our customers would expect no less from a STUDER machine. Precision makes sense only if it carried through to the smallest detail. That is our philosophy – the standard of excellence for the **A827.**

STUDER

PROFESSIONAL AUDIO EQUIPMENT

Worldwide Distribution: STUDER International, a division of STUDER REVOX AG, Regensdorf-Zurich, Althardstrasse 10, Switzerland, Telephone +411 870 75 11, Telefax +411 840 47 37

Adams-Smith enhancements

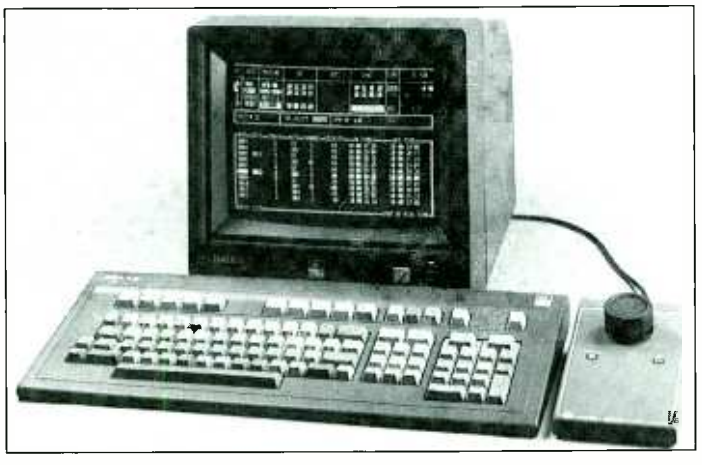
Adams-Smith have announced enhancements to two of their products, the *Zeta-Three* and *2600 E-A/V*.

The new *Zeta-Three Emulator* has been designed to interface with a wide range of video edit controllers; by directly responding to commands, the *Emulator* and its slaved transports behave exactly as a video recorder, either emulating the source VTR or chasing and emulating the record and channel selection capabilities. The *Zeta-Three Emulator* incorporates all the features of a standard *Zeta-Three*, and existing systems can be upgraded to include VTR emulation.

Two new facilities have been made available for the *2600 E-A/V* audio editor: Cross-Lock and Vari-Lock. Cross-Lock allows timecodes running at different frame rates to be directly locked together and Vari-Lock provides time compression or expansion to fit audio into a specific time slot.

Adams-Smith Inc, PO Box 130, 34 Tower Street, Hudson, MA 01749, USA. Tel: (617) 562-3801. Fax: (508) 568-0404.

UK: Marquee Audio Ltd, Shepperton Film Studios, Studios Road, Shepperton, Middx TW17 0QD. Tel: 0932 566777. Fax: 0932 565861.



TimeLine CCU console controller

Timeline have introduced a *Console Control Unit (CCU)* to operate with their *Lynx* synchronisation system. The *CCU* is a mini keypad (5.5 × 5.5 inches) that mounts into the console allowing direct input to the TimeLine *System Supervisor*. As part of the modular system, the *CCU* can interface with up to six machine transports, any of which can be designated as master. Varispeed is available for the master machine allowing pitch change to an entire synchronised machine group. An optional jog/shuttlewheel provides accurate slow speed control of any connected transport.

The *CCU* enables operators to enter and store offsets via the keypad, with the aid of a 16-character alphanumeric display. Status lights for each machine indicate Lock, Code, Busy, Record, Auxiliary; Group and Solo functions can also be selected. Multiple *CCUs* may be serially connected to facilitate multi-operator applications.

TimeLine Inc, 279 Lafayette Street, New York, NY 10012, USA. Tel: (212) 431-0330.

UK: Stirling Audio Systems Ltd, Kimberley Road, London NW6 7SF. Tel: 071-624 6000. Fax: 071-372 6370.

Sigtech Acoustic Environment Correction

Cambridge Signal Technologies (SigTech) have announced the release of their *Acoustic Environment Correction* system (*AEC*), for studio monitoring equalisation. The project started 10 years ago and has only recently come to fruition due to the availability of high speed digital processing hardware. The *AEC1000* uses 25 DSPs to automatically design a corrective FIR (Finite Impulse Response) filter capable of over 1,000 individual frequency adjustments. The filter design and correction process take into account, and differentiate between, the direct sound from the speakers and the reflexions from the room. Using a special test signal picked up by a mic in the listening position, the *AEC1000* can store up to four sets of filters providing correction for more than one pair of monitors, several optimum listening areas, or to cope with major equipment changes that may affect the listening environment.

During measurements and filter calibration, the 2U rackmount unit is interfaced to an *XT/AT/386* PC with EGA or VGA graphics, along with SigTech software and controller board. The unit is inserted via balanced *XLRs* into the line feeds to the monitor amps; it uses 18 bit converters and includes AES and SPDIF connectors for direct digital feeds.

SigTech, 1 Kendall Square, Building 100, Cambridge, MA 02139, USA. Tel: (617) 225-2499. Fax: (617) 225-0470.

UK: Audio Design, Unit 3, Horseshoe Park, Pangbourne, Berks RG8 7JW. Tel: 0734 844545. Fax: 0734 842604.

NTP 477-100 ppm

Danish company NTP have a new high precision, microprocessor controlled ppm meter, the *NTP477-100*. Features include a multifunction, 200-segment display for simultaneous Spot and Bar modes, permanent highlighting of most significant scale marks and a triple-coloured phase meter. Different display modes are selected and indicated from the front of the meter, alternatively the meters may be set up via a PC using the D²B bus. To aid the monitoring of high dynamic range and fast peak signals, an extended DIN scale up to +10 dB and a high overload limit of +23 dB have been included. For increased life, an auto standby function shuts down the display during periods of inactivity. The meter is supplied in a standard 190 × 40 mm module.

NTP Elektronik A/S, Knapholm 7, DK-2730 Herlev, Denmark. Tel: 44 53 11 88. Fax: 44 53 11 70.

UK: Meridien. Tel: 081-293 0909.

Hill Chameleon

Hill Audio have launched the *Chameleon*, which they say is effectively two amplifiers in one and suitable for all applications. The lightweight, 1U unit is capable of delivering a full 2 kW into 4 Ω (both channels driven, transient power — RMS 20 ms/80 ms). Working on the principle that an amplifier will only be called on to deliver its maximum rated power for 25% of the time, *Chameleon* operates in two modes — Headroom and Head-Lok — which are automatically switched as necessary. Headroom is capable of handling severe digital transients and Head-Lok reduces the headroom by 2 dB enabling the total voltage power output to increase.

The front panel features pushbutton gain control with illuminated readouts for each channel that are accurate to 0.1 dB. The rear panel incorporates 4-way binding posts allowing a variety of loudspeaker terminations. Attention has been paid to heat dissipation — heatsinks are an integral part of the amplifier box and a quiet low speed fan has been fitted. A five-level protection system, isolated from the audio path, safeguards both amplifier and speakers, while a soft start circuit avoids problems due to switch-on surges.

Hill Audio Ltd, Hollingbourne House, Hollingbourne, Maidstone, Kent ME17 1QJ, UK. Tel: 0622 880555. Fax: 0622 880550.

DIGITAL AUDIO TIMES

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SUMMER 91

☎ 081 960 2144

NEW LOOK LISTENING ROOM

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

SUMMIT AUDIO

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

YAMAHA DMC1000

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

DIGITAL AUDIO RESEARCH DASS-100

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

FOSTEX G24S

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

SOLID STATE AUDIO FOR VIDEO

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

APOGEE

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

SONY STEREO MICS

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

MORE NEWS FROM EUROPE'S DAT CENTRE

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

AIWA HHB1 PRO KIT

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

SONY DTC1000ES 'PRO'

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

SONY TCD-D3

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

SONY DTC-55ES

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

PANASONIC SV3900/SV3700

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

SONY PCM-7000 SERIES

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

SUMMER SALE BARGAINS

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

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

– All prices exclude VAT.

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

COMING FACE TO FACE WITH THE FUTURE IS A LOT MORE FAMILIAR THAN YOU MIGHT THINK!

We don't ship tape recorders to you in pieces; you won't get our new disk recorders that way either.

Our new DDR-10 digital disk recorder is a fully integrated system that comes ready to roll in and turn on. There's no assembly, no cabling, nothing more to buy. *And because it looks great, clients simply love it!*

If you know tape recorders, you're ready to go the minute you sit down at the DDR-10's control panel. The buttons and knobs are familiar, and they do what you expect. It's the only audio disk recorder in the world designed in this way – designed for the professional.

With up to 60 hours of the highest quality, full-bandwidth digital audio, *and* extensive editing capabilities, the DDR-10 delivers the best "power-to-price" ratio you can buy. And its standard Macintosh® SCSI buss (rather than a proprietary design), means you're free to take advantage of future advances in the computer industry. This way, you're not locked-in to a system which seemed "special" at first, but quickly became *ancient*.

For your convenience, CD replication directly from your DDR-10 soundfile is now available. For more information about this service, and the DDR-10, call your nearest regional office below for information from Otari; a company famous for technical support and customer service, worldwide.

OTARI®

Otari, Inc., Tokyo (0424) 81-8626
Otari Corp., California (415) 341-5900
Otari (UK) Ltd., United Kingdom (0753) 580777
Otari Singapore Pte., Ltd. (65) 743-7711
Otari Deutschland GmbH., West Germany 02159/50861-3

In brief

● **C-Audio** have expanded their *RA* series of high performance MOSFET amplifiers by introducing the *RA500*. This convection cooled, 1U-high amplifier delivers a continuous 100 W/channel into an 8 Ω load; according to the manufacturer it is suited for driving nearfield monitors in studios, and for more general use in theatre, sound reinforcement and installation applications.
C-Audio Ltd, Barnwell Road Business Park, Cambridge CB5 8UY, UK. Tel: 0223 211333. Fax: 0223 410446.
UK: Harman Audio (UK) Ltd, Mill Street, Slough, Berks SL2 5DD. Tel: 0753 576911. Fax: 0753 535306.

● **3M** have added a long play digital cassette for PCM applications to their *AUD* range. The *AUD 80+* provides 83 minutes playing time, which should be of particular interest to mastering facilities working with longer length classical recordings.
3M Magnetic Products Division, 3M Centre, St Paul, MN 55144-1000, USA. Tel: (612) 736-9567.
UK: 3M UK, 3M House, Bracknell, Berks RG12 1JU. Tel: 0344 858551. Fax: 0344 58175.

● Hot on the heels of Cedar Audio, **Sonic Solutions** have announced a realtime version of their 'de-clicking' and 'de-crackling' options for the *NoNoise* system. Sonic Solutions say that the new version will neither sacrifice quality nor reduce the amount of control available to the operator.
Sonic Solutions, 1902 Van Ness Avenue, Suite 300, San Francisco, CA 94109, USA. Tel: (415) 394-8100. Fax: (415) 394-8099.
UK: FWO Bauch Ltd, 49 Theobald Street, Borehamwood, Herts WD6 4RZ. Tel: 081-953 0091. Fax: 081-207 5970.

● **Amcron** have launched two new power amplifiers — the *Geodyne I* and *II* — delivering 300 W and 440 W/channel into 4 Ω respectively. Both amps are 2U units and feature Amcron's patented *Grounded Bridge* design; *ODEP* (Output Device Emulator Protection), which Amcron claim will keep the amplifier working safely under adverse conditions; and *IOC* (Input Output Comparator) indicator, which lights if the amplifier distortion exceeds 0.05%.
Crown International Inc, PO Box 1000, Elkhart, IN 46515-1000, USA. Tel: (219) 294-8000. Fax: (219) 294-8346.
UK: Shuttlesound Ltd, 4 The Willows Centre, Willow Lane, Mitcham, Surrey CR4 4NX. Tel: 081-640 9600. Fax: 081-640 0106.

● The *DCM232* console from **DDA**, can now be fitted with an alternative input module. The *PPR* module has resulted from the development of the post-production version of the console — the *DCM224V*. The new module offers five bands of EQ, which can be split between channel and monitor paths, and independently switched high- and lowpass filters. The auxiliary buses are now fed from two dual concentric controls for the mono buses and two sets of level/panpots for the stereo pairs. Both the original and the new module can be fitted to a 32 bus *DCM*, and existing users can fit the *PPR* modules without modification.
DDA, Unit 1, Inwood Business Park, Whitton Road, Hounslow, Middx TW3 2EB, UK. Tel: 081-570 7161. Fax: 081-577 3677.
USA: Klark-Teknik Electronics Inc, 200 Sea Lane, Farmingdale, NY 11735. Tel: (516) 249-3660. Fax: (516) 420-1863.

● A new mute switch interface is available for consoles fitted with **Steinberg**'s *Mimix* automation. The system comes in 16-channel modular form, for use with any console containing FET channel mutes — other FET switches such as EQ in/out can also be automated.
Audio Arcana, 3 Bardown, Chievley, Nr Newbury, Berks RG16 8TH, UK. Tel: 0635 248078.

● American company **SpeakEasy** have announced the release of new CAD software — *Filter Designer 1.0*. The software is an electronic circuit analysis program primarily intended for use with loudspeaker crossover networks and passive or active audio filters; it can also operate as a basic circuit analysis program for other types of design work. The program will import data from SpeakEasy's *Low Frequency Designer* program and other sources to facilitate the computation of a loudspeaker system's total response, including effect of driver, enclosure and crossover.
SpeakEasy, 46 Cook Street, Newton, MA 02158, USA. Tel or Fax: (617) 969-1460.

Orban digital Optimod 8200

Orban have announced the introduction of a totally new, all-digital *Optimod* for FM radio. The *Optimod 8200* is a true digital processor that incorporates multiple high speed DSP chips, controlled by proprietary software to emulate the effect of conventional analogue processing. Each *8200* is supplied with one or more MVP (Multiple Variable Processor), which determines the type of processing provided by the unit — new MVPs can be added as necessary. The new *Optimod* fits into the same 4U rack space as its predecessor, the *8100*. It features a large LCD display with associated soft buttons, a large scroll knob, cursor direction keys and dedicated Help and Escape keys. With the ability to control the *8200* via modem, Orban foresee a situation where the sound of a station could be changed from a car phone!
Orban Associates, AKG Acoustics Inc, 645 Bryant Street, San Francisco, CA 94107, USA. Tel: (415) 957-1067.
UK: AKG Acoustics Ltd, Vienna Court, Lammas Road, Godalming, Surrey GU7 1JG. Tel: 04868 25702. Fax: 04868 28967.

AMEK BCIII

THE IMPORTANCE OF *EVOLUTION*

Specifications in broadcast audio and video production are continually developing with the emergence of new formats and methods. AMEK's benchmark BCII compact console set a world standard for performance and facilities, but as technical needs evolve so we have evolved a new design generation, BCIII.

Over five hundred BCII installations worldwide have given us the viewpoint of engineers working in all aspects of audio production. We have deep knowledge of what is needed and through an incredibly versatile design with many options from circuit level upwards, we can build from basic components a console to fit your system.

TLA INPUTS
TLA circuitry designed for AMEK by Rupert Neve brings many of the characteristics of transformers to both Mic and Line inputs without the bulk and cost.

MULTIPLE CONFIGURATIONS
BCIII can be configured for up to 32 inputs or for as few as 6 inputs according to operational requirements. With 6 input module variants including 3 versions of the stereo module, 4 group modules and 5 output stages, BCIII lends itself to a host of applications.

BALANCED INTERCONNECTIONS
Pre- and post-fade inserts and all outputs are balanced, operating at full studio level – allowing seamless, trouble-free interfaces to complex studio environments.

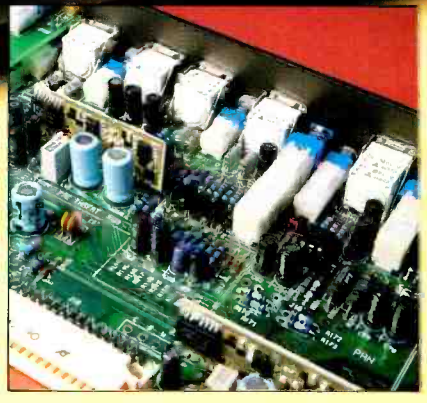
EXPANDABLE MONITORING
With complex multi-feed monitoring for several points including audio and video control rooms and performers' feedback, BCIII's flexible monitoring system allows control in all production environments.

OUTPUT FLEXIBILITY
Outputs can be configured with 4 or 8 mono or stereo groups feeding 1 or 2 separate stereo busses. Multiple clean feeds allow simultaneous handling of outside sources. Four fully-independent Auxiliary sends handle all Effects and Foldback/Cue requirements.

VERSATILE [illegible] DESIGNS
BCIII comes in 3 standard chassis packages including the freestanding SC (studio chassis). The onboard jackfield and extensive 19" rack space (>20U) within the console base allow significant space saving. The meter panel may be supplied with a range of metering options from 200-segment plasma bargraphs to VU meters.

[illegible] CONTROLS
On both inputs and subgroups, the Image control allows the signal to be changed from stereo through mono to reverse stereo – absolutely essential for stereo television audio.

[illegible] AUDIO FOLLOW VIDEO
VCAs may be fitted to faders; the addition of AMEK's ESM/ESR unit allows remote control from Video Editors with a number of manufacturers and protocols being handled. In addition, BCIII is also prepared for DC fader grouping systems.



AMEK

1986

Head Office, Factory and Sales: AMEK Systems and Controls Ltd., New Islington Mill, Regent Trading Estate, Oldfield Road, Salford M5 4SX, England. Telephone: 061-834 6747. Telex: 668127. Fax: 061-834 0593.

AMEK/TAC US Operations: 10815 Burbank Blvd, North Hollywood, CA 91601. Telephone: 818/508 9788. Fax: 818/508 [illegible]619.

Harman Deutschland Clayton amplifiers

A new line of amplifiers has been announced by Harman Deutschland. The *Clayton* range boasts 'American engineering and German production' and is supplied in four models: *CA1* 120 W/channel; *CA2* 300 W/channel; *CA4* 600 W/channel; and *CA5* 1000 W/channel.

The company stress the importance that has been attached to the amplifiers' impulse capability to facilitate wide dynamic range. All models accept the *CX1* proprietary crossover module, which is configured by setting jumpers. Inputs are balanced and *Speakon* connectors are used for the loudspeaker outputs.

Harman Deutschland, Hunderstrasse 1, D-7100 Heilbronn, West Germany. Tel: (0 71 31) 4 80-2 08. Fax: (0 71 31) 48 02 55.

Opcode upgrade MAX

Opcode Systems have upgraded their *MAX* software to *Version 2.1. MAX* is an object orientated programming language for MIDI, that runs on an Apple *Macintosh* enabling the user to create 'patches' that perform a particular function.

The updates include OMS (Opticode MIDI System) compatibility; DSP objects, allowing realtime DSP processing in *MAX* patches running on DSP cards such as Digidesign *Sound Accelerator* and *Audiomedia*; over 20 ready-to-use applications including Yamaha *DMP-11* controller and Lexicon *LXP-5* editor; and newly developed objects including IncDec for incrementing/decrementing numbers, and MultiSlider, which displays many parameters simultaneously and can be used in a scrolling mode.

The updated software will be sent to all registered owners free of charge.

Opcode Systems Inc, 3641 Haven Drive, Suite A, Menlo Park, CA 94025-1010, USA. Tel: (415) 369-8131. Fax: (415) 369-1747.

Peavey developments

Peavey's foray into the keyboard market continues with additions to their *DPM 3* and *3SE* keyboards.

The *DPM SX* sampling expander is a sample input device that permits 16 bit samples to be recorded and loaded into the *DPM 3SE*'s RAM where they can be manipulated and stored to 3.5 inch disk. Samples can also be dumped to any device that recognises SDS files, which opens the scope considerably and includes the *DPM V3*, a rackmount version of the keyboarded instrument including 4 Mbytes of ROM samples.

Still on sampling, Peavey's *DPM SP* is a 1U rackmount sample player, which also handles SDS files with a 3.5 inch drive and bolted to the aforementioned *SX* combines to create a rather attractive sampling package.

Peavey, Meridian, MS. Tel: (601) 483-5365.
UK: Peavey, Northants. Tel: 0536 205520.

Acoustic reproduction

Catering for the broadening need to amplify the bugged acoustic guitar, Trace Acoustic have introduced the *TA50* and *TA50R* combos aimed at the smaller venue and folk artists. Essentially 50 W versions of the existing *TA100* and *TA100R*, the newer models sport only two 5 inch speakers stated by the company to suit the demands of an acoustic guitar better than any other solution. Coupled to Trace Acoustic's, patent pending, Dynamic Correction circuitry, which effectively compares the input to what the speaker is receiving and adjusts accordingly, the *TA50R* has the unusual inclusion of an Alesis *Microverb III* OEM chip with a front panel 16-way selector.

● Trace Acoustic have also released the similarly principled *TA B100* 12 inch combo specifically for upright and acoustic bass reproduction.

Trace Elliot, Unit 7, 49 Braintree Road, Witham, Essex CM8 2BZ, UK. Tel: 0376 517237.

Sessionmaster guitar preamp

Guitar processors are getting smarter but a reassuring trend is the back-to-basics tube and analogue approach being re-injected into the market by some manufacturers. And it needn't cost the earth if the single-channel 1U *Sessionmaster* guitar preamp from Award is anything to go by.

Front panel controls are sparse. Treble, middle and bass controls are presented in that order from the left (back-to-front or is it just me?) together with pots for input gain and output level. Pushbutton switches engage mid and bass boost as well as gain boost, which also has the effect of placing the EQ section of the unit post-overdrive. Connections are via standard jack with the input on the front panel and a so-called recording or treated output socket and normal output socket found on the back. Interestingly, both can be used at once. Powering is via an external power supply.

Construction-wise it's adequate but only just. The 1U rackmount front panel is aluminium and bendy and the switches, pots, audio sockets and power socket are all mounted directly on the PCB with no strain reliefs. While certainly sturdier and better built than it sounds, it doesn't inspire confidence in this unit's ability to survive prolonged on-road or even intermittent abuse. While the *Sessionmaster* is very inexpensive and quite obviously aimed squarely at the home market, its performance is likely to attract admirers from the pro fraternity where the lack of a tougher front panel and surfaced fixed connectors will do it no favours.

However, no matter what attempts are made to criticise this featherweight device, on plugging in there is no getting away from the fact that the *Sessionmaster* sounds quite incredible. It's ridiculously quiet and certainly as good as, if not better than, any game clean feed let alone your average tweed classic with an *RE20* poked into a cone. As no internal gating is used we must assume that there are some damn fine electronics on board. Immediately apparent is the diversity of tonal flexibility — something that can't always be boasted with guitar amps. The controls interact with great fluidity and most actions or tweaks have an immediate effect with the simple layout encouraging experimentation.

Most spectacular is the palette of softly overdriven timbres — the character of which is again infinitely variable. The 'teetering on the edge of graunch' texture is the singularly most elusive sound in most, more costly, alternatives, which often cater instead for out and out overdrive or glassy *Rockman* tones, neglecting the important territory in between.

Clean sounds are nothing if not stunning and in line with everything else the *Sessionmaster* does, guitars still manage to maintain their individual signature. Thus a *Strat* sounds like a *Strat* and doesn't mesh into the oblivion of its humbucked brethren, even at the most extreme settings. This is something which is certainly not the case with many dedicated guitar multi-effects processors.

Overdriven sounds are subject to personal preferences and a guitarist's perception of what overdrive ought to sound like. Here again the shaping offered particularly by way of the mid and bass boost switches is substantial with a pleasing envelope — sustained notes decay naturally without break up or fizz. Tone control is provided by a bridged tee EQ network of the type used in many of the most highly regarded guitar amps around, and the amount of 'choke' that can be added to the three bands is authentic and also permits fine adjustment to be made. The *Sessionmaster* is not a 'two decent settings' type of affair.

As you will have noticed I have taken to this cheap little unit in no uncertain terms. In spite of, and because of, the £149 tag I have to say that it is among the most convincing guitar preamps I have heard. I believe its price has been made possible by concentrating entirely on the business of getting a decent guitar sound this side of a microphone without diluting its attraction with multifarious effects and processing that are seldom needed and often duplicated or available in a more versatile package elsewhere in the rack. And while it may be a little lightly sprung in the build stakes no compromises have been made in the performance — the silence of the unit bears this out.

Even for twice the money the *Sessionmaster* is a steal as it has succeeded in making competing products seem unfocused and rather crude by comparison. Guitarists interested in recording without a loudspeaker will not be disappointed I'll warrant.

Award Design, P O Box 3, Basingstoke RG24 9QA, UK. Tel: 0256 477222.

Software

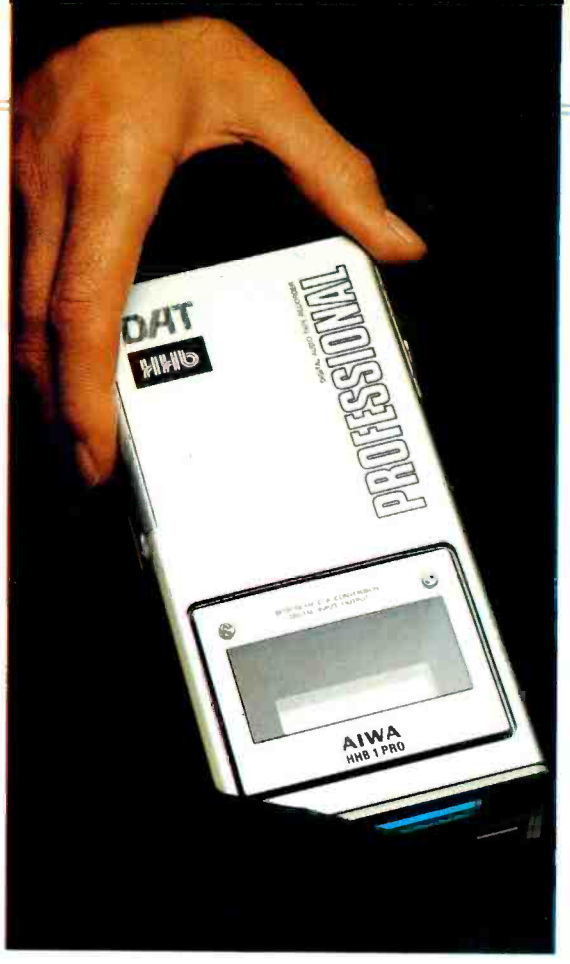
● **Passport**'s *Master Tracks Pro* upgrade *4.5* offers new methods of recording including loop recording and overdub record mode, support of Apple's *MIDI Manager*, on-screen faders and user programmable device name tables.

● *Version 2.0* of *Cubeat*, **Steinberg**'s entry level version of *Cubase* with fewer processing and editing facilities, now offers Drum Edit as well as the latest drivers and MROS files.

● **Opcode Systems** have released *Version 2.1* of *Max* — its object oriented programming language for MIDI and multimedia — as a free update to all registered users. Enhancements include Opcode MIDI System compatibility, DSP objects, applications for the Yamaha *DMP11*, Lexicon *LXP5* and Roland *Sound Canvas*, and objects that handle communication with RS232 devices.

Opcode Systems, Menlo Park, CA, USA. Tel: (415) 369-8131.

Studio Sound's Music News is compiled by Zenon Schoepe



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

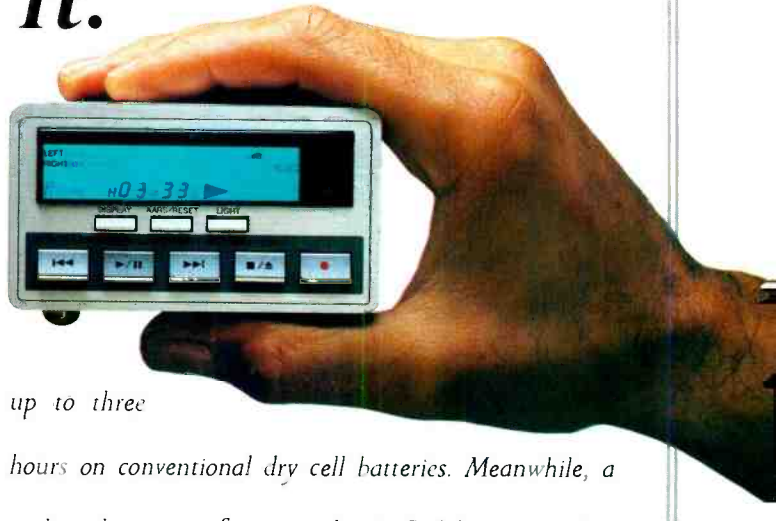
That's precisely why we've joined forces with Aiwa to design our own professional DAT portable – the HHB1 Pro.

In spite of its compact dimensions, the rugged HHB1 Pro offers a wealth of features for the professional user. A single 5-pin XLR switchable mic/line input allows stereo recordings in the field, while audio quality is assured thanks to the latest single-bit oversampling conversion technology. Of course, AES/EBU as well as SPDIF digital interfaces are provided as standard. And because the Pro's informative LCD display can be illuminated, monitoring in low-light conditions could not be more convenient.

The HHB1 Pro stripes tape with 'absolute time' information as it records. So whenever you insert a recorded cassette, you can see precisely where you are on the tape. With Sony's PCM-7000 range of studio DAT recorders capable of editing to absolute time as well as time-code, you can be confident that your HHB1 Pro will function as their ideal low-cost acquisition partner.

The HHB1 Pro records for up to three hours on conventional dry cell batteries. Meanwhile, a multi-voltage transformer and a NiCad battery pack – together with a selection of useful professional accessories including a wired remote controller – are supplied as standard. Since it weighs in at under £1,000 and less than a kilogram, picking up an HHB1 Pro from the world's number one DAT centre just couldn't be easier.

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



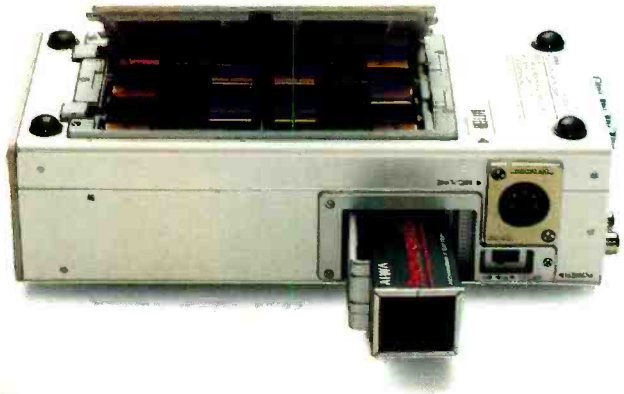


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Photo: Nik Milner

This year's Monsters of Rock concert at Castle Donington in the UK saw the largest Electro-Voice *MT-4* SR system ever assembled for one show. Monsters of Rock is now a touring institution with 19 dates scheduled in Europe during August and September

On tour

● **Britannia Row Productions** ended their summer season with the annual Reading Festival, a contract won from the Mean Fiddler Organisation after a successful performance at the Fleadh Irish rock festival earlier this year. *TMS-3*s were used for the SRS at the flat site, which is notorious for its noise spillage problems. And BRP gave their *Flashlight* system its last outing before the Dire Straits two year bash with a sunny afternoon's music at Crystal Palace Bowl, headlined by Level 42 with John Lemmon at the controls. The company's autumn tours include Whitney Houston, the aforementioned Dire Straits, Cliff Richard in rehearsals, Marillion and a to-be-confirmed Frank Sinatra/Liza Minelli tour.

● **Canegreen** are out until the New Year with Nigel Kennedy. They also have a number of 'raves' using 20 to 30 Meyer *MSL-3*s apiece, in barns, gyms and other suitably cavernous (and cheap) venues. Raves are attracting more SR companies as local councils realise there's money to be made from the parties they once tried hard to ban. In a more highbrow vein, Canegreen had Placido Domingo's open-air concert in Windsor Great Park, the Beverly Craven tour, subrentals on Deacon Blue and a UK tour with hard rock act The Black Crowes.

● **Capital Sound Hire** are fully occupied with the Simple Minds arena tour, using their Martin *F2* system under the eagle eye of FOH engineer Gary Bradshaw. The latest Martin *LE-700* wedge monitors are being used on stage. Martin Audio's David Bearman says the variable stacking configuration that's crucial to the *F2*'s performance in large arrays has been refined even further for this tour to realise the system's full potential.

● **Clair Brothers'** have European tours by Roxette and diminutive Australian multi-media star Kylie Minogue. Troy Clair says they are also scheduled for dates with Sting later this year. In the US, the company is much busier with Sting, Paul Simon, Yes, Prince, Don Henley, Stevie Nicks and Lynyrd Skynyrd.

● **Electrotec**, who have been touring the globe with Rod Stewart since last year, have their hands more than full with a return to the '70s heyday of rock'n'roll mythology — Gun'n'Roses' incident-ridden tour. As we went to press, singer Axl was facing charges arising from a widely-publicised fracas involving fans at a US show; during the riot that ensued a considerable amount of equipment was damaged. No cars have yet been reported driven into swimming pools by the band, but presumably that will all come in due course.

● **Encore**'s John Tinline reports ongoing dates with The Chippendales, many big 'raves', a 'punk revival' with X-Ray Spex at Brixton Academy, and one leg of Status Quo's Rock Till You Drop charity dash around the UK featuring four shows in one day. For the record, the SR participants were Encore Scotland (SECC), Roadstar (Sheffield Arena), SSE Hire (NEC) and Malcolm Hill (Wembley).

● **Entec** had an active late summer with a Clannad tour (using JBL *Concert* series SRS, the new *Accord* monitors and a *PM3000*). A highlight was the group's show at Birmingham's new Symphony Hall at the ICC complex, where Entec's SRS was used in conjunction with the hall's in-house JBL array.

● **SSE** made the most of their relationship with fellow E-V *MT-4* owners **dB Sound** of Illinois, USA, to assemble the largest *MT-4* SR system ever, for this August 17th Donington show on the 19-date Monsters of Rock tour, which was AC/DC headlined and mixed by Robbie McGrath. 140 *MT-4* systems provided around 500 kW and 118 dB, A-weighted, at the desk, said SSE's John Penn. Consoles included *TAC SR9000*s and Yamaha *PM3000*s for FOH and three Midas *XL-3*s for monitors. The SRS was set up and soundchecked in just two days.

● **Wigwam** provided a Meyer SRS for the Symphony of the Spire concert at Salisbury Cathedral, UK, on September 6th. Staged in aid of an appeal (organised by HRH The Prince of Wales) to save the medieval spire, Placido Domingo, Kenneth Brannagh and Phil Collins headed a multi-cultural evening for an 11,500 live audience and a TVS broadcast. Wigwam's Chris Hill explained the site posed major difficulties, being very wide to one side of the stage and dotted with trees, while low arches on the access gates prevented the use of cranes as SR flying platforms. Eventually, Wigwam opted to place the Meyer *MSL-3/UPA-1* system on small left and right towers with further delay towers facing outwards at 40°: "Not ideal," Hill said, "but we had no alternative." Hill says he will realise an ambition when he makes out the invoice — to Prince Charles at Kensington Palace. Wigwam have also been active in rock'n'roll of late with the Chris Rea tour and the Cities in the Park festival in Manchester, headlined by Happy Mondays and featuring pre-production models of the new Meyer *DS-2* low-mid enclosure ("Sound-wise it sits between the *MSL-3* and the *650* sub-bass," explained Hill).

JBL launch Sound Power M series

JBL chose London's International Music Show in July to launch their *Sound Power M* series. Aimed at touring musicians, the *M* series includes three full-range systems: *M360* 3-way with 15 inch cone driver, 8 inch midrange transducer and horn loaded tweeter; *M350* 2-way with 15 inch cone, 1 inch titanium diaphragm compression driver mounted on a biradial horn; and *M330* 2-way with 12 inch cone and 1 inch titanium diaphragm compression driver on a biradial horn. According to JBL, the most popular system is the smallest in the range: the *M330*.

The cabinets are finished in heavy duty grey vinyl wrap with moulded end caps for corner protection. Ribs in the end caps allow secure stacking with other similar units.

Meyer upgrade

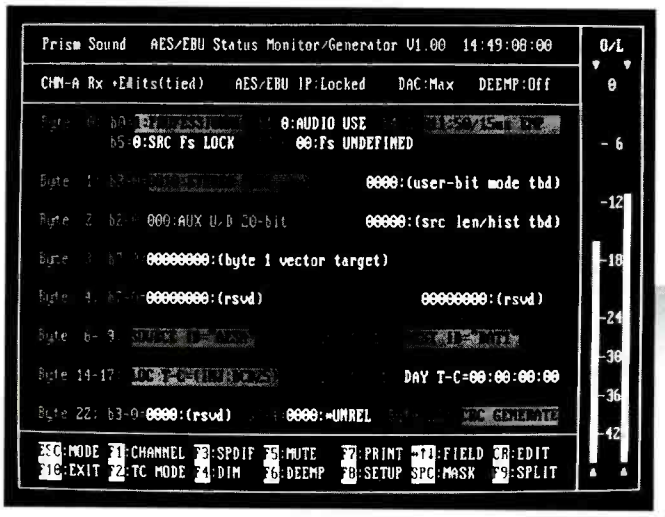
Meyer have announced upgraded versions of two of their speakers. The new products, the *UPA-1B* and the *UM-1B*, incorporate a new HF driver and passive network. The sonic results, state UK distributors Autograph Sales, include higher peak and continuous power handling, reduced upper-midrange distortion and an amplitude and phase response flat to 20 kHz.

In upgrading the widely-used *UPA-1* and *UM-1* designs, Meyer have also allowed purchasers of the new boxes to emulate the sound of their predecessors, using an internal jumper.

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

DAS-90. The AES/EBU-SPDIF Channel Status Editor by Prism Sound

Digital Audio signals are not so easy to monitor as analogue signals; they cannot drive headphones or a level meter. They contain Channel Status data which can be as useful for some as it is problematic for others.



DAS-90 AES/EBU-SPDIF
CHANNEL STATUS EDITOR

Available ready-installed or as a PC-AT expansion card.

The **DAS-90** solves the problem of access to Channel Status, making possible its exploitation and solution of related problems. The **DAS-90** also provides metering and monitoring.

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AES PREVIEW

The 91st Convention of the Audio Engineering Society will be held at the Hilton Hotel and Sheraton Centre, New York during October 4th to 8th, 1991. As usual the Convention will consist of a wide range of technical papers with an associated exhibition. We have compiled this preview from details available to us at the time of writing

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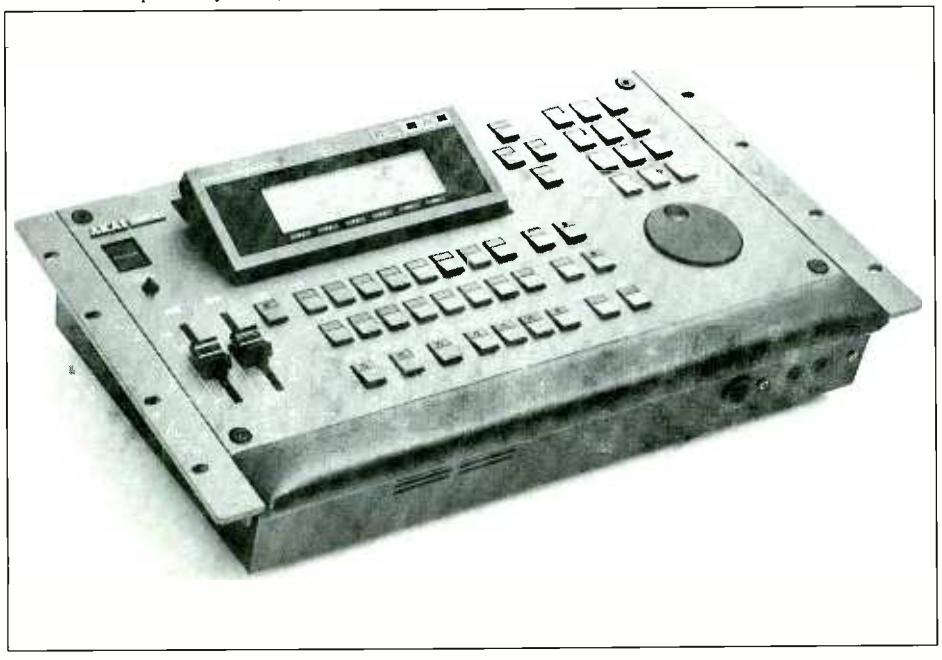
● **AD Systeme:** will be showing the *Optifile 3D* console automation system with the latest updates and also the *Optiview* 24-track previewer that will give a full 7 seconds advance warning of all up and coming cues or drop-ins. ● **AB International:** full line of power amplifiers from 75 W/channel to 1400 W/channel. ● **ACO Pacific:** music and measurement microphones and Acoustic Interface System. ● **Acoustical Physics Labs:** control room and studio designs for recording studios, post-production, broadcast and performing facilities. ● **Acoustical Supply:** display includes loudspeakers, cable, connectors, test equipment and CAD software. ● **Adams-Smith:** will be showing the *2600 E-A/V* audio editor with their new *Super-Sync* software, and the *Zeta* family of products including 'Transport Emulation' for the *Zeta-Three*. ● **Adamson Acoustic Design:** new products on show include the *B118, MH121, FR12* and *S218* loudspeaker systems; also the *AX400* and *DSPX 4000* loudspeaker controllers. ● **ADC Telecommunications:** cable and connection productions. ● **Akai/IMC:** will be showing their *DD 1000* disk recorder/editor along with *DL 500* and *DL 1000* remote control units; *DD-VE1* video editor interface, and *Mac* software providing on-screen editing of all parameters and remote control. Also being shown are the *S1100EX* 16-voice expansion unit and *Mac* remote control software for the *S1100* sampler, the *MPC60 II* updated MIDI Production Centre, and the *ME80P* programmable MIDI patchbay. ● **AKG Acoustics:** will be showing their full range of microphones, headphones and signal processors. New products include a wireless microphone system designed for multichannel operation in sound reinforcement systems. Also on show for the first time will be the Orban *400 A* transmission limiter. Also featured will be the range of dbx signal processors and the AKG *DSE 7000* digital workstation. ● **AKY Recording Supplies:** no information available. ● **Alesis:** full range of digital signal processing, analogue processing, compact mixing console and MIDI based products. ● **Alpha Audio:** featuring the *Boss/2* Automated Audio Editor; the *DR-2* digital audio hard disk recorder and the *DR-2* Edit Controller. ● **Amcron:** full range of power amplifier products. ● **Amek/TAC:** the *Hendrix* multi-purpose multitrack console is the latest addition to the Amek range and is based on design elements from the *Mozart* console including a version of the Amek/Steinberg *Supertrue* automation system. Also on show, the *Motionworker* machine control system, and a software controlled dynamics package both designed for the *Mozart*. Other products include the *B2520* console a broadcast version of the *G2520*, the *BCIII* console, the *Classic* console and the Rupert Neve designed *Medici* equaliser including two new controls, 'Warmth' and 'Sheen'. Products from TAC include the new *SR6000* sound reinforcement console, the *B2* compact professional console which will be shown in three sizes, the *Bullet* console and *ES8* serial interface, which enables remote control from all major edit control systems, and the *Magnum* console with the *MICE* mute switch controller. ● **Ampex:** full range of professional audio tape products including *478* Low Print mastering tape. ● **AMS:** new products will be the *Logic 2* large format, standalone all-digital, recording console with dynamic automation of up to 256 channels; and the *AudioFile Plus 16* hard disk recording and editing system. Also on show will be the *Logic 1*, *SoundField* and *ST-250* mics and the range of signal processors. ● **Analog Devices:** several new products have been added to their range of high-performance signal processing ICs including the *SSM-2018* wide dynamic range/low distortion VCA, *SSM-2142* integrated balanced line driver, and the *SSM-2017* ultra-low-noise preamplifier. ● **Anything Audio:** pro-audio dealer. ● **Apex Machine Company:** manufacturer of audio and video cassette printers, will be showing three new printing machines, the *RS-100*, *RG-1000* and *CA-20*, along with their recently introduced rotary gravure offset print heads. ● **Aphex Systems:** full range of audio processing equipment and interfaces. ● **API:** range of EQ and mic amp modules while featured will be a new API *Discrete* series console equipped with *Touch Reset*, a gas plasma touch screen that allows control of console set-up, storage and console reset. Control of channel settings is through the computer or directly on the channel. ● **Apogee Electronics:** showing the full range of digital converters and OEM modules. Featured will be the *DA-1000-E* D/A converter and the stereo portable *AD-500* analogue to digital converter. ● **Apogee Sound:** new is the *SSM* loudspeaker, which is a compact, high output, wide-bandwidth system intended for front audience fill and monitor applications. New *'P'* series processors for both permanent and portable use with the company's speaker systems will be on show as will companion integrated processor amps and stereo amps. ● **Apollo Masters Corp:** master lacquer discs. ● **ART:** full range of microprocessor controlled digital audio signal processors. ● **Archon Company:** *Arcoset* printing systems, direct on-cassette imprinting systems. ● **Arcoustics:** no information available. ● **Ariel Corp:** featuring their new DSP development systems and pro-audio products. ● **ASC:** will be displaying their range of acoustic materials including the *Trim Trap* lightweight 'glue-up' *Tube Trap*, and the tripod mounted *Studio Trap*. ● **Ashley Audio:** MOSFET power amplifiers, limiter/compressors, noise gates, graphic and parametric equalisers and a line of



Akai DL1000 digital editing controller

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rack-mount mixing products. ● **Audio Accessories:** audio jack panels; pre-wired audio patch panels; patch cords; patch cord holders; polysand and video panels. ● **Audio Action:** production music library of 200 CDs representing 10 libraries. ● **Audio Animation:** featuring the *Paragon*, a fully digital on-air broadcast signal processor, display includes the *Paragon Studio* offering four channels of expandable DSP. Also featured will be the *Muse* digital mastering console. ● **Audio Developments:** their comprehensive range of professional mixers and ancillary equipment including the new *ADO66-11* stereo mic amp and MS decoder. This consists of a small portable mixer type and two phantom powered mic inputs. ● **Audio Logic:** will be featuring their *660* and *266* dynamics units, the *440* Quad Noise Gate, and new range of crossovers. ● **Audio Precision:** the *System One* high performance test and measurement equipment for both digital and analogue domains. ● **Audio-Technica:** full line of wired and wireless microphones including 40 series, *Artist* series, *800* series and *UniPoint* series. Also range of accessories. ● **AudioControl Industrial:** featuring *SA-3050A* 1/3-octave realtime analyser. ● **Audiomation:** will be showing the *Uptown* moving fader automation system with Version 2.5 software. Also on show will be their new 'live sound' system. Sellmark, on the same stand, will be showing its full range of linear and rotary carbon potentiometers, and the *6000* and new *7000* Series faders. Also on show will be the latest *Omnitrac* fader. ● **Audiotechniques/TubeTech:** featuring TubeTech tube limiters, equalisers, and mic pre amps. ● **Audio Teknology:** *Paragon* series live mixing console. ● **Audra International:** displaying audio products manufactured by ARX, Alphaton, Emilar and Etelac. ● **Augan Instruments:** range of multichannel optical recorder/editing systems. ● **Australian Monitor:** will be showing their range of MOSFET amplifiers including the new *K7* and *1K2* power amplifiers rated at 400 and 600 W/channel into 4 ohms each and ruggedised for on-the-road use. ● **Automatic Inspection Devices:** no information available.

Dynaudio Acoustics Professional Monitor range

B

● **Bagend Loudpeakers:** range of live sound speaker systems. ● **BASF:** will be exhibiting their range of tape products including a new ferric cobalt tape *PE 668/968*, a cassette calibration kit with mechanical and azimuth test tapes, a new length LNS voice tape *LNS 17,300*, and the *Eco Shuttle* which is a new recyclable pallet for duplication products. ● **BEC Technologies:** featuring digital audio fibre-optic communication systems. ● **Berklee College Of Music:** with news and examples of their music degrees and professional diplomas. ● **Beyerdynamic:** dynamic, condenser and wireless microphones, dynamic headphones and headsets, gooseneck mics and accessories. ● **BGW:** range of power amplifiers. ● **Bose:** range of speaker systems. ● **Brainstorm Electronics:** will introduce two new products, the *SR-15* Timecode Distributor which includes reshaping controls and a new version of the *SR-1* Timecode Refresher, the *SR-2*. Also on display will be the *TB-4* infrared talkback remote *Communicator*, plus other accessories. ● **Bruel & Kjaer:** featuring series *4000* microphones and the B&K portable digital recording mic system; also examples of the full range of test equipment. ● **BSS Audio:** in addition to their current range, two new products, the *FDS-318* Multimode Variable Crossover offering multipurpose quad biamp or stereo 3/4-way sweepable crossover facilities in 1U, and the *FCS-926* computerised programme equaliser and controller, which further expands the Frequency

Contouring series, launched this year with the *FCS-926* graphic equaliser.

C

● **Cambridge Signal Technology:** *ASC 1000* adaptive signal correction digital filtering unit for automatic analysis and realtime compensation of loudspeaker and room interaction. ● **Cipher Digital:** featuring the *CDI-328* Random Access Recorder available in 2- or 4-track versions. Also full range of synchroniser and timecode products. ● **Community Light & Sound:** expand their range with two new loudspeakers, the electronically controlled *RS660* and the *CSV-8* foreground music system. ● **Concept Design:** exhibiting *DAAD*, the digital duplication system; the *CD 9000* and other equipment and products for the duplication industry. ● **Connectronics:** audio wire and cable, jack patchbays and *XLR*-type patch panels. ● **Countryman:** precision miniature electret condenser microphones. ● **Crest Audio:** range of power amplifiers; sound reinforcement mixing desks; modular amplifiers; and computer control systems. ● **Crown International:** range of microphone systems. ● **Crystal Semiconductor:** will be featuring several new digital audio ICs including the *CS8411* monolithic CMOS device used to receive and decode AES/EBU data, and the *CS5349* complete A/D converter for stereo systems requiring single supply operation. ● **CST Manufacturing:** cassette duplication products including labelling and printing equipment. ● **CTI Audio:** wide range of products from microphones to mixers.

D

● **DDA:** examples from their range of consoles including the new *Interface* mixer, a 4-bus multi-application console, which features a module that sub-mixes six mic inputs, with a single EQ, aux and routing section following the mixed signal path. Also on the stand will be the *DCM224V* post-production console, the *DMR12* and the new 24-track console, the *Profile*. ● **Denon:** showing professional CD players and CD recorder systems and also DAT player. ● **Dialog4:** will be showing the ISDN MUSICAM Codec, a digital fully duplex audio code and decoder able to transmit CD quality 20 kHz stereo audio on switched 64/56 kbit/s networks using terrestrial, fibre optic, ISDN or satellite facilities ● **DIC Digital:** displaying full range of digital audio tape and 8 mm video tape. ● **Digidesign:** manufacturer of *Sound Tools* will debut their *Pro Tools* system, which integrates multitrack digital recording and editing, DSP, MIDI sequencing and automated digital mixing into a *Macintosh*-based workstation. ● **Digigram:** showing the *XTRACK*, which is a 2-, 4-, 6-, or 8-track studio/post-production multitrack system. ● **DOD:** signal and effects processing equipment. ● **Dolby:** new is the Model *430* single ended background noise suppression system, which is based on a modified version of Dolby *SR*. Model *430* has been designed to assist film and video-post production users to reduce unwanted background noises present in field recordings. Dolby will also be showing *DP5500* series Digital Studio-to-Transmitter Link and Model *DP501/DP502* Audio Coding Units. ● **Dorrough:** featuring the stereo signal Test Set, which allows dynamic measurement of level, balance, crosstalk, phase and signal-to-noise. ● **Drawmer:** will be featuring their *DS404* Quad Gate, a 4-channel expander/gate switchable for hard or soft gating, with interchannel linking. Also on demonstration will be the recently introduced *DL241* Dual Auto-Compressor. ● **Dan Dugan Sound Design:** no information available. ● **Duplitronics:** master digital tape duplication equipment. ● **Dynaudio Acoustics:** will be showing for the first time in the USA a range of speakers designed by British acoustician Andy Munro in conjunction with Dynaudio A/S of Denmark. These will include the *M1* nearfield main monitor, the *PPM1* nearfield, the *C2* classical music reference monitor, the *M2* midfield main monitor and the *M3* flush mounting main monitor.

Hybrid Arts ADAP II

E

● **E-MU Systems:** *Emulator III* digital sound production system; *Proteus* sound module family and *Proformance* family. ● **Eagle International:** no information available. ● **East West:** Bob Clearmountain 250 drum samples sound library on CD and pop/rock drum samples also on CD. ● **Eastern Acoustic Works:** professional concert sound loudspeaker systems ● **Electro Sound:** featuring their new *9000* high speed audio duplication system offering a dual slave unit and a new 480 in/s loopbin. ● **Elettronica International:** no information received. ● **Ensoniq:** range of synthesisers and sound modules. ● **Euphonix:** featuring the *Crescendo* digitally controlled production console. ● **Eventide:** will be showing latest hardware and software enhancements to the *Ultra-Harmonizer* series, digital profanity delays for live broadcast, the *VR240* Digital Audio-Logger that stores over 160 track/hours on a single DAT cassette, and the *BD1000* Video Delay for maintaining lip-sync on large concert video projections.

F

● **Fairlight ESP:** introducing an upgraded version of their *MFX* Digital Audio Workstation, the *MFX 2*. The system offers 24-track digital multichannel disk recording and editing aimed primarily at A/V post-production. New features include Turbo SCSI allowing 16 tracks to play continuously from a single hard disk drive, and double resolution colour graphics displaying Clip (cue) Names, Track Names, Audio Waveframes and Fades. All edits such as cuts, pastes, fades, moves and multitrack edits are fast and non- destructive. ● **Ferrofluidics:** showing ferrofluid materials developed specifically for loudspeaker applications. ● **Five Towns College:** no information available. ● **FM Acoustics:** featuring the first of a new generation of power amplifiers, the *FM 801A*. Maximum output voltage is 180 V pp and the continuous output current capability is above 40A RMS. The company claim that due to new discoveries, the amplifier will drive any known load without any form of limiting or compression and that for dynamic signals peak current is unlimited. ● **Fostex:** launching two new products, the *PD2* professional 4-head portable timecode DAT recorder designed specifically for stereo TV and film location work, and the *2412* console, which the company say is the most compact 24-track console ever built. Also demonstrations of video emulation by both the *D20* DAT and the *G24S* 1 inch 24-track analogue machine. ● **Full Sail Recorders:** details on the recording and production courses.

G

● **Gauss:** speaker products being shown include 15 inch and 12 inch high power coaxial loudspeakers, which are retrofitable into existing cabinets or new designs for studio and sound reinforcement applications. Also tape duplication products including the *1100* series test equipment and the *2481* master/bin and *2421* slave. ● **Gefen Systems:** featured will be the *CDJ MacJukeBox*, a *Mac* compatible automated music system with *NSM 2101-AL* CD changer for on-line music applications; the *NSM-2102-AL* CD changer holding 100 CDs with 4 to 6 second access and digital output. Also on show will be the BBC and Digiffects FX libraries and the *M&E* organiser locate system. ● **Genelec:** along with the full range of active loudspeakers up to the *1034A* Genelec will feature the new *1031A* active 2-way DCW (Directivity Control Waveguide), designed for high output nearfield monitoring. Demonstrations of the largest monitor *1035A* are being arranged at nearby Right Track studios. ● **GML:** the series *2000* moving fader automation environment with latest hardware and software updates. Also the GML range of signal processors. ● **Goldline:** are showing the Gold Line range of low cost test equipment. ● **Gotham Audio:** on display will be products from Harmonia Mundi, Audio Design, Cedar, SPL and the Gotham *CDR 90* and *SPOT 90* recordable CD systems. ● **GTS:** no information available.

H

● **Harrison by GLW:** will be showing examples from their range of consoles including the *SeriesTen*, the *ARS-9* automated routing switcher ● **Heino Ilsleman:** automatic cassette packaging

systems. ● **Hybrid Arts:** will be showing the *ADAP II* digital audio recorder and editor; the *ADAP II Portable* which has all the functions of the *ADAP II* in a small unit; and the *ADAP IV* for the digital recording and editing market. This system has four discrete ins/outs and wide selection of features including a choice of magnetic or removable optical disk, realtime and off-line digital filtering and mixing, chase lock to all types of SMPTE; and DAT backup.

I

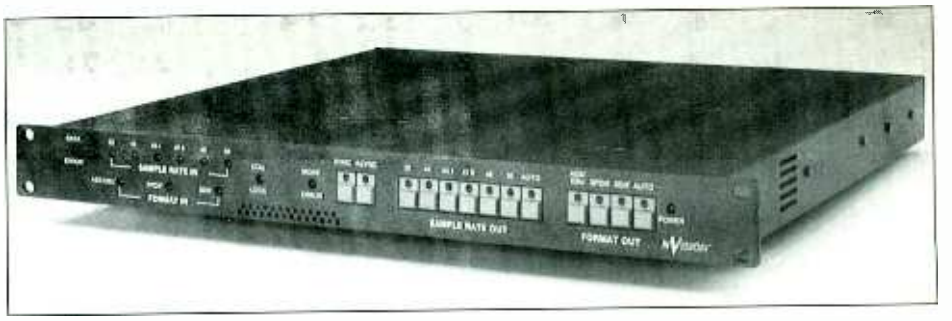
● **Innovative Electronic Designs:** featuring the new *UDAPS* - Universal Digital Audio Processing System. ● **Intelix:** showing the *Studio Psychologist* remotely controlled monitor mixing system and the *MIND Control* distributed remote control system. ● **Intersonics:** no information available

J

● **JBL:** showing the *MR* series of low cost professional sound reinforcement systems. The series comprises 10 models with many new features including a new biradial horn design. Also new are additions to the *Control* series in the form of the *Control SB Micro* speaker and subwoofer ● **Jeanius Electronics:** music production accessories. ● **JL Cooper:** new products on show include *Mixmaster*, an affordable automation system that interfaces to any console, *MLA-1* and *MLA-2* MIDI line amps designed to transmit MIDI data over long distances, *CS-10* an expanded *Control Station* with eight faders recommended for use with Digidesign's *Pro Tools*, *Instant Recall* an external controller that provides 8 channels of audio automation for 250 snapshots.
● **Josephson Engineering:** condenser mics, capsules, accessories and mic preamps. ● **JRF Magnetic Sciences:** full line of replacement magnetic tape heads, alignment tapes and information on relapping services. ● **JVC:** will display the *DS-DT900N* DAT recorder, which reads and writes SMPTE timecode and complies with IEC standards. Also the *DS-LC900* digital parametric controller and S-VHS VCRs and colour monitors.

K

● **KABA:** information on the KABA realtime audio cassette duplication system with tapeless master. Also other duplication products.
● **Kenwood:** CD encoder and writing for recordable CD systems.
● **Kintek:** will be showing the new *KT-904/Plus 2* Stereophonic Converter. This now incorporates the *KT-932* phase meter and *KT-Monogard* within the *KT-904* rack frame. The system will 'intelligently' convert material of a wide variety of types to a stereo format for broadcast applications.
● **Klarity Kassette:** no information available.
● **Klark-Teknik:** will be showing the recently launched *DN735* solid state audio recorder featuring new software to allow synchronisation with a wide range of editors. Also the the series *500* dynamics processors, series *700* digital delay lines, series *300* graphic equalisers, series *400* parametric equalisers, and the *DN60* realtime spectrum analyser. ● **Klipsch:** complete line of professional sound reinforcement speaker systems.
● **Korg:** showing their 8-track hard disk recorder/editor, *SoundLink*. *SoundLink* also incorporates a 16-track MIDI recorder/sequencer and the capacity for full synchronisation to timecode and digital audio signals.



NVision NV4448 sample rate convertor

L

● **Lexicon:** two products will be featured - the *LFI-10* digital audio format interface converts between digital standards including AES/EBU, SPDIF and SDIF-2 and viewing and editing of aux data; and the other new product is *LARES* (Lexicon Acoustic Reverberance Enhancement System) that augments natural ambience in acoustic spaces using mics, MIDI controlled DSP and speaker systems in variable configurations.

M

● **Magnefax International:** audio cassette duplication equipment. ● **Manhattan Productions:** library music catalogue. ● **Mark Of The Unicorn:** range of MIDI and SMPTE products and interfaces. ● **Marshall Electronics/Mogami:** products on display will include the complete Mogami wire and cable products. Featured will be the new range of bantam patch cords. Also on display will be speaker cable products from Sound Runner including flat types, UL listed and quad speaker cable. ● **Media Technologies:** no information available. ● **Meyer Sound Labs:** featured will be a new version of the Source Independent Measurement (SIM) equalisation system that is smaller and faster than the previous model. Other new products include the *USM-1* high power stage monitor, the *DS-2* arrayable mid-bass speaker, and the Meyer *Studio EQ* package. ● **Minim Electronics:** will be showing their range of Ambisonic surround sound equipment for recording and playback in professional situations. Clock systems including the *Presenters* Clock will also be shown. ● **Mix:** pro-audio publications.
● **Motorola:** The DSP operations division will be introducing the *DSP56401* digital audio interface chip that is compatible with AES/EBU and EIAJ *CP340* transmission standards. ● **Music Maker Publications:** music and home recording publications.

N

● **Nady Systems:** wireless products including mic systems, communications and speakers.
● **Nagra:** featured will be the professional open reel *Nagra D* digital recorder. This uses a rotary head and offers two recording formats - 2 or 4 on 1/4 inch tape. Design ensures low tape consumption due to running at 1 7/8 in/s for 2-track and a possible 20 bit record resolution. Also the complete range of portable analogue tape machines will be shown. ● **Neotek:** several new products including the *Broadcast Fader Package* which lets broadcasters configure aspects of the console including VCA or fader control of levels, muting, fader logic, etc. Also new is the *Broadcast Elite* which includes the *Broadcast Fader Package*. Neotek are now offering several machine control options as well as stereo input modules for the *Elan* and announcing the availability of custom colours for their consoles. ● **Neumann USA:** showing the full line of condenser mics but featuring the *TLM 50* omnidirectional mic with a spherical transducer and transformerless circuitry. The capsule design is based upon the historic *M50* tube mic. Also featured are the *KMS 140* and *150* condenser mics for very close work including vocal and instrument - even handheld use. The polar patterns are cardioid and hypercardioid respectively. ● **Neutrik:** are introducing the *A-1* test and service unit available with new RS-232 card. Other test equipment includes the *TT402A* and the *Audiograph* tracing system. Will also be showing the ranges of connectors including the *Speakon* interconnect system. ● **Neve:** featured new products from Neve will include the *44* series compact consoles for broadcast and video-post editing applications; the *HRC-1* digital stereo A-D/D-A converter; and the *2000E* series of TV, production, remote and video post consoles with *Recall Memory System* and a new communications protocol allowing time sharing of 2-channel, *X-86HS* high sampling rate digital 2-track and the *X-880* multitrack digital audio tape machines. Other products will be the Neve *VRP* series and *Flying Faders* automation.
● **New England Digital:** showing DSP options for the *PostPro* and *Synclavier* workstations. Also the *SoundDroid* audio editing system.
● **NVision:** will be showing their first generation of digital products including the *NV2000* audio system, the *NV3512* digital audio routing switch and the *NV4448* digital audio sampling rate converter.

O

OD&ME: following the merger with ODM there will be an enhanced range of CD production equipment on show including the *Monoliner MkII* and information on pending master recorder and electroforming products. ● **Omnimusic:** no information available. ● **Opcode Systems:** featured will be the *Studio 5* MIDI interface, processor, synchroniser and patchbay with microprocessor processing and mapping while offering 128 RAM patch memories. Software displayed will include *Studio Vision*, which combines the *Vision* sequencing program with the ability to recode and edit digital audio; and *Track Chart* which is a studio management program for the mixing engineer ● **Optim Audio:** no information available. ● **Optodigital Design:** no information available. ● **Otari:** full line of tape machines including new versions of *5050* range; mixing consoles including the *Premiere* and series *54/Film*. ● **Oxmoor:** featured will be two new programmable 1/3-octave graphic equalisers. The *DEQ-I* and the *DEQ-II* both support the *PA-422* communications interface and offers constant Q response and sweepable high and lowpass filters.

Q. Who makes the best active loudspeaker?

A.

The Q108 is the first Quested monitor to include integral electronics, with a dedicated electronic crossover and amplifiers.

Designed for a wide range of applications which include broadcast, post production and edit suites, nearfield monitoring and voice overs, in fixed installations and mobiles. The Q108 features a 200 mm (8″) bass driver, 25 mm (1″) dome HF unit and two 100 Watt RMS power amplifier modules.

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P

● **Panasonic:** showing the complete range of Panasonic DAT machines including the *3700* and *3900*. ● **Peavey:** very wide range of products including consoles, mics, monitors, power amps, commercial installation products - speaker systems and mixers, sound reinforcement products, digital musical instruments, keyboards and MIDI products. ● **Penny & Giles:** the full range of professional studio faders and precision controllers. Also jacks and jackfields from Mosses and Mitchell. Featured will be the new range of *Flexipatch* jackfields using PCB technology to provide connections between jacks and connectors. ● **Pioneer Electronics/Technical Audio Devices:** showing speaker systems including studio monitors, subwoofers and compression drivers. ● **Plitron Manufacturing:** designers and manufacturers of toroidal transformers both custom and standard. Also audio power amplification modules. ● **Pro-Bel:** featured products will include the *HD* series 64X64 AES/EBU serial digital audio switcher, the *System 3* switcher controller, the 20 bit oversampling A/D and D/A converters with optional status bit code/decode, the DAWG digital audio waveform generator and the *5245* digital audio reference generator. A wide range of signal distribution modules will also be shown. ● **Pro Sound News US/Europe:** international pro-audio news publications. ● **Professional Audio Systems:** sound reinforcement and studio monitor systems. ● **Pygmy Computer Systems:** showing the *AD-1* analogue-to-digital converter from Pygmy.

Q

● **QSC:** will be showing the new *EX* series of professional power amplifiers which consists of five models with 8 ohm wattage ratings from 175 W/channel to 720 W/channel. Features include improved efficiency, thermal and protection systems that only operate to the degree needed, and open input architecture.

R

● **Rane:** the *Flex* series of signal processors, full size rackmount signal processors and the *MPE* series of programmable equalisers. ● **Raxxness Metalsmiths:** no information available. ● **RE Instruments:** examples of the range of test equipment. ● **REP/SVC:** pro-audio publications. ● **Renkus-Heinz:** new products will include the *EASE* Version *1.1* electro-acoustic simulator program; *EARSS* Electonically Auralized Room Simulation System which allows simulation of how sound will be heard at selected locations within a room; the *C-2* coaxial horn and a weather resistant option for all Renkus-Heinz speakers. ● **Roland:** variety of pro-audio products including the *DM-80* multichannel hard disk recorder and the *RSS* space sound panning system. ● **Roldex:** audio cassette printing systems including the *STF-2* automatic dry offset cassette printer and the *A-5* automatic photopolymer platemaker ● **Rolls Corp:** wide range of modules in a halfrack format including mic processors, mixer, mixer/phantom power, amplifiers and EQ. ● **RPG Diffusor Systems:** will introduce the thermoformed *QRD Diffusor Omniffuser* lightweight, low cost modules for walls and ceilings; the RPG *Studio-in-a-Box* acoustic kit for small installations; the *FRG QRD Diffusor* manufactured from fibre reinforced gypsum; *VAMPS* portable performance QRD modules; and the *Diffractal* full spectrum fractal diffusor. The rest of the RPG range will also be on show.

S

● **Sabine Musical:** no information available ● **SAJE:** will be showing the *Memory* fully automated console for live applications. The console is a control surface for digital control of analogue audio whose racks can be situated remotely. ● **Saki Magnetics:** replacement heads for tape machines including 24-track 2 inch for most machine types. ● **Sam Ash Professional:** the first North American showing of the Digital Audio Research *SoundStation Sigma*, and the updated *DASS 100* multifunction digital audio interface. The *Sigma* has 8 or 16 channels as standard with DSP and built-in optical disk storage. Also new Auto Conforming Package/ Multitrack Emulation for Soundstation. ● **Samson Technologies:** comprehensive range of wirelesss microphone products in both VHF and UHF bands. ● **Sanken:** range of high quality mics including the *COS 11* ultra miniature omni lavalier, the CMS-7S MS stereo mic in addition to the studio cardioid types. ● **Schoeps/Posthorn Recordings:** will introduce the new *KFM 6U* stereo 'Sphere' microphone. Also on show will be the complete range of *Colette* condenser mics including the new *VMS 02 IB* modular stereo mic. This mic is intended for both MS and XY recording with line level outputs and a built-in variable MS matrix. Posthorn will show their *Sonotrim* lavalier mics in conjunction with the Lectrosonics wireless *Pro-Mini* systems. ● **SCV Audio:** showing their full line of signal processing products. ● **Sennheiser:** new products to be introduced will include the model *EK 2012* TV UHF receiver for mounting on portable video cameras; the model *BF 530* supercardioid microphone designed for vocal use; the *HMD25* headphone/mic combination; and the full range of *MKH* condenser mics. ● **Sescom:** wide range of audio accessory products. ● **Shure**

FRIDA
FRIDA
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for portability

for value

in the studio

on air

with timecode

in the OB van

the recorder/editor

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Lyrec

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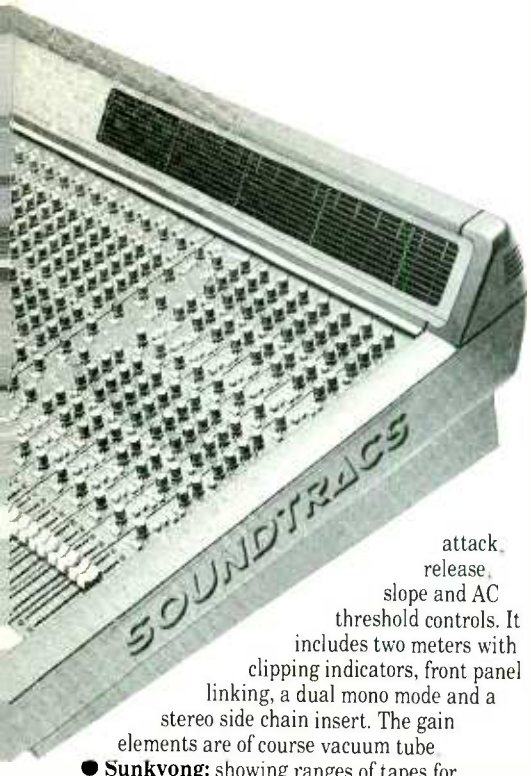
Fax +45 44 53 53 35, Tlx 37568 lyrec dk

Bros: examples from the full range of Shure products including mics, radio mics, portable mixers and amplifiers, auto mic mixers and the Shure *HTS* system.
● **Sifam/Selco:** will be showing selections from their wide range of meters and control knobs. Metering will include VU and PPM to the performance and scale standards of BS/IEC, ANS, EBU and BBC. Featured moulded product will be the *3/03/TPN110* soft touch push-on knob. This knob is black as standard with a integral coloured pointer. ● **Solid State Logic:** featured products will include *Ultimation G* series console automation that allows VCA or moving fader operation to be added to a *G* series console; *ScreenSound* digital audio-for-video editing/mixing system, the *SoundNet* digital audio network for use with *ScreenSound*, the full line of *4000, 5000* and *6000 G* series console, the *Logic F/X* outboard processors. ● **Sonex:** range of acoustic treatment product including Sonex Acoustical foams, building installation foam types, ceiling tiles and Sonex noise barriers. ● **Sonic Solutions:** the *Sonic System* audio workstation with the latest software and hardware updates. Also the *NoNoise* digital noise removal system.
● **Sonocraft:** no information available. ● **Sonosax:** products to be featured include the *FD-M4* mic preamplifier/distribution amplifier which has four outputs and fully discrete circuitry. Includes a vu meter, monitor section with headphones and frame sizes for 12 or 128 channels. Also on display will be the *FD-A100* compact power amplifier; the *JM-3A* compact 3-way active monitor and a selection of other Sonosax products. ● **Sony:** wide range of products including the new *7000* series of professional DAT recorders and editor, the low cost *2700* DAT machine, and the established ranges of large and small consoles, mics, mastering systems, signal processors and digital multitracks. ● **Soundcraft:** featured will four new consoles - the *Europa* sound reinforcement console, the *Sapphyre* in-line console recording console, the *Spirit* range of studio and live consoles; and the *Monitor* version of the *Delta* series. ● **Sound Ideas:** compact disc libraries for sound effects and production music. Featured will be the Lucasfilm sound effects series. ● **Sound Technology:** wide range of test equipment including distortion measurement systems, audio test systems, acoustic analysers and a tape recorder test system.

● **Soundtracker:** showing a production model of the *Soundtracker* digital audio workstation for film and TV post. The system is expandable from four to 24 inputs/outputs.
● **Soundtracs:** *Megas* range of consoles including the *Megas Mix* general purpose console; *Megas Stage 8* bus SR console; the *Megas Studio* recording console; and the *Megas Monitor* complementary to the *Megas Stage*. Also on show will be the *In Line*, *Quartz* and *Sequel* consoles.
● **Spectral Synthesis:** hardware and *Windows 3.0* based software for IBM PCs performing multichannel hard disk recording, sampling, editing, DSP and synthesis. ● **Stage Accompany:** featured equipment will include the *Performer* series - a small system with high output and an arrayable design; the *PPE 2410* programmable parametric EQ that may be controlled by the *SA Net* computer system; the *SA 4525 Blue Box* flexible compact PA system; the *Master* series of studio monitors; the *SA 1310* graphics equaliser; and the *SA Net* computer control system. ● **Steinberg:** Steinberg will be showing *Cubase Mac Audio* which will have the same features as the normal *Mac* version but additionally supports the *Digidesign Sound Accelerator* Card in the *Mac*. This will allow two digital tracks on the arrange page that can be handled as easily as the other 64 MIDI tracks. Steinberg will also have a new software version of the *Topaz* hard disk recorder; and a new *Mac* program for Time Stretching. ● **Stellavox:** will be showing several different models of the *Stelladat* R-DAT portable; several models of the *Stellamaster* studio R-DAT machine; the first showing of the Stellavox outboard D/A converter and the possibility of a completely new product.
● **Stewart Electronics:** range of products including active DI boxes, phantom power supplies, preamps, mixers, power amps and distribution amplifiers. ● **Studer Revox:** new products featured will be the Studer *D740* CD recorder; the Studer *990* analogue console available as standard with up to 80 channels; a new 600 MByte optical drive for the *Dyaxis* together with *Dyaxis MacMix* software Version 3.2 and sample frequency conversion software; the Studio *A623* powered broadcast monitor; the Studer *A728* CD system controller; and new cassette and CD players from Revox. A wide selection of other products will also be on show.
● **Success Speciality:** splicing, leader and hold down tapes.
● **Summit Audio:** will be introducing the *DCL-200* dual compressor limiter with adjustable



attack, release, slope and AC threshold controls. It includes two meters with clipping indicators, front panel linking, a dual mono mode and a stereo side chain insert. The gain elements are of course vacuum tube.

● **Sunkyong:** showing ranges of tapes for duplication. ● **Swire Magnetics:** audio and video cassette shells for duplicators and bulk tape in pancake form. ● **Symetrix:** featured will be the *DPR-44* digital recording/editing station. This system uses a custom graphics tablet to control eight audio tracks with EQ and dynamics, dynamic automation, editing, external control of outboard devices. Other new products include the *524E* Multi-Mode crossover, the *564E* quad expander/gate, and the *SX208* half rack stereo compressor/limiter. ● **Synchrovoice:** no information available. ● **Systems Development Group:** no information available

T

● **Tannoy/TGI:** wide range of studio monitor speakers. ● **Tapematic:** duplication equipment including cassette loaders and slave/masters. ● **Tape Automation:** will be launching a new video tape loader known as the *Delta*. This a fully specified small footprint pancake loader, fully automatic and high speed capable of loading cue-tone, Tap-Code and blank cassettes. ● **Tascam:** large and small mixing consoles, analogue and digital tape machines, DAT and mixer/recorders. ● **TC Electronic:** signal processors including programmable equaliser with remote controller. ● **TDK:** full range of audio and video tape products for bulk applications and mastering. ● **Tech 21:** no information available. ● **Techron:** the *TEF-20* and *SoundLab* software. ● **Telex Communications:** full line of Telex audio tape duplicators, radio mics, intercom systems, wired mics and headsets. ● **THAT Corp:** will be showing the dbx range of VCA products including *2150* IC series, *202* series of modular VCAs, RMS level detectors and noise reduction products. ● **Theatre Crafts:** trade publication for the performing arts. ● **3M:** showing audio recording tape, audio and video cassettes, bulk and special order tapes. ● **Timeline:** new for the show will be the *Console Control Unit* (CCU), a miniature keypad which mounts directly into standard Neve, SSL and other consoles. The *CCU* operates the TimeLine *System Supervisor* multiple machine controller which interfaces to standard console automation software with no changes or updates required. Using *Lynx* Time Code modules the *CCU* controls up to six analogue or digital audio tape recorders, VTRs or sprocketed film transports. Also on show the *Lynx* timecode synchroniser system with controller and system supervisor products. ● **TOA:** showing wide range of sound reinforecement products including the Saori digital processor system and the prototype versions of fully digital mixing systems. ● **Trident Audio:** wide range of audio mixing consoles including the *Vector*. ● **Turtle Beach:** showing the *56K* digital recording system for installation within IBM compatible PCs.

U/V

● **UltraAnalog:** will be displaying range of 20-bit digital converters. ● **UREI:** ranges of power amplifiers, signal processors and studio monitoring systems. ● **Valentino:** compact disc libraries for sound effects and production music. ● **Vega Wireless:** wide range of radio microphone systems including the *600* UHF pro system, the *SP-20* miniature receiever system, the *R-42 Pro Plus* wireless system and the *67B* portable diversity receiver.

W/Y/Z

● **Walters-Storyk Design:** full details of recent acoustic design projects. ● **Westlake Audio:** will be showing the Westlake range of *BBSM* professional studio monitors in configurations from portable to installation types. ● **Whirlwind:** new will be the *HotBox* active reference DI unit with wide frequency bandwidth and choice of battery or phantom powering. The *Presspower* is an active splitter to provide multiple feeds for journalists at press conferences. ● **Wireworks:** showing multipair cabling systems, stage boxes, racks, cables and mic splitters etc. ● **Yamaha:** wide range of products including digital and analogue mixers, loudspeakers, signal processors and digital recorders. ● **Zoom:** range of digital signal processors including the *9002* guitar effects unit, the rack mount *9010* and the new *9030* instrument processor.

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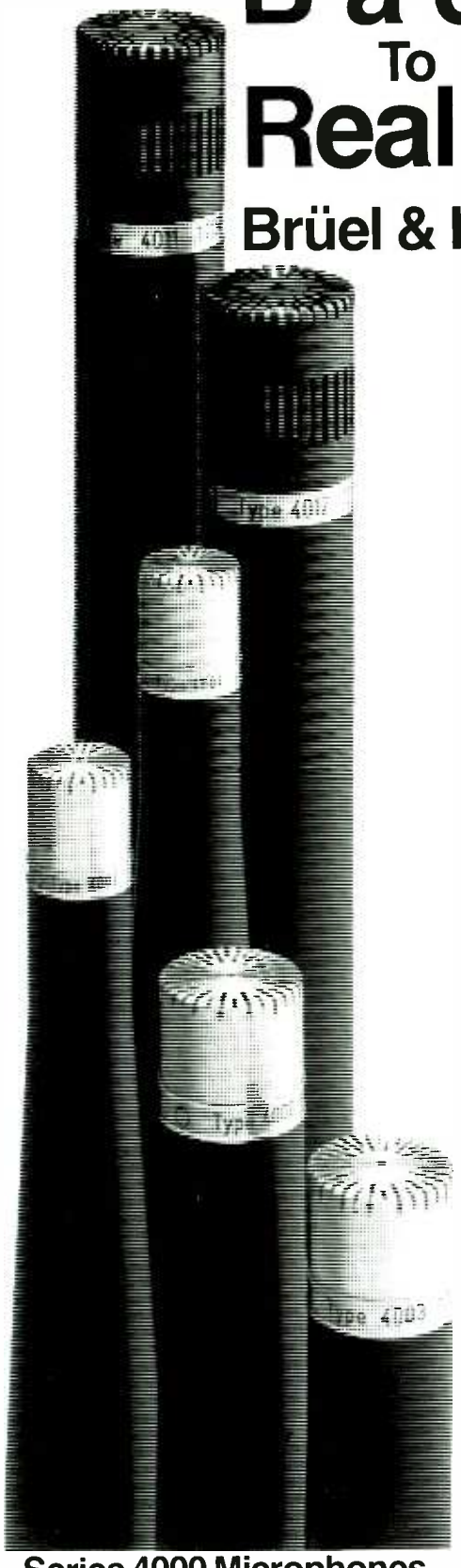

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Control room housing SSL 4000G with Ultimation. Below: A room with a view!

CAPRI DIGITAL

Zenon Schoepe investigates an interesting addition to the world's list of residential studios on the Isle of Capri

Few places on earth conjure up the picture of Eden as vividly as the Italian island of Capri does. Positioned a 35 minute hydrofoil ride from Naples, the rich and the famous, not to mention the artistically inclined, have been flocking to the small green and rocky isle since the time of Augustus some 2,000 years ago. Capri is now poised to make its name in the world of recording with the opening of one of the most ambitious studio projects of the decade and presenting Capri Digital Studios as Europe's answer to the call for an island resort residential facility geared towards the world market.

Brainchild of Capri-born Carloquinto Talamona, owner of small Italian record label Polyxena, the site aims to plug the gap left by the closure of Air Montserrat's residential wrecked by a hurricane some years ago and is blatantly touting for the rock mega-star long-term recording project sector that has been somewhat neglected of late.

Unusually the one studio site with extensive accommodation and recreational facilities, is funded almost entirely by the Italian government as part of a directive to encourage new business and work for the young people of the country's South — a region in danger of becoming totally overshadowed by the more industrialised and affluent North, which has been drawing the young blood away from their home towns in search of employment.

Qualification and approval for the scheme is difficult with strong demands made on the applying party to provide clear documentation and proof that the project will work. With the number of applications received by the Government running into the thousands, less than 200 projects have obtained approval and Capri Digital's glamorous stance as a world class recording studio has been supported by the same hand that has launched floriculturalists and specialist cheese makers in the region. With a 60% Government contribution, 30% Government loan, and the remaining 10% sourced privately by Talamona, Capri Digital is not so much a state studio as magnanimously state funded. The studio investment alone on a hotel site owned by Talamona stands at more than £3 million. Add to this £1.5

Machine room

million spent on the redevelopment of the accommodation on the plot — new building is forbidden on Capri due to the constraints of space on the island — and the proposition is a serious one by any standards. This imparts a definitive feeling of confidence in Capri Digital, safe in the knowledge that things have been funded adequately and correctly run with proper business plans that accommodate such things as the complete refurbishment of the studio's outboard equipment in three years, for example.

Placed on the foothills overlooking Capri town on the quietest side of the island, the studio, its recreation areas and the accommodation is tiered against the side of the hill offering staggering views of what has been widely regarded and acknowledged as one of the most beautiful combinations of land and sea on earth. Four luxury appartments with jacuzzis, private hi-fi, and satellite TV take pride of place just above the main studio building and there is also a restaurant, a swimming pool and a bar area dedicated entirely to the needs of the client. Cars are restricted to a very small section of the island far away from the studio which while it pays dividends to the tranquility of the prospect caused considerable difficulty when it came to delivering the raw materials that a residential studio is made of. An estimated 250 tons of building materials — including studio structures — were transported by electric trolley to the Via Tuoro, above which the studio stands, and then carried by manpower from the paved lane up the 100 or so feet to the building site. Equipment that could not be carried, such as the mixing console and the tape machines, were delivered by helicopter from the Italian mainland and deposited on the studio terrace in a series of trips that tested the mettle of the pilots and the carrying capacity of their aircraft.

An interesting approach has been adopted in the choice of equipment because while many studios strive to complement the kit list of their geographical competitors by offering something different, Capri Digital's position in the global market causes compatibility with the world's studios to be a greater issue leaving individuality to be imparted by a unique combination of facilities, circumstances and location.

Talamona explains: "It is important for us to be compatible with client needs and with other studios.

"Tom Hidley designed the studio; in my opinion he creates the best sounding rooms but it also gives us a level of compatibility with all the other Tom Hidley studios. A client can start in London, Paris or Los Angeles and finish the project here without any problems with the acoustic design."

And a similar logic was applied to the choice of console: "We made our own investigations into the types of console used in top studios and the SSL is simply the most popular console in the world. It was a simple choice."

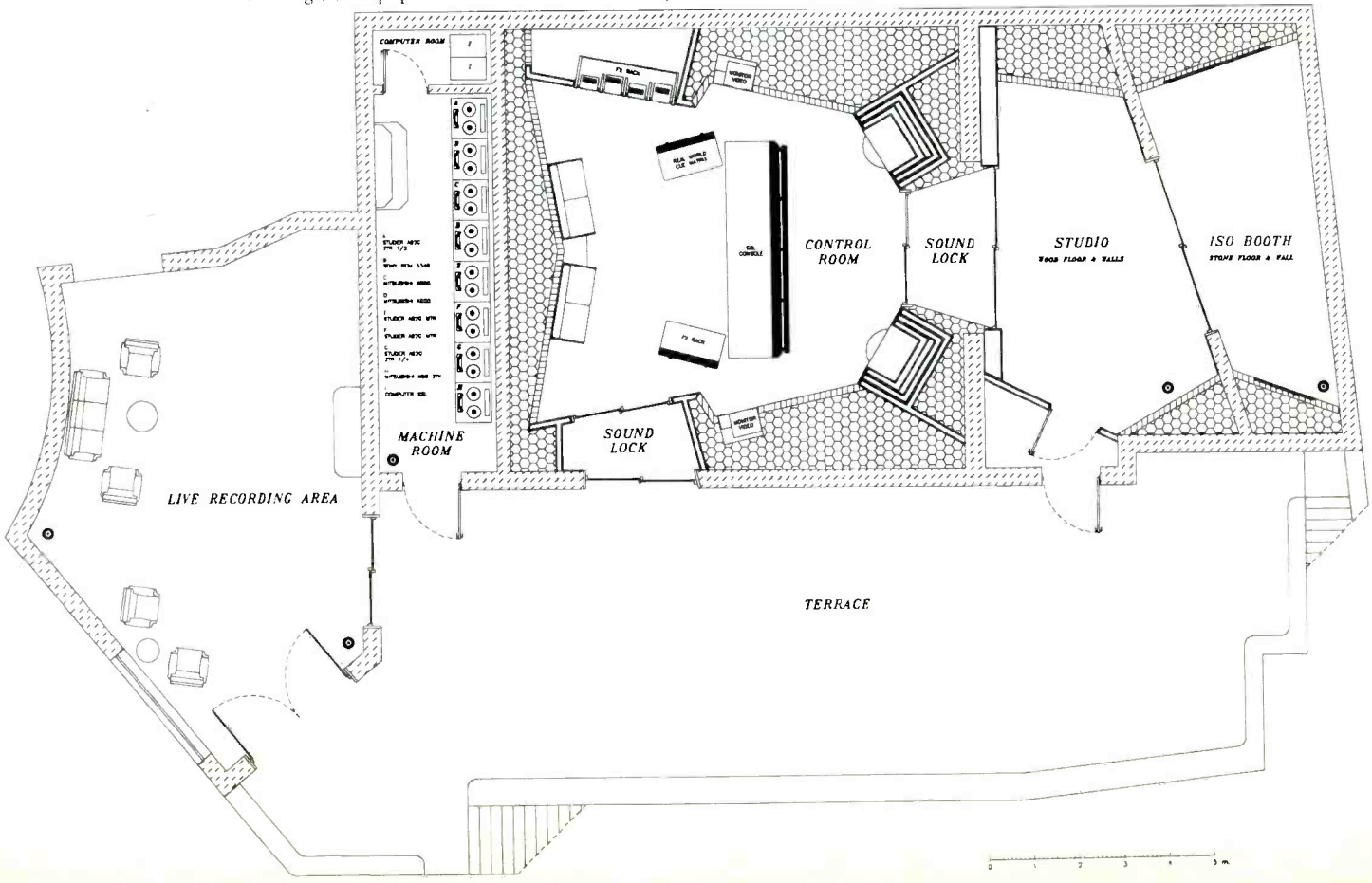
The 64 plus eight stereo channel SSL *4000G* with *Ultimation* moving fader/VCA automation and a Real World *ASM* matrix adding eight aux sends per channel is supplemented by Focusrite *ISA110* and PAST Neve *80* series replica EQs allowing the client to choose freely according to Talamona. The principles of choice and compatibility are taken to the extreme in the machine room tempered with some real-life practicalities.

"We offer every format you could possibly want. That's important because it could be a problem renting a multitrack in Italy as it would have to come from Paris or London and it would still be difficult actually getting it into the studio."

Consequently Capri Digital's machine room shames your average studio complex's sum total of machines. Standing in a row and seemingly stretching into infinity are a Sony *PCM-3348*, two Mitsubishi *X880*s with Apogee filters and DASH-PD-DASH transfer facilities, and two Studer *A820*s with Dolby *A* and *SR* plus a 16-track 2 inch head option all synchronised via *Lynx* and *Motionworker* to the SSL computer.

"We're offering everything to the client while at the same time making it easy for him to compare machine formats in realtime and decide which he prefers for a particular track."

Talamona is referring to a custom-built relay machine switcher panel, which enables remote selection from the control room of the five multitracks in various combinations with



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Studio looking onto the stone clad iso booth

Pod loaded with processing equipment in live area

independent routing to the machines' tracks. Designed by London studio Nomis's Roger Hayler, reponsible for the wiring installation at Capri Digital, it overcomes the unique and peculiar problems encountered when there are 160 tracks to choose from.

Mastering is equally well catered for with ¼ inch and ½ inch Studer *A820*s with Dolby *A* and *SR*, Mitsubishi *X86*, two Fostex *D20*s, a Technics *SV360* and Digidesign *Sound Tools* on *Mac* with 1.2 Gbytes of storage.

The Hidley designed 20 Hz control room sports approximate widest dimensions of 7.5 metres with the expanse of the SSL console and its associated pods overlooked by the brown bill-like protrusions of the TAD loaded Kinoshita monitors powered by 32 kW JDF amplification. There is enough variety of nearfield monitors available to transform the meterbridge of the console into an approximation of the New York skyline. Video and monitor screens are recessed in panels mounted to each side of the desk releasing the more traditionally used area between the speakers to house a mechanically retractable 100 inch screen with Sony *Super-Bright* projector.

Natural wood faced racks built by UK company Human Furniture complement the emerald green cloth covered walls and house just about every popular processor available. Thus pods loaded with Teletronix *LA-2A*s, Tube Tech *PE1B*s, Summit EQs and levelling amps and UREI *LA-4*s vie for attention alongside others brimming with the essential matched pair of TC Electronic *2290*s, Lexicon *480L*, *224XL*, *300* and four *PCM70*s, Eventide *H3000SE*, AMS *S-DMX*s, Roland *R880* and a rack of Valley processors.

A look to the right from the producer's chair gives an uninterrupted view of the island through a glass sound lock and over the large studio terrace reminding you that while the equipment could place you in any recording conurbation in the world you are in fact on the isle of Capri and things could be very much worse indeed.

Entry into the first live room, a comfortable controlled wood floor and walled arrangement, is from the terrace or through a glass sound lock giving excellent visibility into the recording area. From here sliding glass doors give access into a more reflective stone walled and floored room ideally suited to drums and acoustic instruments.

Capturing the sound of these distinctly different areas is made possible by an impressive range of microphones and again a mixture of old and new has been achieved. Neumann valve *M49*, *U67* and *U47*s and AKG *Tubes* are stored along with Sanken *CU44X*s, Schoeps *CMC5*s, Bruel & Kjaer *4006*s and *4007*s, Neumann *TLM170*s, six *U87*s, plus the ubiquitous Sennheiser *MD421*s, *MD441*s, Shure *SM57*s, *SM58*s and AKG *C414*s, *D224*s and *C567*s. Foldback control for the artist is supplied by a Nemesis headphone cue system with four mix stations with the recording possibilities extended by tie lines beyond the studio area. According to Talamona: "You can also play in the bar and even out on the terrace but you might have a problem with noise from the birds and the cicadas!"

Capable of handling video production and post-production and wired for an NED *PostPro* system, which will be purchased if client demand proves high enough, plans include the completion of a dedicated editing room and MIDI suite in the very near future. Talamona believes he is right in offering one studio instead of a multi-studio complex bearing in mind his prospective clientele.

"I feel that it would not be correct to have two studios in the same area because it would make Capri Digital less exclusive for the client. When a client comes here all the facilities are at his diposal and there is no competition from anyone else."

While the proximity to the Italian mainland means that equipment and musicians, including classical players from the Naples Conservatory, can be hired easily, in line with any world class studio Talamona insists that the matter of maintenance cannot be left to outsiders. On site support is therefore provide by two technicians, Jon Mansey — an ex-SSL employee who came to install the console and decided to stay — and native ex-Philips employee Roberto Russo.

"We can repair everything here and we also have a large stock of spare parts," says Talamona but the technical side of Capri Digital will be more than just a repair shop. "We will be developing new facilities for the studio here like the multitrack switching unit with our computer design software. We are also creating strong ties with manufacturers with the view to testing new products."

Ties are being nurtured between Capri Digital and ex-IRCAM employees now working in Rome at the audio digital research centre of IRIS and Italian University Technical Departments. Certain well known Italian companies are also looking to benefit from the audio expertise that the studio can offer.

"We need to be always advancing we can't just offer the island of Capri we have to offer the best technology that is available."

Other staff include MIDI specialist Max Carola and chief engineer Gaetano Ria, the latter drawing on his considerable expertise in the recording industry to advise on the substance of the studio. World market awareness and the aggressive stance that Capri Digital is taking has been reinforced by the appointment of Air Studios' John Burgess, Record Plant founder Chris Stone and Masashi Katagiri as representatives for the territories of Europe, the US and Japan respectively.

Talamona: "We want to become a production centre with clients coming here not just for the studio but for our sound and our team of musicians and technicians. It's the total package.

"It's not just Capri that we're offering but an opportunity to record in Italy, which is famous for its culture and art and maybe also for its wine, food and sun. It's a very attractive place for the artist."

The project that Talamona presented to the Italian government four years ago and received wholescale support for has been realised. It has relaunched the concept of the residential resort studio and in doing so has placed Capri Digital firmly on the map of the planet's first division recording venues.

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MEGABYTES PER MINUTE

A guide to the maze surrounding the conversion of megabytes into available recording time. Storage capacity varies with the types of media and the management of bits. Francis Rumsey provides some useful data

Sound recordings are increasingly stored on computer media such as disks and in solid-state RAM (Random Access Memory). The capacity of these is quoted in megabytes (Mbytes) and it is useful to be able to convert quickly between the number of Mbytes available and the potential recording time at different resolutions and sampling rates. It is also useful to know how the situation changes in the case of multiple channels.

A number of removable media now exist, such as optical and magnetic cartridges, and it helps to know the capacities of these in terms of minutes of recording time. Not all media are suitable for every recording system, and the manufacturer will be able to supply information concerning the compatibility of different disk drives for a particular system, but with the almost universal adoption of SCSI as a means of interfacing storage devices with workstations some flexibility may be had in the choice of medium.

How to work it out

Firstly a couple of reminders. Don't forget that 1 kbyte is 1024 bytes, not 1000 bytes, and that 1 Mbyte is 1024 kbytes. Also don't forget to divide bits by eight to get bytes.

To work out the approximate amount of disk or RAM space required for a file of a known duration it is necessary to know the audio sampling rate and number of bits per sample used by the system in question. Most current commercial systems only store 16 bits per sample on disk, even if they claim to use 18 or 20 bit converters, since this makes optimum use of the storage space available (exactly 2 bytes per sample). If a manufacturer were to allow for, say, 20 bits per sample on disk the additional 4 bits per sample would normally be stored in an extra data byte for every two samples, increasing the amount of space required for a given duration of audio in the ratio 5:4. Manufacturers have tended to stick to 16 bits per sample on disk since any more would increase the already stretched demands on transfer rate, reducing the number of channels that could be replayed in realtime, and affecting flexibility of operation. This situation may change as storage devices improve.

To calculate the amount of space required for a certain number of minutes of recording use the following formula:

Mbytes = sampling rate (Hz) × bits per sample × time × 60 ÷ 8 ÷ 1024 ÷ 1024

For example, to work out the space required for 60 seconds of audio at 44.1 kHz, 16 bits:

44100 × 16 × 60 ÷ 8 ÷ 1024 ÷ 1024 = 5.1 Mbytes

If you know how much storage space you have and want to work out how many minutes recording time it will allow, use this formula:

Time (min) = Mbytes × 1024 × 1024 × 8 ÷ bits per sample ÷ sampling rate ÷ 60

For example, to work out the number of minutes recording time at 44.1 kHz, 16 bits, available from 45 Mbytes of storage:

45 × 1024 × 1024 × 8 ÷ 16 ÷ 44100 ÷ 60 = 8.9 minutes

Table 1 shows the number of Mbytes required per minute of storage at a number of rates and resolutions, assuming that the system makes optimum use of storage space. If the storage space were to be used non-optimally (say by using a whole extra byte per sample for resolutions above 16 bits) then considerably more space would be required than shown here. **Table 2** shows the total recording time for a single channel that could be obtained from 10 Mbytes of storage.

Tape versus disk capacity

Recording engineers are not used to considering the storage capacity of a tape in terms of Mbytes, but to give an example, one hour of stereo recorded at 16 bits, 48 kHz sampling rate on an open-reel digital tape (eg DASH) corresponds to around 660 Mbytes of audio data (ignoring the overhead required for error protection).

Sampling rate (kHz)	Bits per sample	Mbytes/min
32	16	3.7
44.1	16	5.1
44.1	18	5.7
48	16	5.5
48	18	6.2
48	20	6.9
96	16	1.1

TABLE 1: Mbytes required per minute at a selection of resolutions

Sampling rate (kHz)	Bits per sample	Recording time (min)
32	16	2.7
44.1	16	2
44.1	18	1.8
48	16	1.8
48	18	1.6
48	20	1.5
96	16	0.9

TABLE 2: Recording time available from 10 Mbytes of storage

Clearly a 24-track tape recorded on all tracks for the complete tape length would contain 12 times the amount of information as a stereo tape of the same length — in other words, nearly 8 Gbytes for an hour of 24-track. The difference between disks and tapes when storing multiple channels is that disks only use up storage space in the sections where the track has been recorded. Tracks on a disk recorder can't be likened too closely to tracks on a tape recorder, since really the disk just contains a library of recordings that can be played from any of the available outputs at any time, whereas the recordings on tape tracks are fixed in their assignment to outputs and in their timing relationship. In multitrack music recording, sections of tracks are often 'dropped in' for short periods where a particular instrument is playing but left unrecorded for much of the time, thus much of the tape over a ½ hour reel is actually unused. Unfortunately this spare capacity is not really much use because it is not available to be used elsewhere, but on a disk system there is no such limitation and the spare capacity can be redistributed to other recordings.

Some misunderstanding arises concerning the claim that disk space need not be used for silent passages of a track. Disk space is certainly used if the silence is recorded but not if the track is dropped out of record mode for the silent section (see **Fig 1**). A recording of silence takes up just as much disk space as a recording of pneumatic drills.

The effect of more tracks

The calculations and tables given apply to mono recordings. For stereo the available time should be halved, or the capacity required per minute doubled. As far as a greater number of channels is concerned, the recording time will drop pro rata as the number of channels increases, although one should bear in mind what has been said above about multitrack recording on disks. On a disk system, a greater recording time on one 'track' can be traded against a smaller time on another if required. Clearly, not all systems allow for multitrack recording, many being limited to stereo. Some systems (for example the Akai *DD-1000*, which uses optical disks) offer four tracks, but arranged as stereo pairs to optimise disk transfer.

Capacity for error protection

There is again a major difference between tape and disk when considering the overhead required for error protection. Disks have come from the computer market where error-free data storage is mandatory, thus disk drives are designed to look after their own error correction. A hard disk will be formated to avoid writing data to so-called 'bad blocks' where errors would arise, and the capacity of a disk drive is usually quoted by a manufacturer as the formated capacity, having taken error protection mechanisms into account. Therefore all

the available storage space can be used for raw audio data, without needing to worry about adding some percentage for coding overheads.

Digital tape recorders have been designed from scratch as audio systems, and thus the audio data recorded to tape is combined with redundant information and coded to protect the data against errors. Often a large additional percentage of data is required for this purpose, perhaps being as much as 50% on top of the raw data rate. Thus although the raw data rate per channel of 48 kHz, 16 bit audio is around 0.75 Mbit/s, the actual data rate including overheads may be well over 1 Mbit/s.

Effects of data compression

Increasing use is made of proprietary audio data compression systems, such as Dolby *AC-2*, *MUSICAM* and *apt-X 100*. These systems reduce the data rate required for a single channel of audio by at least a factor of four, and often by considerably more than this, with minimal effects on sound quality. The merits or otherwise of these systems will not be discussed here, except to say that they can be used to increase the available recording time on a disk by the appropriate compression factor. Thus a 4:1 compression system such as *apt-X 100*, will squeeze four times the recording time out of a given number of Mbytes of space.

A commercial example exists of a floppy-disk version of a broadcast cart machine that uses data compression. Using 4:1 compression, around 1 minute of audio can be recorded on a conventional 3.5 inch floppy disk if an initial 32 kHz sampling rate is used.

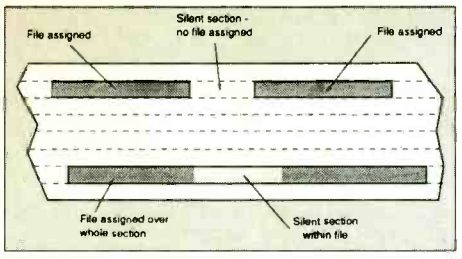

FIG 1: Two silent sections on different virtual tracks of a disk-based workstation. In the upper track the silent section uses no disk space because no recorded file exists here. In the lower section the silence has been recorded and thus consumes disk space

Storage capacities

A number of disk drives suitable for audio storage purposes with many systems are now available. It is important to check with the manufacturer concerned about the compatibility of each with a particular system.

Removable magnetic cartridges: There are three principle types of removable magnetic disk cartridge available, often called 'removable hard drives'. (These are to be distinguished from floppy disks, which are only suitable for samplers and low-capacity data-compressed applications, as described above.) The three types are those that use the SyQuest drive, those that use the Ricoh drive and those that use the Bernoulli drive. The media all look rather like large floppy disks and have a disk encased in a protective cartridge with a sliding access to the surface for the drive heads. The Syquest and Ricoh disks are rigid, while the Bernoulli disk is floppy and actually consists of two platters sandwiched together.

The speed of the three types approaches that of an average-to-slow Winchester disk, with the Bernoulli drive usually being the fastest. Capacities are all around 45 Mbytes, and thus might be useful for medium-capacity purposes. As indicated earlier, at 44.1 kHz, you might expect to get just under 9 minutes of mono recording time. The price of the media is moderate per Mbyte at £50 to £100 for a 45 Mbyte cartridge.

Removable optical cartridges: There are quite a number of different optical drive formats (see 'Optical disks — a recording future?' *Studio Sound*, June 1990) and it is not possible to cover them all here. Standard ISO-format magneto-optical disks (which are re-writable) have a capacity of 594 or 650 Mbytes if both sides are used, depending on whether they are formated to 512 or 1024 byte sectors. This will allow around 1 hour of stereo recording time, depending on the sampling rate.

There are some non-standard optical drives that offer capacities of around 1 Gbyte, and these would provide more like 1.5 hours of stereo recording. Not all optical drives are double-sided, and may require the user to turn the disk over to make use of the full capacity. The price per Mbyte of optical storage is low but the drives can be quite expensive.

Winchester disks: The conventional non-removable Winchester hard disk drive continues to develop and capacities well over 1 Gbyte are now available. A 1.2 Gbyte drive could store nearly 2 hours of stereo.

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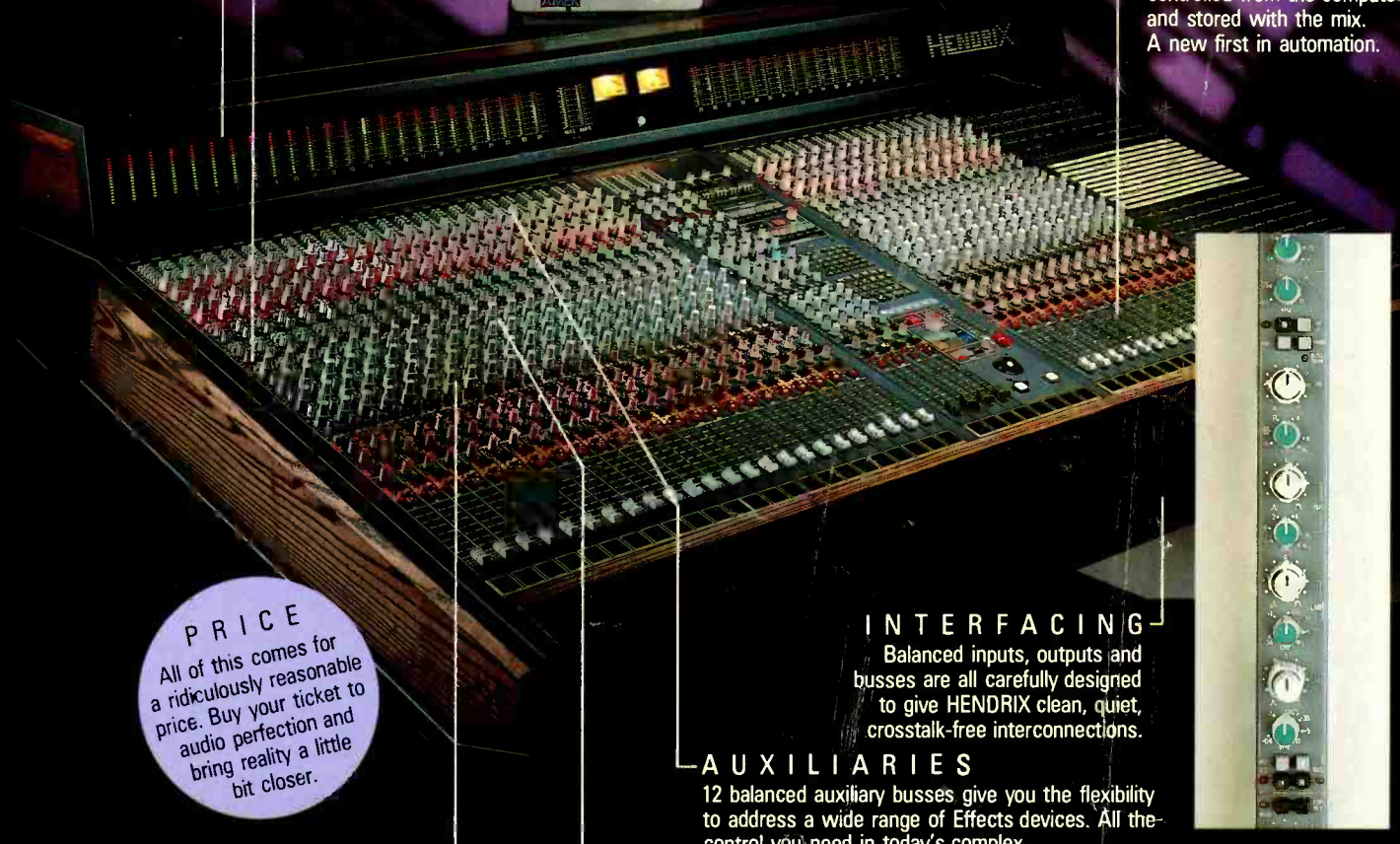

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A DAY AT THE OPERA

Patrick Stapley follows a live recording session: a faithful performance of Mozart's The Magic Flute is digitally recorded for release by Telarc

"Can you hear it?" Jack Renner asks engineer Joseph Magee during recording. The Ramsa is installed in the temporary control room with Abffusor treatment

Edinburgh, Scotland, on a wet Saturday afternoon. Jack Renner turns from the console towards Joseph Magee with a concerned frown.

"Can you hear it?"

Magee cups his hands over the headphones, head bowed listening hard.

"Yes... it's there all right, but I think the music should cover it — we're going to notice it on dialogue though."

From the producer's desk, James Mallinson casts a cursory glance in the direction of the engineers, a signal that he wants to press on. Jack Renner responds with a grave nod to Magee. There is the familiar clicking as the U-matic laces up and all the machines go into record. Magee checks the machines are properly engaged.

"OK James... this will be take... 241."

"Two four one," repeats Mallinson into the slate mic, his English accent contrasting the American voices. With a well practised flick of the finger, he switches on the red light.

The burble from the hall quickly dies down. On the small colour monitor Sir Charles Mackerras can be seen hands poised, orchestra, soloists and chorus at the ready. The hands suddenly jerk into action and music fills the Edwardian room... only to peter out a few seconds later leaving the reverberant voice of Sir Charles hanging in the air.

"Mark that one a false start," says Mallinson scribbling a note with his pencil.

The second attempt results in an excellent take; as Mallinson sends congratulations down the talkback, Jack Renner fixes Joseph Magee with a long, questioning look over the top of his glasses.

"I couldn't hear it," says Magee shaking his head.

"No, neither could I, but make a note on the box anyhow."

In bold letters next to take 241, Magee writes: 'RAIN'.

The anecdote below is typical of the attention Telarc pay to detail: every constituent in a recording from the choice of musicians to the type of *XLR* connector is treated with the same exacting care. Jack Renner co-founded the company in 1977 with producer Robert Woods; the following year they made America's first commercial digital classical recording. From the beginning Telarc's philosophy has been 'less is more' and their approach to recording is both minimalist and purist, without being narrow or stuffy. As the company has grown, so too has its reputation for producing high quality recordings, for capturing great performances and for bringing the listener a remarkable sense of realism. This is borne out by the 24 Grammy awards Telarc has received since 1980 — five of these have gone to Renner for 'Best Engineered Classical Recording', which he has won for the last four consecutive years — creating a record of a rather different kind. Joseph Magee is an engineer based in Los Angeles, he specialises in classical and acoustic jazz recordings, and regularly works for Telarc.

The Magic Flute is the first full length opera that Telarc have recorded; it's also the first time Mackerras has recorded the work, although he's conducted it on numerous occasions. We asked James Mallinson what would make this recording stand out from other versions?

"First of all Mackerras; he is probably the most experienced, but more to the point most knowledgable, Mozart conductor around — he's put a great deal of research into the opera in particular to the tempi and their inter-relationships, and a lot of the tempi will be a little different to those people are used to, although more faithful to what Mozart intended. The cast is the best anybody could have hoped for — they were chosen very carefully as people who could fit together in the overall conception of the piece. Not only are they all excellent singers but they also really understand the characters and impart a real sense of drama to the work. Add to that a first class chamber orchestra and the excellence of the Telarc sound, and I think you will have a very special recording."

Venue

It seems appropriate, then, that the release date will be very close to the bicentenary of the opera's first performance on September 30th, 1791 — two months before the composer's death. This scheduling is in fact accidental — the recording was originally meant to have taken place a year and a half earlier in a Southern Bohemian castle, but "since democracy broke out", as Renner puts it, the castle closed to be renovated as a tourist attraction. Having already recorded the Scottish Chamber Orchestra at Edinburgh's Usher Hall with great success, the decision to relocate to Scotland was not a difficult one.

The three tiered Usher Hall with its domed roof, seats just over 2,200 people with room for a further 333 in the organ gallery behind the stage. Considering its central location, the hall is normally surprisingly immune to external noise, however, on this occasion, because of extensive building work next door, the sessions during the week had to take place in the evening. The hall was booked from July 12th to 22nd: the first day was devoted to rehearsing the orchestra and the following 10 to recording the entire opera. The average day comprised of a three hour music session starting at 6.00pm, with dialogue recording afterwards. On the previous occasion

The choir on stage look out over the 'orchestra pit' towards covered stalls

Renner recorded here, he had set the orchestra out on stage; for *The Magic Flute* he removed the first five rows of the stalls and placed the orchestra off stage, to mimic the effect of an opera house pit. The soloists were arranged on stage singing out over the orchestra, with the chorus behind them in the first rows of the organ gallery. How did the hall's acoustic suit this arrangement?

Renner: "The acoustic is generally pretty good and works well; it certainly has great detail. We have made a couple of adjustments though — to add more bloom to the sound we've covered the seats in the stalls with sheets of plywood right the way back to the start of the balcony — not only does this help to open up the acoustic but it also adds greater warmth — it's a technique we've been using for some time now. We've also hung a large curtain in front of the organ chamber, which successfully removes the 'boxy' effect it was having on the stage acoustic."

In addition to this broad treatment some more focused acoustic adjustments were made. An RPG *Diffusor* panel was placed behind the French horns to deal with 'hard' reflexions coming from a side wall; and a vent, directly behind the basses, which was acting as a bass trap, was covered with more plywood to greatly improve the low end balance.

Recording

True to Telarc's minimalist approach, the bulk of the opera was recorded on just seven mics. The orchestra was picked up by a pair of Neumann *M50* valve (tube) mics placed behind the conductor's rostrum and elevated to about 11 ft — these also acted as ambience mics for the ensemble, so their positioning was critical. The soloist's mics were arranged in a line across the front of the stage — each of the five stands was fitted with a T-bar holding a Schoeps *M221B* valve mic with an *MK5* cardioid capsule, and a Sennheiser *MKH40* — the latter acting purely as a backup. This all vacuum tube set-up formed the basis, to which a couple more Schoeps were added for the chorus parts, and a Neumann *TLM50* to provide a little extra definition from some solo flute passages. Magee explains how this arrangement was arrived at.

"We started off the sessions by putting out Neumann *TLM50*s in the same position as the *M50*s and we also put up a central *TLM50* in the traditional position in front of the conductor. We then listened, very critically A/B'ing between the *M50* and the *TLM50* pairs, while introducing the centre *TLM50*. What became obvious very quickly was that the centre mic wasn't needed — the left/right mics produced a very accurate and detailed picture by themselves. Comparing the pairs, the *M50*s sounded sweeter in the high end with a warmer more rounded low end than the *TLM50*s. We also did some comparisons between the *M50*s and the *TLM50*s and found their response to be incredibly close. In fact, if the hall had contained a little more low end boost we would have used the solid-state *50*s instead — their low end tends to be just a little better defined and tighter.

"As for the mics on the deck, we also did A/Bs between the Schoeps and the Sennheisers and found the Schoeps produced a warmer more flattering vocal sound. This simple miking technique is pretty unusual on a major opera like this — most other companies would be using at least double the number of mics. The problem with that kind of multi-miking is that as soon as you start introducing multiple mono sources, the soundstage collapses and you take away the feeling for the consumer of having a live event occurring in their living room in an acoustic space."

Another unusual factor is that the soloist mics remained open the whole time; apart from occasional fader adjustments of a few dB to compensate for singers moving forward or back, the mics were left untouched — they were never faded out, cut, or manually panned. The stereo positioning of the mics was evenly spread across a slightly reduced stereo picture, and the appropriate effect, whether it be panning or distancing, was created by the singers themselves. Telarc feel this approach produces a more natural as well as a more musical result. To aid the singers, the stage was marked out with tape to give distance and on-axis references to the mics.

The purist attitude was also very much in evidence in the control room (an artist's changing room at the back of the building) where 'high performance' cable was omnipresent, 16 bit converters were bypassed and EQ buttons were firmly switched out. High performance cable in particular has attracted a mixed response over the years, from the highly sceptical to the highly ecstatic and Renner has, in the past, belonged to both camps.

"I was one of those unbelievers who thought that high performance cable was a cure for a disease that didn't exist. It was during some sessions in London's Walthamstow Hall in 1985 that I thought I had the perfect opportunity to prove once and for all that this cable would make absolutely no difference. We had two B&K *4006*s that were essentially a matched pair — they were put out on the same stand in the wind section, one with the highest quality standard cable and the other with Van Den Hul cable. We first listened to the mic with the standard cable and everyone was

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pleased with what they heard, we then switched to the other mic and it was just as if somebody had gone out and taken a cloth away from the front of the microphone — everything changed, the sound opened up and there was far greater detail. Just to make sure we didn't have a mic or desk problem, we swapped over cables between mics — the sound followed the swap. When I got back to the States I ran some more tests using Monster cable, which left me in no doubt that high performance cable really did work. People come up to me all the time and say how can you believe in this stuff look at the maths, look at the physics, it can't possibly work — it does and the proof is in the listening. We use a mixture of Monster cable and MIT [Music Interface Technology] *Proline* cable — the MIT is used for our main mics, connections between the console and machines, and for the loudspeakers where we also incorporate MIT *750 Shotgun* with terminators."

The console, a Ramsa *824* cut down from an *852* had been modified for Telarc to include three main outputs from the stereo bus, a single stereo master fader rather than independent left/right, and professional level on all 2-track returns. The two main criteria for choosing the console were its sonic performance, in particular a short transformerless signal path, and its robust construction. It replaces the Neotek console previously used by Telarc. The console was positioned to the left of the control room facing the side wall, allowing the producer's desk pride of place in the centre of the room. Renner feels that having the console in the middle interferes too much with the direct sound.

For this recording a total of six digital stereo machines were used to provide both direct stereo and a multitrack backup. The main stereo mix output was sent to a Sony *1610/DMR-2000* set-up and a Panasonic *3700* DAT (back-up); both these machines' ADCs and DACs were bypassed being replaced by the DCS 20 bit ADC and the Madrigal DAC. Telarc normally use their own 20 bit ADC unit, which incorporates Ultra Analog chips, but unfortunately the trip over had disagreed with it. An Audio Design *Pro Box 3* was used to convert between SPDIF and AES/EBU.

The remaining four Panasonic *3900* DATs were fed from the console's eight groups: groups 1 and 2 for the orchestral mics, 3 and 4 the chorus, 5 and 6

Mackerras, Mallson and Magee (at console) during playback

Further notes on Telarc's equipment

With so many possible recording and production consoles available it may seem surprising that Telarc opted to use what is, to all intents and purposes, a touring console. Panasonic, however, say that the Ramsa *WRS-852* was designed to offer the ruggedness and flexibility needed for sound reinforcement but not to the detriment of the audio quality.

"I've compared many consoles," explains freelance engineer Joseph Magee (who provided technical assistance on *The Magic Flute* sessions), "and find that the Ramsa *852* offers spectacular performance. In terms of subjective differences I find that mixes through the *852* capture a wider soundstage left to right, greater depth front-to-back, and increased frequency response top to bottom. The result is a much wider, more detailed 'window' into the performance.

"I attribute that transparency in performance to several factors including the electronic design — John Windt had a great deal to do with the component selection and board layout — plus an intelligent grounding scheme. Also, no transformers are used on the mic inputs, and the internal signal path is kept as short as possible — the EQ sections can be hardwire bypasses if appropriate. I've used the *852* on dozens of classical recordings and live performances and find it an excellent choice."

The other innovative aspect of this Telarc session was the use of a quartet of serial controlled *SV-3900* professional DAT machines to record the stereo master mix plus submixes of the orchestral mics, chorus and soloists. All four machines were linked to a *SH-MK390* remote controller, which 'talks' to each *SV-3900* on a network via ESbus protocol commands.

"All these DAT elements can be loaded into a workstation such as the Lexicon *Opus*," says Joseph Magee, "and resynced with one another. Even though we only recorded a sync reference at the head of each take, the inherent speed stability of these *SV-3900s* means that each discrete track will stay in tight sync for a long time. This is obviously very important if we decide to go back and change the timing relationships between the tracks at a later date."

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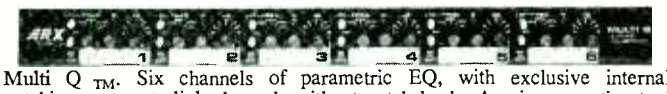

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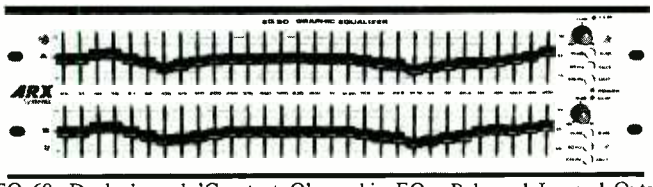

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the soloists, and 7 and 8 a mix of everything. This basically produced an 8-track recording spread across four stereo machines; the idea is that if at a later stage the balance on the U-matic needs to be changed, it can be remixed by transferring the DATs to a multitrack hard disk system (Telarc favour the Lexicon *Opus*) where the three stereo sections would be time slipped back into sync and rebalanced as necessary. To provide a sync reference point, a click was recorded on the machines before each take — this was done rather primitively by tapping a specially EQ'ed mic. Panasonic's *MK 390* remote provided simultaneous control for the four machines.

This system can also be used for overdubs, for example if a singer's part had to be put on later, the U-matic tape containing the orchestral backing would be played back through the console, the soloist's performance would be recorded to its own DAT, while the mix of the two would go to the mix DAT (groups 7 and 8) for a control room reference — the soloist's performance on DAT and the orchestral recording on U-matic would then, as before, be married together during post-production. They will be using the *Opus* to add some sound effects — thunder, a lion's roar and the sound of a lock — as well as adding digital reverb to a section of dialogue requiring a cavernous space.

Telarc have their own editing suites back home in Cleveland, Ohio, where the masters will be assembled using a Sony *DAE-3000* editor; another Cleveland studio provides the *Opus*. Wherever possible complete takes are used for editing purposes, Renner feels strongly that piecing together many short segments destroys the natural flow and continuity of the music. The tape used on these particular sessions was Ampex *467* for the U-matic, and TDK *DA-R120* for the DATs.

Monitoring

Monitoring was from a pair of the very latest limited edition, Anniversary model B&W *801 Matrix IIs*, driven by a Threshold *SA 4E* amplifier. Renner uses B&W *801s* whenever he records in Europe, because it's the standard classical monitor that everyone, including himself, has become very familiar with; this is not the case back in the States where Telarc have been using ADS *1530s* for a number of years.

"We're currently looking for a new reference standard. Although I feel comfortable with the B&Ws, we've got six other people to satisfy, and they have trouble with the *801s* — they don't find them detailed enough, and they feel the low end isn't deep or tight enough. Actually I have a little trouble with that, too. However I think the combination we've got here of amp and cable makes these a very, very useful monitor. The version we're using at the moment is also a lot smoother than the earlier *Matrix IIs* which tended to be very aggressive in the upper mid."

Positioned between the speakers, and in the front corners of the room, were RPG *Abfusors* (Absorptive Phase Grating) panels; these were complemented by RPG *Diffusor* panels at the back of the room. "We like to absorb at the front and diffuse in the rear," Renner says with a grin. The panels were stacked in twos, so that the lower has wells running vertically, and the upper horizontally — an arrangement that Telarc have found produces optimum results.

"We were actually among the first people in the States to take RPGs on location, and we worked very closely with Peter D'Antonio, who developed the system, in field testing and researching various uses and results. They can drastically change the acoustics in a room, and in this case they enable us to hear very comfortably from many different positions — actually this room contains a lot of natural diffusion of its own from relief decoration, woodwork, etc. Apart from acoustically treating the control room I often use *Diffusors* in performing spaces — I think they're the most innovative acoustical material that has come along."

Equipment

The equipment for these sessions came from a number of sources. The heart of the set-up — Ramsa console, Sony digital, monitor amp, cabling, mics, etc — came over from Cleveland. Other items came directly from manufacturers. B&W sent in the *801 Matrix II* monitors. RPG Europe supplied their acoustic material, the Neumann *TLM50s* came from Bauch, and Panasonic supplied the *3900* DAT machines and remote controller. Some equipment also came from Tony Faulkner with whom Telarc have a special working relationship, and yet more originated from Audio FX in London. Organising everything to be in the right place at the right time turned out to be quite a logistics exercise — plus getting it all back again.

Renner stated that this is the most expensive classical project undertaken to date by Telarc with only the Liza Minelli project being more costly. He is confident that Telarc can recoup costs in the first one to two years. As *The Magic Flute* begins selling its first copies, he will once again be busy managing a second opera-like project for Telarc, this time *The Mikado*.

THE CHURCHILL TAPES

The question as to whether it is Churchill's voice to be heard on his recorded wartime speeches has been raised before. Now Barry Fox has come up with some new and seemingly conclusive evidence

Winston Churchill delivers one of his many after-dinner speeches

Heated correspondence in the *Guardian* newspaper earlier this year brought an old audio dispute out into the open again. Are the world famous recordings of Winston Churchill's speeches really of Churchill? Or were at least some of them made by an impersonator?

This is not a nitpicking issue. Winston Churchill's wartime speeches are generally accepted to represent some of the best oratory ever recorded. Historians analyse them. Discs and tapes have been sold to the public round the world for nearly 30 years. They have been wonderful money-spinners for Decca, and now EMI. The recordings have many times been used in films, and radio and TV programmes. But 15 years ago Norman Shelley, an actor famous for his BBC Children's Hour broadcasts, claimed to have made the recordings in place of Churchill. Shelley's claim has since been reported several times and received reasonably wide coverage. But no warning has ever been given to people who buy or use the recordings. Neither the announcer on the recordings, nor the biographical sleeve notes, even suggest that some of the speeches may have been recorded by an impersonator, or by Churchill at later dates.

Both Shelley and Churchill are long dead and cannot set the record straight. The correspondence in the *Guardian* generated conflicting claims and counter claims. Quite simply no-one now knows who they are listening to or when it was recorded.

The record companies involved in releasing the material have adopted a singularly cavalier attitude to historical accuracy.

Unhappily for the record companies, but happily for their customers, it looks as if speech experts at Sensimetrics in Cambridge, Massachusetts have finally discovered the truth. They have been using Churchill's speeches to test the power of new technology designed to analyse speech and check the forensic pedigree of tapes brought to court as evidence. In May, Sensimetrics completed several months work on 20 of Churchill's speeches. Their results support Shelley's claim and contradict public reassurances of pedigree recently volunteered by the BBC.

The analysis also supports belief that some of the recordings made by Churchill have been heavily doctored with artificial echo to create the false impression that they were made in Parliament during the war, when in fact they were made years later in a small room in Chartwell.

The record companies can no longer afford to play ostrich. Even if they do not take action voluntarily, members of the public are likely now to complain to Trading Standards Officers about the record labelling. I already have following failure of first Decca, and now EMI, to make any useful comment.

In 1964 the Decca Record Company issued a set of LP records labelled, 'The Voice of Winston Churchill...authorised by Sir Winston Churchill'. In 1983 Decca's subsidiary Argo re-released them on cassettes. These were sold in record shops and by the Imperial War Museum. When Polygram bought Decca, EMI took an exclusive licence to sell the Argo recordings on the EMI/Music for Pleasure label. EMI is now selling the Churchill cassettes widely. At the Midem international music industry festival held in Cannes earlier this year, record company Thames Distributors promised a digital CD version of the same recordings this spring. At Cannes, Thames said they knew nothing of the pedigree dispute and it was "Argo's problem, not ours".

Although Churchill made broadcasts and 78 rpm disc recordings for the BBC in London during the

Speechstation Notes

Following Barry Fox completing this article, we have found more details about the Sensimetrics *Speechstation* speech analysis hardware and software. It is possible to purchase the board and software to run *Speechstation* as a result of a joint project between Sensimetrics and Ariel Corporation. Potential users will require an IBM *PC* or compatible with a hard disk, mouse, VGA graphics card and monitor. DSP on the board completes most of the processing and therefore the speed of the computer is relatively unimportant. On the commercial side, Ariel manufacture and distribute the board in the USA and jointly support with Sensimetrics.
Sensimetrics Corporation, Building 100, One Kendall Square, Cambridge, MA 02139, USA.
USA: Ariel Corporation, 433 River Road, Highland Park, NJ 08904. Tel: (201) 249 2900
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war, no recording was allowed in the House of Commons. But recordings of his Parliamentary speeches appear on the commercial releases. In the '70s Norman Shelley broke silence and told what happened after Churchill made his rallying speech on June 4th, 1940, at the time of the Dunkirk disaster. He spoke on a TV chat show and mentioned in passing that he had recorded the speech for Churchill. I followed through by phoning his agent who put me onto Shelley. I spoke with him by phone and wrote the story in a hi-fi magazine in 1977.

Shelley told me how the British Council wanted to send a recording of the Dunkirk speech to the US for broadcast to generate support there, too. Churchill told the Council that he was "too busy" to go into a studio and record it, but "an actor fellow" could do it, as long as he approved the voice.

Shelley recorded a test and later recalled how Churchill listened and gave approval. "Very good, he's even got my teeth," Churchill told the Council representatives in front of Shelley. Shelley also recalled how he then went on to do more impersonations. I heard all this, first hand, from Shelley.

The BBC later confirmed that Shelley did indeed record the Dunkirk speech in the Transcription Service studios near Regent's Park but said recently that Churchill made his own re-recording for Decca, in 1949, at Chartwell. At the same time he recorded other missing Parliamentary speeches. But on the Argo releases these sound as if they were made in a large hall, like Parliament, not a small room.

> In the '70s Norman Shelley broke silence and told what happened after Churchill made his rallying speech at the time of the Dunkirk disaster. The British Council wanted to send a recording of the Dunkirk speech to the US to generate support. Churchill told the Council that he was 'too busy' to go into a studio and record it, but 'an actor fellow' could do it, as long as he approved the voice

Robert Berkovitz of Sensimetrics was previously with Dolby Laboratories and Acoustic Research. He has always been fascinated by the Churchill tapes. Sensimetrics currently work for the US Government's National Institutes of Health on computer speech synthesis and analysis. They have developed a PC-based analysis system called *SpeechStation*, which uses a radically new method of speech analysis.

Conventional analysis of speech tabulates the energy at different frequencies. But variations in the talker's health or mood, the weather, the technical quality of the recording and the room acoustic can all colour the sound too much to make such measures reliable as a so-called 'voice print'. For *SpeechStation*, Sensimetrics looked for a speech characteristic that is more stable and difficult to disguise.

Humans produce the vowels of speech by expelling air through the glottis, a slit between two folds of tissue stretched across the path from the lungs to the throat. Air forced through the slit causes the tissue to vibrate with a buzzing sound, which is rich in harmonics spread over a wide range of frequencies. These excite resonances, called formants, in the vocal tract shaped by the tongue, lips and jaw.

Typically there are three or four formants, all below 3.5 kHz. These are enough to make speech clearly intelligible even over telephone lines of limited audio fidelity. The formant frequencies do not vary greatly between individuals but the patterns in which they are combined reflect individual speech habits. There are always two or three formants at the same time during a vowel. And formant patterning is relatively immune from coloration.

So, for *SpeechStation*, Sensimetrics adopted the approach of statistically analysing how often certain combinations of formants occur during speech. The system (hardware and software for use with a PC, costing around US $2,000) converts analogue speech into digital code and analyses the code 50 times a second (faster than formants are produced) to identify the formant frequencies. Then the formant combinations are plotted by frequency on a graph, the first formant frequency plotted against the second and the second against the third. As each Churchill speech lasts eight or nine minutes, many thousands of dots accumulate during each plot, forming clusters or clouds where pairs of formants appear most frequently. The computer sorts the dots into boxes, 125 Hz square, and adds colour to represent cloud density; yellow for high density and blue for low density.

The result is a coloured patch pattern on the computer screen, which shows how often the talker uses certain tongue, lip and jaw configurations. There is very little chance that the clouds generated over a period of minutes by one talker will match the clouds generated by another.

Five of the 20 speeches Sensimetrics analysed were recorded on public occasions, eg after-dinner speeches, and can thus be taken as a reference for Churchill's genuine voice. The other 15 were obviously made without an audience and are thus suspect. The formant concentrations on 12 of these match the reference recordings. But on the other three the patterns are quite different.

All three 'alien' recordings are of speeches originally made in Parliament in 1940. One is the speech Churchill made in Parliament on May 13th, when he became Prime Minister and promised the nation nothing but "blood, toil, tears and sweat". Another is "their finest hour" of June 18th, when he predicted the Battle of Britain and warned of a "new Dark Age" for Europe and the US if Germany won. The third is the most famous of all Churchill's speeches, the Dunkirk rallying call "A colossal military disaster" which was made on June 4th, 1940, and contains the oft-quoted pledge, "We shall fight on the beaches, in the fields, in the streets and in the hills."

The announcer on the recording variously introduces these speeches as "Winston Churchill's speech to the House of Commons...", "Here is his speech..." and "A speech to the House of Commons...". The sleeve note carries a picture of Churchill and the title "Winston Churchill, a selection of his wartime speeches".

The analysis now done by Sensimetrics clearly suggests that when people listen to these three

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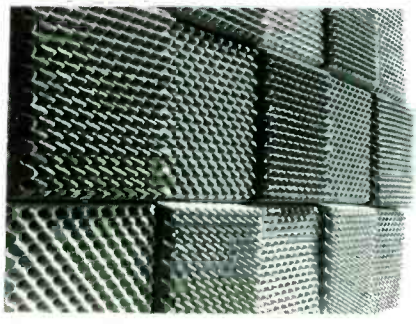

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speeches they are unwittingly listening to an impersonation, almost certainly by Norman Shelley.

The question must now be asked, how can the fake recordings have survived so long without complaints from customers, historians and programme producers who use them?

Even if people believe that the Parliamentary speeches were recorded live in the House of Commons, perhaps as a result of some special wartime dispensation, there is none of the characteristic background sound of a debate. Artificial echo is a very poor substitute. Also the voices on the alien speeches sound different to the casual listener, because in many respects the human ear is still the fastest, most wide-ranging and accurate analytical tool.

But the more and the closer you listen to the recordings, the more confusing it gets. Churchill's voice changes significantly over the five year period of the war, charting the physical toll on the man. And Shelley does a remarkable impersonation, complete with traits and mannerisms.

Decca's role in all this has been far from exemplary. The BBC only has documentary evidence to support the pedigree of the broadcasts that Churchill recorded on 78 rpm discs. But when Decca released the speeches on cassette on the Argo label in 1983, the sleeve notes described them as "historic recordings, taken from radio transcriptions dating from 1939 to 1945".

The BBC acknowledge that Norman Shelley recorded his impersonation of the Dunkirk speech in 1940 for the British Council at the BBC's old Transcription Service Studios near Regent's Park. But the BBC have always believed that all the Parliamentary speeches issued for commercial use were private recordings made later by Churchill at Chartwell.

In 1984, after I had published articles and made broadcasts questioning the role of Shelley, the BBC asked Decca to stop implying that all their commercial releases were sourced from the BBC, when the Parliamentary speeches were not. The BBC have since then worked on the assumption that all these, including Dunkirk, were recorded after the war at Chartwell.

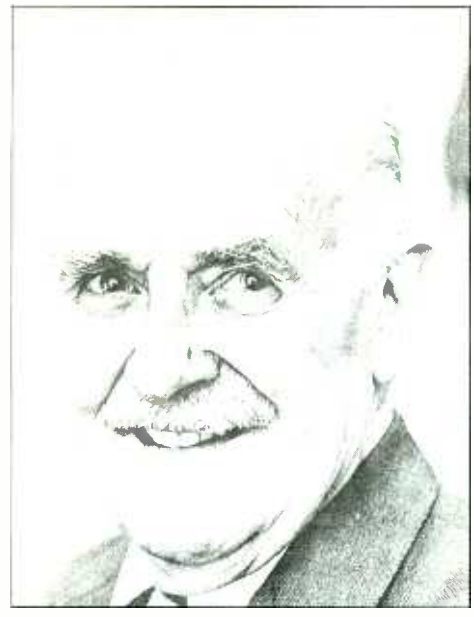
By neat, but clearly unplanned, timing Argo's release of the Churchill cassettes had coincided with the release by Argo of Norman Shelley reading A A Milne's *Winnie the Pooh* and *House at Pooh Corner*.

Significantly, the record companies did not respond to the correspondence that appeared in the *Guardian* newspaper while Sensimetrics were working on the tapes. This followed from a reader's letter (in April, 1991).

The reader thought that Ralph Truman, an actor with the BBC Repertory Company, had recorded Churchill's speeches. David Irving, author of a biography of Churchill replied and told how Shelley had confirmed to him that "several times in 1940" millions of radio listeners were tricked into believing they were hearing Churchill's voice when it was in fact Shelley's. Irving says he re-checked with Shelley when Martin Gilbert, another Churchill biographer, refuted Shelley's claim.

Sally Hine of the BBC's Sound Archives "irritated" the original questioner when she wrote that Shelley's recording was "not really an impersonation, simply an actor reading in Churchill's style.

"Churchill recorded this speech [the Dunkirk speech of June 4th, 1940] together with other Parliamentary speeches in 1949 for Decca," wrote Hine.



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"Several myths have arisen. It has now even been suggested that most of Churchill's war broadcasts were made by Shelley. Here in the Archive we have watched this story grow with great interest and some irritation."

Sally Hine was clearly distressed to hear the results of forensic analysis. "You're joking!" she exclaimed. "This just goes round in circles. What I wrote was what we at the Archives believed and still believe. But we only have documentary evidence on the broadcast recordings we supplied to Decca. We have always referred queries on the Dunkirk speech to Decca."

In 1984, when the Shelley rumours first peaked, the British Council said they were "intrigued" but had no files. Like the BBC, the British Council referred the matter to Decca. But Decca's spoken word producer for Argo, Peter Orr, said that even though he had by coincidence previously been head of the British Council's recorded sound section, he knew nothing of the Shelley impersonations.

"I do not know whether Norman was Churchill or Churchill was Norman," wrote Orr, "and I should be most grateful to be excused from playing any part in this, to me, quite baffling business."

Orr now works for EMI on the Argo label. The latest releases on Music for Pleasure/Argo no longer claim that the recordings are all sourced from the BBC. But there is still no reference to Shelley or the impersonations. When I phoned EMI to ask what the company thought about the *Guardian* correspondence and latest round of interest in Shelley's involvement, the Argo label office said it was "the first we've heard of it". Weeks later no-one at EMI had bothered to get back to me with further thoughts or comment. Nor had anyone from Thames Distributors returned my call.

Says Berkovitz of Sensimetrics: "There is now sufficient weight of evidence to oblige Decca and EMI, and anyone who issues a CD version of the speeches, to re-check their historical pedigree. They should also go back to the original recordings and re-release them without the artificial echo, which was apparently added to make people believe they were listening to a live recording made in Parliament, when in fact they are hearing only an impersonation or recording made years later."

Once the *New Scientist* magazine and the *Guardian* had published the story of Sensimetrics' analysis, calls poured in from round the world. Not surprisingly a lot of people want to know just what they have bought and been listening to, in the fond belief that it was Winston Churchill. The most interesting development is the discovery that the BBC have been basing their reassurance on quite different recordings from those released by Decca, now sold by EMI and soon to be issued on compact disc.

In 1985 the BBC issued a compact disc dealing generally with the war years (CD 3005) and this includes excerpts from some of Churchill's speeches. Although the CD is hard to find, the BBC helpfully supplied Berkovitz with a taped copy. He has now compared those speeches, which appear in excerpted form on the BBC CD and in fuller form on the Decca/EMI releases. At least two speeches ("Their Finest Hour" of June 18th, 1940, and "The First Year" of August 20th, 1940) are entirely different. The BBC's versions are almost certainly Churchill's later re-recordings. So the BBC's reassurances on the pedigree of the Decca/EMI releases lose credibility.

Surely EMI now have a clear duty to burrow into Decca's files and look for dates and authentication of these money-spinning releases. If there is no clear documentary proof that all the recordings were made by Churchill himself, EMI are surely now morally and legally obligated to put warning stickers on all product in the shops. The narration on the recordings should also be re-edited to make it clear that some of the speeches cannot possibly have been recorded on the dates given. At the same time it would make sense to remove the appalling artificial echo which Decca added to Churchill's original historic recordings.

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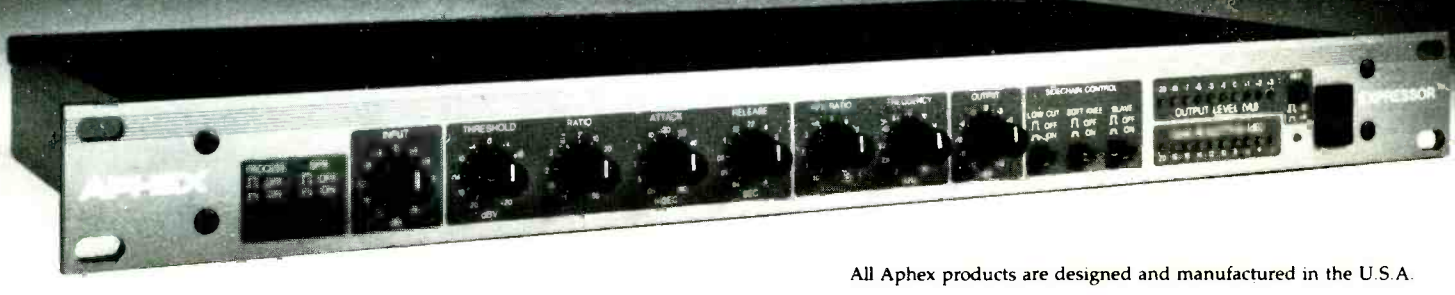

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Chief dubbing mixer Paul Roberts operates TSW's Logic 1

LOGIC 1 AT TSW

The first Logic 1/AudioFile Plus package went to Television South West in the UK. Patrick Stapley finds out how the system has fared one year on

"If I've done my job well," says Paul Roberts, "it should go totally unnoticed." Roberts is chief dubbing mixer at TSW (Television South West) in Plymouth, where he is responsible for post-producing audio for all kinds of material ranging from commercials to full length documentaries. Although his colleagues are only too aware of the painstaking work that goes into laying audio to picture, it is something the viewer takes completely for granted. Last September TSW took delivery of a 12-channel AMS *Logic 1* and a 4 hour *AudioFile Plus*, making them the very first facility to install the package, and Roberts the first operator to use it.

The dubbing suite is small, which makes the compact console and integrated *AudioFile* a good choice straightaway — looking at it now it's hard to believe the control room originally contained a 24-channel analogue console, a wide-bodied Studer *A80* 24-track and cart machines with racks and racks of carts. Roberts says it was interesting to watch the faces of some of the directors as they walked into the new room for the first time: "You could see them thinking what the devil is all this."

The area is split into three rooms — control room, overdub booth and machine room/entrance space. Apart from the AMS equipment the control room contains an outboard and patch rack, two Fostex *D-20*s, a Studer *A810*, a Denon CD with BBC and Digiffect libraries (3,500 sound effects all catalogued onto a database) and of course monitors both audio and visual. The machine room houses a Betacam SP machine, a 1 inch C-format MCI machine, 16 mm machines and an Audio Kinetics *ES 1.11* synchroniser — it also contains the main frame, power supply, I/O rack and so on, for the *Logic 1* and *AudioFile*. The tiny booth with its connecting window contains monitoring and a Beyer *MC734* mic. Redecoration and refitting took three weeks, only a small proportion of which was needed to install the *Logic 1/AudioFile*.

The vast majority of the work Roberts deals with arrives on video; it's rare to get film these days but he has to be prepared, hence the 16 mm machines. Normally the audio arrives on cut Betacam SP or 1 inch C-format, where the two tracks can contain anything — dialogue, music, background ambience — recorded as guide or master audio, in mono or stereo, and often doing some alarming things over edits. These analogue tracks are patched to the *Logic 1* where they are assigned a stereo ADC and transferred flat to the *AudioFile* — Roberts uses *AudioFile* channels 7 and 8 for this. There is also a plan afoot to lift off the high quality Betacam FM tracks during editing to timecoded DAT and so do away with the longitudinal tracks. Depending on the programme, Roberts' job will be to smooth over bad edits, add and edit music, record a voiceover, add effects, and generally repair and enhance what can be less than perfect audio.

How had he found the transition from analogue to digital and how had he felt about replacing tried and trusted equipment with brand new uncharted technology?

"We were buying a system that none of us had ever used before, which was both an exciting and unsettling feeling. Having worked in various dubbing theatres and operated several systems, I suppose I had pretty set ideas of how to do things. Sometimes I think you find yourself working in a

particular way because it's an old habit, rather than the best way of doing the job. When you're forced into the situation of using something unfamiliar, you actually have to form a new habit around it — and often that new approach can be better. This is certainly the case here, and I found myself having to change the way I worked quite radically — the first couple of weeks were mind twisting."

But he did not find that this slowed him up: "...because the system as a whole is so much faster, the learning curve, if you like, is offset against that. What I did find, though, right at the beginning was that I fell into the overkill trap and was getting the *AudioFile* to do absolutely everything — putting in little fades here and ramps there — and that was making jobs take longer. After working this way for a while, I said to myself, 'hang-on you've been dubbing for a long time now and can handle most of this by yourself,' so now I dub normally and only use the power of the *AudioFile* when I run into a problem I can't handle or for something that's going to take a long time. The area that's really going to speed things up and make my life a lot easier is when the automation becomes available. For example on a dialogue track, apart from adjusting levels, I'll be changing EQ all the time during a mix — adding more presence to counter someone moving off mic, ducking low end as the mic pops, changing the EQ setting entirely to suit a different voice and so on — add to that the auxiliaries and the dynamics, which also are adjusted throughout a mix and I think you can see how invaluable a totally automated control surface will be.

"We bought the *Logic 1* having a lot of faith in what we were going to get, and it's actually happening — they haven't let us down; it's just taken a long time. There's such a huge software tree now that any branch you pick at can disturb the rest of the tree if you're not careful, so they're doing it very gently. The desk we have now is very different to the one that arrived last September — originally it was a mono only system, which has become more and more stereo as time has gone on. Because I was the first person to use the system and the only user for several months, a lot of the changes came directly from our experiences here."

An example of this is a change that was made to the control arrangement of the EQ. The EQ, like other functions, is assigned to four Logicators — these are the continuously rotating controls that have built-in light displays around their caps to provide visual reference. The four EQ parameters — frequency, gain, Q and curve — were originally arranged into four pages, so if EQ, Page 1 were selected, the four Logicators would control frequency selection for the four bands, similarly Page 2 would assign the Logicators as gain controls. What Roberts felt would make more sense was rather than spreading a band's parameters across four pages, it would be better to allocate each band a page of its own. So now when Page 1 is selected, the four parameters for Band 1 will be assigned to the controls, enabling all the settings for that band to be viewed and adjusted without having to change pages.

Of course, one of the great advantages of a digital desk over analogue, is that its control surface can be configured to operate in all kinds of different ways — its limitations are bound by software and processing power rather than hardware.

"With a desk of this nature being basically a computer, you can completely change the way it operates. By going to the mixer set-up page, which is displayed on the *AudioFile* screen, I can configure the channel layout as I please. For example, I can make the channel mono or stereo, if I make it stereo I can choose between AB inputs with or without width control, or MS inputs. Then I may consider EQ, and I have the option to add all four bands and both filters if I wish, similarly dynamics can be added as necessary. I may then decide I require an insert and this can be positioned anywhere in the channel. The critical factor is the amount of processing power available in the system, which is determined by the number of TSP (Transputer-controlled Signal Processor) cards fitted — we have five. Obviously if you're using lots of stereo channels you're using up twice as much power, so it's important to set up the desk to provide the necessary functions for a particular job to maximise on processing power — it is not possible to have all the functions on all channels at the same time."

Isn't it possible this could be restrictive?

"The argument is that you don't need four bands of EQ on each channel, you don't need full dynamics on every channel, you may only need inserts on a few channels, and this is absolutely right, but it can be irritating if you have to stop the mix and re-assign the desk because the channel you want to use hasn't got a particular facility attached to it. Having said that, the system does allow you to store desk set-ups, and the more familiar one becomes with this way of working the more likely one will create set-ups to suit the type of session. Individual channel set-ups can also be copied to other channels to speed things up."

There is a store of set-ups that relate to different types of jobs.

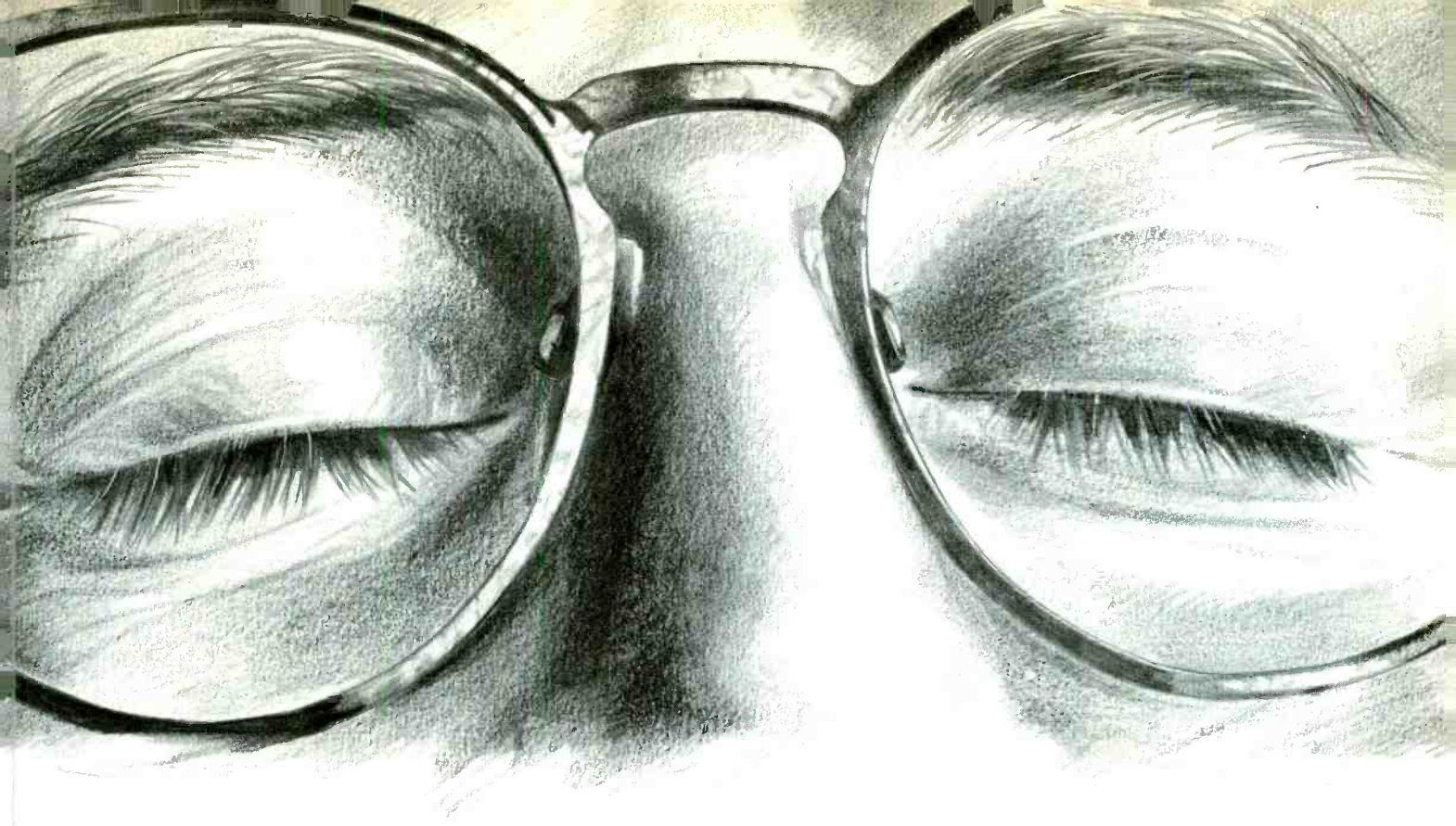
"And I am continually amending and adding to them as time goes on. There is a huge amount of power available to a channel and you really have to consider how much of that is going to be needed. There are four bands of EQ possible, each with a range of 12 Hz to 20 kHz ±24 dB peaking or shelving in either direction, two band filters and full dynamics. Because each EQ control has this full bandwidth, I find that I'll often only need a couple of bands because they'll do everything for me. The same applies to the filters, which can both be either high- or lowpass with full bandwidth selection and adjustable slope — they can be really useful for effects. The dynamics have literally just arrived, so I'm still getting to grips with them but on the face of it they appear very comprehensive, and sound good."

Regarding sound quality Roberts takes the console totally for granted. He finds it quiet and: "So clean that you forget it's a mixing desk. When you mix any number of buses together there's no increase in noise; the only noise you ever hear is from the original analogue, it's fantastic. It's the same kind of percentage jump in quality we experienced going from film to multitrack with Dolby. The EQ sounds excellent but I'm often surprised at how much I have to add to get the effect I'm after — to get a comparable response to analogue EQ, I find I'm sometimes having to add an extra 4 to 5 dB. I think this is because there are no phase or delay errors with digital EQ, so in its purer non-distorted form you need more to create the kind of effects we've become used to with analogue. I also find the high Q settings sound less tight than I would have expected. Another thing that is very noticeable is the difference in quality of the DAT machines when they are input on the AES inputs rather than on ADCs — we always connect them via the AES links."

The *Logic 1* has been supplied with eight stereo ADCs, eight stereo DACs, and two AES/EBU input/output cards — the client chooses the number best suited to his needs. These are assigned as necessary to provide interface to external equipment — so for example DACs might be attached to the four stereo auxs to enable sends to the Lexicon *200* reverb, BEL *BD80*, Eventide *H3000*, as well as a feed for the stereo foldback. Roberts finds that he often uses a single ADC during dubbing and patches machines or mics to it as necessary. The console is totally flexible in layout — anything can appear anywhere or be sent anywhere — but it was noticeable that the last four channel faders were more worn than the others; this is because Roberts always uses these as stereo returns from the *AudioFile*, purely because they're nearest to him.

"The way I'd like to operate the desk is to have all the channels permanently switched to stereo, with the ability to treat each leg separately if need be — at the moment you can't switch to either the A leg or the B leg and dedicate it on the stereo channel. For example, say I have recorded a piece of stereo music to Tracks 7 and 8 on the *AudioFile*, and in a gap in the music I've added a piece of dialogue on Track 7 — there is no way of sending Track 7 to both legs other than bringing it back separately on a mono fader — this means you can end up with stereo and mono returns to deal with one pair of tracks. It would be nice and this is

TSW also work with analogue using the digitally-controlled AMS Assignable console



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GCV KEPPENS

something I've asked for: to be able to parallel a stereo channel so that it can deal with mono or dual inputs."

A very important reason behind choosing the *Logic 1* was its stereo capability, which TSW hoped would greatly speed up and simplify stereo dubs. Has this proved the case?

"By having proper stereo pans, stereo EQ and so on, it immediately halves the number of controls you have to worry about, so this has helped enormously. On our previous desk there were no dedicated stereo channels and things could get a little frantic to say the least. In fact, I was doing very little stereo work before getting the new console: I now produce stereo programme wherever possible. The automation is again going to play its part in making a stereo dub much quicker."

Does this mean all the component parts of a dub are being stereo'ed?

"There are varying degrees of stereo — you can have everything mono with just the music in stereo, you can start building up the background mushes in stereo, or you can start panning things around. Generally speaking dialogue is kept mono — not only because it's disconcerting to have dialogue panning all over the place but because the majority of drama and documentaries we do here use live dialogue. For example, if I get a cut sequence that's been shot in a street in mono, the background is usually pretty good between cuts; you may require a little added mush here and there just to smooth over the occasional edit but normally it doesn't require a lot of attention. Now if the same sequence has been recorded in stereo, it's a very different story: suddenly you find things whizzing about all over the place as the edits go through; the aeroplane that was on the far left is now on the far right, the pram with the squeaky wheel appears to have made a sudden dash, and so on. It sounds horrendous and can cause big problems because you've got to insert a large piece of background to try to disguise it without interfering with the dialogue. Making everything stereo isn't always as sweet and easy as people may think."

How do the methods of working with the *AudioFile* compare with multitrack operation?

"The things you can do on an *AudioFile* are mind blowing. It really is the difference between a typewriter and wordprocessor and I think anyone who's become used to the full power of a wordprocessor would be reluctant to return to a typewriter. I've had occasions where a word's wrong or it's been missed out entirely, using the *AudioFile* I've very quickly slotted in a replacement, time squeezed as necessary and got it into perfect sync with the picture. That wasn't impossible on our old analogue set-up, but it took forever to do and the end result wasn't as good."

Is there ever a need for more than eight channels in the *AudioFile*?

"Generally speaking eight's OK, you can always submix to other tracks anyway but there is one point about the *AudioFile* that we weren't aware of when we bought the system, which may catch other people out. Although it's an 8-track system, it doesn't necessarily allow you to have eight outputs simultaneously. The reason for this is the way the hard disk operates with buffers called FIFOs (First In First Out). There are eight of these RAM packs in an 8-track system that either input or output data. So if you want to playback eight separate sounds, they're buffered up ready to go and as they play through, the hard disk tops up the FIFOs. That's fine, but the problem comes when you do a join on a track, at that point the system requires two FIFOs — one buffer for the outgoing sound and another for the incoming one — but on the screen it looks like one single track, just like a multitrack tape. Say you're working in stereo and you do a join — you've used four FIFOs. The limitation is not the number of tracks but the number of FIFOs. There are two ways out, you either upgrade to a 16-output *AudioFile*, which still suffers from the same problem but has double the number of buffers, or you simply supply the system in the first place with more FIFOs. I hope this is something that AMS will resolve in the future." *(This is common to most systems — Ed.)*

What would Roberts like to see the system do in the future?

"What I'd like to see is a totally automated system so that not only are all the desk functions automated, but all effects, reverbs, delays, etc, are built in as well. A totally automated studio. AMS's reason for not taking this approach is they say different people like different types of effect unit. The thing is that in broadcast, engineers don't often get the choice: they're not perhaps as 'spoilt' as studio engineers, often the main criterion for buying equipment is purely budgetary. It wouldn't worry me who manufactured the reverb or delay as long as it did the job — the important thing is that by being integrated, it would make the system and my job more efficient."

And looking back over this first year what does he think of the system now?

"Since installation the room has been used virtually solidly, and extremely productively with little breakdown time. Although the software has taken longer to materialise than we originally expected, much of it has evolved with us, which has actually been very useful, and to a degree I think this will continue, although, of course, with other users coming on line there will be a wider input. I certainly would have no hesitation in making the same choice again. It really is an incredible concept and quite honestly I can't see anything else doing the job as well for us."

At the time of writing, there were eight *Logic 1/AudioFile Plus* systems installed in studios worldwide. Based on the current interest being shown in the system, AMS are extremely confident about future sales.

CUSTOM EELA

Tim Frost visits a Dutch manufacturer and discovers how they build a digitally controlled configurable broadcast desk

Eela Audio, set up in the late '70s by Pieter Bollen and Frans van Eenbergen, began by adapting recording desks for radio work. They now employ a total of 40 people, with a further 15 sub-contractors working more or less exclusively on the company's products. Not exactly rags to riches but the company have developed a nose for what the market wants and have built enough of the right quality and intelligence into their mixers to convince Swedish Radio to commission them to develop and build a mixer for their local radio network. And now other broadcasters, including the BBC, are also putting the Eela desks through their paces.

The mixer that resulted from the co-operation of the Swedish LRAB local radio network and the Dutch NOS/NOB broadcasting service was the *S240*. LRAB liked the philosophy of an earlier desk, the *S90*, which was Eela's first go at a self-op desk. This was aimed at stations who followed the American radio format, where the DJ does most, if not all his own engineering rather than having someone behind the glass operating the turntables and the faders. The Scandinavian broadcaster's demand for more console but with the same operational convenience, took Eela down a configurable route — towards desks that are not quite fixed format and not quite 'virtual'.

One of the main drives behind Eela's approach to mixer design is the customisation required by their customers. Heading the design team, Frans van Eenbergen expressed a feeling that they produced almost too many versions of the *S240* to meet clients' exact demands.

"We found there are as many different ways of using the system as there are stations — radio is never simply 'just radio' — they all need different equipment and different facilities on the mixer. We had orders for 140 units and we made 25 different versions, with nearly all the changes being on the control side. It was good business but not necessarily economic. Also, although reliability has been very good, each time you change something the opportunity to introduce faults increases."

The *S240* laid down some basic design features including VCA faders, CMOS switched routing and a 25-pin D connector for machine control. Although these have given the system the switching and control flexibility, needed by different stations, the *S240* is primarily a 'standard' product; the different operational requirements being met mainly by factory hard-wire modifications.

But it soon became evident that if Eela incorporated a level of intelligence into the mixer, then these configuration changes could be made by simply altering a bank of internal switches, rather than rewiring. This would help manufacturing, as there would be less special configuration wiring to do, and would also allow the mixer to be reconfigured on-site if operational needs changed.

Having established themselves, with two national broadcasters using the *S240*, Eela's *S440* configurable desk was prompted by further project requests from German and Dutch broadcasters. With a common belief among all radio engineers that DJs and radio journalists all have the technical dexterity of a rhinoceros with gloves on, the demand was for a desk with the minimum of bells and whistles that could be very easily operated in a simple presentation suite. But they also wanted it to double as a flexible production mixer for editing work — two almost contradictory demands.

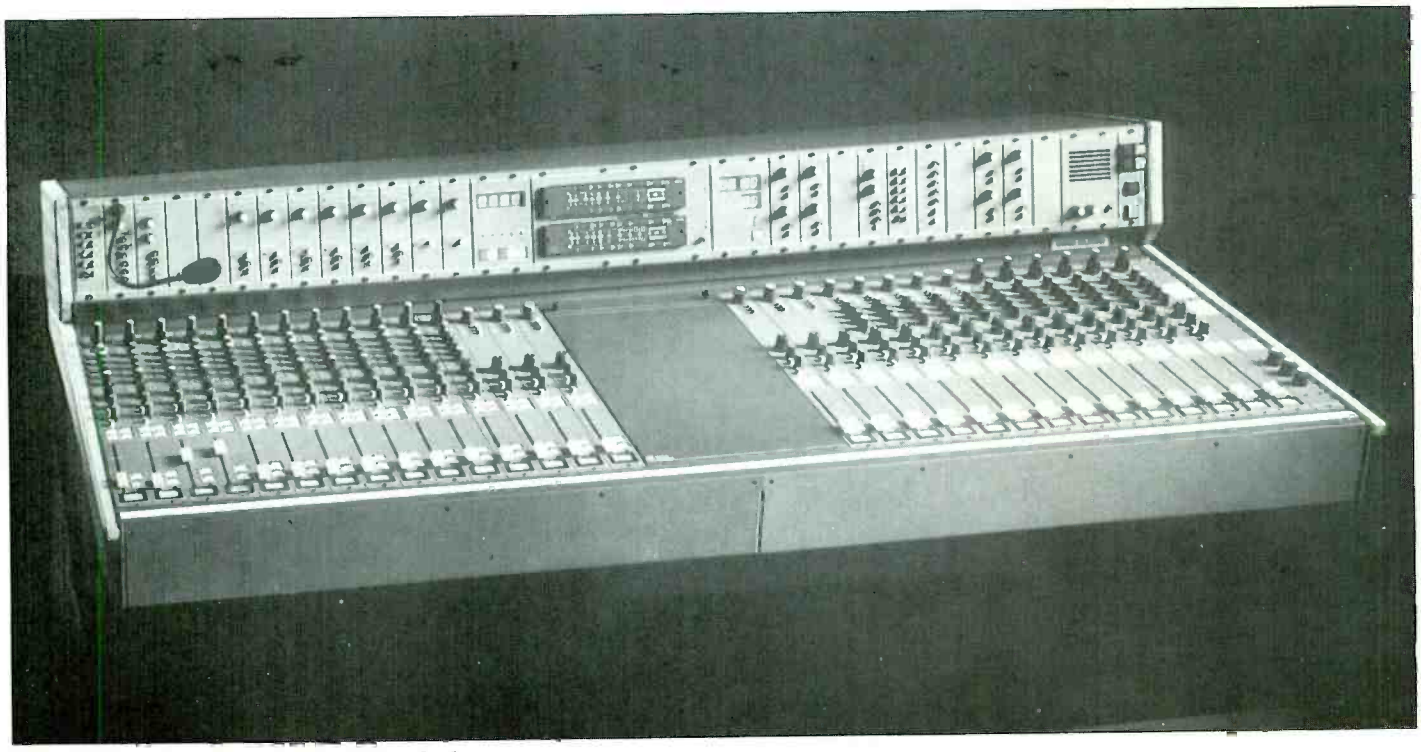
The Dutch broadcasters, still going through a period of dramatic growth in numbers of local stations, had been experimenting with different ways of getting the right type of mixer for their new stations. Originally they used built-to-order desks but while these exactly met the original needs, they couldn't be modified as the station grew and changed the way they worked. So moving from completely customised desks they went to the other end of the market and proposed buying off-the-shelf recording desks and adding outboard sections to provide the necessary control features.

This idea was dropped after just two installations, when it became apparent that even if they could accept the high cost attached to custom modifications, the fact that the engineer had to work on what was virtually two separate mixers ruled it out on a practical level.

"They all want different equipment," observed van Eenbergen, "and the Dutch want it all in one unit."

The answer would be to design a mixer that has all the facilities of the 'virtual' or configurable desks coming onto the recording market. By making all of the switching and level controls remotable, either with VCAs or stepped level controls and solid state switching, the desk can be controlled from any form or control surface and configured in almost any way at the touch of a button.

This total control is almost a by-product of the designs for all-digital mixers but has been more



Eela Audio's S440 broadcast mixing desk

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commercially adopted for digitally controlling analogue signal paths. But in the recording world, these desks are still few and far between and beyond the resources of a small local station, whose total equipment and technical build costs are likely to be under £¼ million.

van Eenbergen took the configurable approach one step further back and designed a desk that combines traditional channel strips, all carrying audio and with a fixed set of controls, with almost total digital control over the operational and routing features.

Since all the station signals go through the mixer then reliability goes beyond crucial. If the desk goes down on-air, then a certain amount of panic sets in while everyone decamps to a second studio. This possibility defined exactly where and how van Eenbergen would introduce intelligence into the *S440*.

"What we wanted to do was combine all of the options into one design and we could only do it with microprocessors, which can be very dangerous. We could use one very clever computer sitting in the corner doing all the work but if there is a problem then the whole desk goes down. So instead, we placed a microprocessor on each channel strip, which simply replaced all of the logic in that strip. Then if something goes down, only that one channel is affected. We went back to DC and logic buses and, as the only connection between the channels is the bus, it is simple to fault find. In fact, as the microprocessor replaces six to 10 CMOS chips, there is an increase in reliability and also if there is something wrong, you only have the one chip to look at."

The software for the processors is all written in-house by a team of developers who rigorously annotate every stage of the program as it is worked on. van Eenbergen insists that this is done after one programmer left having done 90% of a project that no-one else could then finish because they couldn't backtrack though the program.

"From then on I demanded that it is all written and fully documented as we go along so that even I can understand it.

"Software development has been a major investment for us, the engineers all have machines at home as well as here. They are a good team and young — and you need those young dogs for good software."

The processor controls both the initial configuration of the desk and the operational activities. It controls diverse elements from the fader law to the automated routing and answering of incoming phone lines for phoned reports, and that universal feature of stations worldwide — the phone-in.

There are several features unique to radio desks, some of which are slowly migrating to studio format mixers. Transport control, especially of NAB cart machines and more recently CD players, is an absolute essential.

The simple action of remotely starting an NAB cart has developed completely different operational traditions over the years. These split into two main schisms — the fader start and the button start, with several variations within each tradition and new ones being added, especially with the increased use of CD players. The fader start, which fires the machine automatically as the channel fader is brought up, is favoured in Europe and the button start is preferred by US presenters, and variations include simple on/off switching with the fader operating independently, or bringing the channel to air only when both the button is 'on' and the fader is up.

Eela's approach with the *S440* is to build switching for both options into the channel strip,

Frans van Eenbergen demonstrates the S440 fader unit

but which variation is configured depends on the settings of the 8-way DIL switch on the channel circuit board. These eight switches allow each channel to be configured in over 250 different ways.

Since the processor is in charge of all the VCAs and the switching routines, van Eenbergen has also introduced important and subtle features that can be overridden or altered by the customer if necessary.

"On all the 'hostile' switches we put in additional safety, and by hostile I mean any function that will lose the on-air signal," van Eenbergen explains.

"When a channel is on, a hostile switch simply won't function. Even when the channel is off, you still have to hold it down for a user-defined amount of time before it switches. The action of the main fader VCA is damped down so that any small changes that take less than 40 ms are ignored, so that channels aren't accidentally switched on by a nervous hand. The VCAs can also be used for automatically 'ducking' channels or with remote extensions so that the engineer can have final control of levels."

The current production versions of the *S440* use less than 25% of the processor's power and the next stage will be to introduce RS232 and RS422 ports. By using a serial computer interface it will be possible for users to alter every element of the desk's configuration on an instantaneous basis. Exactly how this will be implemented, whether there will be some form of dedicated unit or a standard computer interface, will depend entirely on what the stations decide they want, Eela are taking a wait-and-see approach on that one.

The desks are modular, including the frames, and are assembled at the company's new custom-built site, which they moved into less than a year ago. Being based in Eindhoven, the centre of Philips' operation, Eela is well served by electronics sub-contractors. Nearly all the boards are built off-site and brought in for assembly and testing, which is all in-house.

van Eenbergen explained his construction and test process: "Because of Philips the standard of the sub-contractors in the area is very high and they do all the basic board making and quality control. Each desk takes around two weeks to complete. For each one there is a complete set of documentation produced before we start building and these are generated by computer from standard drawing modules. We need to produce the paperwork first so we have full wiring details; in the early days we started by doing the drawing afterwards but that isn't really practical."

The desks are constructed traditionally using ribbon cable to connect the module buses. On each desk being constructed there is evidence of van Eenbergen's preoccupation, or perhaps fanaticism, about earthing. He has put a lot of work into the earthing format on all of the Eela desks. The *S440* is no different, with balanced signal buses and the audio and digital returns separated to keep the analogue ground as clean as possible.

As each unit is built, the engineer attaches a simple continuity tester that remains connected until the desk is ready for quality control. Earthing problems are picked up as and when they happen, rather than facing the major task of tracking down an earthing loop on a fully loaded desk. The desks are QC'ed using Audio Precision test gear, soak tested and then re-QC'ed before despatch.

Further developments of the *S440* are going in two directions, an increased level of transport data control and a move to a multitrack production format.

"With the advent of CD machines like the Denon broadcast CD, which can report back useful information about their status, we are putting more intelligence into the machine operation control. We are also looking at the diagnostics, although we already have a watchdog circuit, which resets the channel to the default settings in the event of a problem."

With more production work being done in-house by broadcasters, a multitrack desk is very much on the cards. With the spare capacity available within the *S440*, a recording/production version would not be a major problem to develop. It would have to be a different layout but have the same look-and-feel of the *S240* and make the most of Eela's distributed intelligence.

"We still have six DC buses left over and five control signals available so we have lots of facility to expand," van Eenbergen concludes. " We could go multitrack by adding another routing board. Although this would be a dedicated machine, most of the facilities would be the same as the *S440*. Then at least the engineers who are doing both production work and on-air presentations will be dealing with a recognisably similar desk."

A BNS monitor hovers over on-air console.

Tim Frost visits a monitor manufacturer in The Netherlands

A Dutch company that is spreading wider abroad from the springboard of Dutch broadcast is monitor manufacturer, BNS. Established at the tail end of the hi-fi boom of the '70s, BNS survived the early '80s by introducing low cost domestic hi-fi speakers, capturing a notable share of the domestic market from the mainly UK produced competition. They captured the attention of Dutch national broadcast engineers from NOB, with a Neoprene moulded cabinet shown in Paris in 1984.

NOB were at that time looking to upgrade and standardise their monitoring systems. NOB speaker engineer Tom Magchielse felt that between them, they could develop and manufacture a range of monitors that would meet all their requirements.

Why, with the market full of monitors, did NOB go to all the effort of developing their own? Magchielse pointed out that it is almost traditional for national broadcasters to commission their own designs.

"There are lots of monitors available but somehow they do not seem satisfactory. It is a problem all broadcasters seem to face, the Germans have their own designs built by Klein and Hummel, the BBC have their own designs built to their spec and it is the same here.

"One of the key reasons is that you want some consistency in sound, even though the size of the speakers varies. So whether it is a big speaker, a small speaker or a very small speaker, there must be some family relationship where only loudness and bass response will be different; that is what we have to aim for."

The *Professional* BNS range developed by BNS/NOB started with two self-powered compact monitors, the *A3* and *A4*, to which they have just added a sub-bass box. NOB's most used system is the larger *A3*, in which the only standard component seems to be the mid/bass driver, an Audax unit. The cabinet is a polyurethane moulding and the top end uses an HF ribbon driver.

While looking good and being non-resonant, BNS haven't used the flexibility that moulding offers to introduce soft edges to the cabinet design, something that Magchielse believes is relevant primarily to very large free-standing cabinets like the Genelec.

"We do not use large speakers like that at NOB, if we need higher levels or more bass, then you can use the sub-bass box. There we have used a bandpass design with two chambers, which is the only way to extend the response below the free air resonance of the bass driver. The driver has to be good and we are using the most powerful of the Volt range which has a linear excursion of 7 mm."

Unusually, BNS use a ribbon tweeter. The ribbon is based on a magnet structure produced by the sound reinforcement company Stage Accompany, and the ribbon itself is made by Philips (there's a surprise), to which Magchielse has added a foam 'lens'.

"This ribbon unit has quite a complicated history but it has two important properties: it is fairly efficient at 92 dB, and it can sustain about 30 W continuously. You can cross it down to 1 kHz but sounds better if you run the crossover at just over 2 kHz. However, the ribbon is definitely not an isophase device, so we use the foam lens, which evens that out and has some damping influence on the response."

The electronics built into the speakers provide the active crossover and bi-amping from amplifier sections of 35 W and 15 W respectively. Since the monitors are not designed for high level rock monitoring, BNS have set an SPL target of 105 dB steady state; the amplifier powers have been set so that it is not possible to over-power the drivers. The same philosophy applies to the sub-bass unit where the internal 200 W MOSFET amplifier is designed to take the bass driver to the maximum power it will sustain and no further.

BNS have been manufacturing high value hi-fi speakers for several years now and they use Van den Hul cables in the domestic units. What influence had ideas from the hi-fi market had on the *Professional* series?

"We use Van den Hul on the consumer products, but one of the advantages of the active speaker is that with the amplifier in the box and close to the drivers we don't have the wire problem.

"But we are aware of cable differences, and we have a speaker and interconnect cables designed especially for the NOB. We think this cable sounds better and the secret is not so much in the copper but in the insulating material.

"There is a lot going on in high end domestic audio and there is an obligation for professional monitor manufacturers to look into those areas. We are not insensitive to what is happening in these fields, although they are not all applicable to monitors."

The BNSs are used extensively by Dutch broadcasters both in TV and radio. As most of the local radio stations have all been built in the last few years, the studios look stylish and often have the *A3s* suspended by mounting brackets from the ceilings, both looking and sounding good.

But while radio has created the right environment for listening to what is going on, even with the move towards stereo, the situation in the TV studios, Magchielse regrets, is not quite so good.

"People at home are listening to TV through big speakers and finding there is all sorts of bass thumps and noise around, so broadcasters have to put in better monitors.

"But, of course, the rooms have lots of glass and low ceilings because they like it that way. They are saying all they want to do is take out the video monitor and slot in a speaker — they will have to learn that it just won't work that way."

Pete Townsend – "Abbey Road"

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A FUTURE OF MIXING?

Simon Sanders takes a second look at the Yamaha digital mixing processors with their developing range of uses

When Yamaha introduced the *DMP7* in 1987, they were very unsure of what the audio industry would make of it. MIDI and digital audio were still relative newcomers and here was a product that combined elements of both in an innovative way. Three years later, the audio industry still seems a little unsure but, with the rapid growth of MIDI and digital audio, more and more engineers and producers are discovering the power and flexibility that these mixers can offer.

The *DMP7* is an 8/2 line mixer, with in-built equalisation and effects. However, there are two major differences between this and a run-of-the-mill line mixer: firstly, all the mixing processes are carried out in the digital domain, and secondly, all functions are addressable over MIDI and not just the mutes.

The *DMP7* was designed as a desktop mixer and is about the size of an office typewriter. The unit is supplied with a rackmount kit and takes up 8U of rackspace, although a further 2U is required as all the socketry is on the top.

It offers eight unbalanced line inputs with input trims that cater for line levels of between –20 and +3 dBu. For mic lead inputs you need to add the *MLA7* — an 8-channel mic/line amplifier, with balanced inputs, 48 V phantom power, a 20 dB pad, +4 dBu unbalanced outputs for direct connection to the *DMP* and gain controls with peak LED indicators for each channel — all housed in a 1U rack unit.

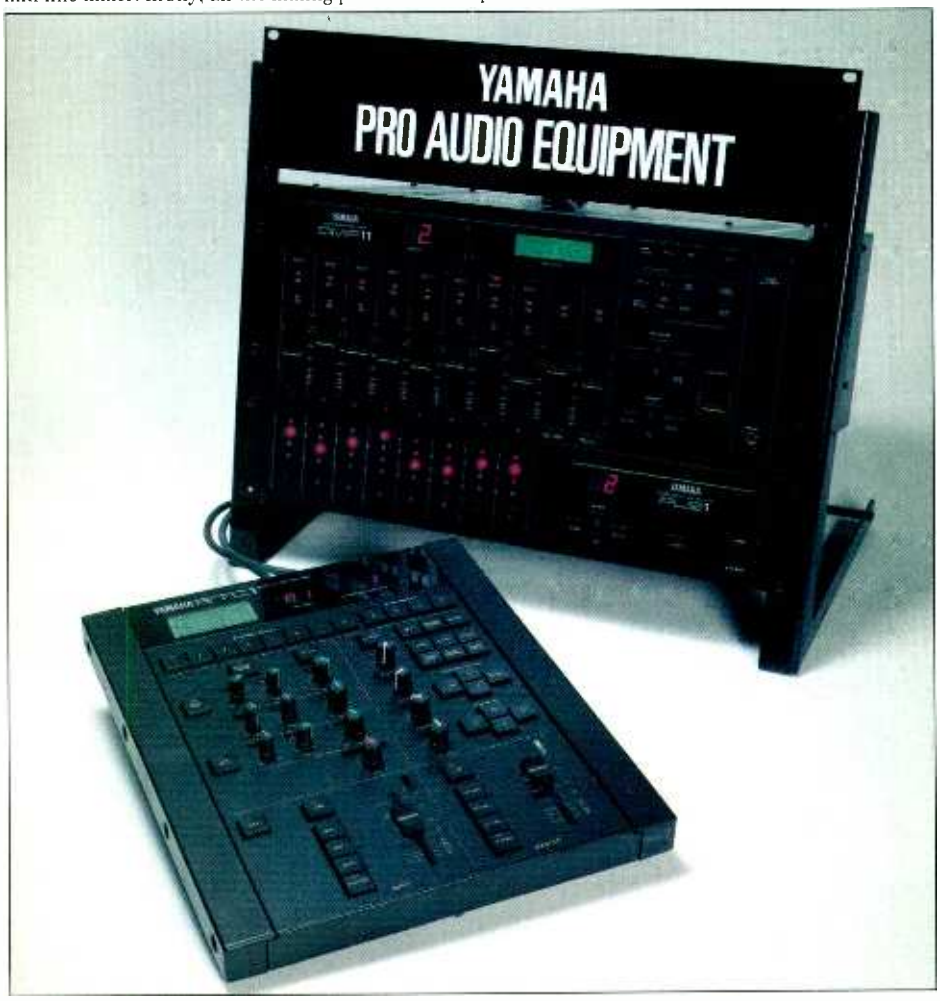
If more than eight inputs are required, two or more *DMP*s can be cascaded. Cascading is achieved digitally (using 8-pin DIN connectors) so there is no degradation of the signal and all 16 inputs, in the case of two machines, are available for use. This option means that the *DMP7* can be expanded as your system develops, with none of the problems associated with cascading conventional line mixers.

Stereo output is via either balanced *XLR*s (for 600 Ω lines) or unbalanced ¼ inch jacks (for 10 kΩ lines), both at +4 dBu. A headphone socket is also provided.

To give increased flexibility on the inputs, Yamaha have introduced the *PLS1* — a programmable line switcher. The *PLS1* has eight channels, each with four inputs and one output. For each channel, any one of the four inputs, marked A to D, can be routed to the common output by pressing an increment button. The patching for all eight channels can be saved to any of 99 internal memory locations, which can be switched over MIDI using Program Change commands.

At first, I could not see how I would make effective use of the *PLS1*, so I began to experiment. What I wanted to achieve, was to be able to switch the outputs from the 8-track recorder to the main console during recording sessions, and to the *DMP* during subsequent mixdown, so I tried running the *PLS1* in reverse, using the outputs as inputs and the inputs as outputs, and lo and behold, it worked. So now, with just two button presses, I can route output of the tape machine to either mixing system.

The *DMP11* looks very similar to the *DMP7* in most respects, but the *DMP11* was specifically designed as a rack unit (4U) without the motorised faders of its sister machine (fader positions are indicated on the LCD). Its inputs/outputs, operating system and mixing parameters are almost identical — the few differences being outlined below.



How do the DMPs work?

On entering the *DMP*, the analogue signal is converted to a linear 16 bit digital signal at 44.1 kHz, using successive approximation A/D converters. The earlier models of the *DMP7* were infamous for producing quite a lot of noise in the A/D conversion stage but Yamaha have introduced an upgrade in later models, giving a 4 dB reduction in the noise floor. For those who are using the older models, this modification is available as a retrofit (the EK board).

Once the analogue signal has been converted, it remains in the digital domain throughout the mixer, being converted back to an analogue signal at the output after any cascaded signals have been digitally summed.

The signal path is much the same as for any other mixer (see **Fig 1**), however, there are no monitor or recording buses, so if you plan on doing any recording, a conventional console will be needed for laying down material (unless you record in stereo pairs and don't mind frequent use of a patchbay). I have been using the *DMP*s for mixdown only, using an analogue console for actual track laying — I have found this route entirely satisfactory and wholly suited to my 'desktop' approach to post-production.

Another common approach is to use the *DMP*s as submixers, mixing the material in the usual

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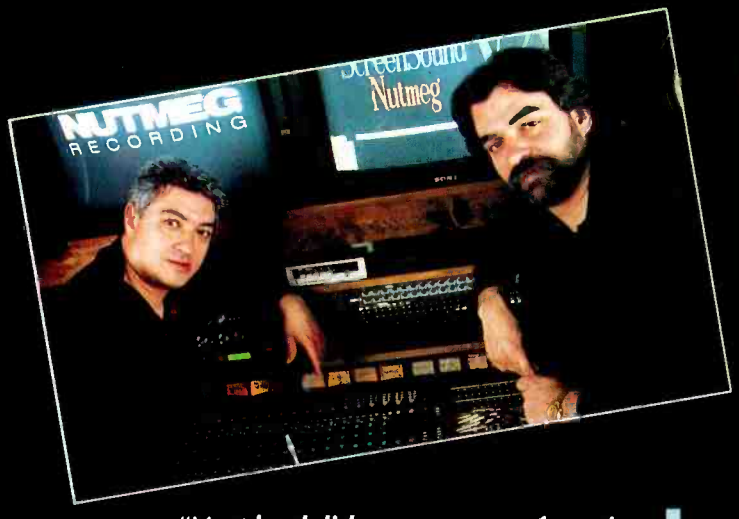

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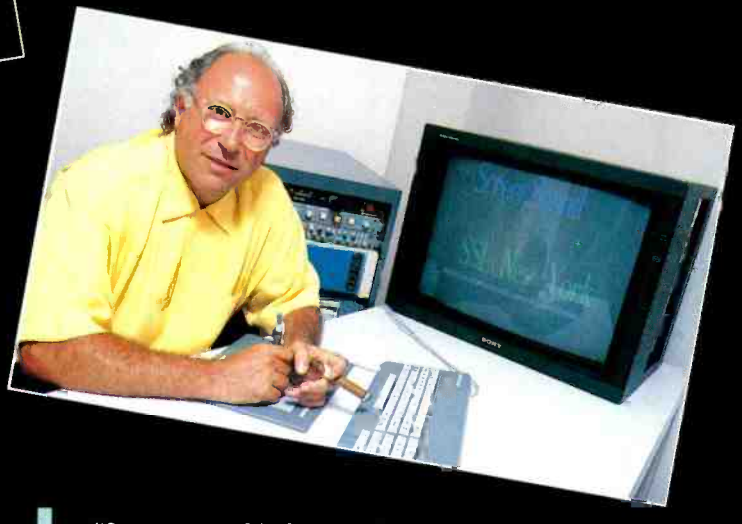

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DMP11
Centre frequency ranges
Low EQ 32 Hz - 18 kHz
High EQ 32 Hz - 18 kHz
Bandwidth Q = 0.1 - 5.0
Gain ±15 dB

TABLE 1: Equalisation characteristics

DMP 7		DMP 11	
Sends 1 & 2	14 Gate rev	**Send 1**	14 Gate rev
1 Rev 1 hall	15 Reverse gate	1 Rev 1 hall	15 Reverse gate
2 Rev 2 room	16 Delay L & R	2 Rev 2 room	16 Delay L & R
3 Rev 3 vocal	17 Stereo echo	3 Rev 3 vocal	17 Stereo echo
4 Rev 4 plate		4 Rev 4 plate	18 Pitch change A
5 Flange A	**Send 3**	5 Flange A	19 Pitch change B
6 Flange B	1 Stereo echo	6 Flange B	20 Pitch change C
7 Chorus A	2 Flange	7 Chorus A	
8 Chorus B	3 Chorus	8 Chorus B	**Send 2**
9 Phasing	4 Phasing	9 Phasing	1 Stereo echo
10 Tremolo	5 Panpot	10 Tremolo	2 Flange
11 Symphonic	6 External low EQ	11 Symphonic	3 Chorus
12 Early ref 1	7 External mid EQ	12 Early ref 1	4 Phasing
13 Early ref 2	8 External high EQ	13 Early ref 2	5 Panpot

TABLE 2: Effects programs

way, then using them to control all or part of the final mix from the subgroup outputs of the main console.

Parametric equalisation

The *DMP7* offers three bands of parametric EQ, while the *DMP11* only offers two (see **Table 1**). The high and low bands are also switchable between peak or shelving characteristics.

The equalisation is versatile and surprisingly warm for a digital system. It can be used for subtle sweetening or drastic correction, giving powerful control over the entire audio spectrum. One point where many engineers will find fault, is that setting the EQ is not as intuitive as traditional rotary controls — fine tuning entails moving the display cursor from one parameter to the next, which can be a frustrating and time consuming process compared with analogue EQ (although the *RTC1* does offer rotary controls for the EQ and makes the equalisation process far easier).

Effects processing

Following the equalisation, the *DMP7* has three effects sends (the *DMP11* only offering two). All effects sends can be configured as pre- or post-fader.

A wide range of internal effects is available (see **Table 2**). Effect Send 3 (or Send 2 in the case of the *DMP11*) can be used as a send to an external effects processor from a mono jack socket on the rear panel (+4 dBu) the signal being returned via the stereo effects returns (again +4 dBu). Effect Send 3 can also be used as a monitor bus if you wish to use the *DMP* for laying tracks.

If using Effect 3 for an external send, the *DMP7* gives you the option of equalising the return signal, using one of three bands of parametric EQ. The *DMP11*, however, has no external effects return so the external effect signal must be returned on an input channel.

The internal effects are of the same type and quality as the *SPX90* and offer much the same facilities. The effects are more than usable, the reverb algorithms being particularly versatile. As the effects section is integral to the unit, there are no cables to pick up extraneous noise (or to trip over) and no extra rack space is required for effects. On-board effects can be limiting having access to only one external processor, but I have found the effects flexible enough to cover 90% of situations and tend only to use external effects for specific applications (mostly compressors and gates when recording to tape).

The channel functions are very much as you might expect — a pan control, solo button (post-fader), channel mute and channel fader. The channel faders of the *DMP7* are motorised

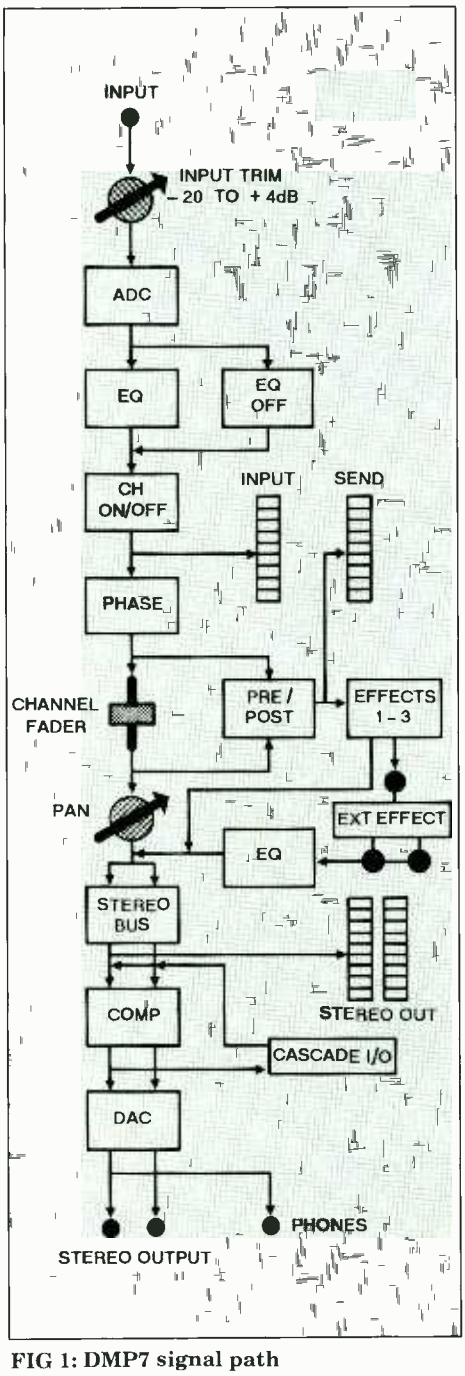

FIG 1: DMP7 signal path

however, and to those used to other motorised fader systems (for example Neve's *Flying Faders*) the response of the *DMP* faders will seem slow and jerky but you have to bear in mind that the faders are acting purely as visual indicators — the channel level itself is under software control and the actual audio response is fine. When setting levels the faders have a very heavy feel but are sensitive enough for most applications.

An autofade time for the channel faders to travel their entire length can be set. Fade times can be set from 0.1 to 10 seconds and are invaluable for smooth fade ins/outs.

The *DMP7* has a compression feature on the stereo output, just prior to D/A conversion. The compression ratio can be set from 0% to 100% and can be turned on or off. I have not found this to be a particularly useful feature, as I usually only use compression when recording to tape, but it can come in handy as a peak limiter when mixing particularly dynamic material.

The rear panel sports a socket for a foot controller, which can be used to control the stereo output level, or alternatively, assigned to control any parameter in the unit. This can be extremely effective, allowing 'realtime expression control' (according to the manual) of, for instance, reverb time, or centre frequency of EQ, without using up one of those valuable hands.

Memory functions

Once a mix is set up, it can be saved to internal memory or RAM cartridge. Set-ups are referred to as scenes and saving is a very simple procedure, as is recalling a mix. There is a memory protect feature to stop accidental overwriting, which, unlike most other devices, does not default to On when powering up so it is advisable to reset the memory protect after each save.

Being able to save entire mixes and recall them with a single button press saves a considerable amount of time — clients do tend to get a little restless when valuable minutes are spent setting up a mixer.

All data can be dumped to external storage over MIDI through System Exclusive Bulk Dumps. In theory, the bulk dumps can be saved to any MIDI sequencer, although I have found that most sequencers have problems accepting the *DMP* bulk dump data, as the files are very large. For the *DMP7*, one memory location contains over 200 bytes of data, and there are 30 memory locations, giving a total file size of over 6 kbytes. The *DMP11* has 96 memory locations and its data is transmitted as 500+ bytes per block, giving a file size of over 48 kbytes. You can, however, dump single memory locations but this requires the sending of a System Exclusive Dump Request

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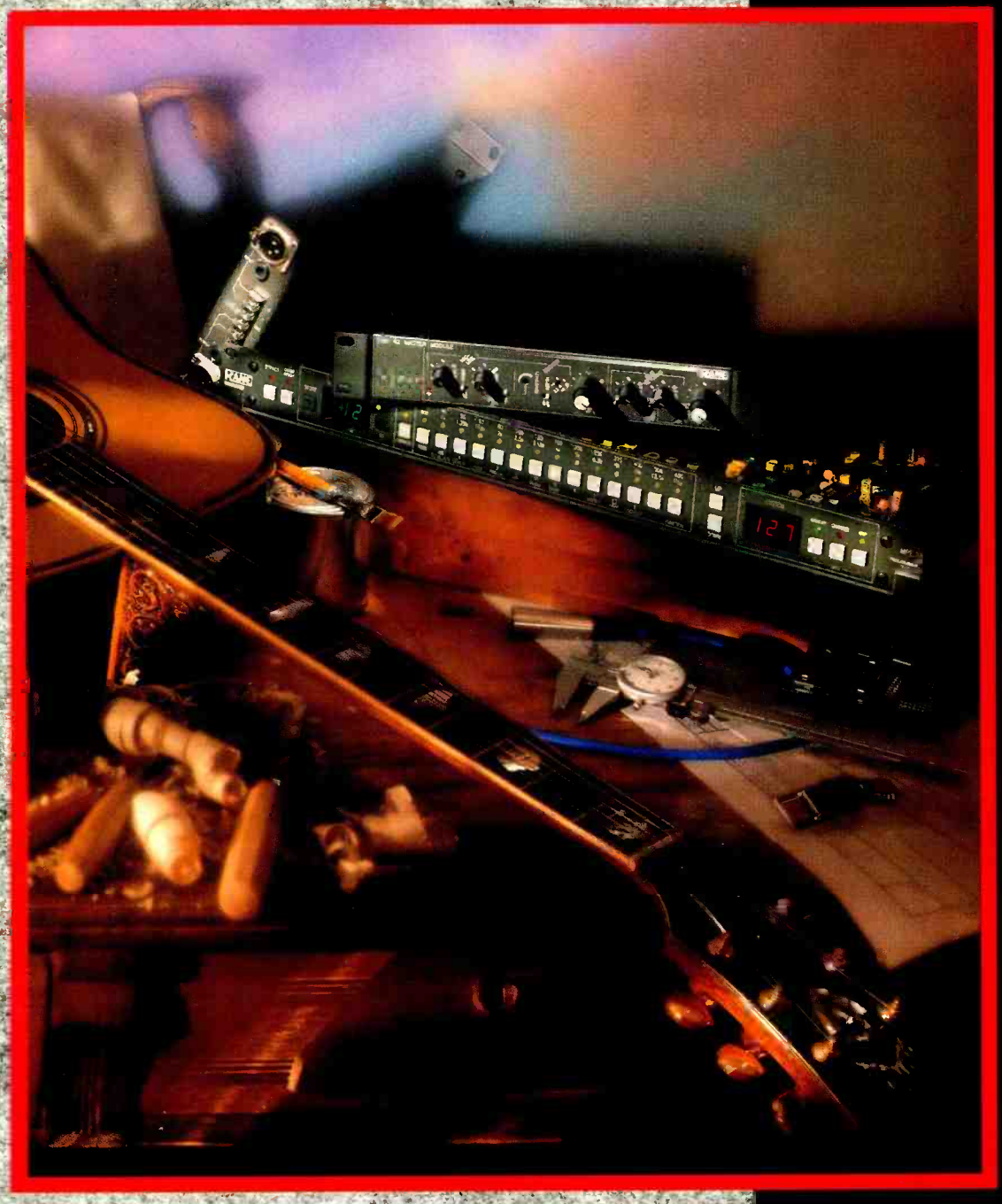

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message, which may present problems if you are not familiar with building SystEx messages.

MIDI control

This is the area where the *DMP*s really come into their own. In its simplest form, MIDI control can be a simple case of selecting memory locations using the Program Change command, either from a keyboard or a sequencer. This can be used in performances to select different keyboard mixes during the set by simply selecting the relevant Program Change on the master keyboard. Program Change numbers are assignable to enable set-ups for a song to correspond with voice selections.

In a studio session, the mix for various parts of a song can be stored to memory locations and recalled by the sequencer at the relevant points. A mixer set-up for 'the chorus' will usually be consistent, so once saved to a memory location, 'the chorus' set-up can quite easily be recalled at the beginning of each 'chorus'.

The Program Change aspect of MIDI control is, however, only of limited use as, much of the time, the engineer is more concerned with changing levels, etc, on-the-fly, and switching between memory locations only selects static scenes. This

brings us to the realtime aspects of MIDI control.

Every parameter within the *DMP*s is assigned a parameter number, and each parameter number can be assigned to respond to a MIDI Note or Controller command via an assignment table. Setting up assignment tables is a laborious process but once done there should be no need to repeat the procedure. I have found that it is better to stick with the default assignments, using the remapping facilities on the sequencer for performance control.

So, how do the *DMP*s use MIDI commands to control their parameters? In the manual the parameter for Channel On/Off for Channel 1 is given as 0. In the default assignment table, parameter 0 is assigned to MIDI Controller number 8. In MIDI terms, there will be three bytes transmitted: the Controller Status byte ($Bn) followed by the first data byte denoting Controller number (in this case $08) followed by the second data byte, carrying the data value (since Channel On/Off can only be in one of two states, the data values will be 0 = Off, 1-127 = On). Using the assignment table (or Controller remapping in the sequencer) Channel On/Off could be re-assigned to MIDI Controller number 64 — the sustain pedal. This allows Channel 1 to be demuted by pressing the sustain pedal attached to the master keyboard, and muted when the pedal is released (using a 'normal On' pedal would operate the other way round and would probably be more appropriate in this particular example).

As you would expect, the *DMP*s will also transmit the relevant MIDI data from the MIDI Out, allowing a realtime mix to be recorded directly into the sequencer from the *DMP* itself. This has numerous advantages in the studio in that any mistakes or timing errors in a recorded mix can easily be rectified on the edit page of the sequencer.

It is only really practical to record the mix one channel at a time but when you consider the advantages of having the sequencer execute a fully automated mixdown, this is not as inconvenient as having to run numerous takes on a conventional console.

One thing to watch out for is that many sequencers will reset the commonly used Controllers to zero when playback is stopped. This usually manifests itself as Effect Return 1 being reset every time you stop the sequence (as Effect Return 1 is assigned to Controller number 1 — Modulation Wheel). This means that the first thing you must record in the sequencer is a Controller command to set the affected parameters to their required value at bar 1.

Remote interface

After the introduction of the *DMP7*, Yamaha found that one of the most heavily criticised aspects was the user interface. People were used to traditional consoles, with rotary pots for most functions, and the ability to visually assess the mixer set-up at a glance. So, the *DMP7*, with its multipage menus and assignable faders, etc, was not being accepted as its operating system was seen as complex and time consuming. In response to this, Yamaha developed the *RTC1*.

The *RTC1* is a remote control unit designed specifically to give *DMP* users a more conventional user interface. The *RTC1* is connected to up to four *DMP* units via their MIDI interfaces (one MIDI Out and four MIDI Ins are the only socketry on the rear panel).

The front panel is divided into five sections: across the top are 12 buttons for selecting the eight channels on the four machines, (Channels 1 to 8 and Banks A to D), selection of a bank and channel deciding which input channel is being processed from the front panel of the unit.

At the foot of the front panel are two sections marked Input and Master, each with a set of buttons and a fader. The Input section contains the Solo button (press Solo, followed by the channel[s] and/or effects return[s] required) the Channel On/Off button, a fader and four buttons for fader assignment. The fader can be assigned to Channel level or Effect Sends 1 to 3. The Master section also has four assignment buttons — Effect Returns 1 to 3 and Stereo Output, which assign functions to the Master fader and Master On/Off button. When using several *DMP*s cascaded together, the Master fader controls the stereo output level of the master machine but has no effect on the slaves.

Above the Input Channel is the equalisation section, which offers rotary controls for the bandwidth, centre frequency and gain for each EQ band, with buttons for Phase reversal, EQ On/Off and Peak/Shelf selection for the high and low bands. There is also a rotary pan control.

All the faders and rotary controls have null indicators, so when a new channel is selected for processing, the operator can set all the controls to their previous positions, to enable visual confirmation. This may sound a long-winded way of working but once you are used to the idea, it

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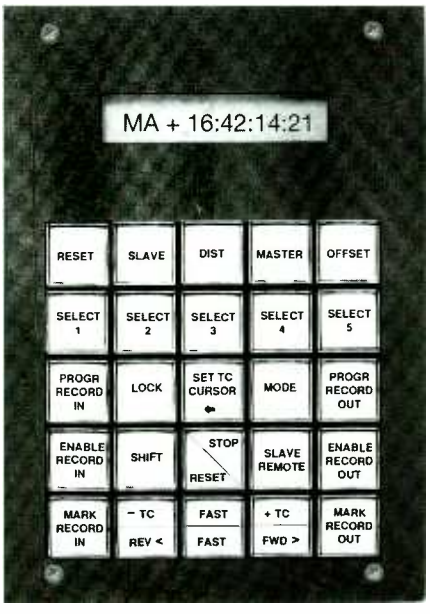

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takes no time at all to set all controls to their null points, in fact, only the controls on the particular functions being used need be set.

To the right of the equalisation is the utility section, which duplicates all the utility functions of the *DMP* itself, and the controls are laid out in a very similar way. There are a few extra functions. Four rotary controls marked Control 1 to 4 (again with null indicators) are assignable to virtually any function you want. In conjunction with the Fader Grouping button, each control can be used to gang together several fader functions. Thus, it is possible to group together Channel levels, to perform multichannel fades from one control. These controls can also be used to access all the effects parameters so that reverb time, etc, can be set from the *RTC1*. The *RTC1* will also allow you to gang together two channels of the *DMP* so they will act as a stereo input channel, duplicating EQ settings, etc, between the two channels.

When setting levels, it is often necessary to be able to confirm settings for all channels visually and the *RTC1* allows this by displaying fader levels on its LCD. It displays all eight channel or effects send levels for the selected machine (plus effect returns and master levels), or up to 32 channels for when you are using more than one *DMP*, (channel levels only). I have found this to be a vital feature, despite the fact that the display is not very high resolution.

The display also shows the actual data for equalisation, etc, which enables you to set values numerically if required.

When communicating with the *DMP*s, the *RTC1* uses the default MIDI command assignments to control the various parameters. When I first received the *RTC1* and *DMP7/11* system which I currently use, the *DMP* assignment tables had been set up for some other use, and I could not get the devices to work together. At first, I was resigned to having to set up the assignment table of the *DMP7* for all 205 parameters one at a time, until I discovered that the default assignments can be recovered by holding down the Fader Flip and MIDI buttons while powering up the *DMP* — thus saving about two hours work. A note of caution, however, because other power-on key combinations can cause the software to crash, so this is one of those rare occasions when it is not at all wise to experiment.

One important point to remember is that when turning on the *RTC1* it is necessary to press the button marked Request. This is so that the *RTC1* knows which machines are connected to which MIDI Ins. When the button is pressed, all displays will shut off for a few seconds and when they return the main display of the *RTC1* will show a table of which machines are connected to show that all is OK.

The *RTC1* will also call up the relevant memory locations on the connected mixers by sending a program change command. This is operated by simply selecting the relevant program number on the *RTC1* and then pressing Recall.

Several of the *RTC1* display pages use abbreviations to denote the *DMP* parameters, which can be very off-putting, especially if you are not used to the machines. To counter this, there is a button marked Help, which will show the title of the highlighted parameter in full.

The *RTC1* makes the *DMP* series a lot more user-friendly and, more importantly, much faster to work with. The analogue-style control of equalisation, coupled with the equalisation itself, gives you a very powerful and intuitive command over the whole audio spectrum. Although there are still several multipage menus, the main mixing functions are far more accessible and the *RTC1* makes it possible to work on several parameters at once (invaluable, since all aspects of a mix are interdependent).

One of the greatest advantages is that the *DMP* units themselves can be mounted in a rack, with the *RTC1* sitting on the desk alongside the tape machine remote and computer, allowing more free space for all the self-replicating pieces of paper that build up when working on a piece.

Using digital mastering

When using the *DMP*s with a digital mastering system (many studios are now mastering to R-DAT) it seems a shame to convert the *DMP* signal to analogue, only to convert it back to digital at the master machine input.

Using the *FMC1*, it is possible to take the digital signal from the *DMP*'s cascade output, which is in Yamaha's proprietary digital interface format, and convert it to either AES/EBU, CD/DAT or SDIF formats. This allows digital connection of the *DMP* output to the digital inputs of the mastering machine, by-passing the D/A/D conversions.

The *FMC1* also allows the user to select 44.1 kHz or 48 kHz sampling rates, to ensure complete compatibility. If using the 48 kHz sample rate, the equalisation frequencies of the *DMP*s will be shifted up by about 10%, and the delay settings of the effects will be slightly shorter than normal.

Using the *FMC1* in conjunction with a DAT recorder, I have found that the quality of my masters has improved dramatically (especially as I use the sequencer to mute all channels with no signal passing through). This is not really surprising, as once the signal is converted at the *DMP*'s input, it remains in the digital domain right through to the master. This system gives you a true 'WYHIWYG' (What You Hear Is What You Get) recording, which is, after all, what engineers/producers have always been trying to achieve.

Conclusion

The *DMP* mixers and their associated products have been around for several years and have not yet achieved the status within the professional recording and production industry I feel they deserve.

If you are prepared to adopt a different approach to mixing, you can benefit enormously from what the *DMP*s have to offer. They are powerful tools and can enhance the quality of your audio production. Used with the *FMC1*, they offer the only digital mixing/mastering system currently available for less than an arm and a leg.

If you are already using MIDI sequencing for your audio production, then the *DMP*s will make the job of mixing the material an extension of the creative process and you should not have too many problems adapting to them.

For my own work, the *DMP*s have given me exactly what I required of the mixing aspect of my system — portability, flexibility, transparency and quality. They have been instrumental in my adoption of a 'desktop' approach to music production — something which, only a couple of years ago, was totally against my musical principles. The *DMP*s do have one or two minor foibles but when you consider the many advantages the system offers, they seem almost too good to be true.

Barry Fox

Blumlein biography; Churchill tapes' authenticity

How much do the people who edit some learned journals for the sound and picture industries really know about their industry and its history? The answer is: precious little, I fear.

Alan Blumlein invented stereo, made the first stereo recordings on disc and film in the mid '30s, fathered all-electronic television and died developing the airborne radar system that helped win World War II. The *Journal of the Audio Engineering Society* (June 1991); *Television*, the journal of the Royal Television Society (June 1991); *Image Technology*, the journal of the British Kinematograph, Sound and Television Society (July); and *Nature*, the science journal that published the first proposals for television back in 1908 (May 23rd, 1991) have all carried a call from Francis Thomson of Watford for biographical material on Alan Blumlein. By the time you read this, there will doubtless be more appeals to add to the list.

Presumably the editors of these journals are unaware of the background to all this, because if they *were* aware, they surely would never have published Thomson's appeal without editorial comment.

With so many members of the AES in the UK knowing the background to Thomson's promised biography, and distressed by what has happened, or more accurately not happened, it is particularly depressing to see the AES journal actively contributing to the problem.

The AES appeal refers to 'a revised biography'. Well, although Francis Thomson has been collecting biographical material on Blumlein since 1972, there is still no trace of anything he has ever published on him (other than calls for material) and he has ducked every request that he identify anything he has written. Thomson refuses third parties access to his collection and will not catalogue what he holds, even to Blumlein's family.

To the unwary editor, Thomson's letters carry credibility because he is able to describe himself as a Member of the Institution of Electrical Engineers. The IEE knows the score and fears that the 50th anniversary of Blumlein's death in June 1992 will pass unmarked by the long-awaited biography. Last year the IEE asked Gordon Bussey, author of meticulously researched books on the history of radio and television, if he would help Thomson write the Blumlein biography. Bussey met with Thomson and the IEE's publisher in November but Thomson later refused to co-operate.

"As a historian, the idea of monopolising biographical material is anathema to me," says Bussey. "It will be a tragedy for the nation if this biography is permanently blocked."

Blumlein's son Simon despairs. He is now resigned to the fact that the anniversary will pass without the publication of the biography for which he and his mother agreed Thomson could collect material. His mother died without ever seeing any tangible results of her trust in Thomson. Simon worries, as do many others, about what will happen to the collected material in the long term. This fear deters those who could bring peer pressure to bear on Thomson from doing so.

Although the IEE expresses 'concern' and has tried to organise the joint venture between Bussey and Thomson, it has so far exercised no professional sanctions.

And this leaves Thomson free to continue collecting material, and not just on Blumlein, either.

In 1979 Thomson wrote to one of the inventors of the heart and lung machine, saying that he expected soon to finish his biography on Blumlein and would then like to have a go at writing their life story. Perhaps wisely, they did not bite.

Thomson then appealed (again as Blumlein's biographer) in the January 1980 edition of *Wireless World* magazine (now *Electronics World and Wireless World*) for 'papers, notes, photographs, etc' from anyone who knew S G Brown. Sidney George Brown, who died not long after Blumlein, had patents on ground-breaking technology such as the gyrocompass and audio headphones. There is no sign of a biography on Brown, either.

The Spring 1990 newsletter of the Special Forces Club carried an appeal from Thomson for memories, photographs and names and addresses of anyone who was at Gorhambury House, St Albans, during the war.

"I have been asked to compile an exhaustive record of SOE and War Office use of this stately home," he assured SFC members. As with his 1972 and 1991 calls for material on Blumlein, Thomson again created a sense of urgency with a tight deadline for contributions.

Returning to the Blumlein theme, Thomson's latest appeals variously explain the years of delay with his talk of a 'revised biography' and a puzzling tale of his attempts to 'shield the Blumleins from the belated discovery that a person close to them had supplied me with letters . . . stolen from the late Mr A K Van C Warrington'.

In a tragic vicious circle, few people know enough about Alan Blumlein to realise the significance of the missing biography. And while there is no biography only a few people will even have heard of the man and understand the significance.

Some more notes on the vexed question of whether some of the most famous recordings of Winston Churchill were in fact made by Norman Shelley.

Or is it vexed? When an American TV programme wanted to investigate the question, the Churchill family refused to co-operate. Before writing any articles I asked EMI Records several times for comment on the pedigree of what the company is selling under Churchill's name, and what is soon to be released on CD. Back in the '80s I had asked Decca. EMI just ignored my questions. Decca had brushed them off.

The BBC is obviously sick of the whole business. The Archives acknowledge that the controversy has been running since 1979 and reconfirms that Decca admitted to wrongly labelling their cassettes (by suggesting that they were recordings of BBC broadcasts). But the BBC refuse to release copies of their authenticated recordings for computer analysis.

After a piece of mine was published in the UK's *Guardian* newspaper, one reader wrote to say he remembers the authenticity of some of Churchill's broadcasts being questioned as early as 1946.

Another suggested that the BBC check back on fees paid to Shelley at the time for recording soundalike speeches.

But it seems unlikely that there would be any money for digging through 50 year old accounts.

A broadcast engineer at LBC recalls how, back in the early '70s an older colleague pointed out a studio in Bond Street as "where Norman Shelley recorded Churchill's speeches".

Another *Guardian* reader has more concrete evidence. He tells how, around 1948, Churchill approached Oscar Preuss, the Artists Manager of Parlophone (part of EMI). Preuss suggested that they be made on EMI's new tape recorder. The writer, who had been working at Abbey Road on the tape deck, spent three months at Chartwell. During that period Sir Winston recorded all the war speeches published in his book of war speeches and some extracts from his war memoirs. On completion these became the property of Sir Winston. Decca presumably got their material from Churchill.

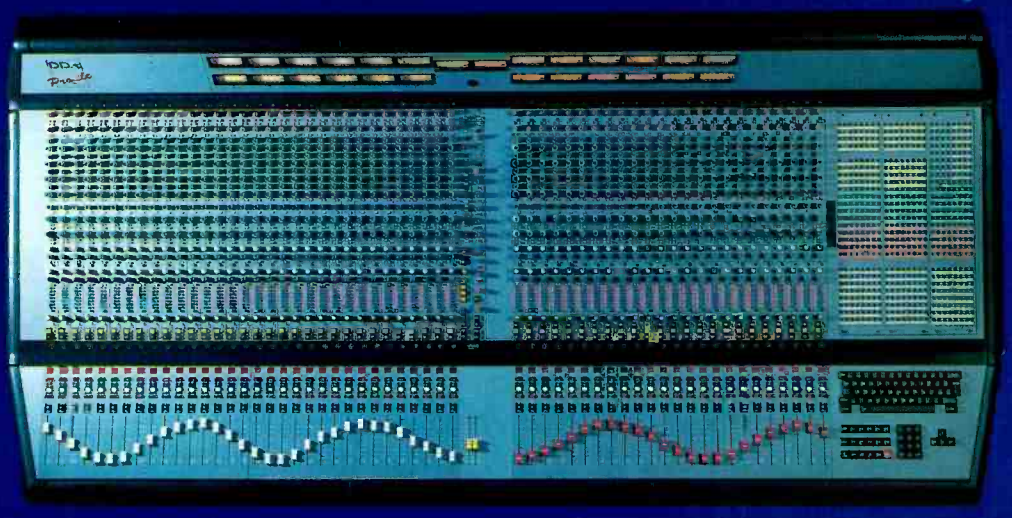
Yet another reader has studied Harold Nicholson's diaries for the year 1940. To cut a long story short, they show that the speeches in doubt were recorded after the event, whether by Shelley or Churchill, with the BBC and Decca/EMI having different versions.

When I failed to get any response from EMI Records I forced the issue by lodging a formal complaint with the Hillingdon Trading Standards Officer (the TSO with responsibility for EMI's record operation at Hayes). The TSO subsequently told me that EMI's legal department had become involved and would be providing a full response.

I heard nothing from EMI's legal department but I did hear from the Hillingdon TSO who reported that EMI Records have "taken expert opinion and concluded that the recordings are not misleading the public". The TSO refers to "full documentation" but for reasons of confidentiality is, of course, not free to copy it to me.

The fact that a Trading Standards Officer does not spend public money on a prosecution cannot, of course, be taken as proof of the pedigree of the recordings but it does suggest that, better late than never, EMI has adduced some useful evidence to rebut the evidence I published.

I have asked EMI for a copy of the same material. I will pass it on when I get it. Even if the record companies have not until now thought the matter worth considering, there are a lot of people out there who would like to see this piece of audio history set straight.



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Martin Polon

I Did It All With My Gold Tipped Thing-A-Ma-Bob. Comment from our US columnist who asks can we really improve audio with more tweaks?

The new studio was magnificent. Not one penny, not one 'bob' had been spared. Alan prided himself on that. He had inherited $13 million. He knew that Grandma Ray would have wanted him to build the biggest and best recording studio in town. Who said Iowa could not be big league. He would put them on the map with something besides 'prairie oysters'. The very thought of it and the not-so-delicate aroma from the nearby meat packing plant made him quiver. He hated the term 'meat packing plant'. How much better to call it an 'abattoir'. The French certainly knew how to give things style, he considered.

As to the studio, well this was to be Alan's finest hour. If he had heard about it . . . he bought it or did it. He read all the magazines that had anything to do with audio. Not just the professional ones but the home hi-fi ones and the ones for the 'tweako' trade. He loved that word. He had built the world's first tweako recording studio. Not that he always believed exactly in everything but it was kind of like a talisman. That's it, he thought. I'm buying good spirits. Good vibes. Alan after all was rather suspicious. Or make that superstitious or make that both and all of the above.

His chief engineer, Alice, was a sensible woman. She had apprenticed with one of the TV networks, fresh out of music-recording school. She had studied at night for a Bachelor's degree in engineering while raising a child. She had paid her dues at the TV empire and now was able to do things as she saw them. All in all, a formidable choice Alan thought. He and Alice sat down for yet another of their facility planning meetings.

"We are just about ready to go on the air?" he asked.

"The only thing yet to be done in Studio A is terminating the gold tipped oxygen-reduced virgin copper 14 gauge microphone cable. The special connectors with the sputtered gold pins have not come in yet. The vendor indicated that we could have platinum pins to NASA standards, as used on the Space Shuttle. They are a stock item at considerably less cost. But I told him no — you had said gold and gold it will be."

She sighed and continued: "The concrete pedestals have really worked out very well. The digital tape transports mount perfectly and the spring-hung floating mounts work like a charm."

Alan queried, "The bugs are finally out of the hydraulic absorber mounts under the console?"

Alice replied, "The trick with that is to adjust the hydraulic supply very carefully. The viscous oil dampers and oil reservoir are very sensitive to system commands. I thought that moving the pumps to a central machine room would have played havoc with the system but in fact it works better for the distance between the units. The engineer for the hydraulics company said it had something to do with the equalisation of pressures by the lines. And we were right to use a separate oil supply for the absorber bank underneath the studio floor. Remember, in Berlin, these babies are used to isolate the convention centre complex from nearby traffic noise. They have these huge springs that carry the floor. The hydraulics simply achieve balance with the load."

Alan pondered for a moment. "And the machine room concept really worked? I must admit, I am rather proud of that idea!"

Alice's spunk was aroused by that. She echoed, "Yes, it works. I admit I hated the idea. In fact, I still hate the idea. But in actual usage with all the things we have under our roof, it really does work. The architects loved it, since it made this just like an office building project. A central core with studios all around like a hub and its spokes. In addition to all the hydraulics, the power supplies for all the electrostatic studio and control room speakers work perfectly with the distance."

Alan responded, "Yes, I was concerned about that long wire run for the electrostatics."

Alice continued, "The 14 gauge *Teflon* covered shielded cable solved the linkage problems. And with the refrigeration units in the machine room, putting the power amplifiers next to their cooling source was very sensible. It was difficult to route the No 2 gauge speaker feeds but we finally found a supplier for the raceways the power company uses in their substations. The only real problem we have had is with the super cooled tape vault."

Alan looked puzzled.

Alice retorted, "It's the nincompoop factor. The staff go in for tape and forget to put on their parkas and face masks. They get frostbite!"

Alan asked, "When do we finish the repairs on the building's Faraday cage and lead foil shield?...

Well, fellow rock and rollers . . . there you have it. The story of the world's first Ultra Tweak, Yuppie, New Age, Holistic, Ying and Yang, Natural Balance recording studio. But let's keep those mellow fluids in check, fellow occupants of this planet. Just remember you read about it first . . . here. However, lest you think I am being just the teeniest bit critical of such incredible tweakdom as described above in this month's fairy tale, perish the thought. I have an honest curiosity about anything that can make better recordings and better reproduction of music and even if something doesn't work for sure on a scientific level, it may well enhance perception on other levels.

Seriously, I have a theory about all the enhancements that people contemplate or actually use for recording and reproducing the tuneful performances of the muse. Basically, that is to throw money at problems is to solve them. Since everything we do in life is in one way or another a compromise and financial factors usually are the most compelling, we rarely, if ever, achieve an optimum condition. That holds especially true for audio, where cost factors as well as profit factors mediate against something being done to perfection. Many of the so-called esoteric or 'tweak' enhancements of the recording and/or playback environment really are just a case of doing something in absolutely the best way possible. This is usually unaffordable or unobtainable for most who settle for something a little bit or a lot less.

Perhaps one reason I exhibit a significant tolerance for the so-called idiosyncrasy of the high end audio set is that as a teenager in Los Angeles, I built my audio system the same way they do today. I had the good fortune as a teenager to hold down three jobs at the same time while going to high school. One of my jobs was with a military surplus electronics company. I spent 20 hours per week reconditioning automatic telephone switchboards from US Navy destroyers so they could be sold to pot bellied, cigar smoking dictators in Third World countries. I chose not to be paid in the coin of the US but rather in the coin of the realm; in that case, surplus electronics.

At that point in time, there was a virtual cascade of high quality electronics being sold off by the US Government from surplus stocks accumulated during the Second World War and, more especially, from the Korean War, and including state-of-the-art from area contractors in aerospace and military projects. I had the pick of the litter, one could say, of the finest products made in the audio and electronics industry. My sound system was housed in a steel electronics rack made by Western Electric. The power to the rack was fed through a Triad autoformer/filter that kept the voltage constant and the input power pure. All switching was done via gold contacts on premium grade Western Electric patchbays with Western Electric double plus patchcords. (For the true cognoscenti, the name Western Electric was like the name 'Sterling' on silver; the manufacturing and supply arm of the Bell Telephone System, backed by the resources of the Bell Telephone Labs.)

My 40 amplifiers were built by Altec Lansing to Government specifications but I went ahead and rebuilt them. I used 1% plastic resistors and metal and glass sealed capacitors filled with special insulating oil (PCBs, I think and that's why I have 12 toes but that's another story). All the valves (tubes) were specially selected military equivalents with special bases and enhanced performance over similar civilian valves. I built 16 speaker systems using 1 inch board and Altec Lansing 8 inch bi-cone drivers and fed amplifier energy via 12 gauge stranded cable with Teflon jacketing. I used a Grado cartridge that did not come from surplus and a Rek-O-Kut turntable that did. My turntable lived in a massive sliding contraption in the rack. I could go on and on about vu meters and record cue amplifiers and so on and so forth. I never had it so good and in fact I surely have not

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been able to afford such indulgences since — either in time or in money. And if you must know the truth, I'm really not sure that I am having half as much fun with audio!

The point of all of this is not a roam down memory lane but the fact that each time I changed something for the better, I felt I could hear a difference. In point of fact, I probably could. Superior components do influence sound quality. Let us examine some of these areas of technical controversy in audio today.

Wiring: One can barely contest the fact that the larger the wiring gauge you use, the lower the loss. Many argue that specific kinds of wiring in terms of alloys of copper and precious metals, or the process of cable winding and interleaving can affect frequency response in a positive way. It is suggested that these improvements can influence microphone connection as well as speaker connection. They can surely impact AC power feeds to professional or other large audio systems. I can think of no good reason not to use the best and the largest cable or wiring in an application that the finances and logistics will tolerate.

There is a small coterie in audio that feels small is beautiful and too much wiring is overkill. To these good fellows and dames, I say this: using wiring and cabling smaller than what is allowable is like boar hunting with a 22 calibre pistol. Yes, it is challenging and there is the thrill of the sport, but I prefer not to sing soprano and find the Smith and Wesson 44 magnum a more suitable tool for the job. So it is with the right wire.

Electronics and amplifiers: It seems very logical to me and many others to use amplifiers that have separate power supplies for each channel or massive power supplies for both. Torroidal power transformers, oversized capacitors, under-duty-cycled rectifiers and hefty regulators all bring a smile to my face. I love to see oversized output transformers on line level devices and on amplifiers feeding high voltage speaker systems. I still size up amplifier quality by performing the world famous 'hernia' test. An amplifier that can cause significant medical damage to your body via lifting is an amplifier well worth knowing. Premium components under the hood as it were, will yield superior performance and long life. When you buy a watch, do you want it to say made in Taiwan or made in Switzerland?

Of course, analogue vacuum valves have returned with a flourish for those who desire the 'warmth' of subtle distortions over the 'by-the-numbers' precision of digital electronics. Unfortunately, the price tags for many of these babies nearly match the national debt. To some extent, that is unavoidable since vacuum tubes have achieved a rarity equalled only by baseball cards. Do valve units sound better or warmer than digital electronics. I am not sure anyone is qualified to make that interpretation for someone else, but isn't that what the audio business is supposed to be all about?

Precious metals and connections: It appears that the good folks at the UK Ministry of Defence (MOD) and the US Department of Defense (DOD) have spent literally tens of millions of taxpayers' pounds/dollars verifying the superiority of precious metals for the purpose of interconnecting electronics. It would appear that all the research has to be the final word on the subject.

Mountings and bases: Since we are dealing with acoustical environments and any coupling of signal between components is totally undesirable, as are vibrations from speaker systems coupling to their environment, how can there be any question of using the most robust bases and mounts possible? The only downside is that the prestigious *Ladies' Home Journal* has yet to do an article on 'Decorating your living spaces with concrete'. To be blunt, the use of concrete bases, enclosures or mounts rarely scores points with one's spouse, children, landlord or downstairs neighbour. It makes sense to use the sturdiest materials you can afford and that you can get away with. And, oh

yes, despite the mathematical abstractions of some speaker designers, 1 inch thick virgin plyboard speaker enclosures make reproduced music very nice indeed!

Speaker systems: A basic rule of thumb is buy the largest components you can possibly afford and house them in the biggest box you can find and tolerate in your working and/or living spaces. There is much to be said for smaller systems and satellites and subwoofers but the bottom line is that we are talking ultimately about a piston moving air. The larger the piston and the larger the chamber, the more air is going to be moved. If something unique or exotic seems to please you in a speaker design, buy that device by all means. That includes any exotic materials including spun metal cones.

After all, the bottom line here is a very subjective decision based on personal perception. Do we buy studio monitors because another studio uses them or a lot of people like them. If we do, and continue on that path of logic to purchase all our studio gear, there will be nothing to differentiate us from the rest-of-the-pack. That has happened in our industry. There was a time not so long ago that artists recorded in certain facilities because of the chef and kitchen, or the chateau, or the horse riding. Most of the studios had the same or similar equipment and sounded quite the same. Nothing essentially wrong with that but today major facilities sell the acoustical personality of their management and staff through decisions as to the 'sound' of a place. Back to subjective perception again!

Now the proverbial bottom line here is that even if there is no reputable explanation for a hypothesised audio phenomenon or if I am indeed full of that famous peasant dish, Road Apple Pie, or all of the above and then some . . . it still contributes to my second major theory on the subject. That is the very human need to be able to tweak our audio systems. The history of audio reproduction since the Second World War has been full of constant experimentation, innovation and change . . . both in the home and in the studio. It is only recently that the incredible precision of microprocessors, integrated circuits and digital electronics has made audio components so perfect as to defy easily accessed or accepted improvement at the hands of mere mortals.

The whole range of compact disc modifiers is pointed at giving the user some hands-on power again over his or her listening environment. The user/listener cannot change 99% of the electronic and acoustical environment in the studio or the home. Electronics have reached the stage of complexity that only the factory can fix them with test jigs and dedicated computerised diagnostics in many cases. Less than 1% of the known professional or home consumer universe build their own loudspeakers. And so on, and so forth. That is why audio professionals, audio practitioners, audio enthusiasts and just plain listeners use oversized speciality cables and wiring; gold plated connectors; concrete bases and mounts; tube limiters and equalisers and power amplifiers and even microphone amplification; speakers made from aluminium, plastic, recycled glass, etc. That is why CD users stretch green rubber 'condoms' around their discs or freeze their discs cryogenically. That is why a 'cherry' copy of the Burt Bacharach soundtrack score for the movie *Casino Royale* fetches as much as £300 ($500) in the hands of committed audiophiles who consider that LP to possess fidelity 'clean as a whistle'.

What all this is ultimately about is the return of humanity to audio technology. In the little ways that people can use various sundry articles, they can 'humanise' their contact with what has become a very computerised and automated process. Nobody is 'right' and nobody is 'wrong' when it comes to these acoustical, electrical and/or thermal departures from the norm. They are subjective as is our entire industry — what it sells — and the experiences of the end users. So be it!

BSS TCS-804

Sam Wise presents a technical review of the BSS Dual Time Corrector

As audio mentor Don Davis would say: you can't alter time but you can delay a signal. The BSS Dual Time Corrector does just that (delays the signal) but in a cost effective, high performance and useful manner, so perhaps we can forgive the misnomer.

Unlike studio effects delay units, which provide for a multiplicity of applications, the BSS *TCS-804* (and its little sister *TCS-803*) have only one main purpose — to align the signal arrival times from loudspeaker cabinets. The *TCS-804* does this in a way that makes it eminently suitable both to mutually align the individual devices within a single cluster or group (time- or phase-aligning a loudspeaker); and to delay this group (or a complete cabinet) with respect to other sound sources. The *TCS* series is mainly applicable for signal alignment of multiway loudspeaker systems, either in a tightly spaced cluster, or distributed around a building but its remote control and memory capabilities will find it additional applications.

Operational features

The two principal applications of a signal alignment delay are to make small adjustments between the signals within an array of loudspeakers, and to make relatively large adjustments between spaced sound sources. The first use mainly helps to ensure that the final acoustic signal isn't affected by big holes in the response created by cancellation between drivers, the second helps to prevent echoes from delay towers and the like, and enables use of the Haas effect to make the sound image appear to come from the performer rather than the nearby loudspeaker system.

Cluster alignment requires very small steps of delay adjustment. Typical high quality delays use an audio sampling rate of about 50 kHz, giving a time resolution of 20 µs and require a steep roll-off anti-aliasing filter. The filter itself causes a further signal delay at high frequencies and produces ringing from steep input waveforms. The *TCS* series improves on this, using a 100 kHz sampling rate, giving advantages in each of these areas. In addition, the resulting 10 µs resolution enables driver alignment errors as little as 3.5 mm to be compensated — giving potentially smoother very high frequency output from multidriver arrays. These effects are easy to see using

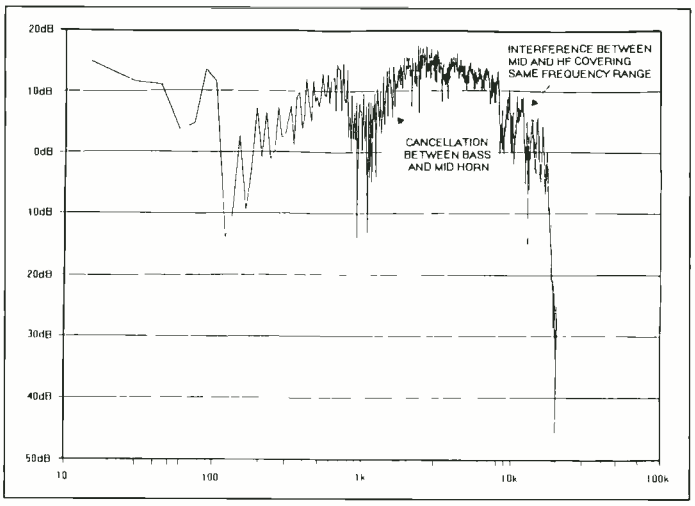

FIG 1: Response of JBL Concert series cabinet with BSS FDS 360 crossover. Major improvements were made as a result of this measurement

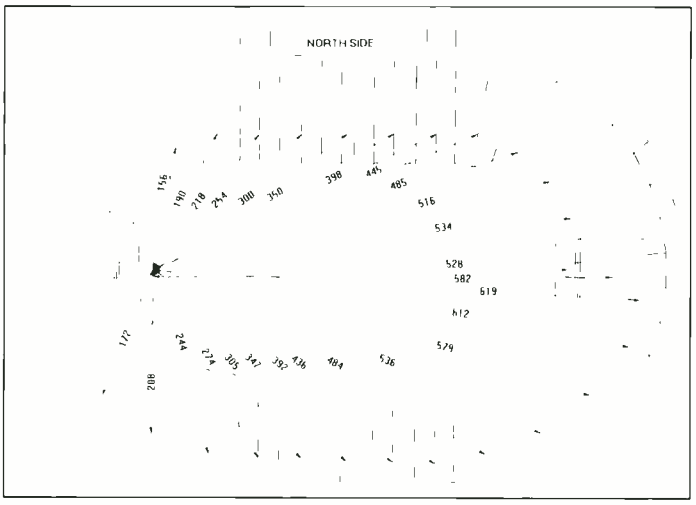

FIG 2: Wembley Stadium—delay times (ms) from west stage to loudspeaker aiming points

Manufacturer's specification

Input section: 10 kΩ electronically balanced
Output section: electronically balanced and floating, capable of driving +20 dBu into 600 Ω

System performance
Conversion method: dynamic floating window PCM
Frequency response: ±0.5 dB, 20 Hz to 20 kHz
Dynamic range: typically 105 dB, no pre-emphasis
Distortion: <0.03%, 20 Hz to 20 kHz at +10 dBu
Group delay deviation: ±7 µs from 20 Hz to 20 kHz
Stereo synchronisation: ±0.5 µs

User memory type: EEPROM — no batteries required

Parameters
Delay range
stereo: 75 µs to 650 ms (1.3 s optional)
mono: 75 µs to 1.3 s (2.6 s optional)
Minimum step size: 10 µs (3.4 mm at 21°C)
Input level: -9 to +20 dBu digitally adjustable in 1 dB steps
Output gain: ±6 dB digitally adjustable in 1 dB steps
Temperature: -19°C to +44°C, manual or automatic entry

Display
Input headroom: 7-step linear showing 0 dB to +20 dB
Parameters: 4-digit plus units/status LEDs

Power: 120/240 VAC, selectable, 50 VA, 50 to 60 Hz
Dimensions: (whd) 19×1¾×9 inches/482×44×228 mm
Weight: 5 kg net
BSS Audio Ltd, Unit 5, Merlin Centre, Acrewood Way, St Albans, Herts AL4 0JY, UK.
USA: BSS Audio, PO Box 1388, Pleasant Valley, NY 12569.

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"Choke" by the Beautiful South, recorded and mixed exclusively on Mike Hedges' DynaudioAcoustics M1s

The prototype of our M2 midfield monitor produced a number 1 album

"Listen Without Prejudice" by George Michael, recorded and mixed almost exclusively on Chris Porter's DynaudioAcoustics M2/SubWoofer system

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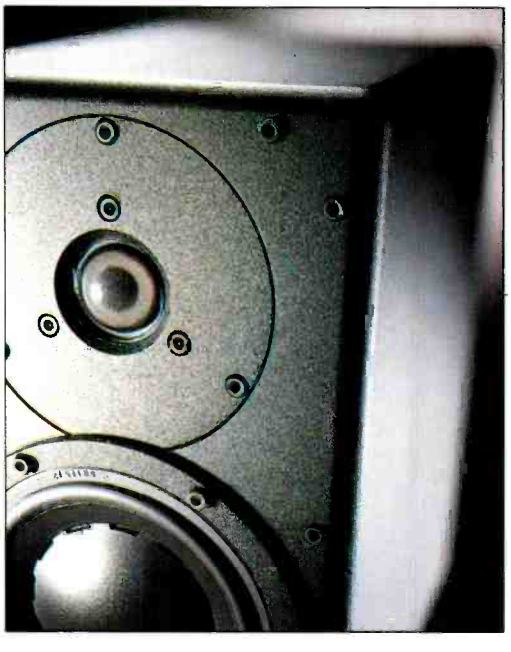
The new M3.3 Main Monitor



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measuring instruments, as shown in **Fig 1**, but can become less relevant when many drivers are in use. Then, small changes in listening position can give large high frequency interference variations that cannot be compensated for. When engineering sound reinforcement systems, the signal delay is a very useful tool but only when combined with careful selection and aiming of the loudspeakers themselves. The sampling rate of the *TCS* series requires twice as much RAM to provide the same delay time as a more conventional unit.

Delay tower alignment — delaying remote loudspeaker signals with respect to a stage or main loudspeaker array — can require long delay times. **Fig 2** is a plan view of Wembley Stadium — a famous English sports and concert venue. The drawing shows the delays required at various terrace audience positions with respect to the normal end stage position. At the opposite end of the stadium, the overhead loudspeakers require a delay of at least 619 ms. The standard *TCS-804* has a maximum delay in 2-channel mode of 650 ms, conveniently fulfilling the requirements of most large venues such as Wembley. In mono mode, 1.3 s is available. These delay figures can be doubled with the addition of a memory module.

Often, both types of alignment are required. A localised collection of loudspeakers need aligning with respect to each other, then the whole must be delayed with respect to a main sound source. Traditional delays have made this operation somewhat difficult. Each delay output is first adjusted to get the intra-cluster alignment right. Then, to delay the whole local system with respect to other systems, further delay must be added identically in each output. The *TCS-804* provides a tracking mode, which maintains the delay times used for alignment within the cluster automatically adding the same offset to each output to move the whole cluster together.

Variation with temperature

Another problem with the delay of audio signals is caused by the physical fact that the speed of sound varies substantially with temperature and less with humidity. If the speed varies, then the time of arrival varies. In a large stadium, this can easily take a local delay tower from apparently inaudible (but nevertheless providing the main local sound energy) to the cause of a severe echo. Take for example the variation in sound speed caused by a temperature change from 15°C to 35°C. This variation could easily occur between a soundcheck in the morning and the middle of a concert mid-afternoon with thousands of perspiring bodies present. The result is a 3.4% increase in the speed of sound or a 3.4% decrease in the delay. With a long delay of 600 ms such as at the rear of Wembley, the difference in arrival time is 20 ms. This would be unlikely to produce an echo but the previously (apparently) inaudible local cluster now becomes the main source of attention, distracting the audience. Where sound is travelling over even longer distances, as at pop festivals, the problems only get worse.

The *TCS-804* series provides a way to automatically adjust the delay in line with temperature variations. Each unit is provided with a temperature probe input connector and, of course, BSS provide the probe as an accessory. Then internal software allows the delay to track temperature variations — keeping everything in alignment. One probe can serve a number of *TCS* units by use of MIDI or FSK remote control interconnections.

Long distance

One common element in major rock and roll shows is the sheer scale of operations. Take Wembley again for example. It is a 200 metre walk from stage to the front of the opposite terrace, and that is if you can go direct. Through the stands it is nearer 500 metres. Imagine you have a delay tower, or in Wembley's case a built-in delay system, located there. System wiring requirements dictate that the amplifier racks must be near the loudspeakers. If a delay is used to align the units within this distant cluster then it is most convenient that it is housed with the amplifiers. For one thing, the delay usually requires less input cables than output cables, so wiring is minimised by locating the delay at the rack. Therefore, this unit, which must be set up differently for each venue, requires a long walk to adjust. Or maybe you decide to locate it near the mixing console for operational reasons. But to adjust the distant loudspeakers it is still necessary to go to the far side of them to listen. A series of long walks, or two people and a radio link are required for the setting up process.

As an electro-acoustic consultant, mainly working in indoor performance spaces, I have plenty of opportunity for long hikes up and down stairs to tune the sound systems in buildings. The amplifier racks are often in the roof near to the loudspeakers but most of the audience is in the stalls or circle and that is where the testing and listening take place. Usually the lift isn't working and the tuning work is done late at night to avoid the noise of contractors working to meet a deadline. This is also the time when my mobile radios decide to pack up. Remote controlled delays, equalisers and amplifier gains are just what I need to concentrate on the real job instead of frustration and tired legs.

Again, BSS have an answer. The *TCS-804* provides a remote interface, which can optionally be MIDI, PA422 or FSK. This allows a single daisy-chained cable between units to be used for remote delay adjustment. It is possible to control all aspects of the unit from your own software but BSS supply their own remote control unit with some nice features of its own. This remote can even be used to control the delays by transmitting via a radio microphone, giving wireless operation.

Stereo tracking

The *TCS-804* can be used as a true stereo device, with automatic tracking between the two halves. In this mode, there are two delay outputs for each of the two stereo inputs. Should more delay outputs be needed in stereo, then two complete units can be set up to track each other. All these features make adjustment easier and more intuitive.

Adjustment units

As becomes obvious above, there is a direct relationship between distance and delay time. Most alignment delay units provide adjustment only in µs. BSS have recognised that it would be useful to be able to input data in feet, inches or metres and have provided all options. Since the relationship between distance and delay time is affected by temperature, this effect is automatically dealt with, either by manual temperature entry, or automatically by using the optional temperature probe.

Background noise

In indoor performance spaces, particularly in concert halls, background noise is very critical. For example, in the new Symphony Hall at ICC Birmingham, the specification is essentially that noise should be below the threshold of hearing, and it very nearly is. Large amounts of money have been spent to achieve this, including placing rubber dampers into the nearby railway track construction.

Presently in that hall, one of the noisiest items is the sound system, largely due to amplifiers that do not meet the manufacturer's electrical noise specification having been installed. In that installation, there are no signal delays. Had there been, the signal to noise ratio would have been further degraded by 4 to 6 dB, since most delays have a total dynamic range of about 90 to 92 dB, compared to a typical power amplifier dynamic range of 105 to 110 dB. Here again, BSS have produced an answer, providing a delay with a dynamic range of 105 dB.

Not only that, but the design allows the user to set the internal headroom (or clipping point) to match the rest of the system. The gain control only adjusts the input to output *through gain*, leaving the internal drive level fixed. Any later changes in the 'headroom' also maintain the system gain settings. As with delay, the gain can either be adjusted in each output separately, or ganged for stereo use.

Operational controls

The *804* has four delay taps, which enable configurations of: true stereo linked operation with two taps per channel, a dual mono 1/2 format, or a 1/4 format. Selection of DUAL or MONO modes is by a rear panel mounted pushbutton. Parameters are adjusted by a large control knob, which has a useful two-speed action. A parameter is selected for adjustment by a series of seven

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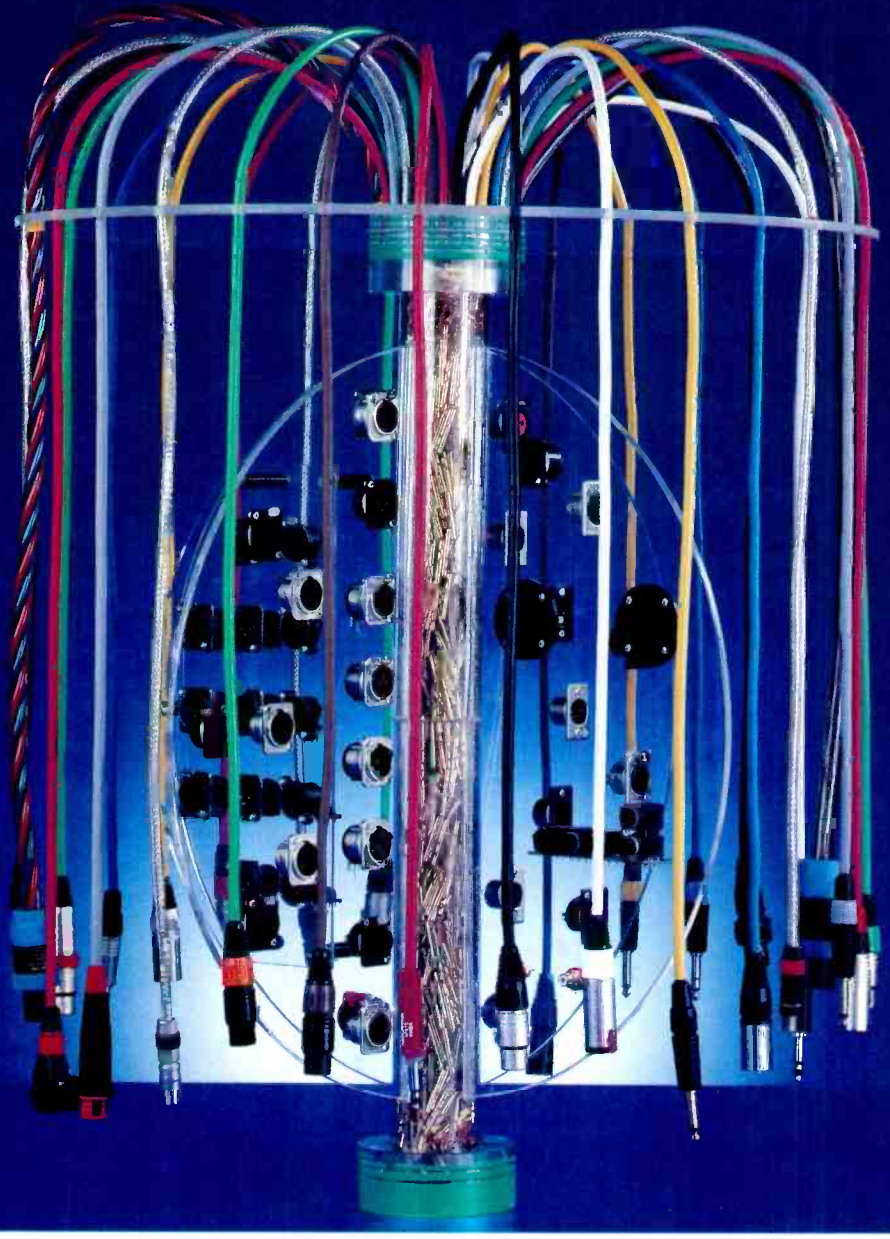

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pushbuttons which run across the front panel of the unit.

When first powered up, the mode of the *TCS-804* is DELAY ADJUSTMENT. Pressing the UNITS button selects the adjustment units between ms, ft, inches and metres. Pressing one of the four DELAY selectors illuminates the corresponding green LED and allows the selected delay output to be adjusted. In DUAL mode, pairs of outputs can be linked for simultaneous and closely matched adjustment.

RELATIVE changes the readout to indicate the relative delay of one output with respect to another. The first press of one of the four DELAY channel buttons selects the reference output (indicated by a red arrow). The second DELAY selects the delay output to be adjusted (indicated by the usual green LED). The numeric display indicates relative adjustment between the two outputs. The following example will help to explain the benefit of the RELATIVE delay adjustment function.

Imagine setting up a delay cluster in a mobile situation. The first step is to create a time-aligned loudspeaker array. Suppose you know the required delay times for the mid and high frequency units relative to the bass driver, so you enter these directly. These might be 0 µs for the HF horn connected to Delay Channel 1, 150 µs for the mid on Channel 2 and 70 µs for the LF on Channel 3. Pressing the RELATIVE button will light a series of flashing red triangles. Pressing DELAY 1 indicates that Delay Channel 1 is the reference. In this case no further button presses are made, indicating that the outputs will be moved together. Changing units to metres with the UNITS switch will allow you to enter the distance from FOH to the delay tower, say 25 metres. Changing back to ms enables you to add a little extra delay to reinforce the Haas effect if you want. Pressing the RELATIVE switch again disengages the relative mode display. The outputs can now be individually selected to verify that the three outputs now read 72.71, 72.86 and 72.78 ms respectively.

Level

BSS have rightfully recognised that when gain controls on delays are adjusted, one of two purposes are in mind — altering the gain through the unit, or changing the effective operating level, and thus headroom, in order to optimise noise performance. In fact, that is exactly what happens when adjusting levels on the *TCS-804*.

Headroom adjustment affects the gain settings of input and output gain controls. One is effectively setting the level of input signal, which will drive the unit to clipping. Digital gain control allows complementary adjustments to be made at the inputs and outputs of the delay to keep the gain constant as the headroom level is altered. Increasing headroom reduces input gain and increases output gain on all associated channels, while a decrease does just the opposite. It is possible, therefore, to adjust all six input and output attenuators together, with no fuss and with all the thought being done by the designer, not the operator.

GAIN on the other hand maintains the input drive level, and therefore headroom, but alters the output drive level to give the required gain through the system.

The function described here is for DUAL mode. In MONO mode, all outputs track the one input. Pressing the front panel button labelled LEVEL puts the unit into headroom adjustment mode for the left channel. The *TCS-804* UNITS indicator changes to dB, the input headroom setting is displayed and the four DELAY selector LEDs flash. Initially, the input headroom adjustment can be made over a range from –9 to +20 dBu in 1 dB steps. Turning the knob clockwise causes the headroom to decrease, since the knob is effectively increasing the gain. Pressing the LEVEL switch again accesses the right channel. A further push couples left and right channel together for simultaneous stereo adjustment.

Associated with the input headroom control is the headroom level display. This reads from 20 to 0 dB in seven steps, going from green, through yellow to red in the process. It indicates the input's signal margin remaining before clipping. In stereo or 2-channel modes left and right displays are given, in mono mode MONO is illuminated and only the top display is activated. As on most digital equipment, these are peak

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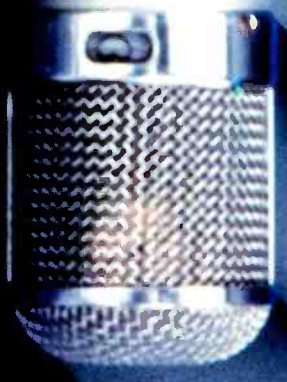

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responding meters but in this case they are not instantaneous, requiring about an 18 cycle burst at 5 kHz to read 3 dB below the actual peak level. The indicators also retain the highest display for about 2 seconds, keeping the single relevant LED illuminated.

Altering the input to output gain is accomplished by selecting one of the DELAY channel buttons. The system gain can then be adjusted in 1 dB steps over a ±6 dB range.

Pushing the control knob and rotating it anti-clockwise while in gain setting mode will mute the selected output. Muting is confirmed by a display of OFF when the control knob is pushed and by a 'I I' sign when it is released. Pushing the knob and turning it clockwise will restore the output, confirmed by an ON display.

Memory functions

The settings of all the above, including the rear panel mode control, can be stored in any of 12 available memory locations. Interestingly, input gain (headroom) information is not stored, to prevent feedback or damage when recalling a widely different setting by mistake. In most situations, the headroom setting would not change between presets since the same input and output equipment is likely to remain connected.

Saving set-ups is very easy, initiated by pressing the PROGRAM button. Rotating the 'knob' selects the program number required, then STORE or RECALL saves or recalls the complete unit set-up except for HEADROOM. STORE or RECALL alone can also be used to control memories by using a two-press action.

A D-type connector labelled REMOTE PROGRAM SELECTION on the rear panel provides for a simple relay interface or switch to be built to recall any of the 12 set-ups.

Data security

To the right of the control knob is a small hole through which may be found a 'safe' switch, providing an 'electronic security cover'. Pressing this with a 'blunt instrument' such as a ball point pen, will lock the front panel. Program changes may still be made with the various rear panel interfaces. When selected, a SAFE indicator lights in the display.

Bypass mode

Finally, on the extreme right of the panel, there is a bypass switch, with an attendant IN/OUT display. When switched out, the delays are set internally to 0 ms, while retaining the gain structure. The settings are still available for adjustment on the display, however. An option is available that will perform a total hard-wire by-pass via relay contacts for all four outputs.

Utility functions

Press and hold UNITS to enter UTILITY mode. The display first shows MIDI channel or RS422 address (depending on the option fitted). Another press displays temperature, which will be either the reading of the temperature probe (if fitted) or manually entered temperature (no probe fitted).

Another press displays master or slave transmit mode. This allows the MIDI or FSK interface to transmit information between master and slave units. Different modes allow the transmission of temperature information only, temperature plus program change, or complete set-ups.

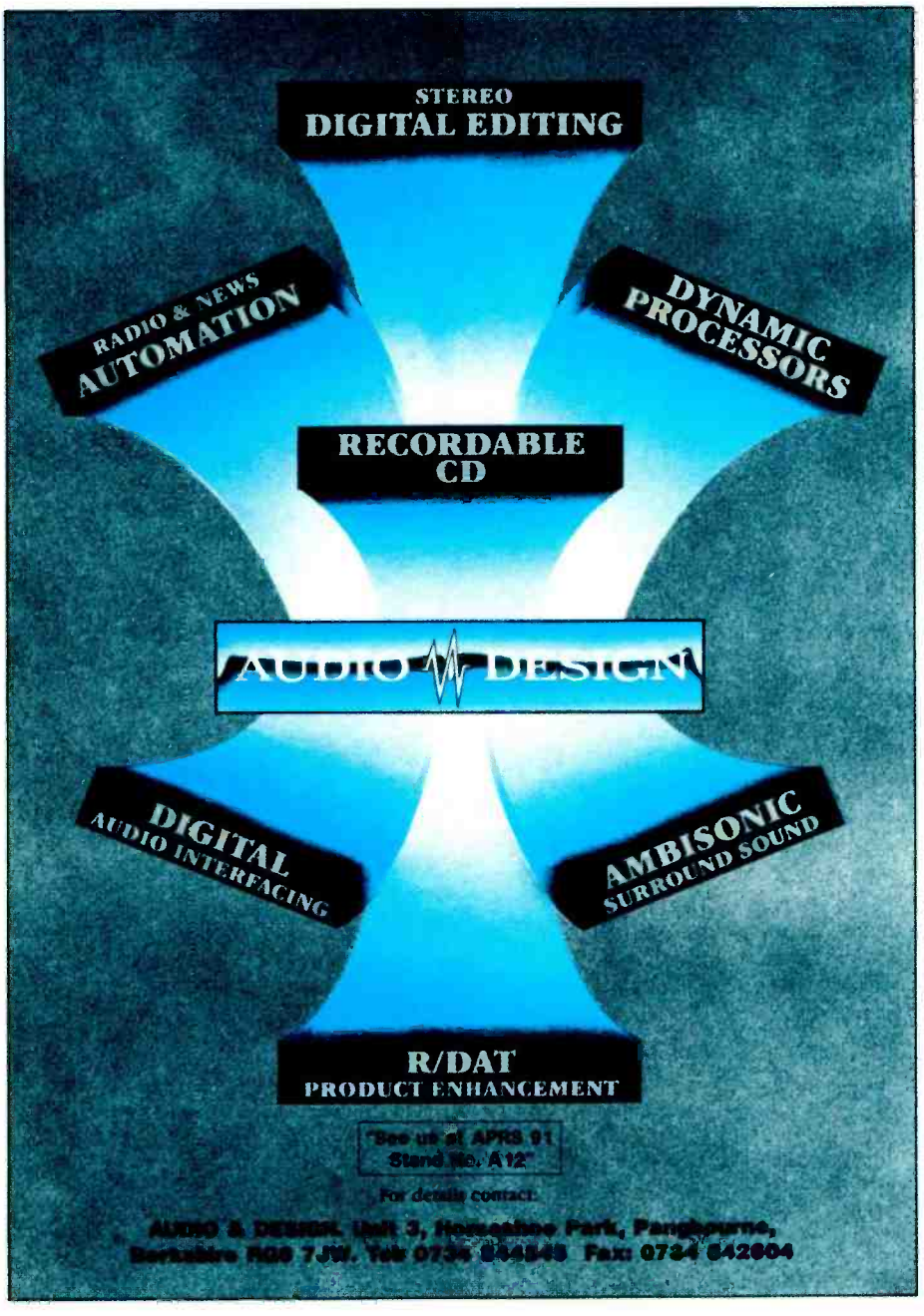
User manual

The test sample came with a 30-page user manual, numbered 1 to 35. Where pages 29 to 34 went we don't know. The manual is clear and apparently complete, with useful diagrams making it more interesting than many.

Performance

Inputs and outputs

Audio signals are connected via standard 3-pin *XLR* connectors. Inputs and outputs are



electronically balanced and wired Pin 2 hot. Pin 1 of the input is not connected to earth in order to help avoid earth loops. BSS have decided for you, that in any connection between equipment, the screen should be connected at the output. It is a pity there is no option, as it is almost certain that 50% of sound engineers will use the exact opposite convention. The input may be transformer balanced if required with a simple modification.

Input impedance is about 10 kΩ, as specified. Output impedance is about 50 Ω with a 600 Ω driving capability. Input common mode rejection is better than 70 dB at 100 Hz, decreasing to about 50 dB at 3 kHz which is maintained up to 20 kHz — a good performance.

Acceptable input levels range from 0 dBu to +20 dBu, depending on the setting of the headroom control, with corresponding output levels up to +20 dBu.

Operating levels

The point to remember when setting up the *TCS-804* is that the headroom level actually refers to the system clipping or safety level, not the general operating level. Thus, if the goal is to obtain an SPL of 96 dB with an amplifier and loudspeaker safety margin of 6 dB, then the system limiters should be set to act at the 102 dB level but the mixing engineer will try to operate 6 dB below this on the console meters. Let's say that the limiters are good, so we set the system up for amplifier clipping with a +12 dBu input level, corresponding to only 1 dB over the limiting level. The headroom of the *TCS-804* should then be set to match this +12 dBu clipping level. The meter of the *TCS-804* will then read in dB of safety margin below clipping and the *TCS-804* will also be optimised automatically for noise performance at this level.

Now to investigate performance variation with headroom and input/output gain settings. With a +20 dBu input signal, headroom at +20 and input/output gain set to +6 dB, the output level measures +25.98 dBu. At this level, THD+N (80 kHz BW) is 0.04% and the HEADROOM meter reads 0. A slight increase in input level sends the unit into hard clipping. Turning off the input signal gives a 22 Hz to 22 kHz RMS noise reading of –105.5 dB below maximum output or –85.5 dBu absolute.

Reducing the headroom level to 0 dBu and output gain to –6 dB gives an output level of –6.04 dB, THD+N of 0.04% and a resulting output noise of -92.6 dB below output or –98.6 dBu absolute.

Typical pro-audio applications are unlikely to have a clipping level of less than about +8 dBu, which with unity I/O gain gives a dynamic range of 100.5 dB. This is a considerable improvement over 16 bit delays and meets the published specifications.

Noise and distortion

The ⅓-octave BW noise spectrum at the +8 dBu headroom level is shown in **Fig 3**. This is an acceptable performance, with a rather larger 100 Hz component than we have found on other BSS products, which are typically among the best in the industry.

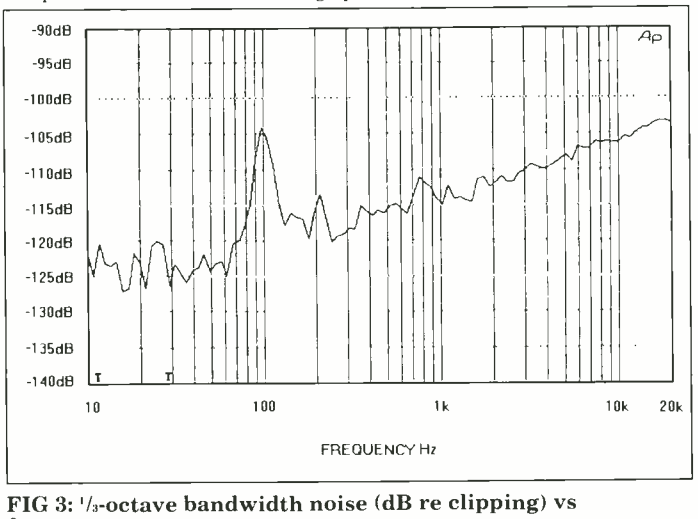


FIG 3: ⅓-octave bandwidth noise (dB re clipping) vs frequency
Output 1 set at 20 ms, 0 dB gain, 0 dBu headroom setting

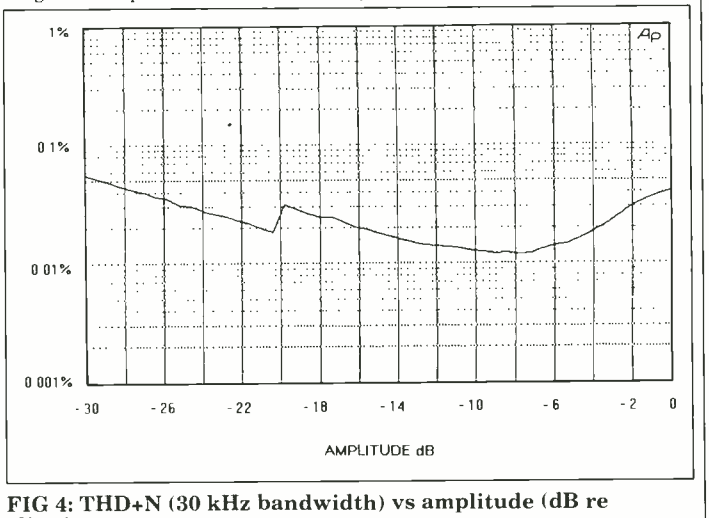


FIG 4: THD+N (30 kHz bandwidth) vs amplitude (dB re clipping)
Output 1 set to 20 ms, 0 dB gain, 0 dBu headroom setting

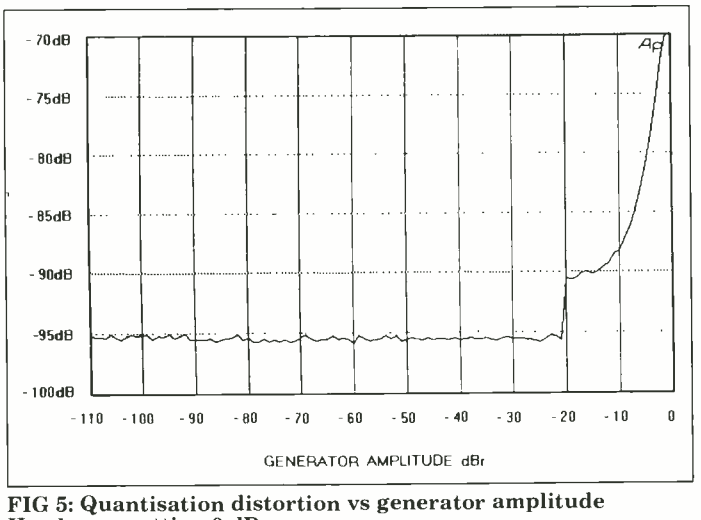


FIG 5: Quantisation distortion vs generator amplitude
Headroom setting 0 dBu
Output 1 set to 20 ms, 0 dB I/O gain, 0 dBu headroom setting

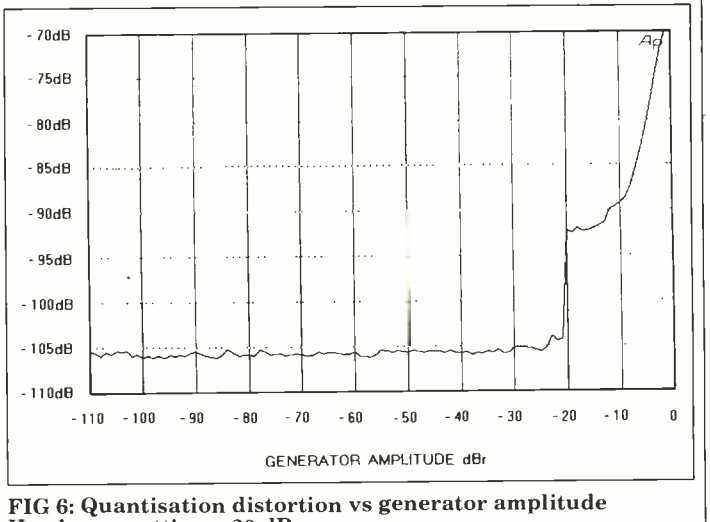


FIG 6: Quantisation distortion vs generator amplitude
Headroom setting + 20 dBu
Output 1 set to 20 ms, 0 dB I/O gain, 20 dBu headroom setting

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The THD+N distortion performance versus level shown in **Fig 4** is interesting. The gains are set up with headroom at 0 dBu and I/O gain at 0 dB. Starting at the right hand end of the curve, the distortion at just below clipping declines from 0.04% down to about 0.01% at –8 dB, then begins rising again as noise starts to predominate actual distortion.

But then at –20 dB, the THD+N drops by about 6 dB, rising again as level decreases. **Fig 5**, showing quantisation distortion, also has the characteristic bump at –20 dB, when it is even more clearly about 6 dB in level.

Changing the headroom setting to +20 dBu results in **Fig 6**. Here the blip is nearly 12 dB. It appears that what BSS are doing is sensing the signal level, and at about 20 dB below the internal clip point they are automatically boosting the input gain and simultaneously reducing the output gain — thus improving the noise performance without sacrificing headroom. This is what BSS call a 'Dynamic Floating Window PCM' converter system.

Modulation noise is typical of a high quality converter, being 2 to 3 dB at mid frequencies. Input/output linearity error is exemplary, showing no measured error down to –80 dB, with a worst case of 1 dB error at –105 dB referred to the clipping point.

Response, gain and phase

Frequency response is flat to within ±0.1 dB from 20 Hz to 10 kHz, reaching –1 dB at 20 kHz. The anti-aliasing filter is much gentler than other products measured to date, as shown in **Fig 7**. This is possible because of the higher sampling rate of the converters used, 100 kHz instead of a typical 50 kHz. One benefit of this is seen in the group delay measurement where the group delay remains more constant at high frequencies than in 50 kHz systems.

Gain control accuracy is shown in **Fig 8**, which reveals that over the ±6 dB gain range, gain is usually within 0.2 dB of the expected setting, with a worst case error of 0.37 dB. Interchannel gain difference tracked much better than this and outputs remained within 0.05 dB of each other.

Interchannel phase difference is also interesting (see **Fig 9**). In stereo mode phase matching between linked outputs is excellent. In mono mode, with all outputs set to the same delay, significant phase errors are introduced, roughly 18° at 20 kHz between adjacent outputs or up to 60° between Outputs 1 and 4. This phase shift is caused by a delay in digital samples from output to output. The delay is 2.5 µs, or one quarter of the sampling rate, ie the sampling interval is divided equally between the four outputs. The question could be asked: does this matter? Why have a 4-channel delay if all outputs are going to be set the same? And, if they do happen to be set the same, will they be driving loudspeaker units that are covering the same frequency range and may therefore interfere? In my view this is unlikely, making this engineering compromise a good one.

Remote control

There is a MIDI interface option, with the standard 5-pin DIN In, Out and Thru connections. This enables master/slave operation between *TCS-804* and *TCS-803* units alone, or via a central computer or sequencer. Instead of MIDI, you may have an RS-422 interface operating the PA-422 IED/ALTEC protocol, or you may use BSS's *FCP-800* handheld remote control unit with the FSK interface option.

The *FCP-800* Time Correction Controller, remote control unit for the *TCS*, is a robustly built

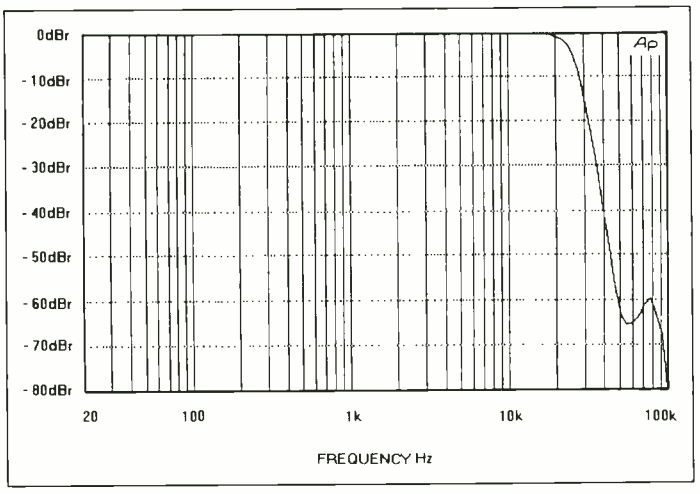

FIG 7: Output level vs frequency
Wideband frequency response showing anti-aliasing filter
0 dBu input, Output 1, 0 dB I/O gain, 0 dBu headroom
Delay set at 40 ms
Curve typical of all settings and outputs

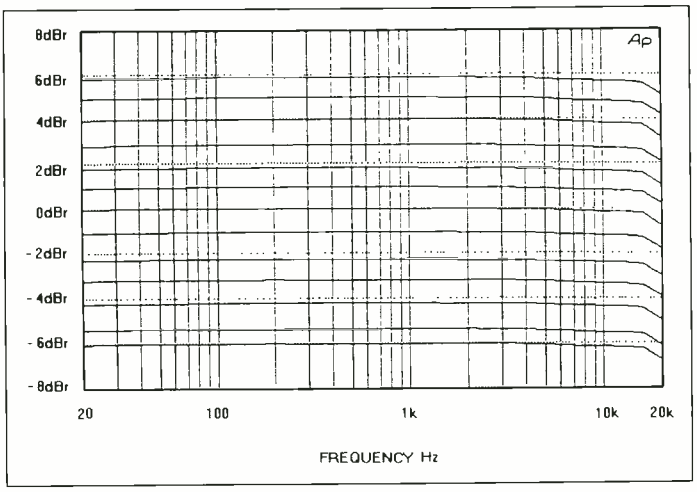

FIG 8: Amplitude vs frequency with varying I/O gain, delay set at 40 ms
Curves typical of all outputs
Top curve represents +6 gain, with 1dB gain change between curves

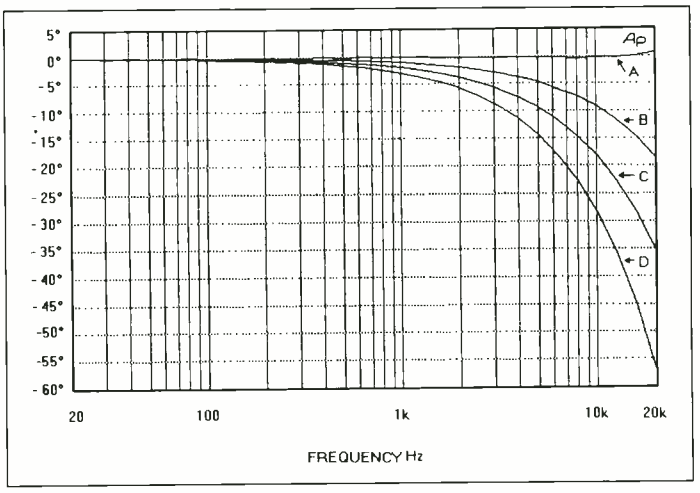

FIG 9: Phase vs frequency
Delay set at 30 ms on all ouputs
Reference output is 1
2-channel mode: A = the phase of Output 2 referred to Output 1 when they are stereo linked. Measurements of Output 4 referred to Output 3 are similar
Mono mode: B, C and D = Outputs 2, 3 and 4 referred to Output 1. The increasing phase shift reveals that the output converters may be shared or that there is significant delay between RAM access times

FPC-800 Time Correction Controller

unit housed in a plastic enclosure. It comes with its own padded carrying case. The unit contains a four-digit numeric read-out used for display of delay distances and levels only, signal delay time is not shown. Further LEDs indicate the function to be controlled, CHANNEL select, DELAY, MEASURE (explained later), LEVEL and MUTE. The SELECT key loops around these functions. Four further buttons select the required delay output and have associated LED indicators. DOWN and UP alter the chosen parameter. STORE and RECALL upload or download set-up to unit memories. ON/CHECK turns the unit on and performs a functional self-check. A nice feature is that if it hasn't been used for some time it turns itself off, saving the battery.

To use the remote, the *TCS* units must be fitted with the FSK remote control option board. Presently this prevents the simultaneous use of either MIDI or PA-422 remote control methods, partly due to tightness of space on the rear panel, but BSS mention that they are investigating a combination of the interfaces most commonly sold.

The general operation of the remote is easy. It is connected daisy-chain fashion (like MIDI) through the required *TCS* units, each of which have FSK IN and OUT sockets. The FSK IN on the remote is only required during the process of downloading the existing memory contents of *TCS-804* units. General operation is straightforward and will not be described here.

The MEASURE function is useful and unique. The *FPC-800* contains an internal microphone. It can generate a special test tone which is sent down the FSK remote control line. On the last *TCS-804* in the circuit, this tone can be taken from the FSK output and plugged into the FOH loudspeaker system. The *FPC-800* channel is selected to match the delayed system being adjusted. Then MEASURE is pressed followed by the button for the delay output being set. A tone is emitted from the FOH system and the signal transit time measured using the internal microphone. A second measurement is made automatically and compared to the first. If the two are within tolerance of each other, the data is accepted and the calculated delay value is sent to the *TCS-804* to set the delay in the selected channel. (I did not have the opportunity to test this function during the review.)

The *FPC-800* is designed so that it can be used 'wireless', transmitting its control commands via radio mic to remotely located *TCS-804* equipment.

Summary

The *TCS-804* is an impressive, easy to use product. The 'anomalies' that appeared under detailed examination look like engineering decisions rather than accidents, and for the application are unlikely to cause audible errors. Indeed, the benefits delivered — excellent dynamic range, flexibility, easy operation and value for money — reveal a manufacturer who puts much thought and care into design. The only real criticism about BSS products is that if they are mounted in road racks by the rackmount ears only, they break: BSS warn against this in the manual. Perhaps some hardware to assist with supporting the rear would be a nice option to overcome this criticism. Having dealt with BSS recently on several occasions, and finding them helpful, it is also possible to recommend their customer support.

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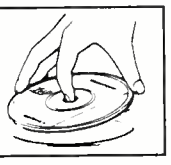

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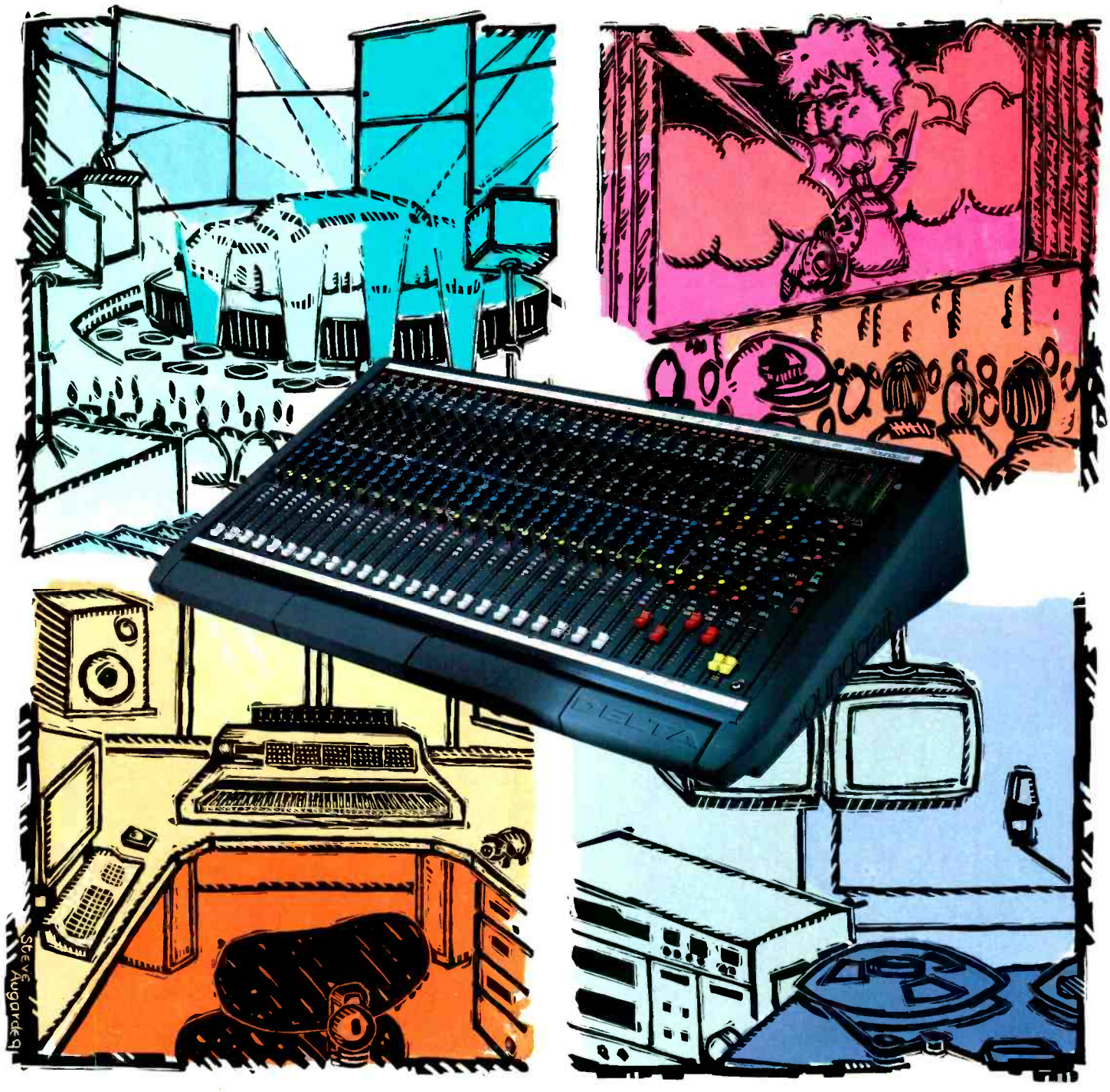

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