

July 1992

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AND BROADCAST ENGINEERING



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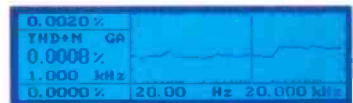
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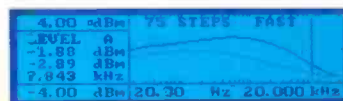
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The media divorce

Our industry loves buzz words. Take 'multimedia', for example — we are going to be hearing a lot more of that one. But this ongoing collision of various forms of media is, to my mind, rather misleading. We are constantly being told of new technical developments which, 'bridge gaps' or 'bring together' different forms of communication, yet it is our initial application of technology that is largely to blame for separating them.

It probably all started with Thomas Edison's wax cylinder, since it first provided the means for separating a man or woman's speech from their physical presence. The media divorce was certainly exacerbated by the arrival of the telephone as this interactive media robbed the participants in a conversation of all the visual cues which otherwise assist communication. The advent of radio made disembodied voices a part of everyday life and the popularisation of musical recordings made music without sight of musicians the rule rather than the exception.

Today we are seeing the other side of this particular technical revolution: the reconciliation. Television has done much to reunite images with speakers, actors and musicians — even if they are only two-dimensional at present. Videotape and video discs have helped consolidate this in much the same way records and audio tape consolidated radio. 'Videophone' systems already exist, although most of us have yet to see them enter everyday use. And now we're talking about multimedia.

Admittedly, given current levels of technology there are now aspects of audio and visual media which can not be attributed directly to nature, and the developments we are seeing in assembling music and dialogue for film work are genuine advances in their field. Similarly, the world of computer-generated images is unprecedented in the natural world and we must take full responsibility for its development and application. But the fact remains: technology has caused us to discriminate between our senses and now it must provide us with the means to reunite them.

So what of the future? Looking at the long term, the development of three-dimensional image systems are almost a formality. But will technology see us recognise the omission of other senses from our communication systems? After all, smell is supposed to command the strongest power of association.

One aspect of this somewhat esoteric discussion is directly relevant to our industry today, however, and that is the blurring of these previously-distinct areas of expertise and activity. The day of the audio-only engineer, for example, is almost certainly coming to a gentle close. ■

Tim Goodyer

Cover: The graphic display of the BSS FCS 92 Dual Varicurve Equaliser-analyser

Radio star.



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The new PCM-7010 – a brilliant radio performer

Ideal for broadcast radio applications, the new PCM-7010 professional DAT recorder from Sony offers quick loading, instant start, fader start and simple remote or direct control. The recorder has four heads for simultaneous confidence monitoring and its modular architecture provides options for digital I/Os, memory start, timecode interface and an RS-232 interface option for PC control.

Like all Sony products, the PCM-7010 is robust, reliable and versatile and, like all DAT recorders, offers substantial cost and performance benefits compared with 1/4" analogue.

DAT is a well established and proven professional format, now being used across a wide range of broadcast and production applications.

Backed by the EBU recommendation of DAT as an exchange medium, broadcasters have been quick to realise its benefits.

Sony has pioneered DAT technology and in 1991, with the PCM-7000 series, launched its first DAT recorders designed purely for the professional. Since then over 1500 units have been delivered to customers in Europe alone, a clear sign of success.

Now, with the PCM-7010, Sony can offer an unmatched range of five studio models, two portables and two remote controls, to cover all the requirements of field and studio recording, post production and transmission.

**Sony DAT – brilliant performance,
unmatched range.**



A sound choice!

One stereotype you can't ignore.



MS Stereo from Sennheiser

A superb combination from Sennheiser. That is both versatile and effective.

The MKH 30 is a pressure gradient mic with figure of eight directivity, optimising wide frequency response, lateral sound rejection and extremely low inherent noise. Matched with the remarkable directivity and sensitivity of the MKH 60 supercardioid microphone.

And to enhance the stereo image, low frequency ambience and vibration pick-up is minimised by highly efficient roll-off filters.

So everything you record sounds natural, with an accuracy no other method can achieve.



For operational flexibility, using a Y connecting cable means only one multiway cable is necessary.

Of course, when MS stereo isn't required, each mic can be used independently.

Important, when you consider the variety of tasks that you have to face in the field.

Sennheiser have produced an informative brochure by Manfred Hibbing on MS and XY stereo recording techniques which is available free.

For this and details of other great MS 'stereotypes' from the Sennheiser range phone (0628) 850811.

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Parsons knows

Heavy metal riffing is not what you expect to hear as you approach a beautiful 16th century house set in the depths of rural Sussex, nor is it particularly what one associates with Alan Parsons. So it was with some relief that I discovered the noise was being made by Parsons junior rather than his 'headbanging' father, who I found taking refuge, soldering iron in hand, in his newly finished studio.

Since moving out of London six years ago Parsons has owned three country properties, and has built a studio at each, although the second was never completed. This new studio, housed in a barn previously used as a garage on the 100-acre estate, is the first he has built without employing a designer. I asked Parsons what had inspired him to design his own studio?

'It's not so much that I chose to design it myself, more that I chose not to take on a designer. The cost would have virtually doubled and recognising that this is not the best time to be spending money, I've been cutting corners all the way.'

Having moved house twice in reasonably close succession, Parsons has become only too aware of the rather ephemeral nature of the home studio and this has played a major part in the design.

'I've decided that as long as I'm going to build home studios they should be designed to accommodate moving. It would be absolutely pointless for me to spend £100,000 on this place and leave it behind two years later. My last home studio, The Grange, was sold to someone who had no interest in the studio at all, so we ended up salvaging as much as we could without destroying the place in the process. The idea with

the new studio is that it can be dismantled with the least disruption, and leave a perfectly usable space behind — for that reason I purposefully avoided building studio-shaped rooms that a subsequent buyer would have no use for.'

The studio consists of a 5.5 m x 4.5 m control room with a high, pitched ceiling, and a galleried area at one end beneath which is an isolated 2.5 m x 4.5 m recording area with a floating floor; there is also a machine room, and a long galley kitchen. Work on the building started last November with the majority of time and expense going into general repairs to the structure and soundproofing. Parsons employed a local firm of builders who had been involved with his previous studios and kept a close eye on the project, directing it on a daily basis. Many of the items from the last studio, like Shone acoustic doors, air conditioning, trunking, line boxes and so on, were incorporated along with all the equipment — including the 51 input Amek *Angela* and two Sony 3324 multitracks. Monitoring is from a pair of 4 x 12 Questeds from which Parsons has removed the high and mid-frequency drivers.

'At the Grange the Questeds were mounted on these big purpose-built concrete plinths which worked very well, — but again to save money I've taken a different approach this time. Having seen what some speaker manufacturers are doing lately, I literally unscrewed the mid and top units from the main cabinets (which are placed on the floor a few feet in front of the console), rehoused them in new enclosures and placed them on top of the nearfields. Roger Quested was slightly horrified when



The new studio is in a 16th Century house in rural Sussex

I told him about it saying that the speakers were designed to work that way, but I think he'd be surprised just how good they sound.'

At the time of my visit the control room had little in the way of acoustic treatment, remaining quite live. Although Parsons was generally pleased with the sound and relieved that it was not 'an acoustic disaster', he admits to being no acoustician, and consequently had arranged Andy Munro to come and cast a professional ear over it.

'The good thing about what Andy is doing at the moment is that he's almost specialising in this kind of studio, because he's building studios within studios. He's designed a system of modular units that fit in very neatly and can transform the sound of a room relatively cheaply. The other advantage, of course, is that being modular they're easily removable.'

I called Munro just after he'd been to the studio and asked him what needed to be done?

'The control room acoustic is remarkably well balanced considering the lack of an acoustic treatment. There's a lot of low mass panelling (plaster board and rockwool) which gives damping, and the pitched ceiling works a treat

acoustically. What I intend to do is to add some low frequency absorption to the room's boundaries and focal points and a little diffusion to the rear wall using *System Z*, which is a modular acoustic system we've developed over the last couple of years. The advantages of working with a modular system like this is that you're always working with a known quantity, it's quick to install and it's cheap. Once the control room is treated and we have a good reference point, then we'll have a look at the studio.'

Parsons rather modestly regards his design as 'just good common sense', as he says there's nothing particularly fancy about it and it relies on tried and tested methods, but it fulfills his requirements perfectly and after all he is the client.

'It's been a valuable exercise both in terms of experience and cost — the whole thing has come to no more than £30,000, and I was quoted over £50,000 by a couple of studio design companies. I'm not saying that consultants aren't necessary — they actually do a very good job, but if you want to save money and you know what you want, this is certainly one way to do it.' ■

Patrick Stapley



Mr Parsons at his *Angela*

Innate role

Pete Dolan and Barry Hilton's company Innate Management announced three marketing and PR clients at last month's APRS show in London. They are to handle all European business for BOP Studios, the new residential studio in Southern Africa. Also they will handle the business development for Ian Sylvester, and are to handle the marketing and PR of Unique Audio Engineering, whose initial work includes installation of studio equipment into various Irish country houses.

Intimate

Paul Madden (ex-Madhouse Recorders) and Errol Henry of Intimate Productions, have opened their new facility at the former premises of The Smokehouse Studios, East London.

The studio is based around a custom-built, 64-channel fully automated Harrison console and two MTR 90s.

Installation was completed by Paul Madden in three days in order for Errol Henry to mix the new Jones Girls album for ARP records. Intimate Studio. Tel: 071 702 0715.

Dedication

In a move to provide a more efficient and faster sales service for Bruel & Kjaer Series 4000 studio microphones, a new company has been set up in Denmark to handle worldwide distribution. The new operation is called Danish Pro-Audio and it promises some new B&K product soon. UK distribution has not been changed.

Danish Pro Audio APS, Concordiavej 1, DK-2850 Nauerum, Denmark. Tel: +45 42 800 588. Fax: +45 42 800 578.



Give The Best People The Most Advanced Tools, And What Do You Get?

Results. That's why Vice President of Sound Operations Bill Varney (*on the left*) and Chief Engineer "Doc" Goldstein (*at right*) installed SoundStation II at Universal Studios Sound in Hollywood. "DAR has engineered this system so well, you never run out of power" explains Doc. "And SoundStation has the best user interface in the business. Editors love its intuitive work style."

"What's more," adds Doc, "...we get just as many compliments on DAR's WordFit® software that automatically synchronises replacement dialogue, its Stereo TimeWarp® time compression/ expansion software, and its Optical Disk subsystem that cuts loading and saving time substantially."

"Although we originally bought our SoundStation for dialogue editing," says Bill Varney, "...before long, editors were using it just as often for music and effects." Satisfied clients have been keeping the SoundStation room busy 16 hours a day. No wonder Bill and Doc have just installed an additional SoundStation at Universal.

The SoundStation room is one unique part of Universal Studios Sound's complete package of post-production services. You can find the same level of technical excellence, creative problem-solving and efficiency in every sound service at Universal. And of course, it's where you'll find professionals like Doc and Bill and their dedicated staff. From ADR and Foley to transfers, editing and the hottest new effects, their commitment to great post production is truly Universal.

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Royal Opening

May saw the official opening of Neve's new British headquarters by His Royal Highness The Prince Edward. During the formal opening ceremony, at which the 'Thespian Prince' unveiled a commemorative plaque and planted a Japanese Maple, he was serenaded by the 'Neve Singers' who sang of the company's pride in their new rural 'abode' and pleasure in being 'one big family'. One slight drawback of the country setting, though, has been the strong smell of pig manure picked up by the building's air conditioning, and in deference to the royal nose, a halt was ordered on muck spreading during HRH's visit.

The new 7-acre site is only a few miles from Neve's old premises, and the 4,500 m² complex allows the

company to centralise R&D, Sales and Marketing, Special Manufacturing, and Administration under one roof.

A major feature of the Litlington complex is a purpose-built recording studio that will be utilised for both training and demonstration purposes. In keeping with the company's long term development programme, room has been left for 25% expansion, although there appear to be no plans to integrate AMS. 'AMS are not going to move into the new premises,' said Neve MD Laci Nester-Smith, 'just how AMS and Neve are going to combine is very much an open question at the moment.'

Neve Electronics International Ltd, New Cambridge House, Bassingbourn Road, Litlington, Hertfordshire SG8 0QD, England. Tel: 0763 852 222.

DSP Conference

The future of audio development can be summed up in just three letters: DSP. Recognising a need for information, the AES is running a two-day conference on 14th and 15th September 1992 at Kensington Town Hall, Hornton Street, London W8, UK, which will bring together eighteen experts in the field of DSP for audio.

The Programme:
14th September.

The Digital Signal Processor: Introduction to Digital Signal Processors; Where's the Point? Architectural issues for Parallel DSP Audio System.

DSP and Psychoacoustics: The Human Hearing Process; Implementation and Measurement with respect to Human Auditory Capabilities; Sensitive Methodologies for the Subjective Evaluation and Assessment of Perceptual Coders.

Developing Audio Processing: Filter Design; Filter Topologies;

Options for DSP Code Generation. 15th September.

DSP Applications 1: Developing an Open-Ended DSP System for the PC; Practical Adaptive Room and Loudspeaker Equaliser for Hi-Fi use; Multichannel Signal Processing Techniques in the Reconstruction of Soundfields; Signal Processing Considerations for Acoustic Environment Correction; DSP in the Active Control of Sound and Vibration.

DSP Applications 2: Application of DSP to Audio Restoration; Examples of DSP use in TV Sound; Cost-Effective DSP for Audio Mixing; An Overview of Signal Processing used for Automatic Dialogue Synchronisation.

The Conference fee is £245.00 plus VAT for AES Members and £285.00 plus VAT for nonmembers. The registration fee includes Conference documentation, lunch and refreshments for both days.

For information contact: AES Ltd, DSP Conference, PO Box 645, Slough, SL1 8BJ, UK. Tel: 0628 663725. Fax: 0628 667002.

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spectral compressor

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When selected, the Enhancement section 'dynamically' boosts high frequency energy relative to the amount of gain reduction taking place in the compressor section. Variable 'Dynamic Spectral Enhancement' also allows the engineer to add additional 'Spectral Energy' to programme material offering unlimited scope for creative processing.

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IBC Preview

A list of major IBC audio exhibitors for this year's new venue in Amsterdam.

● **Adams-Smith**: will be featuring the *O-Gen* time code cue system, a multistandard video pulse generator with genlock. ● **AKG**: will demonstrate new software for the *DSE-7000* RAM workstation, plus their *Blue Line* microphone system, new BSS programmable *FCS-926* equaliser-analyser and other products. ● **Amek**: showing the *BCIII*, *Media*, and *Hendrix* consoles, plus *TAC SR6000*, *B2*, *Bullet*, and *Big-by-Langley* desks. ● **Ampex**: full range of professional tape. ● **AMS**: featuring the *Optica* low cost editor, *Spectra* colour interface for Audicfile, and *Logic 2*. ● **Audio & Design**: showing own brand CD-R, *Smart Box* format converter and SCMS defeat box. ● **Audio Developments**: *AD261* and *AD146* mixers designed for location recording. ● **Audio Precision**: full range of test and measurement equipment including *Portable One Plus*. ● **Audio Processing**: launching the *ACE 100* digital audio PC expansion card, and featuring the *DSM 100* two-channel full duplex digital audio transceiver. ● **Audix**: featuring the *ALB* live broadcast mixing system. ● **Avid**: first European showing of the *Audio PixStation* 24-track layout and editing system, plus audio enhancements to the *Media Composer* video editor. ● **Beyer Dynamic**: new UHF and VHF radio mics. ● **Calrec**: first showing of the *T-series* digitally controlled production-postproduction console. ● **Canford**: showing wide range of products including Sonifex Discart cart replacement machines. ● **DDA**: featuring the *DCM224V* postproduction console, also the *Interface Series* of general purpose consoles and processing equipment from Klark-Teknik. ● **Dialog 4**: the ISDN MUSICAM Codec, a digital fully duplex encoder-decoder designed to transmit CD quality audio. ● **DAR**: featuring Internal Digital Mixing for *SoundStation Sigma*, and the *Signet* networking system. Also enhancements to *DASS 100*. ● **Philip Drake**: showing range of audio and video distribution products, plus AD converters. ● **Dolby**: featuring the *DP5500* studio to transmitter link, and

models *501* and *502* bit rate reduction units. ● **Ela Audio**: a number of new products including the *S 120* broadcast mixer, *V Sound* mixing system, *EA 944* Audio Level Inserter, and the *Active 2* mini loudspeaker. ● **Elliott Brothers**: showing examples of their broadcast centre project management, design and installation work. ● **Fostex**: working demonstrations of the *PD2* DAT machine, plus the *G 24S 1* in multitrack and *D20B* DAT machine. ● **Future Film Developments**: full range of interconnection products, featuring new data cables and cable-matching transformers. ● **GML**: showing their new *HRT* rackmount modular mixing system, and demonstrating Version 6.6 automation software. ● **Graham-Patten Systems**: upgrades for the *D/ESAM 800* digital mixer, and European introduction of the *D/ESAM 400* digital mixer. ● **Gresham Wood**: manufacturers of technical furniture, will be debuting their *Production 2000* broadcast console. ● **Harrison**: the first showing of the fully digital *MPC* film and postproduction console. ● **Hes Electronics**: featuring the *HDTB* Intercom System. ● **HHB**: the *Yamaha DMC 1000* console,

Pro-Audio & Light Asia '92

A list of major audio exhibitors, for this year's venue in Singapore.

● **Altec Lansing**: will be showing their range of speakers and power amplifiers. ● **Amcron/Crown**: exhibiting full range of amplifiers and microphone systems. ● **Amek**: showing range of consoles including *Hendrix*, *Mozart*, *BCIII*, *Media*, *TAC SR6000*, *B2*, *Bullet*, plus the *Medici EQ*. ● **Ampex**: full range of professional tape products. ● **ARX**: launching the *MicroMAX* foreground loudspeaker system and the *DI-1* direct inject box, plus current range of processors, amplifiers and speakers. ● **Australian Monitor**: showing their full range of power amplifiers. ● **Dynacord**: featuring processor-controlled amplifiers, delays, EQs, and speaker systems. ● **Electro-Voice**: range of speaker and microphone systems. ● **Gauss**: showing a selection of speaker components. ● **Genelec**: their established range of loudspeakers. ● **JBL**: the new *M Series* comprising crossovers and dynamics processors, plus established range of



Matrox Studio at the IBC

Marantz CDR1 recorder, *Cedar DC1* de-clicker, and a range of DAT machines. ● **IAC**: debuting the *Quick Build* range of voice-over booths along with their range of acoustic components. ● **Lightworks**: will launch *DIGStation* and show new additions to the *Lightworks* editor. ● **Lindos**: new software for the *LA100* audio analyser. ● **Matrox**: first European showing of *Studio*, a desktop video system incorporating 32-bit DSP-based digital audio. ● **MBI**: featuring the *Series 30* stereo production console and *Series 20* broadcast desk. ● **Michael Stevens**: exhibiting audio products by *Artisan*, *Bel*, *Chromatec*, *Onyx*, *Sponsor* and *Total Systems*. ● **Neve**:

World premiere of the *Neve 55 Series* analogue broadcast console. Also showing the established *66* and *44* series broadcast consoles. ● **Otari**: enhancements to the *Pro-Disk* hard disk recorder and editor, new range of R-DAT, and video duplication products.

● **Penny & Giles**: launching new endless-belt controller. ● **Plasmec**: featuring the *ADAS* family of hard disk recording systems and the latest *Mosses & Mitchell* patchbays. ● **Rank Cintel**: demonstrating their full range of telecine and graphic products including first European appearance of the *Turbo 2* telecine with redesigned audio section. ● **Racom**: featuring the *DAMS 2* system.

● **RTW**: featuring the *8800* DAT remote control unit, *PLM 300V* program level meter, and *MLT 1255* mic and line tester. ● **Seem Audio**: showing *Frigg* and *Select* consoles along with *Seemon* nearfield monitors. ● **Sennheiser**: new *EM1046/SK50* radio microphone system. ● **SSL**: featuring their new integrated postproduction system *Scenaria*, along with *ScreenSound* and *SoundNet*. ● **Sony**: fully functional production and postproduction systems will be demonstrated using the new standardised method of Serial Digital Interconnection, plus full range of video products. ● **Soundcraft**: showing a wide range of consoles for broadcast applications including *BVE100*, *Europa*, *Venue Theatre*, *Sapphyre*, *3200*, and *SAC200*. ● **Studer**: full line of analogue and digital recorders, plus the new 2-channel digital audio workstation, *Dyaxis Lite*. ● **Trilogy**: will show the *Ouija* digital storage and playout system designed for radio applications. ● **3M**: full range of professional tape including enhanced version of *275* digital audio tape. **IBC will be held from 3rd to 7th July at the RAI Congress and Exhibition Centre, Amsterdam.**

SPL EXCITEMENT

Exciters and enhancers can be tricky to evaluate for many reasons, not least the tendency of otherwise rational manufacturers to become vague and mystical about such devices. Exciters vary from enigmatic black boxes whose acceptance requires an act of faith on the part of the user to complex esoteric contraptions laden with so much pseudoscientific jargon that not even the manufacturers seem quite sure of what they are actually doing.

A while ago the German company of SPL released a product which fell into a couple of the usual traps, but the rationalised, streamline version, the *Vitalizer*, appears to have got the balance between mystery and claptrap about right. That is not to say that it ever becomes fully clear what the *Vitalizer* is doing — the manual only says that it 'combines dynamic equalisation, phase shift manipulation and harmonic enhancement' and that it is 'based on established psychoacoustic principles' — but it provides enough user control, via sensibly-labelled knobs, to give the user the feeling of being in charge.

Most of the suggested applications involve the processing of a complete stereo signal during mixing, remastering, transfers, postproduction, radio broadcasts and live events, and the overall aim is to increase clarity and subjective loudness without introducing unwanted side-effects

such as harshness and boominess.

The 'equalisation', which is all boost, covers two areas: upper and mid, with a TUNE control to determine how far down the spectrum the effect will operate, and bass (curiously labelled SUB BASS) where a single centre-off control offers two boost curves, Soft to the left and Tight to the right. This is complemented by a DEEP button which takes the operating frequency range down lower still. Strangely, it is suggested that the Deep effects will only be heard on full-range monitors, which does not sit well with many of the real-world applications suggested for the unit such as the processing of radio material to make it sound better in cars. Having said that, the effects produced are extremely useable and SPL's claim to accentuate the low bass without the disadvantages of conventional LF EQ is vindicated. The contrast between the two responses is quite marked, both having immediately obvious applications and fitting their descriptive labels well.

The overall amount of bass, top and mid treatment is set by a single PROCESS DEPTH control; as a result everything is a bit interactive when setting up, although this is not a problem in practice. The Q of the top and mid section is adjustable, but as this can introduce some pretty unpleasant effects including self-oscillation it is wisely mounted

behind the panel as a screwdriver preset. In fact it seems not to be a Q control at all; it introduces a narrow peak which sweeps down as the "Q" control is advanced, and is then unaffected by the setting of the hi and mid tune control. This is supposed to allow fine tuning of the processor to lift, say, a hi-hat, but from what I heard it is best left alone or even taped over.

In addition to all this each channel has a HARMONICS control which, like all good enhancers, is supposed to 'restore harmonics that have been corrupted by the recording process'. That is as may be, but it certainly does 'increase the subjective brightness of a sound' in a way which sounds neither like conventional EQ nor like the other section of the *Vitalizer*. It also sounds different from certain other enhancers since it does not use the controlled distortion methods often employed elsewhere. It does, however, tend to bring up a little noise, which can become quite noticeable at extreme settings.

Despite the clear intention that the *Vitalizer* should be used for stereo processing, there is no facility for stereo linking the two channels. This is a little unfortunate, since the phase shifts the process partly relies on could cause problems if the two channels were set differently, deliberately or otherwise. A quick test showed that identical physical settings of the sub bass controls in

fact produced a phase discrepancy between the two channels from a mono source. The implications of this for stable stereo images, for mono compatibility, and for those still involved with black records are rather worrying, although on the bulk of the material I put through it such problems only became significantly audible when the controls were deliberately misadjusted.

An unusual feature is the PROCESS SOLO switch, which removes the direct signal from the output. Not only does this provide a quick check on what the processor is doing, but it allows the unit to be used in an aux send-return configuration so that several channels may be processed in varying degrees. A further curious facility is a STEREO WIDTH control, which uses phase shifts to exaggerate the stereo image. This would be more at home on a portable cassette player, and would seem to have little to do with the unit's main function; SPL suggest (quite rightly) that it should only be used sparingly, and I for one would be quite happy to see it removed altogether.

What would be left would be a versatile, controllable and effective processor. Its overall effect is not unlike the old hi-fi 'loudness' switch, but under precise control and much more flexible, and quite unlike anything which could be produced with conventional EQ. I have found nothing which could not easily be sweetened with it, and I shall be sorry to see it go back. ■

Dave Foister



Its overall effect is not unlike the old hi-fi 'loudness' switch, but under precise control and much more flexible

UK: The Home Service, 178 High Street, Teddington, Middx. TW11 8HU.
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ON TOUR

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

- **Audiorelease** have a large tour with Erasure, including a mammoth run of shows at London's Hammersmith Odeon, along with Joan Armatrading, Lou Reed and indie chart-toppers Carter USM.
- **Britannia Row** headed into summer with two of the season's largest tours under way and a pair of European festivals. The Cure began their *Wish World Tour* in May with a string of UK club 'warm up' dates before commencing sports halls and stadiums in the US and Canada. For the outdoor dates, BRP is using a 96-pair Turbosound Flashlight system — the company recently added another 12 pairs to its existing stock of 144. Also touring were Dire Straits, Frank Sinatra, Barbra Dickson, while the Glastonbury and Roskilde Festivals, unusually, were both held on the same weekend — June 16th to 18th. BRP also confirmed it has been appointed to work on the Olympic Games opening and closing ceremonies.
- **Canegreen** have a full summer with a Finish rock festival, Van Morrison, ELP in rehearsals for three months, a long tour with David Byrne, Stevie Wonder in the UK and Europe and the Crystal Palace Blues Festival. They also assisted Audiorelease on Lou Reed and provided racks for the excellent Crowded House. Mr Stille says The Radio Station in-ear system has 'sold out' with Erasure taking the last five and David apparently acting as an unofficial salesman at the Freddie Mercury Tribute. U2, Genesis, Grateful Dead and Joan Armatrading are among other recent buyers.
- **Capital Sound Hire's** Phil Anderson has Status Quo out on 'various festivals' starting with the Isle of Man in late Spring and tours until later this year; Crowded House in June and July and Natalie Cole's dates.
- **Clair Brothers'** summer season is full of substantial tours. Among those winding up in late July were Prince, Elton John's Europe dates (some, including Wembley, in conjunction with Eric Clapton), Roxette, Michael Jackson, Joe Cocker and the first European leg of U2's *Zoo* world tour.
- **Concert Sound** had The Music of Andrew Lloyd Webber (UK tour) as their main contract in July. It



U2's Zoo tour with Clair Brothers

plays theatres and arenas, ranging from the NEC, Manchester G-MEX and the Albert Hall to the Sunderland Theatre. They have ongoing involvement on the Dire Straits tour, while August sees Chris de Burgh in rehearsals for a major September to December tour. Other work included involvement in the *Save The Planet* show down in Rio featuring Placido Domingo.

● **Electrotec** had Cher's recent tour of Europe and the UK, with Dave Zammit on FOH and George Barnes mixing monitors.

● **Encore** spends five weeks at the Hackney Empire with *Black Heroes & The Hall Of Fame*, and has a large number of shows with muscle market leaders The Chippendales — plus Primal Scream dates, Orb at Brixton Academy and the London Philharmonic Orchestra (in conjunction with EML) in Werchter, Belgium.

● **Entec** ship 80 k of JBL Concert System to Ireland for Neil Diamond shows, and have the WOMAD festival at Reading's Rivermead site. They are also busy preparing for the 'official' Reading Festival on August Bank Holiday.

● **Funktion One** the new audio partnership formed by Tony Andrews and John Newsham, made its 'live' debut at, appropriately, Glastonbury Festival. In a far-flung field the boys installed a surround-sound *Flashlight* PA, central experimental music area (with live contributions from Underworld and DJs). Explained Tony beforehand:

'It's the most intense soundfield we know how to create; it's what we've always wanted to do and we hope the people who come along will enjoy it as much as we will — it's completely unfettered.'

● **GB Professional Audio** provided Shermann Audio PA systems for the recent Glasgow International Festival as well as shows by Donovan and Marian Montgomery.

● **Showco's** general commitments include the Chicago and Moody Blues tour, Harry Konick Jr, Linda Ronstadt, James Taylor, Skinny Puppy, Liza Stansfield (US), Santana, Genesis, Ozzie Osbourne and Slaughter, Spinal Tap's Albert Hall dates, Willie Nelson and The Beach Boys.

● **Sound Hire** have their Meyer *MSL-4* and *MSL-10* systems engaged throughout the summer on the amplified opera work in which specialise. July sees a lengthy run of work in both indoor and outdoor venues with singer Jose Carreras; most of the dates are in Germany and Switzerland.

● **SSE** are in the midst of Simply Red's stadium dates (June to September). The Wembley shows see the debut of their new slimline delay tower 'mast' — a neatly packaged self erecting device capable of flying around 25 kW of PA with, says Chris Beale, 'very little sightline intrusion.' Other shows include Wet Wet Wet; the London and Glasgow Fleadh's; Nirvana; the Canon Hill Park, Birmingham orchestral festival with Simon Rattle. **SSE** has added

another 24 systems of *MT-4* to its hire stock.

● **Star Hire's** John Cooper reported a 'very busy summer'. The Nice Jazz Festival headed their events along with some 22 stage-and-sound packages for orchestral events including the Midland Symphony, Wren Orchestra and Bournemouth Symphony. Slightly more unusual is an Elkie Brooks outdoor show on the Isle of Wight on July 25th, using Elkie's distinctive purple SLI system and Star Hire's control and 1978-vintage Turbosound Festival System subwoofers. Of this latter system John describes the Festival system's long throw nature (now reboxed by Star Hire) as 'quite brilliant for classical events', while the stackable design suits their mobile stage concept. ■

In Brief

- **Adamson** full range *MH121* cabinets and *FM121* passive floor monitors have been specified in a number of major contracts including two New York Broadway shows and a concert by producer and composer Daniel Lanois at Toronto University.
- **BSS** has sold its *DPR-901* Dynamic Equaliser to a number of leading SR companies and tours, including Dire Straits (Brit. Row-Concert Sound), Rod Stewart (Electrotec), Simply Red (SSE), Bryan Adams (JSI) and The Cure (Brit. Row).

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Trident

Trident have introduced a new midrange console to replace the Series 80. The Series 90 in-line console has been built to satisfy the increasing requirement for large numbers of inputs and incorporates three independent inputs per I/O module. Each of these has separate level, automated cut, and pan controls, as well as independent access to a three-way splittable EQ section and 10 aux sends which include two stereo pairs. Both long and short throw faders are VCA automated with the option for moving faders in the large fader position. The proprietary automation system also controls 12 switches per module, and provides 16 switch master groups. Integrated machine control and synchronisation are standard, controlling two tape machines and a virtual MIDI transport. The desk also incorporates 4 stereo effects returns and 2 separate foldback outputs. A virtual dynamics system is scheduled for later in the year, which will allow processing on all channel or monitor paths plus the main stereo bus.

Trident Audio Developments Ltd,
Trident House, Rodd Estate,
Govett Avenue, Shepperton,
Middx TW17 8AQ.
Tel: 0932 224665.
Fax: 0932 226721.
USA: Trident Audio, 2720 Monterey
Street, Suite 403, Torrance,
CA90503.
Tel: +1 310 533 8900.
Fax: +1 310 533 7072.

Big-by-Langley

Named after Amek cofounder Graham Langley, this new low-cost, in-line console features onboard automation and recall. The standard

configuration comprises 28 I/O modules, 4 stereo effects returns and an additional 4 routable stereo line inputs which double as fader controlled subgroups. The I/O module equaliser may be split between primary and secondary paths resulting in a total of 64 equalised inputs during mixdown, and the 8 auxiliaries can be rerouted to provide up to 20 sends.

The *Minitrue* automation system is closely related to Amek's *Supertrue* system and will control faders, and muting for the monitors, channels and auxiliaries 1 and 2. The recall system allows manual matching of controls either visually via a screen display or aurally by spoken instructions — the console shown at APRS incorporated the voice of Rupert Neve!

The 'Big' system can be optionally upgraded with Amek Virtual Dynamics, which offers software controlled compression, limiting, expansion, gating and autopanning for each channel which integrates to *Minitrue*.

Amek Systems, New Islington
Mill, Regent Trading Estate,
Oldfield Road, Salford M5 4SX.
Tel: 061 834 6747.
Fax: 061 834 0593.

SADiE

The Cambridge-based company Studio Audio & Video Ltd, have introduced the *Studio Audio Disk Editor (SADiE)* for IBM PC-AT or compatibles. The 2 or 4 output audio editing package runs under *Windows 3* with all features being mouse selectable. The system utilises the company's X-S floating point digital audio processing card which includes AES/EBU and SPDIF I/Os and supports up to 24 bits of audio data per channel. Analogue I/Os are provided from an X-ACT card that



SADiE — the screen display

performs 64 times oversampling delta sigma conversion as well as providing a SMPTE reader-generator.

The screen display is split into three areas: a transport control window provides traditional tape machine control including instant locate; a level control window provides fader control of the outputs and peak metering; and an edit window graphically represents the audio. All editing functions — move, cut, paste and copy — are nondestructive, and the system maintains two independent EDLs. **Studio Audio & Video Ltd, The Old School, Stretham, Ely, Cambridge CB6 3LD.**
Tel: 0353 648888.

Soundtracs

Soundtracs showed three new mixers at the APRS — the *Jade*, the *Solo Logic*, and the *Exiom*.

The *Jade* in-line console features fader automation on both channels and monitors, mute automation on all inputs plus dynamic gate processors on every channel. Independent EQ is provided for both primary and secondary signal paths, and inputs, outputs and buses are balanced. The desk has extended bandwidth electronics and a comprehensive TT patchbay.

The *Solo Logic* shares all the features of Soundtracs' *Solo* MIDI mixer with the addition of fader automation. The system has a 12-bit fader resolution, is frame accurate for faders and 1/2-frame accurate for

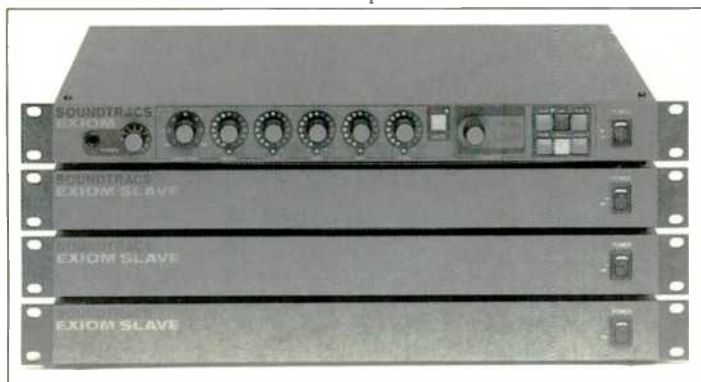
muters. Automation control is via an integrated LCD and rotary dial, and from local channel read and write keys. Mixes can be saved internally or to a sequencer which can also control *Solo Logic* via the MIDI Manager page.

The *Exiom* is a mixer in a rack. Each 19 in 1U contains 8 stereo channels with individual gain, 2-band EQ, aux send, pan, mute and level control. Expansion via rear ports provides up to 64 stereo channels all with recall and mix automation from a MIDI sequencer. The system has been designed for programming suites and MIDI studios where the mixer can be controlled on screen. **Soundtracs, 91 Ewell Road, Surbiton, Surrey KT6 6AH.**
Tel: 081 399 3392.

Perfect pitch

New UK company Perfect Pitch have launched the *OM 3D* Sound Processor, which is a competitively priced (£3,500-\$1,944 approx) rival to the RSS and Qsound systems. The company claim the system can control 2 channels of 180 degree continuous pan in an arc to the front of the listener, and 2 channels of 300 degree pan that extends from the left speaker to the right in a continuous arc to the rear of the stereo listening position.

Select Systems, Hanworth Trading Estate, Hampton Road West, Hanworth, Middx TW13 3DH.
Tel: 081 893 8662.
Fax: 081 893 4318.



Soundtracs Exiom — a mixer in a rack



Lexicon LFI-10 links incompatible equipment

Lexicon

The Lexicon *LFI-10* is a digital interface unit that links incompatible equipment by viewing and modifying the digital bitstream. Auxiliary information such as format, sample rate, count and time to CRCC is interpreted and corrected as necessary between digital audio devices.

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

Raindirk

Raindirk Audio have expanded their *Symphony* console into three types. The original desk will be called the *LN1*, a new improved version will be known as the *LN2*, and a mobile 48-group/dual mic input version will be named the Mobile Series *LN*.

The *LN2* features a redesigned parametric equaliser that can be split between channel and monitor paths. The 8 auxiliary sends, previously controlled by 4 level controls, now have 6 controls each with mute switches and a paralleling facility for aux 5 to 8. The cue sends have also been fitted with mute switches, disconnecting the circuit from the mix bus to keep noise and leakage to a minimum. The channel insertion point is now selectable pre-post EQ, and the original *Stereo 1/2* button has been replaced by two routing switches. A VU reverse facility has been added per channel allowing local line in-line out meter switching. LEDs have been added to all areas, and a bi-colour LED shows signal present and peak indication for each channel. The normal destructive solo mode can now be locally overridden to AFL, and global switching is provided for AFL/PPL switching — a Solo Clear facility is also provided for clear-down selections. Raindirk have replaced their patchbay system with standard 19 in rackmount 1U units, and have

included Mogami oxygen-free copper cable between studio circuits and patch and patch to modules.

The Home Service, 178 The High Street, Twickenham, Middx TW1 8HU. Tel: 081 943 4949. Fax: 081 943 5155.

Focusrite

Focusrite have released two new products, *ISA 230* Stereo Compressor-Gate and the *ISA 215* Dual Mic Pre-Equaliser. The Compressor-Gate retains the transformerless balanced output stages and same racking and power supply system as used with the *ISA 130* dynamics unit. The compressor section offers six switched ratios including limiting, continuously variable Attack (300 μ s to 90 ms) and Release (100 ms to 4 s plus Auto). Gain makeup is provided from 0 to 20 dB. The gate includes Range (0 to 80 dB), Hold (20 ms to 4 s), Release (100 ms to 5 s), and Attack (1 ms or 50 μ s), plus KEY IN and KEY LISTEN switches.

The Dual Mic Pre-Equaliser retains the familiar *ISA 110* controls including phantom power, switched mic-line gain controls, 20 dB gain trim, 4-band equaliser (HF and LF shelving, HMF and LMF peak), high and low-pass filters (18 dB/octave), in a 2 U rackmountable unit which features a newly designed integral power supply.

Focusrite, Unit 2, Bourne End Business Centre, Cores End Road, Bourne End, Bucks SL8 5AS. Tel: 0628 819456. Fax: 0628 819443.

Creation

To address the problems incurred by long cable runs between microphones and amplifiers, the British company Creation have produced a Microphone Amplifier System that places a variable gain preamplifier close to the microphone, sending the signal at line level using an usually

powerful line driver stage. The 8-channel system consists of a 3U Audio Rack that connects to a 1U Controller providing gain, phantom power and metering for each channel. The system features all discrete, fully differential audio circuitry custom designed toroidal transformers and locally regulated power supplies. Creation claim exceptional results even over cable runs of 300 m with input noise measured at -128 dB and bandwidth exceeding 200 kHz.

Creation, Unit 3, Mount Farm, Churchill, Oxfordshire OX7 6NP. Tel: 0608 658946.

ART

ART have superceded their *Multiverb LT* processor with the *Multiverb LTX*. This updated version is capable of four simultaneous effects without the need for programming, and offers over 250 presets of multiple effects including reverb, chorus, flanging, delay, reverse effects, stereo imaging and so on. The 16-bit unit provides stereo inputs and outputs, and is MIDI addressable. Existing LT systems can be fully upgraded to include the new LTX features.

UK: Harman Audio, Mill Street, Slough, Berks SL2 5DD.
Tel: 0753 576911.
Fax: 0753 535306.

C-Audio

C-Audio have launched the *IA* System comprising a range of high

performance MOSFET amplifiers featuring a range of options such as plug in active crossover, limiter, speaker processing and AES/EBU and SPDIF input modules. The optional onboard processor provides control for additional modules including gain, phase, mute and delay. An interface module allows control via RS232, RS422, MIDI or a high speed network. The current range of *IA* amplifiers offer power outputs of 250, 450 and 650 watts per channel into 4 Ω , and are to be joined later in the year by 1000 and 1500 W bipolar versions.

Also new is the 1U high MP1 monitor amplifier (2 x 75 W/8 Ω) designed for the compact studio and broadcast market. The company have also upgraded their *RA* Series of power amplifiers to include the option of Speakon Connectors or jacks and binding posts, plus new styling in line with C-Audio's new corporate identity.

UK: Harman Audio.

Offset A-DAM

Following the recent price reduction of the *A-DAM* 12-track digital recorder, Akai have announced the availability of the *DS1200* Digital Offset Editor. The rackmount 1U allows programmable offsets between master and slave machines, thus enabling spin-ins and editing.

UK: Akai Digital, Haslemere Heathrow Estate, Silver Jubilee Way, Parkway, Hounslow, Middx TW4 6NQ. Tel: 081 897 6388. Fax: 081 759 8268.



C-Audio RA 3001 recently upgraded

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Electronic Musician
May 1992 Issue

"The SP offers ambitious programmers the potential for creating new signature sounds. Particularly considering its low price, expandability and first-rate storage and loading capabilities, the SP gives a musician more than just an introduction to sampling. With the SP, Peavey moves the flexible-architecture philosophy to new frontiers."

EQ Magazine
February 1992 Issue

The DPM® SP/SX sampling system is a phenomenal value. Costing thousands less than comparable units from our competitors, and hundreds less than most low end systems, the SP/SX combination represents the most powerful, yet affordable, full-featured 16-bit sampling system on the market today!

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The SP is capable of handling up to 32 megabytes of internal sample memory. The sample RAM is expandable with low-cost industry standard SIMMs expansion boards.

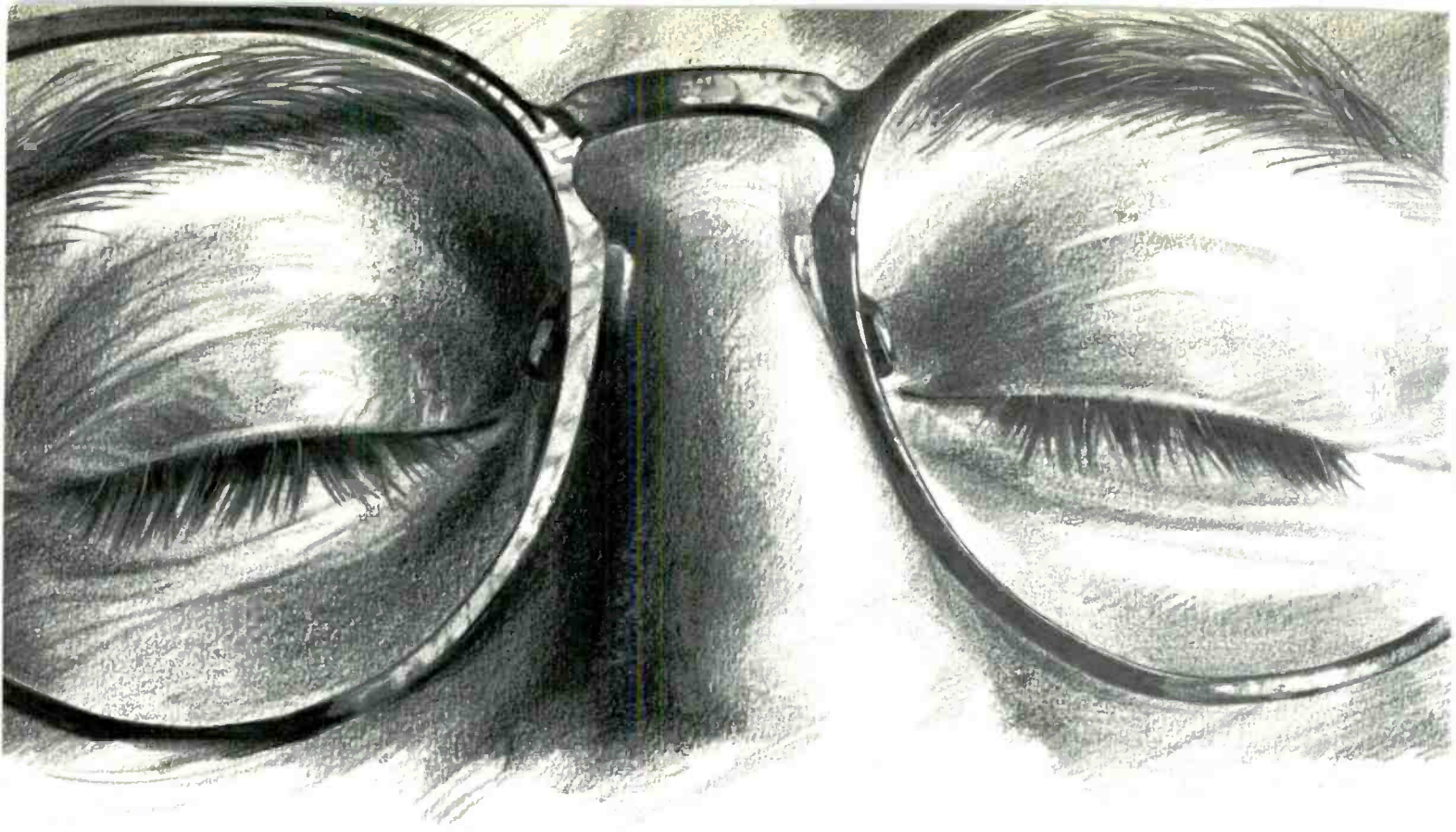
The DPM® SX Sampling Xpander module allows you to digitally record your own 16-bit samples and send them over SCSI to the SP or in the standard SDS format to your DPM 3 or other compatible instrument.

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Scenaria is capable of accessing both audio and video instantly

SSL SCENARIA

Patrick Stapley on the ideal streamlined companion to *ScreenSound* and *SoundNet*

The surprise unveiling of SSL's *Scenaria* at April's NAB show, is a further indication of the company's commitment to the audiovisual market. *Scenaria*, which was described to me as 'everything you need for postproduction of audio to picture', represents the culmination of a long R&D period and is the first hardware controlled, digital product from the company since the ill-fated *01* Digital Production Centre which appeared in 1987. SSL seem to have repeated the concept they pioneered for mixing console design of integrating a number of elements into one 'Master Studio System', or in this case 'Master AV System'. The result is a well equipped and ergonomically well thought out package. Although *Scenaria* is very much a stand-alone system, great importance has been attached to its compatibility with SSL's *ScreenSound* and *SoundNet*, thus providing the advantages of networking and shared facilities. A fundamental

philosophy has been to produce a system that will actually increase efficiency, boost productivity and ultimately raise profits — which after all should be the bottom line for any equipment purchase.

So what exactly is the *Scenaria*? Broadly speaking it is five things rolled into one — a 38-channel digital mixing console, a 24-track hard disc recorder, an audio editor, a machine controller and a random access video store (*VisionTrack*). By virtue of the fact that video is stored within the system, *Scenaria* is capable of accessing both audio and video instantly, thereby removing the locate time associated with VTR transport.

Scenaria comprises a compact console — unmistakably SSL in styling — containing three angled bays which connect to a remote processing rack. The central bay contains all the assignable mixer controls such as EQ, dynamics, auxs, motorised faders and so on, as well as transport-locate controls and various menu and

function keys (including five User Preset keys). Also centrally placed is a high resolution monitor that can be interacted with from the console control surface, or by using the pen and tablet as with *ScreenSound*. Another monitor, built into the right-hand section of the console is used to display and control the video element of the system. The left bay houses a magneto-optical drive and a digital patchbay which is part of the computer controlled routing system. Lastly, there is a qwerty keyboard that neatly slides beneath either the left or right desktop depending on user preference.

The audio in *Scenaria* is referred to hierarchically, starting with a Clip which is a piece of audio placed against time code in what is referred to as a Reel (can be thought of as a track). Eight of these Reels makes up a Desk, and 3 Desks (24 channels of disk-based playback) constitutes a Project. In addition to these 24 Reels of audio there are other related data such as automation, routing, time code offsets, EDLs, cue sheets, scripts and so on, which are all stored at the Project level and can be backed up to the internal MO drive, in rather the same way that mix data is backed up from a *G Series* console to floppy.

The system is capable of storing up to 48-track hours of audio, although 9 hours (3 hard disks) is standard. Audio inputs and outputs are via 16 (optionally 32) analogue I/Os, and 16 (optionally 32) AES/EBU pairs. Routing is controlled from a computer driven audio routing matrix, and all selections can be stored and recalled. In addition, ▶



Compatibility with ScreenSound and SoundNet is of fundamental importance

fully edited and automated ScreenSound Desks may be imported directly into the system.

Scenaria will store up to an hour of random access video in a compressed format, and this can be catalogued in a list of Video Clips. The loading of audio and video can be carried out as an off-line operation thus saving time, and machines may be controlled directly from the console via computer routable RS 422/232 serial interfaces. Up to four machines can be simultaneously controlled via Sony 9-pin, VPR 3 or Laserdisc protocols achieving a frame-accurate picture 'hard lock' of sound and picture; individual machine record enables are provided for layback of the finished mix either to picture or tape.

The main screen displays 34 channel faders (35 to 38 have to be scrolled), a main stereo mix fader, and a main monitor fader (doubles with the monitor pot on the console). The default condition is that Reels 1 to 24 will replay to faders 1 to 24, and faders 25 to 38 will be used to return effects, external machines, etc — however this can be reorganized to suit the user. Once audio exists in a Reel the associated fader head will turn from grey to yellow providing a useful indication of recorded and empty tracks. The main screen also displays metering, fader movement, and solo and cut status for each channel. In addition, there is an Overview display that shows the audio clips on each Reel as blocks of sound passing over an imaginary playhead rather like a piece of marked multitrack tape. This vertically scrolling display provides a very clear indication of where audio segments start and finish as well as pinpointing silence. It also indicates gain profile by incorporating a linear representation of fader movement, giving useful visual feedback of automated level changes against recorded audio.

The 38 screen faders and main stereo fader can be assigned either individually or subgrouped to the eight hardware moving faders and their associated CUT and SOLO buttons, where (if previously named) they will be identified on the

electronic 'scribble strips' above each fader automatically. Alternatively, track titles can be added or amended directly through Scenaria's qwerty keyboard. To make life easier, eight banks (accessed from a column of keys directly to the right of the faders) are available in which fader assignment groups (Logical Groups) can be stored and reset — the default condition of these banks is to assign consecutive groups of eight screen faders to the motorized faders (that is Bank 1: 1 to 8, Bank 2: 9 to 16, Bank 3: 17 to 24, etc.).

Processing

From the hardware fader, assignment can then be made to the other processing sections. These are arranged into three separately assignable areas comprising two identical EQ, Filter, Aux, Pan and Delay sections and a Dynamics section — each section contains an ID window showing what is currently assigned. The idea of having two EQ sections is a nice touch in that it allows simultaneous control of separate channels, which is not always possible with assignable technology.

The default EQ is four-band parametric and this can be reconfigured to a number of different topologies; it can also be configured to provide high

and low-pass filters and a two-band EQ. Dynamics (Expander-Gate, Compressor-Limiter) feature a positive-negative delay for the sidechain; there are eight Auxiliaries available per channel (toggled 1 to 4 and 5 to 8) with On-Off, Pre-Post switching; and the Panning (left-right) control doubles up as a Delay (± 200 ms) control. The sections include a FLIP key that allows two different settings to be toggled, a CLEAR key that removes the last move but keeps the others, and a RESET key that removes all previously stored values. The rotary controls each have a crescent-shaped display that tracks the pot position, but as yet the system does not include any form of parameter identification (that is frequency, dB values, etc.) — I understand this is to be added. However, both EQ and Dynamics can be graphically displayed on the main screen, and user setups may be stored within the system — SSL are also in the process of adding a number of factory presets.

There are no limitations set by the amount of processing power within the Scenaria, and full dynamic processing is available on all channels at all times.

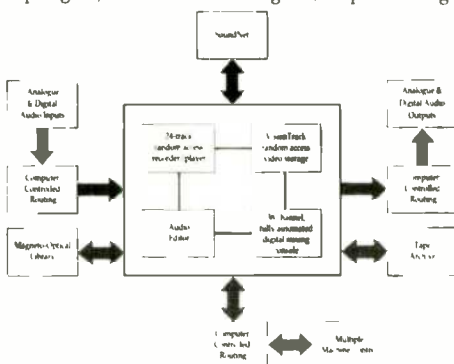
Automation

Any number of Channels can be grouped together in the traditional master-slaves arrangement and apart from level and cuts, groups will also respond to EQ, dynamics and delay changes. The system provides a total of 24 groups which can be nested.

The console is fully dynamically automated and can operate either in the conventional moves against time code sense or on a per Audio Clip basis. For example, Clip Fill mode will write the current parameter values for the duration of the Clip, while Absolute will write them for the duration of the Reel. Dynamic fader movement is displayed in three different ways: firstly, the eight moving faders will track assigned movement; secondly, the screen faders follow moves; and thirdly, the linear display on the Overview screen will trace level against Audio Clips. Automation data is automatically updated when edits are performed, and an imported project from ScreenSound will include the automation data.

Editing

Editing can be performed on single or multiple Reels down to sample accuracy, and is similar in operation and facilities to ScreenSound — in fact the editing screen looks almost identical. Butt and crossfade edits, track slipping, inserts (Audio Fill) and time compression-expansion are all available and will simultaneously affect the video if applicable. Complete Scenes can be added or removed with ease while retaining continuity, and the system will import EDL files and Autoconform. The editor includes a Dialogue Sift function that automatically scans a Reel for silence, replacing it with digital silence so providing quieter mixes and saving on disk space. All editing is nondestructive and like processing can be undone at any time. Of course, the other great benefit is there is no degradation involved when copying or moving audio around within the system. ▶



A schematic of the Scenaria system

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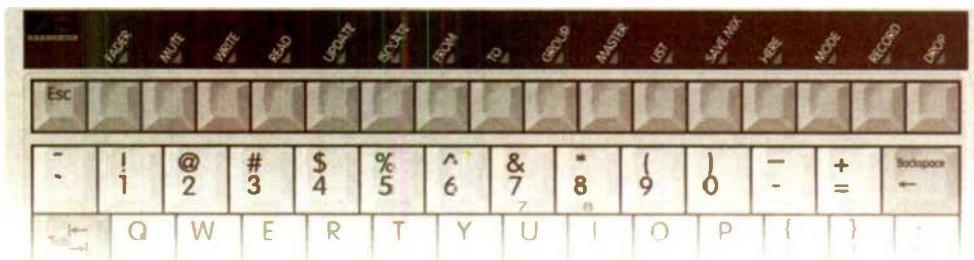
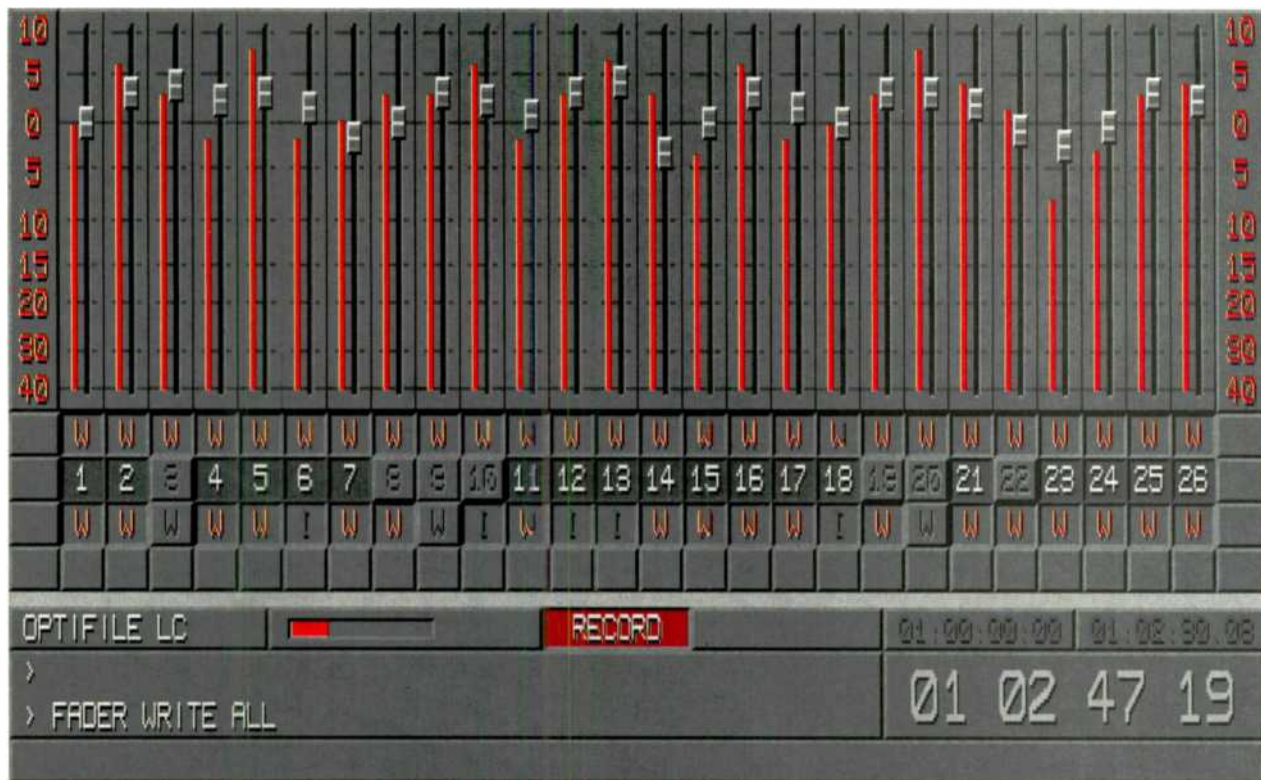


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Locating positions, or rolling back, is achieved instantly for both audio and video. Time code readouts are displayed on both screens as well as in the console, and there are a number of controls associated with moving around a mix. Firstly, there are the normal TRANSPORT keys one would expect to find on a tape machine, and these may also be assigned to provide external machine control; additionally, there is a Jog-Shuttle wheel and mechanical feedback, that locks at the extremities of the mix. Five MARK keys are included for capturing-locating time code positions, and these also perform as programmable ROLLBACK-ROLLFORWARD keys. The MARK keys and TRANSPORT keys are also represented graphically on-screen where they may be controlled using the pen and tablet if preferred. Another method is to grab the hands of the clock display (shows elapsed time), again using the electronic pen, and physically move them to the desired time. There is yet another way, which is rather ingenious and involves *VisionCue*.

VisionCue is a time-based display that allows the operator to see video events in a graphic form. It works by comparing colour and brightness information between frames and so identifies scene changes. The display, which is superimposed over the video, shows a series of horizontal marks on a vertically scrolling display — the bigger the

mark the bigger the scene change, the broader the mark the longer the crossfade. Not only does this provide useful visual feedback, but it also operates as a visual autolocator by allowing the user to pick a position using the pen and tablet. *Scenaria* provides the best of both worlds, enabling the user to reference to either audio or visual cues.

Networking

The networking element of *Scenaria* has been central to the overall product plan from the outset, and compatibility with *ScreenSound* and *SoundNet* is of fundamental importance. Although *Scenaria* is a fully integrated stand-alone system, it is nevertheless the policy with most postproduction facilities to separate editing and mixing into different areas, and it is here that the networked system comes into its own. By having satellite editing rooms that link to a central mixing room, a production can pass from the editing stage within *ScreenSound* via *SoundNet* to the mixing environment within *Scenaria* — by virtue of the fact that *Scenaria* will import all the necessary data as an off-line operation, the process can be organised in an efficient, seamless fashion.

Another important aspect of networking is shared facilities, and through *SoundNet* up to 16 SCSI devices can be accessed. This could

comprise of a bank of hard disk storage or a mixture of hard disk and optical drives. Obviously, through this arrangement there is no need for a facility to duplicate sound libraries as all audio will be equally accessible to all operators. The system also provides a system-wide database for stored audio, including clips from off-line optical disks, and will search and sort audio-based on a Keyword identification process. Off-line back up of audio from selected disks can be achieved directly from the screen display, which also shows which disks are being used and from which station.

Another facility available through networking is the ability to slave *ScreenSound* to *Scenaria* in order to provide extra space.

Conclusion

Scenaria offers a great deal from a remarkably compact design. It is indeed a fully integrated system, and the combination of random access audio and video will, I am sure, appeal strongly to users.

As the system stands at present, there are still a few details to be finalised and some finishing touches to be made. SSL, who claim they have already received considerable interest in *Scenaria*, are forecasting its commercial availability by the autumn. ■

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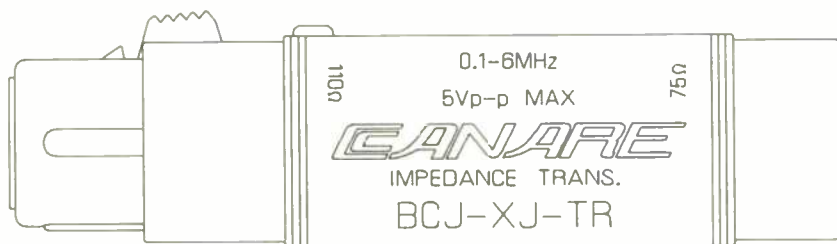
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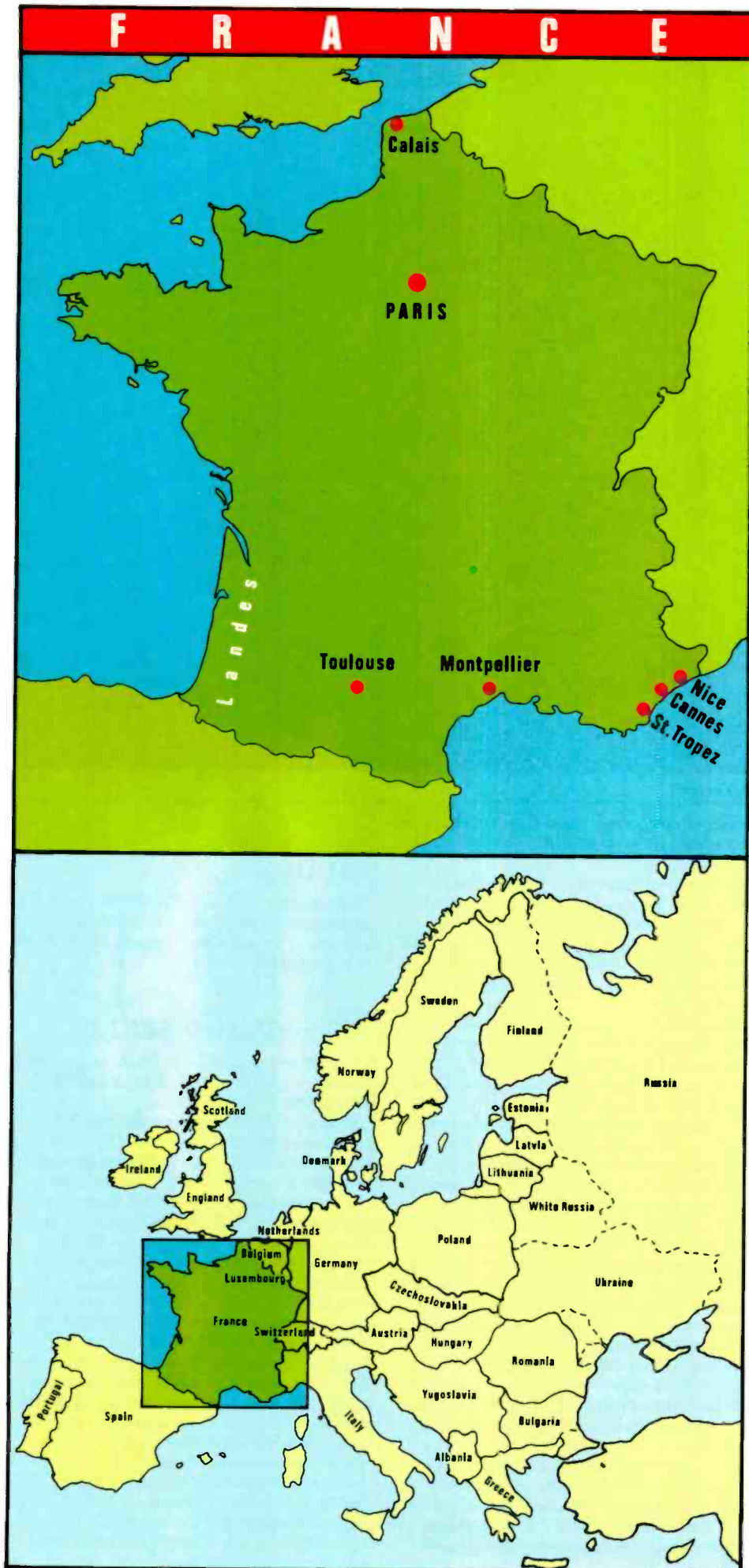
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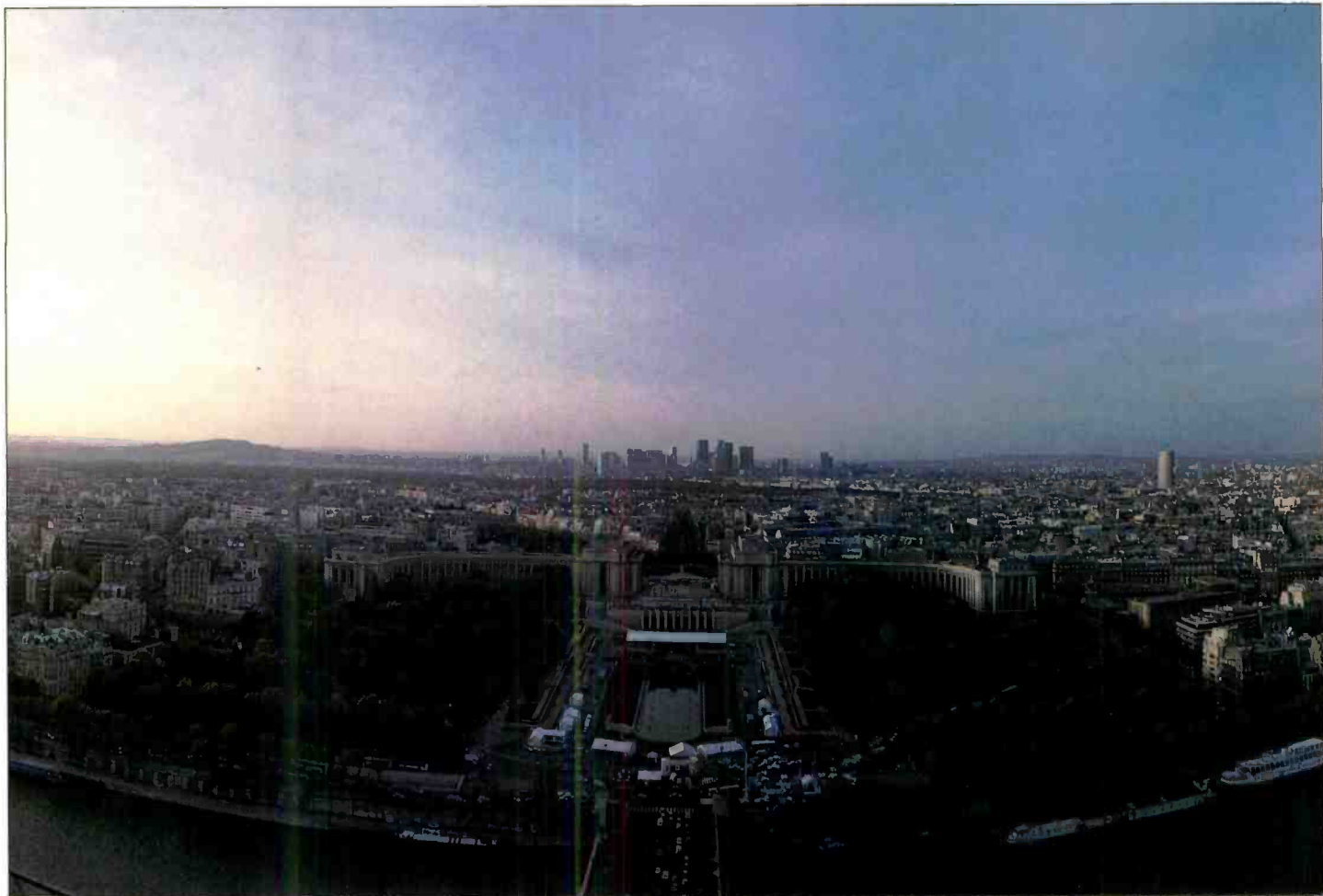


FRENCH LEAVE

Most recording in France is based in Paris but there are exceptions spread across the countryside. Sue Sillitoe provides the tour

For most people France should not need much introduction. It is only a short hop across the English Channel and there can be few people who have not made that hop at some stage in their lives.

Compared to the UK, France is a large country. It has three times the land mass for about the same number of people — roughly 55 million. As it covers such a large area the climate varies dramatically from very hot in the South to temperate in the North. The weather in Paris, which is where most producers and engineers are likely to find themselves, is similar to London although it can be one or two degrees warmer. There are so many treats in store for visitors to France that it would be impossible to list them all here. Everyone knows about the gastronomic delights of this country. Food to the French is almost a religious experience and wherever you find yourself in France it is well worth sampling as many of the local specialities as you can. For those who don't speak French, don't despair. The French take great delight in putting everyone else to ▶



A panorama of Paris from the top of the Eiffel Tower

shame with their mastery of the English language, although you may find that the further south you go the more you will need to rely on a few basic phrases. Generally, the French love it if you attempt to speak the lingo and most of the main studios will have at least one English speaking member of staff who will help you overcome any language barriers.

Paris has history, culture, art, romance, excitement and energy in such a seductive combination that most people are instantly hooked. But it also has its downside — traffic, for one, and increasingly beggars who will stop you in the street for the price of a cafe-cognac.

If you are planning to spend time in Paris you would be better off leaving your car at home because parking in the centre is even more of a nightmare than parking in London. Instead rely on the Metro which is cheap, quick, clean and efficient (it knocks spots off the London Underground), or if you want to get a real flavour of Paris life take a taxi — that way you get to see something of the city and experience first hand the unique rudeness of Parisienne taxi drivers.

Getting to France is very straightforward and there are numerous ways of doing this. For those wishing to travel by car, ferry crossings are frequent and inexpensive. Unless you plan to travel at the height of the holiday season you shouldn't need to book more than a few days in advance.

An average crossing from Dover to Calais takes one hour 15 minutes and costs roughly £250 for a standard return. This includes the cost of transporting your car and up to five passengers.

For an even speedier trip check out the Jetfoil or the Hovercraft which take 35 minutes on average but are sometimes subject to the weather. Once in France there are good interconnecting motorway

links to all of the major cities. These are signposted as A roads and the speed limit is 130 kph or 81 mph.

Most major airlines such as Dan Air, British Airways and Air France offer flights to major French airports including Charles De Gaulle (Paris), Toulouse, Montpellier, and Nice. An APEX fare to Paris works out at £98 return, provided you book two weeks in advance and stay over a Saturday night.

Flying time to Paris is roughly one hour, but remember there is an hour's time difference between the UK and France — and don't forget that Charles De Gaulle airport is about 30 minutes from the city centre by bus or taxi.

Transporting equipment to and from France will require an EC carnet. These are free and are available from the Customs and Excise office. Drivers will need a 'green card' available from your insurance company. You will not need an International driving license but a UK driving license is essential along with your vehicle registration documents, a GB sticker and a warning triangle which must be carried by all drivers, French police are not renown for their sense of humour and will not be very tolerant if you are caught without the right documentation. On the spot fines are a fact of life, as are motorway tolls. Obviously you will also need a passport, and some non-EEC citizens may need a Visa which can be obtained from the French Consulate General in London.

Mains power in France is 220 V, 50 Hz and unless you have the round two-pin continental plugs on your gear you will need a suitable adaptor. Most studios have these but it is worth picking up a couple of spares before you leave the UK just in case.

Money

Exchange rates fluctuate all the time but reckon on 10 francs to the pound for quick exchange conversions. France is more expensive than the UK so be prepared.

The studio scene

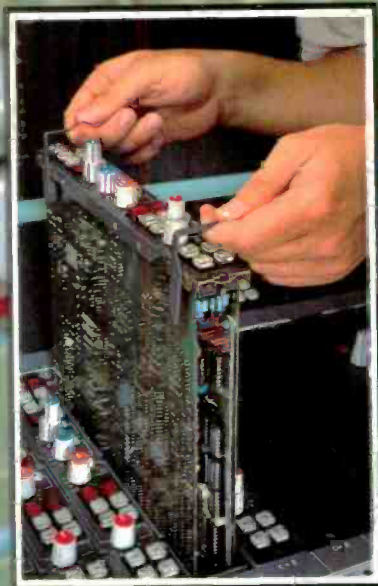
There are as many studios in France as there are in London, ranging from small 16-track facilities to fully digital 48-track complexes.

Unlike the UK where countryside residential recording studios are a large part of the scene, nearly all the main French studios are situated in or around the centre of Paris. There are a few glaring exceptions such as **Studio Miravel** near Cannes and **Studio Polygone** in Toulouse, but in France the capital is the heart of the recording industry — virtually all international record companies have offices in Paris which has always been known for its avant-garde attitude to the arts and music. At the moment the French recording scene is suffering from many of the same problems as its UK counterpart. Fewer projects are finding their way into studios because record companies are not signing so many new artists and are relying instead on CD compilations for their profits. To counter this France has been doing its best to attract international work and has been aided by relatively low rates in its studios and swinging tax disincentives in the UK. Projects from the US and Japan are finding their way into French studios although at the time of writing this type of work was thin on the ground. ▶

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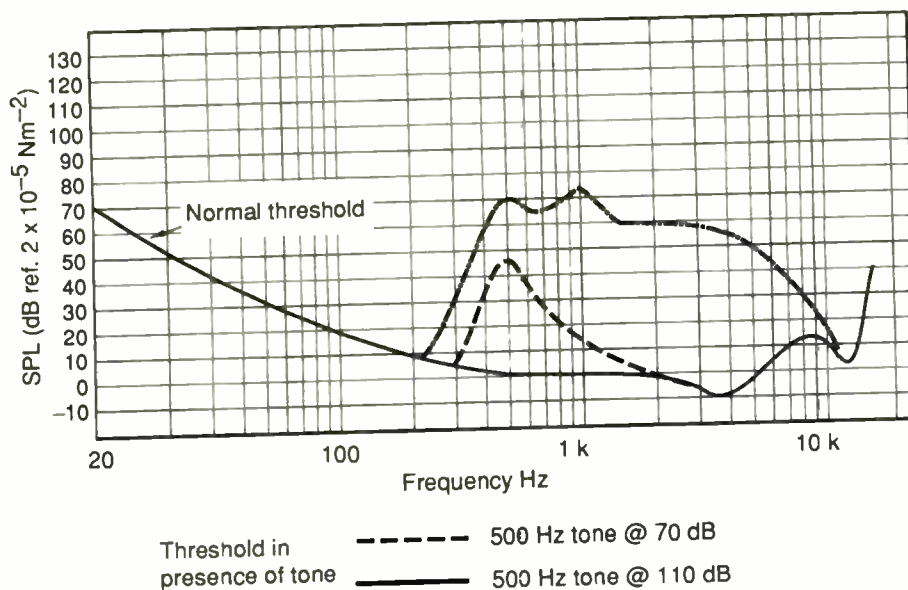


Fig 1: Masking effect of 500 Hz tone at two levels

in operation in broadcasting and telecommunications for years, but the current interest centres on what are commonly called 'perceptual coders' which rely on the use of a detailed mode of the human hearing mechanism to decide what is perceptually 'redundant' in the audio signal. These systems offer previously impossible degrees of reduction in data rate, and operate in real-time with a short coding delay. They are possible because the speed and cost of digital signal processing have now reached a point at which the required operations can be performed without waiting a day and using a room full of computers.

Approaches to perceptual coding

Comparison with lossless coding

There is an important distinction to be made between the type of data reduction used in some computer applications and that used in perceptual audio coders. The distinction is really between 'lossless' coding and coding which involves some loss of information. It is quite common to use data compression on graphics and text files in order to fit more information onto a given disk, but such compression is usually lossless in that the original data is reconstructed bit for bit when the file is decompressed. Methods are used which exploit apparent redundancy in the information, such as coding a string of eighty zeros by replacing them with a short message stating the value of the following data and the number of bytes involved. This is particularly relevant in single-frame bit-mapped picture files where there may be considerable runs in one state in each line of a raster scan and nothing is changing. One may expect files compressed using off-the-shelf PC data compression applications to be reduced to perhaps 25 to 50% of their original size, but it must be remembered that they are dealing with static data, and do not have to work in real-time.

The alternative to lossless coding is to exploit perceptual redundancy in a signal, this being especially relevant for data which represents situations that change with time, such as sound or moving pictures. Here one is normally dealing with real-time data reduction, rather than with the concept of a static file which may be 'unpacked' at a later date. The original data is not reconstructed accurately on decoding, but the

intention is that the information which has been lost will not be missed, since you either cannot see it or cannot hear it. Clearly this is guaranteed to arouse the emotions of professionals working in either pictures or sound, since they base their reputations on their ability to hear things which other people cannot, and thus any form of data reduction which cannot guarantee lossless reconstruction will be the subject of heated debate. If the potential user can appreciate how such systems work then he or she may be better able to make a judgement over whether or not the loss of information is important, since if it is really true that the lost information cannot be perceived then why not reap the benefits?

As usual, though, it is not a straightforward argument, since there is always a trade-off between data rate and sound quality, but there is considerable evidence from detailed subjective tests by trained listeners around the world (as exemplified in ¹ for example) to show that at the more moderate levels of data reduction (for example 192 kbit/s per channel) there really is no perceptible effect on audio quality, even after, say, five generations of coding and decoding. If you really want very low data rates such as 64 kbit/s per channel then you pay the price of some side effects.

Psychoacoustic principles of perceptual coders

The key advance in perceptual coders is in the full exploitation of auditory masking. Perceptual coders rely for their success on an analysis of the audio signal in the frequency domain to determine the energy present in different frequency bands, and, based on a model of the human hearing mechanism, determine the level of masking in each band. In other words, by looking at the audio signal's spectrum regularly it is possible to decide how much noise may be tolerated in each narrow frequency band before it can be heard. An explanation of the success of this approach requires a short excursion into psychoacoustics.

For some time now it has been appreciated that the phenomenon of masking exists. Intuitively we all know that it is difficult or impossible to hear some sounds in the presence of others, and this is the essence of masking. The exact nature of the masking effect depends on the frequency spectrum and level of the louder sound, it being clear that signals at a high level within one frequency band will effectively mask signals below a certain

threshold within the same band. Fletcher ² wrote about it in the 1940s, and the concept was refined particularly by Zwicker ³ in the 60s. It is related to the existence of so-called 'critical bands' in hearing, these having been determined by Zwicker as a number of discrete frequency bands (24 between 20 Hz and 15 kHz) within which the masking effect is constant and independent of the separation in frequency between the 'masking' and the 'masked' sound. Once the two sounds become separated in frequency by more than a critical band, the masking effect of one on the other is reduced, (although it may still be

present, especially above the frequency of the louder signal). The critical bands are not uniform in width, but a reasonable approximation of $1/3$ -octave bands is often used in models ⁴.

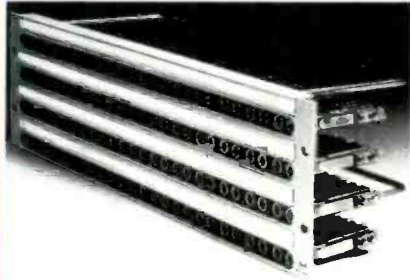
Forerunners of today's coders

Systems such as NICAM (near-instantaneous companding), ADM (adaptive delta modulation) and ADPCM (adaptive differential PCM) are the forerunners to today's perceptual coders. Indeed NICAM is now used widely for stereo television transmissions around Europe and Scandinavia, even though the degree of data reduction involved seems small compared with what is now possible. (An example of a typical dilemma in the fast-moving field of digital audio processing, since, although NICAM already seems dated, it should be realised that had we waited for lower bit rates we would not have had a digital stereo TV service for at least another five years or so.) The difference between such systems and perceptual coders is that they operate principally on the *broadband* audio signal, although they can still exploit perceptual masking to a certain extent. NICAM, for example, works by reducing 14-bit samples to 10 bits for transmission using a sliding window to determine *which* ten bits are transmitted, resulting in lossless coding at low signal levels (which don't need more than ten bits anyway) and an increased level of quantising noise at high signal levels. The increased noise at high signal levels is normally masked by the audio signal, and pre-emphasis is used additionally to reduce the annoyance of HF noise.

The problem with broadband systems is that the degree of data reduction cannot be made particularly great because the increased noise and noise modulation becomes too audible. Since the increased quantisation noise is spread over the whole audio band it may not be masked by certain signals which are constrained to fairly narrow bands, such as pure musical notes.

Fig. 1 shows an approximation of the masking effect of a 500 Hz tone at a sound pressure level of 70 dB and again at 110 dB, compared with the normal threshold of hearing in the absence of the tone. It will be seen that with the tone at 70 dB, lower-level sounds of a similar frequency will be masked unless they rise much above 45 dB, and that the masking effect is still present, but to a smaller extent, up to around 3 kHz. At 110 dB the tone masks lower level sounds below 60 to 70 dB over quite a wide frequency range (up to around ►

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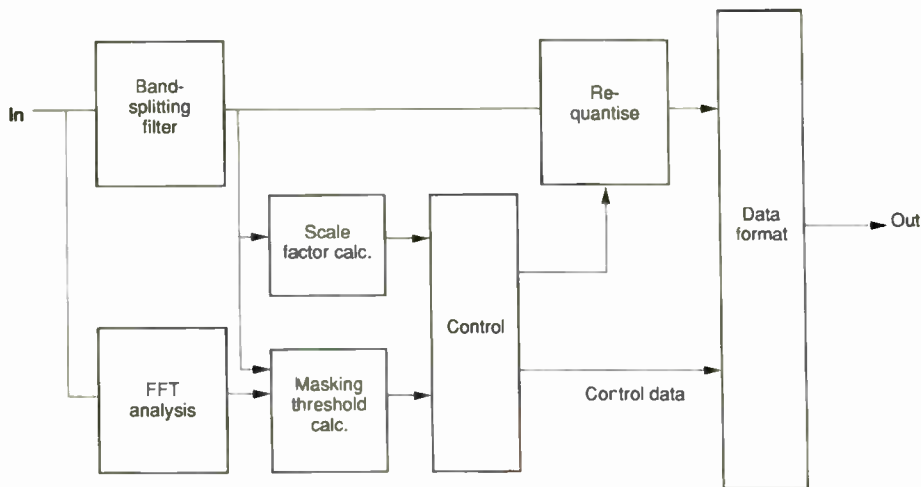


Fig. 2: Block diagram of basic perceptual coder

3 kHz), with the effect ceasing completely at around 11 kHz.

Time-domain masking also occurs, in that a sound may also be masked by another sound occurring a short time either before or after it. The effect is most noticeable over a period up to around 50 ms either side of the sound in question, but this too can be exploited in a perceptual coder.

Areas of concern with perceptual models

Most of the models of auditory masking are based on tests using pure sine tones, since it is difficult to develop exhaustive models based on complex sounds such as music or speech, and this has been a factor of potential worry since most perceptual coders will be used with complex signals, but in fact the problem turns out to be of only secondary importance since the pure tone situation is actually more critical than the complex signal case (in other words, better masking usually results from complex signals). Among others, Ehmer⁵ has produced data on the difference between the two cases, and manufacturers of perceptual coders have not relied solely on sine tone models but also on exhaustive subjective tests to determine the efficacy of their models.

Another important factor, highlighted by Gerzon⁶, is that the directionality of sounds also influences their masking effect. He suggests that in some cases 'unmasking' effects may occur when sounds arrive from certain directions, and that this is clearly of importance when designing systems for surround sound applications such as HDTV.

The question of how to *measure* the performance of a perceptual coder also rears its head, since it would be useful to have an objective test as well as a subjective one. It is a problem because clearly the measurement system also requires a model of the masking process, and to date this is most likely to be the same model as that used in the coder. Hey presto, all the systems pass with flying colours! If a more effective model were to be developed for the measuring system the chances are that it would immediately be used in the coder, in order to improve performance, with the same result as before.

Implementation in perceptual coders

Earlier it was stated that the problem with reducing the bit rate of an audio signal was that one either had to reduce the audio bandwidth or allow an increase in the noise level. The masking phenomenon described above is useful because it

allows for quantising noise to be increased considerably, *provided* that it is constrained to a relatively narrow range of frequencies. In fact the noise can be allowed to rise by amounts previously considered totally unreasonable, provided that it remains below the masking threshold of the audio signal concerned.

In a perceptual coder, the audio signal is split up into a number of frequency bands (sub-bands) using a bank of digital filters. Quite commonly, these sub-bands may correspond closely to the critical bandwidths described before, although some systems employ additional processes such as linear prediction which allow for the number of frequency bands to be reduced considerably. In a simultaneous side-chain the audio signal is subjected to a fast fourier transform (FFT) or equivalent transformation, the result of which is a collection of values representing the energy present in each of a number of frequency bands. These values are then processed by the masking model to determine, for the short portion of audio concerned, what the likely masking effects will be across the audio spectrum, with the result that a threshold curve can be determined above which quantisation noise must not rise. This control side-chain then acts on a requantiser in the audio signal path which allocates the bits available to each of the filtered bands (see Fig. 2). A delay results in this coding process, and this is between around 4 ms in the fastest system and around 55 ms in the slowest.

The requantisation is coarser than the original quantisation (because the data rate is to be reduced), and thus the noise level rises considerably, but the system aims to keep the noise well below the masking threshold in each band. Because the noise and distortion produced by the requantisation process is confined to narrow bands it is much more effectively masked than in broadband systems such as NICAM.

Since the output data rate of such systems is to be at a fixed rate, the system is constantly juggling a fixed pool of bits between bands, depending on the masking threshold. Consequently, the more bits there are in the pool the greater the 'headroom' which will result between the actual noise level in each band and the masking threshold. If there is a lot of masking headroom then the noise will be a long way below the masked threshold, and thus a number of repeated coding and decoding stages may be 'daisy-chained' before the noise becomes audible. A system operating at an output data rate of 192 kbit/s per ▶

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channel will have a greater coding margin than one operating at 64 kbit/s per channel. Indeed, in a typical system at 192 kbit/s (a compression factor of 4 when sourced from 48 kHz, 16-bit initial conversion) there is considerable masking headroom, and thus the sound quality would be very high. To all intents and purposes this could be considered 'transparent' coding and would be used for high quality applications in which some postprocessing was anticipated. The same system at 64 kbit/s shows evidence of some side-effects, with noise modulation clearly audible on some signals, but it should be remembered that this represents a compression factor of 12! Such a data rate is suggested more for speech, or where the highest sound quality is not the primary concern, although the quality is by no means poor on music.

At 64 kbit/s considerable advantage has been gained in some perceptual coders through the use of 'joint stereo coding'. Under such a scheme additional redundancy may be found between the left and right channels of a stereo pair, and again this can be exploited to increase the coding margin. Clearly this can only be used when the

two channels involved carry related programme material, but the result is considerably improved sound quality when compared with the dual mono mode. Although of greatest value at low bit rates, such a scheme can also be used to advantage at higher rates.

ISO/MPEG-Audio standard layers

The ISO/MPEG-Audio standard was developed alongside the group's main task of compressing the bit-rate of moving pictures. Without going into all the history, it is sufficient to say that a standard has now been formulated which is based principally on the French MUSICAM system, devised by the CCETT, incorporating elements of the ASPEC system, devised jointly by a large number of companies. The result is a number of so-called 'layers' of the standard, each of which has different merits. Sound quality can be traded off against complexity, data rate and coding delay. A useful summary of the standard and the

technology may be found in Brandenburg and Stoll⁷.

Essentially the three layers (I, II and III) are of increasing complexity. Layer I is the least complex, and is based on a simplified version of MUSICAM. It is intended for applications involving relatively low levels of data reduction, and is suitable for high quality consumer and professional applications at between 192 and 96 kbit/s per channel. (The data compression process used in Philips DCC is quite similar to ISO/MPEG Layer I.) It involves only a relatively short coding delay of around 8 ms. Layer II is more complex, being basically a refined version of the MUSICAM system and intended for use at data rates between 192 and 32 kbit/s. Layer III, incorporating the best parts of the ASPEC system, is the most complex, involving a longer coding delay (around 24 ms) and having a number of additional mechanisms to improve sound quality at very low bit rates, by dividing the spectrum into narrower bands, using nonuniform quantisation and incorporating Huffman coding (similar to that described earlier for computer file compression) as

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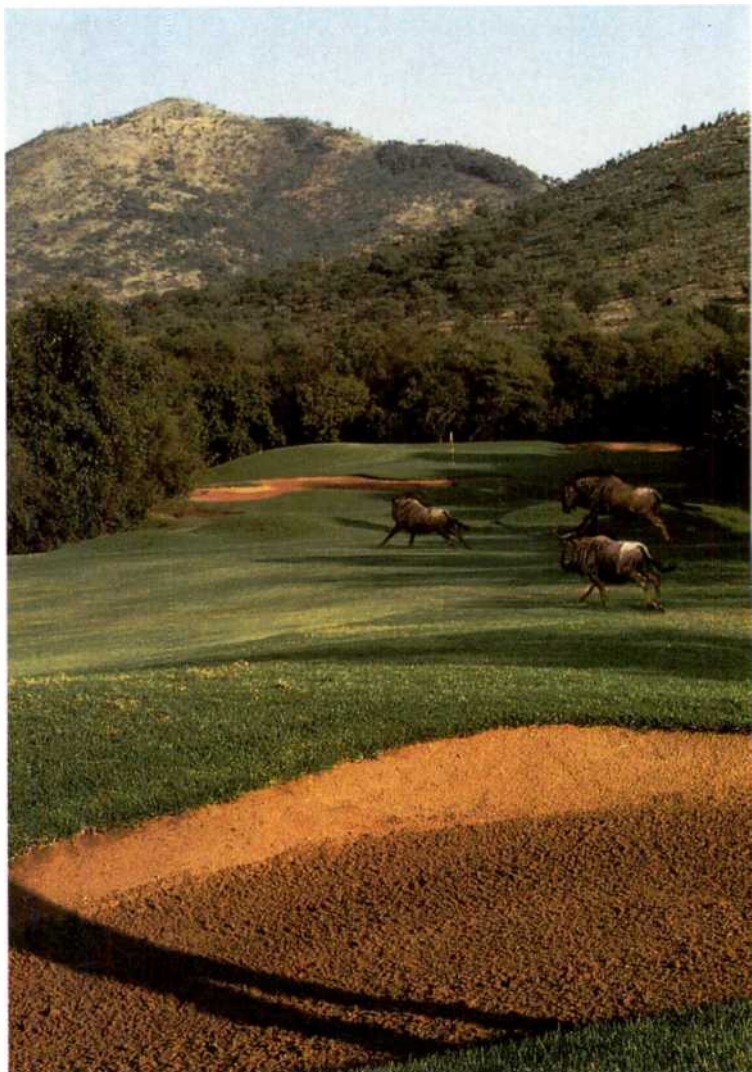
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Clearly, since the higher layers are more complex they require more computing power, and will thus be more expensive, but they allow coding at lower bit rates with less effect on sound quality. It is intended, for example, that Layer II will be used for Digital Audio Broadcasting (DAB) at 128 kbit/s per audio channel. Layers II or III are considered good for ISDN applications at 2 x 64 kbit/s for stereo, with Layer III providing the higher sound quality at this data rate.

Non-ISO commercial systems

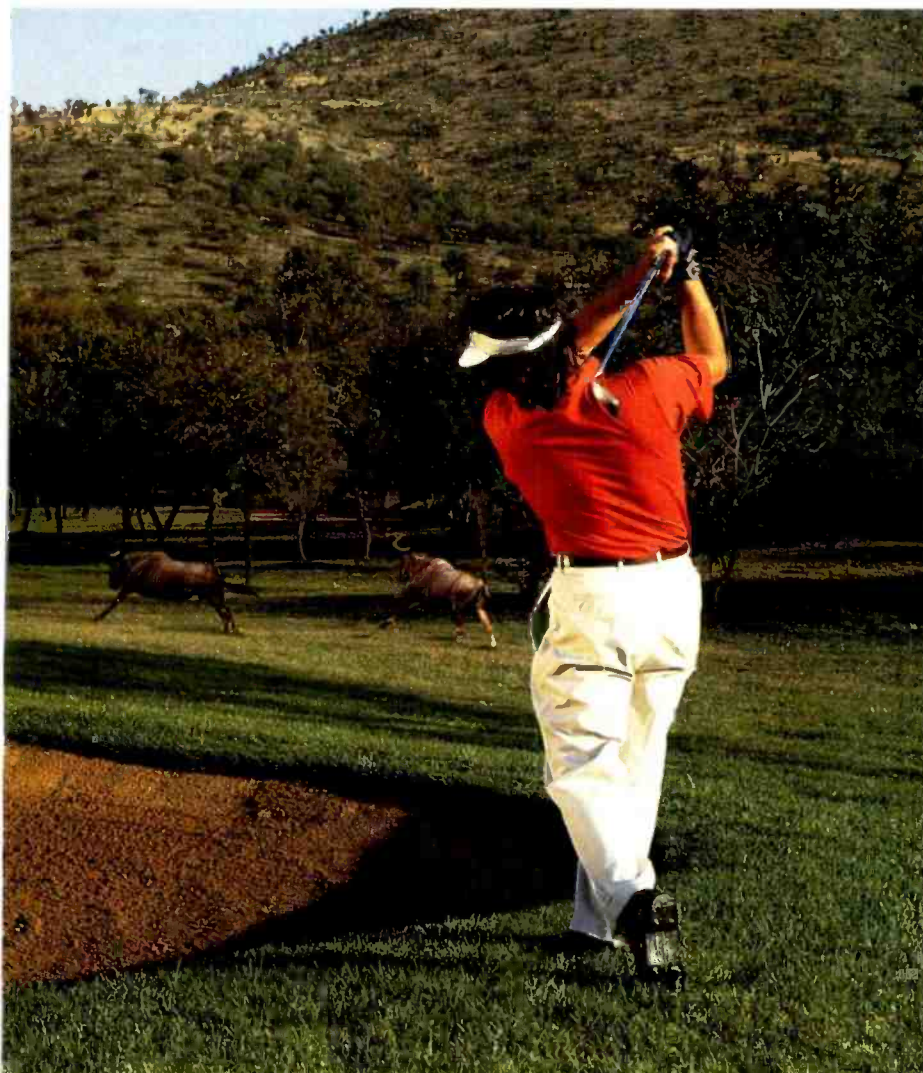
The standardisation process alluded to above has left companies such as Dolby Labs and Audio Processing Technology (APT) with commercial systems that do not conform to the ISO/MPEG standard. Often manufacturers are content to sell a system on its merits, without it necessarily conforming to any international standard. These

systems should be mentioned here because they are probably the most widely adopted commercial coders on the market, having been available for a few years now, either built into third-party products or as stand-alone codecs.

Both the Dolby AC-2 and the *apt-X-100* are high quality coding processes based on some or all of the principles described (although the *apt-X-100* system differs most from the others since it relies principally on linear prediction and ADPCM techniques derived from speech coding, rather than using a model of the masking process), and currently offer data rates of either 128 or 192 kbit/s per channel. The coding delay of the *apt-X* system is very low at around 4 ms, and, although it does not use a masking model specifically, the resulting noise levels have been shown to lie well below masking thresholds in examples⁸. Dolby manufactures both a 'moderate delay' coder (55 ms) and a 'low delay' coder in the AC-2 family, and again the resulting sound quality is very high. Both these technologies are capable of being extended to lower data rates, but this has not yet been implemented commercially to date.

ISDN applications

A single ISDN line from the local telecoms company offers two so-called 'B' channels at 64 kbit/s each, amounting to 128 kbit/s per line. The great advantage of ISDN is that the cost of the data transfer is the same as that of a phone call, and the destination can be dialled like an ordinary phone call, although the installation costs of a line are currently high. Terminal adaptors are becoming available commercially which will convert data from virtually any source into a format compatible with the ISDN link. One such adaptor, the *BR13*, which has been used by APT with their *DSM100* coder, is from a company called Ascend. This useful box synchronises three ISDN lines, therefore providing a data capacity of 384 kbit/s which is ideal for the highest quality compressed audio, either from a Layer 1 ISO/MPEG coder, or a Dolby or APT coder. The unit accepts an *RS449* data input from the audio coder and outputs to ISDN connectors. In a recent operation using the *apt-X-100* coder, such a system was used to transfer a high-quality stereo ▶



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The Dolby AC-2 system does not appear to have been used yet with ISDN applications, although Dolby Labs in the UK cites the use of digital T1 links (capable of up to 1.5 Mbit/s) in the United States between studios which need to transfer audio data regularly. These are not dial-up links, but fixed digital circuits. Lucasfilm has used this approach extensively for transferring high quality audio between its locations in the USA, avoiding courier fees for tapes or films.

A small number of MUSICAM coders are also available, allowing for output data rates to be switched between a number of rates. Again these can be interfaced to ISDN using a suitable terminal adaptor. The BBC World Service is experimenting with such coders for distributing programmes to remote transmitters at 64 kbit/s per channel. In a recent AES demonstration in the UK they demonstrated the coder at a number of data rates, showing how joint stereo coding was particularly beneficial at 64 kbit/s.

Clearly a 128 kbit/s ISDN link would provide a high quality circuit for mono audio programmes using, say, ISO/MPEG Layer II coding with no audible loss of audio quality on almost all programme material. For stereo operation a single ISDN link would not be adequate for the highest quality material, since some degradation would normally be apparent at 64 kbit/s per channel, although by using a Layer III coder in joint stereo mode the results can be surprisingly good. An alternative would be to use 2 ISDN lines and a terminal adaptor capable of synchronising them, which would then allow a total data capacity of 256 kbit/s, and thus the possibility for using Layer II coding at 128 kbit/s per channel, or using one of the other commercial coders.

Conclusion

Over the next few years the audio community will witness an expansion in the availability of low bit-rate coders for high-quality audio applications, especially now that the ISO/MPEG standard is all but finalised. There will also be an expansion in the availability of ISDN links as a means of transferring digital data over distances at phone-call costs. High quality audio can be transferred over such links with little or no reduction in sound quality, and this will clearly have important applications in broadcasting and studio environments. ■

Francis Rumsey

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LINE INPUT

Simon Croft talks to ISDN users



One of the latest commercial codecs. Dialog 4's Musictaxi

In computer orientated industries ISDN is held to be an acronym for Innovative Solutions users Don't Need. Amusing though this may be it seems quite wrong when the broadcast sector is considered. It allows a more listenable signal than the telephone line and presents an economical alternative to the full-time leased line. On the surface at least, ISDN is almost 'something for nothing'.

The BBC makes extensive use of ISDN, particularly for relaying overseas news. It has codecs installed at BBC locations in New York, Tokyo, Paris, Berlin, Singapore, Hong Kong and Rome, as well as links with other broadcasters including CBS, ABC and several in Europe.

For these applications, the BBC uses well established G722 technology as its 4/1 compression and resultant 7 kHz bandwidth is quite acceptable for speech.

James Mandlecott of the BBC World Service Projects and Planning department also thought it 'all right' for music where the transmission medium was shortwave radio.

But the World Service is starting to use the considerably superior MUSICAM standard for some applications. It is establishing a permanent link from Bush House in England to stations in France supplying a mixture of pop and news via a 128 Kb MUSICAM link. 'It is cost-effective and the quality is excellent', noted Mandlecott.

The World Service hopes to use the same compression technology for transmissions via satellite and is also considering the use of ISDN for OB links between the Albert Hall and Bush House some miles away during the Proms season of classical music.

The idea would be to use 286 Kb or 348 Kb for maximum quality. 'That is the equivalent of six local telephone calls for three hours. It's peanuts.' The BBC would even get the benefit of the Off Peak rate levied in the evenings.

Although the initial investment in interfaces was high, Mandlecott considered the technique 'a bodge that works' and a realistic alternative to spending vast sums to achieve greater bandwidths.

Peter Jackson, chief engineer at London-based Capital Radio, agrees. The station runs several services and regularly uses ISDN for sports commentary and sports news, which are normally transmitted on AM. It has encoding equipment installed at five football grounds in the UK and also runs regular links with fellow stations Radio Clyde in Scotland and Radio Piccadilly in Manchester.

Jackson said that although G722 was 'an internationally agreed standard' it was also 'a little old fashioned'. Capital Radio opted for the four-band apt-X100 system rather than the two-band alternative because 'we think it sounds better'.

Compared to the BBC, independent Capital Radio is a small operation. Jackson said the company had made the investment in codecs and adaptors 'there is a pure economic return compared to leased lines. There is a good return over the lifetime of the equipment, about five years'.

Jackson pointed out that for the independent,

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ISDN spelled flexibility, particularly for new services where lines were used for a matter of minutes a day.

In the longer term, Capital Radio will be looking at 'individual occasions' where it might be possible to use ISDN with a greater bandwidth to relay complete programmes including music.

Before the technology had reached its current level of sophistication Capital Radio had broadcast a series of five breakfast shows from Eastern Australia. 'The equipment was not available at the time', said Jackson. 'But if it had been we could have bought the equipment on the back of the programme and still had cash in the bank.'

Next time round there was 'every likelihood' that Capital Radio would take the plunge into MUSICAM or ISO Level 3. On the other hand, 'a 15 kHz bandwidth is perfectly conceivable with two 64 Kb dial-ups in parallel'.

In fact with a high compression codec that will 'squeeze' the voice into 4.8 Kb, 'the prospect exists of full stereo B with stereo reverse-cue plus four-way talkback into two 64 Kb channels, which is a good deal'.

Jackson pointed out that ISDN 2 gives the capacity for two simultaneous links. This is not, he added, a line each way but two full duplex lines, just like a normal telephone call.

This also raises the issue of the processing time penalty paid on higher compression rates, which is the main drawback in going from 4/1 to 8/1 in some applications. The delay end to end from a 4/1 ration is about 9 ms. That, said Jackson, is a 'round trip' of 18 ms, 'about the most you can get away with without a clean feed'. Commentators are aware of some effect but it is not enough to distract them while talking.

MUSICAM or ISO Level 3, on the other hand is likely to create a delay more like 40 ms end to end, necessitating more elaborate cueing arrangements and wiping out some of the benefits of ISDN.

In reality, there is a greater danger as compression is adopted for storage as well as transmission. For example, the MUSICAM compression system is used in DCC technology and many hard disk or optical disc-based system also use high levels of compression.

Where the same type of compression used, the build up of artifacts is not too drastic. Mandlecott's experience at the BBC has been that material can survive a couple of passes through G722 before it becomes unlistenable. At 384 Kb, MUSICAM encoded material remains presentable for around 15 cycles, although this plummets to 'three or four' cycles at 128 Kb.

But nobody is making any predictions as to what happens when material is put through a random succession of disparate encode-decode cycles.

Some sources suggest that programme that sounds quite acceptable when it is decoded by the correct process can exhibit 'disastrous' degradation if subsequently put through a differing encode-decode cycle. Which raises the possibility that while ISDN is a tasty proposition for broadcasters, problems with different compression routines could prove that there is no such thing as a free lunch. ■

Simon Croft



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MEASURE FOR MEASURE

When we listen to stereo in a room, huge errors in the sound are created by the inevitable defects in both loudspeakers and in the room itself. Philip Newell's extended series¹ has dealt with the problems of finding a way through the necessary compromises and trade-offs in getting adequate monitoring in the studio, but a new actor is now emerging: the use of digital speaker and room equalisation.

This new technology has become commercially affordable due to the recent strides in bringing down the cost of and increasing the processing power of digital signal processing (DSP). This has allowed new kinds of equalisers to be devised that are not viable using analogue technology, and has also allowed the process of measuring the defects of a speaker and a room, and of setting up the equaliser to be largely done automatically by the digital signal processing, with little human intervention apart from placing the measurement microphones.

The basic idea of digital room and speaker equalisation is easy: one simply measures the frequency and phase response of the room by using a measurement microphone, and compensates for any measured defects by devising an equaliser to feed the loudspeaker which exactly undoes the measured frequency and phase response errors. Such an equaliser is very complicated, but with modern DSP technology, implementing such an equaliser is possible at a cost that is already within the price range affordable by a top-end studio, and this cost is coming down very rapidly.

If things are this easy, and if the result is a perfect frequency and phase response, what is the problem? The problem is threefold. First, the above procedure only equalises one point in the room, and in general may well make the frequency and phase response in other points in the room a great deal *worse* subjectively. Secondly, the required equalisation may attempt to compensate for deep 'suckouts' in the measured frequency response by correspondingly huge boosts, which may well exceed 30 or 40 dB, and such boosts will overload any amplifier and loudspeaker, and be hugely audible at other room positions. Thirdly, even at the one point, the so-called 'inverse' room equalisation has to be done accurately; even tiny errors that are, say, 30 or 40 dB down can be highly audible if of the wrong kind as noted in a previous article by the writer², and the kind of errors produced in digital room equalisers are typically of the most audible kind.

All these problems mean that the design of a digital equaliser for speakers and rooms involves considerable subjective and psychoacoustic know-how. Because of the above-listed problems, not all defects in loudspeakers and rooms can be equalised away completely, and no one should

Recent developments in DSP and knowledge of the psychoacoustics of room perception mean digital room equalisation technology is about to become widely available. Michael Gerzon probes the problems and benefits

expect miracles from the new technology. However, the new technology can give significant benefits not available using analogue technology, and this article reviews some of the problems and benefits, both as they affect studio monitoring applications and domestic applications. Within the next ten years, with falling costs, we can expect most domestic and in-car systems to incorporate room-equalisation technology, and this technology is also likely to become widely used in studio monitoring.

The players

While it is not known how many companies are working on room equalisation products, it is known to be an active research area, with a lot of competition and secrecy. The US company Sigtech is, at the time of writing, the only company with a commercial product in the field, specifically targeted at studio monitoring systems. B & W loudspeakers are about to launch (provisional launch date is September) a room and speaker equaliser incorporated into a high-end hi-fi preamplifier aimed at the high-end domestic market, although it may well also be useable in professional applications also. Marantz has launched a high-end domestic hi-fi 'effects' processor in Japan which incorporates a limited degree of room equalisation capability, although this is thought not to be of the same sophistication as that of other products.

There is also a consortium of US companies jointly funding a room equalisation research and development project headed by Floyd Toole, but little is known about their work, except that it may still be a year or so away from being implemented in commercial products; however, with the calibre of people involved, one may expect interesting results from this group. Bob Stuart of Meridian, an English high-end hi-fi company, have launched a loudspeaker incorporating digital compensation for loudspeaker defects (but not room defects) onto the market. While this product does not incorporate all the potential benefits of digital equalisation, it does illustrate the degree of

improvement obtainable by designing a loudspeaker and associated digital equaliser as an integrated system.

In order that the reader not be misled, the writer must declare an interest; he has been employed as a consultant on the B & W project, although the main work was done by Peter Craven and Colin Bean. However, I hope that my comments are not biased in favour of particular contenders.

It remains to be seen whether other room-equalisation contenders emerge from the woodwork either in Japan or from continental Europe. In particular, it is not known how close participants, notably KEF and Bang & Olufsen, in the Eureka Archimedes room response analysis project are to devising such products.

What can be equalised?

Traditionally, loudspeakers and rooms have been equalised using graphic (or, more rarely, parametric) equalisers, but it is widely recognised that at best these can give only a limited improvement, and at worst can make the sound markedly worse. The problems of loudspeakers in rooms are not merely that the frequency response is not flat, but that there are time-domain problems with delayed reflections and reverberation. Graphic equalisers generally cannot tackle the time domain problems, and even the frequency response errors can only be corrected to a limited degree due to the 1/3-octave resolution of typical graphic equalisers, which is coarser than the critical-bandwidth resolution of the human ear. Empirical experience of equalising PA systems with parametric equalisers suggests that equalisers with a Q of about 10 are required to tame simple frequency response effects well subjectively.

If one wishes that the speaker and room equalisation not introduce audible side effects that are worse than the original problems, one must ask what aspects of the sound should be equalised, and what should be left alone. This is the problem

that all who have worked on room equalisation have had to tackle. There seems to be a broad consensus on many aspects of what should be done^{3,4}, but the fine details still are the subject of research and possible controversy.

Even after deciding what aspects of the sound to equalise, the fact is that the ideal equaliser required is still very complex and expensive (typically using of the order of 50,000 taps per channel), so that there is also the problem of how to approximate the ideally desired equaliser with cost-effective DSP technology affordable to the customer, and this also involves practical trade-offs that vary between different systems.

We shall first look at two problems that affect what we can in practice equalise. One thing we want an equaliser to do is to flatten the frequency response, but *not* at the expense of boosting dips or notches in the frequency response to the point where the boost causes amplifier and speaker overload or massive amounts of the boosted frequency at other listening positions. To overcome this problem, one needs to measure the original room frequency response, and not to equalise for the actual measured response, but for a 'regularised' version of the room response which, in some manner, has filled in deep troughs of the measured response.

It is a fortunate fact of life that the ears are much less sensitive to the odd trough in a frequency response than to peaks of similar magnitude, so that equalising in a manner that leaves some of these troughs in the final overall response, while equalising out the peaks, gives a subjective improvement, while avoiding the problem of excessive boosts. One must therefore recognise the necessity of not attempting to equalise fully the troughs or dips in the original frequency response.

Digital room equalisation can, in principle at one room point, not only equalise the frequency response, but also correct time-domain problems such as reflections off walls and reverberation. The trouble is that this correction can only be done for one point and that the time domain response is made much worse elsewhere. So the problem arises of how much of the room's time-domain problems can in practice be corrected, and how. Again there seems to be broad agreement on some aspects of what can and cannot be done, but there are some surprises which need a degree of detailed explanation to be fully understood.

For those who understand the concept, we shall conclude in the following that the basic room equalisation needs to be of minimum phase type, although there are significant exceptions to this if the best possible results are to be obtained.

Preresponses and minimum phase

One of the basic principles of science is the law of causality, that asserts that an effect cannot precede its cause. A *causal* filter is one in which the output does not emerge earlier than the input, and all real-world filters are of this type. In the digital domain, by adding an overall time delay so that one can have responses occurring before the ►

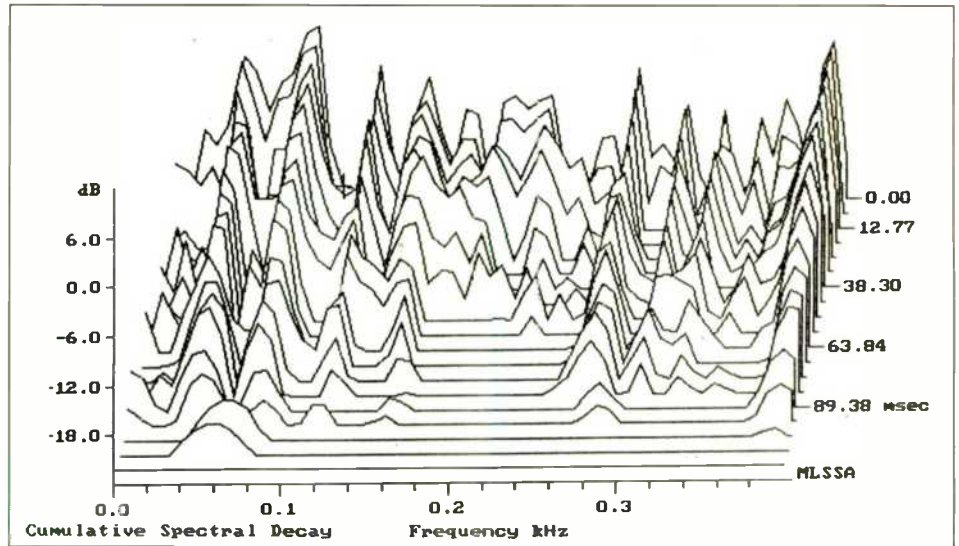


Fig. 1: Waterfall response of a loudspeaker in a room

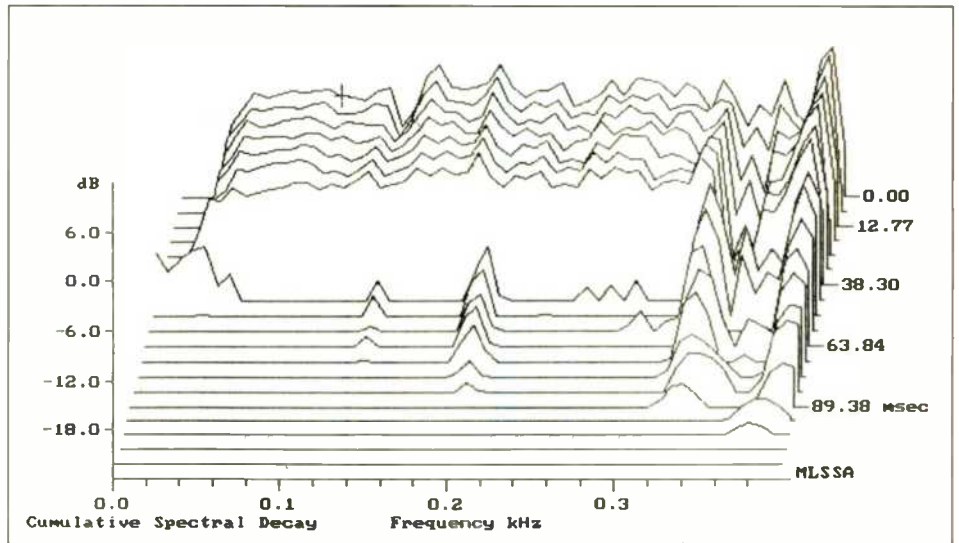


Fig. 2: Waterfall response of the system shown in Fig.1 after acausal equalisation. Note the removal of most of the decay artifacts

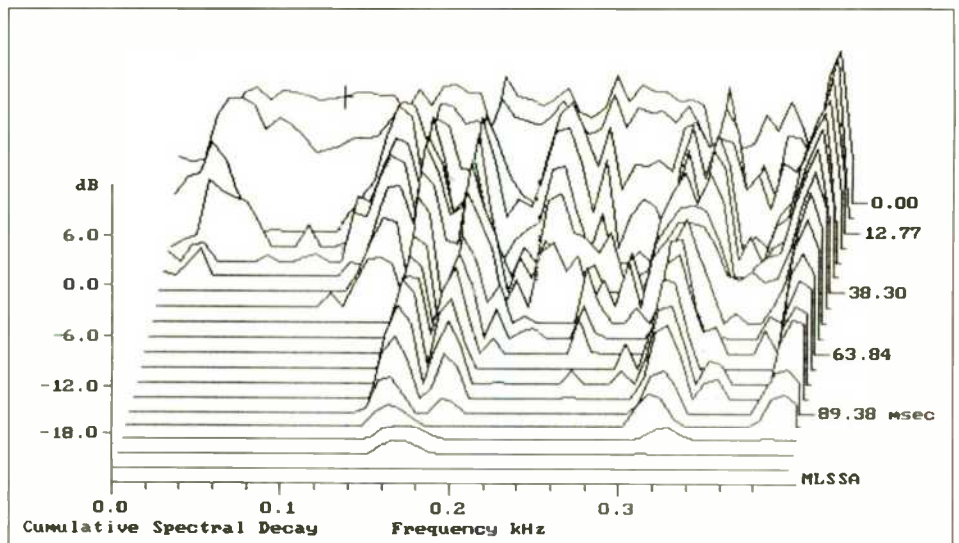


Fig. 3: Waterfall response of the system shown in Fig.1 after minimum phase equalisation. Note the flat initial frequency response and the removal of the lowest frequency decay artifacts

delayed input (although after the actual input), one can simulate the effect of an acausal filter (that is one whose output responds before the input arrives), at the expense of having to wait a little.

A filter is said to have an inverse filter if the result of cascading the filter with its inverse is to give back the original input signal, that is the inverse filter exactly compensates the effect of the original filter. A causal (real world) filter is said to be *minimum phase* if its inverse is also causal. The type of filters found in most graphic and parametric equalisers are minimum phase, but it is found that, except for almost anechoic rooms or very close to the sound source in normal rooms, the measured room response is almost never minimum phase⁵. This means that any equaliser that attempts to compensate fully for the response of a room will be acausal, and will have a prereponse occurring before the arrival of the input. This in itself is not a serious problem with digital equalisers, as we can incorporate an overall time delay in the equalisation that allows time for this prereponse to occur. The real problem is that at points in the room other than that of the measurement microphone, this prereponse and the room response do not completely cancel each other out, so that one hears a 'pre-echo' or prereponse before the main sound arrives. As noted in refs. 2 and 4, such prereponses sound highly unnatural in general, and are very audible. Some writers³ claim that not more than 2 ms of prereponse should be allowed, although in the B & W project, at least one important exception to this rule has been identified.

While I shall deal with exceptions later, generally, room equalisation should be of minimum phase type to avoid problems with audible prereponses when cancellation of the room response is imperfect. Minimum phase filters have a lot of special properties that arise from long-known deep mathematical results, and there are many ways of ensuring that a room equalisation filter is minimum phase.

Many workers in room equalisation have derived minimum phase filters using what are termed LMS algorithms (Least Mean Square algorithms), based on work in the theory of linear prediction. Roughly speaking, such algorithms attempt, for a given complexity of filter, to minimise the error energy caused by the approximate inverse filter. While the theory of LMS filtering is widely available and understood, it is unfortunate, as noted in reference 1, that the ears do not seem to respond to the total power of the error in equalisation, but rather seem much more sensitive to some kinds of very low level errors (even those as much as 80 dB down) than to other errors of a very much larger magnitude, which seem almost innocuous. Thus LMS algorithms, while they minimise an objective measure of the error in equalisation, do not minimise the subjective error, and in fact are found to be very poor indeed subjectively.

For this reason, the B & W project has adopted an alternative approach to deriving minimum phase equalisation filters that incorporates subjective criteria. One of the remarkable things about minimum phase filters is that the overall frequency and phase response of the filter can be computed, by a mathematical method involving the Hilbert transform, just from a knowledge of the amplitude frequency response alone. In other words, if one knows what frequency response one wishes to correct, the phase response aspects of the filter follow automatically.

This has the very convenient side-effect that one can devise the equaliser by computations in the

frequency domain rather than in the time domain. This allows one to regularise the measured room frequency response to eliminate dips in the response, and also to smooth the measured frequency response at higher frequencies, where it is found that one does not wish to equalise too fine details in the frequency response (since this varies from point to point in the room). While the $\frac{1}{3}$ -octave equalisation of conventional graphic equalisers is too coarse, Sigtech have found that equalisation to $\frac{1}{4}$ -octave resolution works well in their system at higher frequencies.

Another advantage of being able to derive minimum phase equalisers just from a measurement of the amplitude of the frequency response is that one can measure the spectral power (frequency) response at several points across a listening area, and equalise for the averaged frequency response at all these points, rather than for the frequency response measured at just one point. This gives better results across the whole listening area, at the expense of less good results very near one single point. It has been found in practice that, under domestic conditions, averaging for six points across, say a settee, can give useful room equalisation across the listening area. Sigtech's studio system, on the other hand, uses a single measuring point, and is designed to equalise the optimum monitoring position in a studio situation.

One advantage of averaging the measured response at several room points is that this tends to 'fill in' measured dips in the frequency response. Nevertheless, it is found that regularisation of the troughs is still necessary to avoid excessive boosts, since by no means all troughs are adequately filled in by averaging over just six points.

Time-domain effects

By using minimum phase equalisation, which depends only on the measured amplitude frequency response, and not on the measured phase response, one appears to have thrown away any possibility of compensating for time-domain effects in the room. One of the surprises revealed by B & W's research was that minimum phase equalisation can significantly improve the time-domain performance of a room as well. This is an interesting phenomenon that is not completely understood. It appears that the fine detail in the frequency response, especially at frequencies below 300 or 500 Hz, contain a great deal of hidden information about a room's time-domain response. Conventional minimum phase equalisers, such as graphic or parametric equalisers, ignore this fine detail so do not have much effect on time-domain behaviour, but a minimum phase equaliser compensation for fine details of frequency response appears to cancel out about half of the unwanted time-domain information. In particular, minimum phase equalisation approximately halves room reverberation time at low frequencies near the measurement positions — a useful improvement at those lower frequencies that are most difficult to control by standard treatments.

This improvement in reverberation time behaviour only occurs if the length of the response of the equaliser filter is significantly longer than the reverberation time of the room; the B & W filter has, at low frequencies, a total length of about 1 s, so as to cope with most domestic rooms, whereas the Sigtech filter has a length of about 50 ms, and so is unable to respond long enough to improve the reverberation time. This observation brings us to the problems of implementation of room equalisers, since the reason why Sigtech's

processor has such a short filter is to do with the limitations of affordable DSP technology.

DSP complexity

The ideal equaliser for a room would use a filter with about 50,000 taps per channel, but such filters are extremely expensive at present, although costs are falling fairly rapidly. Because of the high costs, Sigtech's current filters use a more affordable 2200 taps, which is enough to do a useful LMS equalisation of at least a monitor room acoustic, but not enough to control effects more than 40 dB below the wanted signal, such as the decay tail of reverberation in a domestic room.

The B & W approach has reduced processor complexity by observing that it is only at low frequencies below about 500 Hz that one requires a long filter length, and that the reduced frequency resolution needed at high frequencies means that a much shorter filter length can be used for the higher frequency equalisation. Oversimplifying somewhat the actual complex process used, by using a process well known to communication engineers called *decimation*, the signal is first split into two or three frequency bands, one below 500 Hz, one between 500 Hz and for example 3 kHz, and one above 3 kHz, and each frequency range is represented at a different sampling rate. For example (and these figures are illustrative and not necessarily those used in the forthcoming commercial product), if the input is sampled at 48 kHz, so is the band above 3 kHz, whereas the band between 500 Hz and 3 kHz would be sampled at just 6 kHz sampling rate, and the band up to 500 Hz at a 1 kHz sampling rate. At a 1 kHz sampling rate, one only needs one forty-eighth of the memory to store information for a filter of a given length as compared to what one needs at 48 kHz, and also, one has to do computation only one forty-eighth as often, since the samples occur that much less often in time. This means that one only needs to do $1/(48^2)$, which equals about $1/2300$, of the amount of computation that is required at 48 kHz sampling rate for a given filter length below 500 Hz. A significant saving is also made (by a factor 64) for the shorter filters required in the mid-frequency band, and only in the highest frequency band, where the full sampling rate is used, is no saving obtained. However, in this highest band, only very short filters are required for equalisation purposes, so that great processing power in the DSP chip is not required.

The use of decimation to reduce the processing requirements to fit available DSP chips is a key to both the relatively low cost of the B & W processor and its very long filter length. The price paid for this cost saving is a considerable increase in the development effort required to get such a complicated signal processing algorithm to work. There is a considerable amount of work in getting high-quality decimation algorithms to work well, and it is doubtful that the B & W system would have been practical without the use of some very powerful software tools devised by Peter Craven, who is almost unique in combining such advanced software skills with a detailed knowledge of the subtleties involved in high-quality audio.

Room measurement

The measurement of the room response is not a trivial operation, and considerable work has been done on attempting to do this well. In principle one could simply feed the loudspeaker with a sharp impulse, and measure the impulse response at the microphone, and then use standard well-known ►

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mathematical methods to derive from the impulse response the room response. The problem with this is the fact that the power in an impulse that can be fed to a speaker and amplifier without overloading them is limited, and any room noise will degrade measurement accuracy rather badly. Even in a really 'quiet' control room, the amount of low frequency noise due both to air conditioning and, surprisingly, to natural variations in atmospheric pressure due to high-frequency components of the weather, is enough to make impulse measurements totally useless. The situation in domestic rooms, where noise levels are often much higher, is even worse.

For this reason, workers in this area use test signals that have much higher power than an impulse, usually either a pseudo-random test sequence or a chirp test signal. By a process known as de-convolution, it is possible to derive an impulse response from the output of the measurement microphone when such a sequence or chirp is fed to the speaker. Such test signals often last one or several seconds in duration, and have many thousands of times as much power as a single impulse, and give a correspondingly improved signal-to-noise ratio in the measurement. Even so, averaging over a number of measurements is required to minimise noise errors.

Another problem in making the measurement is that if too high a level of test signal is fed to the speakers, they will produce distortion which will cause significant errors in the measurement. Especially in the B & W case, where errors 60 dB down can affect the reverberation time of the equalised room, it has been found important to use measurement procedures that minimise the effect of nonlinear speaker distortions.

Even after all these precautions are taken, noise problems remain severe at the lowest frequencies, especially below about 50 Hz, and sophisticated methods are required in deriving the room equalisation to minimise the effects of such errors on the derived room equalisation. Equalisation of the low frequency behaviour of both the room and the loudspeaker is important subjectively, and cannot be done well unless a very long filter length of the order of a second is used — another reason why the B & W system uses decimation in order to achieve the required low-frequency resolution and filter length.

Bass response

The problem of bass response is particularly important with digital equalisation. All loudspeakers have a roll-off in the extreme bass, almost always of a minimum phase type. Minimum-phase bass roll-offs produce very severe phase distortions, not only near the roll-off frequency, but several octaves higher, and these phase distortions cause a 'woolly' quality that hitherto has been unavoidable. The only way in the past to reduce these phase distortions has been to lower the frequency at which the roll off starts, but even lowering the roll-off to 5 Hz, which is difficult, is not enough to eliminate audible phase distortion.

Any attempt to lower the low frequency roll-off by minimum phase equalisation requires enormous degrees of bass boost, which can soon result in speaker or amplifier overload. While the digital signal processing algorithm can be designed to incorporate a limiter to prevent speaker or amplifier damage, such a limiter results in a loss of sound accuracy at high levels.

While it is not practically possible to compensate the low-pass phase distortion by analogue filters

(such filters would use several thousand precision analogue components!), it is possible to design in the digital domain a phase-equalisation all-pass filter, which does not affect the low-frequency amplitude response, but which does eliminate the phase response problems caused by the low-frequency roll-off. The actual length of such phase-compensation filters is a substantial fraction of a second, and so would be very expensive to implement as an FIR filter at full sampling rate, but such a filter is quite economical to implement at a lower decimated sampling rate.

Since the very low frequency performance of a speaker in a room is very difficult to measure, even in most anechoic chambers, the phase compensation is derived from the theoretical bass alignment of the loudspeaker, possibly supplemented by near field measurements. Adjustments can be provided for standard speaker bass alignments on the control unit, with preset settings for particular models of loudspeakers.

The subjective effect of phase compensation of the bass from loudspeakers is very marked, giving a much tighter and more 'punchy' quality, with greater transparency, and interestingly a subjective extension of bass response of at least half an octave. The improvement is audible even on loudspeakers with a very high cut-off frequency, such as Quad electrostatic designs. The bass phase compensation, which as already remarked has a significant effect even in midrange frequencies, is a very nonminimum phase filter, and has very substantial preresponse. Despite this, it is found not to be subjectively disturbing in the way that spurious room preresponses are.

The benefits of bass phase equalisation are considered, by those who have heard it, to be a substantial improvement over what was hitherto possible with analogue technology, and digital equalisation provides a way of improving bass performance without going to ridiculously large, space-consuming, power-hungry, monster speakers, and is certainly a much cheaper route.

Crossover compensation

While the room requires minimum phase equalisation to avoid preresponse effects, phase anomalies in loudspeaker systems can respond well to digital nonminimum phase compensation. Besides the poor phase response caused by bass roll-offs, the other main cause of phase distortion in loudspeakers is the cross-over networks. With analogue circuitry, it is impossible to design a crossover network that has both an adequately rapid cross-over rate and a flat phase response⁶. Again, experiments have shown compensation for the phase response anomalies of the crossover networks, while ignoring any phase response anomalies at the same frequencies of the room, gives a marked improvement in subjective quality. The all-pass behaviour of cross-over networks is not minimum phase, so that the compensating all-pass filter is acasual, with a preresponse.

Such digital equalisation of nonminimum phase speaker phase anomalies is subjectively important, and is another exception to the earlier rule that speaker and room equalisation should be minimum phase. Many high-rate crossover networks have a Q higher than 0.6, which causes audible colouration. Correction of this colouration by digital equalisation gives a much cleaner sound even on recordings, such as UHJ encoded material, that have passed through low-Q all-pass distortion networks.

In general, equalisation of known sources of

phase distortion has proved to be subjectively much more beneficial than attempts to flatten the amplitude response of loudspeakers. It is quite easy, using digital equalisation, to obtain a ruler-flat textbook perfect frequency response from a loudspeaker, but in general this will not sound any better (and may sound considerably worse) than the unequalised speaker. When digital equalisation comes into widespread use, be extremely suspicious of loudspeakers incorporating digital equalisers that have a measured ruler flat on-axis anechoic response; such a measurement may please the advertising boys, but is likely to be at the expense of more important speaker characteristics!

Combining room and speaker EQ

Given that the requirements of equalising the speaker and the room are so different, there is the problem of separating the effects of the speaker and the room when measuring the room. Naively, one might think that all one has to do in measuring the room is to send a test signal through an equaliser for the speaker before sending it to the speaker and room and that the result picked up by the microphone will be the room response. Unfortunately, it is not quite that simple, partly because one has compensated the low frequency phase portion of the speaker response, but not its amplitude, and any attempts to compensate for the amplitude error in the minimum phase room equaliser will also add additional unwanted compensatory phase distortion. The method of measuring the room component of the equalisation requires careful modelling of the speaker equalisation within the measurement algorithm, and involved quite a lot of thought in the B & W development programme.

The additional problem of excessive measurement noise at low frequencies requires careful design of the equalisation algorithm based on in-room measurements, so that the benefits of speaker bass equalisation are retained and so that room bass resonance effects are controlled without spurious noise artefacts.

In the B & W system, it has been found that attempts to equalise the loudspeaker and the room as a single system by overall minimum phase equalisation do not give optimum results, and that quite a sophisticated separation of these two components is required, with quite a complicated joint equalisation strategy. There appear to be significant psychoacoustic differences between equalisation errors generated over a small volume such as that of a loudspeaker and those generated over a large volume such as a room. There is still a lot to be learned about the precise dividing line between loudspeaker and room effects.

Tolerances

Providing a good average equalisation over a chosen listening area does not guarantee that results will always be good. For example, variations in air temperature and humidity change the room response, and one might fear that the room equalisation will no longer work. Such fears have proved largely unfounded, as have fears that the effect of moving a small item of equipment or another person coming into the room might be to wreck the effect of room equalisation. While obviously, it is best to determine the room equalisation under the actual conditions of listening, it is found that the results are fairly tolerant of typical small changes over the useable ►

listening area.

This tolerance to normal small changes in room acoustics arises largely because the equaliser does not attempt to correct fine details of the room response at higher frequencies, which are most subject to change.

Careful room equalisation will give significant improvements over the listening area it is designed to serve, which in domestic use may be around 2 metres wide, but in practice it is also important that the equalisation should not significantly degrade results away from this optimum listening area. It has been found in the B & W programme that well-behaved equalisation algorithms can still give improvements away from the optimum listening area, to the improvements given by speaker phase equalisation, and partly due to the equaliser 'sucking out' obtrusive low frequency room resonances.

It is obvious that room and speaker equalisation cannot turn a sow's ear into a silk purse, and the benefits of such equalisation are best appreciated with systems that are already basically good. There is no excuse for bad speaker or control room design. Rather, digital equalisation will help make already good systems a little more consistent, and will reduce system-dependent colourations.

History

It is significant that the two companies, B & W and Sigtech, who are launching speaker and room equaliser products first both involve people who were involved with early pioneering investigations into sophisticated equaliser algorithms. Peter Fryer, the technical director of B & W, published the first detailed work on using FIR filters for speaker and room equalisation as long ago as 1980. At that time working for Wharfedale, he

described the use of an analogue charge-coupled FIR filters aligned to compensate for measured room and speaker responses. Ronald Genereux a Vice-President of Sigtech, was a participant in the earliest commercial digital FIR filter room and speaker response equalisation project at AR in the early 1980s, a project that was handicapped by, amongst other things, the inadequate DSP technology of the period.

The involvement of people with around a decade of prior experience in equalisation algorithms for rooms and speakers is indicative of the degree to which empirical experience of the problems is a necessary prerequisite to designing subjectively acceptable equalisers. This is not an area where mere engineering skills are adequate. While high-level engineering skills are certainly necessary, other subjective skills are vital.

Quality

Although the processing in the equalisers takes place in the digital domain, it is necessary to convert the results into the analogue domain to feed the amplifiers and speakers. The quality requirements for the D/A converters are particularly stringent in this application, and far more severe than for the D/A converters in CD players or digital tape machines. This is because the material on CD or tape is generally of a restricted dynamic range, say the 92 dB of 16-bit digital audio. However, the output of the digital equaliser may well have been subjected to digital gain control to turn the level up or down, and will also have been subjected to frequency-dependent boosts and cuts which will cause the output signal to have a far wider dynamic range than the input source signal.

For this reason, even when the input is

conventional 16-bit source material, the output D/A converters must work well over a far wider dynamic range, and 20-bit performance is desirable and it would be useful to have converters that exceed this, especially in terms of linearity, where even a 24-bit performance would not go amiss. Since all source material, whether it originated from CD, analogue tape, digital tape, live microphones, or off-air broadcasts, is being passed through these final D/A converters, it is desirable that these converters should perform better than the quality of any material being thrown at them.

Thus it is not surprising that companies that are developing digitally equalised speaker products, such as Meridian or B & W, take the problem of getting the best performance out of D/A converters particularly seriously. It is a paradox that loudspeaker companies, whose skills are traditionally the furthest from those of hi-tech digital technology, should actually be those companies who are pushing the quality aspects of digital conversion and processing the hardest. This is precisely because their applications are the most stringent as regards quality.

While having done little on room equalisation, the case of Bob Stuart at Meridian is particularly interesting. Rather than develop a digital equaliser for an analogue speaker system, he has developed a speaker where the crossover design was also implemented using digital equalisers. In other words, the speaker design, from the beginning, is conceived in terms of digital signal processing. Such an integrated digital speaker design permits the crossover performance to be optimised for the speaker units without any of the compromises forced by the use of analogue crossover networks, whether passive or active. In particular, not only is it possible to ensure on-axis

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phase linearity (something that overall digital equalisation of a speaker system can give), but the crossover rates and the off-axis phase performance can also be optimised.

Stuart also has the reputation of designing some of the best D/A and A/D converter systems around, based on commercial chip-sets, but skillfully optimised in subjective and objective performance. This kind of integration of historically disparate skills will be increasingly important as technologies like digital room and speaker equalisation become more important.

Equaliser types

Most attempts at sophisticated digital room and speaker equalisation have been based on FIR (finite impulse response) filters, that is tapped delay line or transversal filters, since the theory of designing such equalisers is well understood. However, it is also possible to design speaker and room equalisers using recursive filters (that is filters with feedback paths), for example as described by Greenfield and Hawksford⁸. In general, recursive filtering might be expected to suffer less from undesirable effects due to limited filter complexity, but it is not obvious which of recursive or FIR types of filters is best in room and speaker equalisation applications. The Sigtech product is based in FIR filters, whereas in some situations, a combination of FIR and recursive filters, or an equaliser based on recursive filters, might be used in other products.

Among other unknowns at present is the optimum kind of microphone that should be used for room response measurements; both Sigtech and B & W use an omnidirectional microphone, and certainly omnidirectional types are the most accurate for measurement purposes. However,

there may be advantages in using directional types, for example cardioids, for measurements so as to discriminate which part of a room's acoustic is being equalised. This and many other areas are still the subject of further research.

Both Sigtech's and B & W's products are based on separate monophonic equalisation of each of the speakers in a stereo pair, and some kind of stereo matrix equaliser, cross-feeding the two channels between the loudspeakers, may give improved results at a single listening position. However, there is no evidence that stereo matrix room equalisation offers any advantage over extended listening areas. However, once one has multiple loudspeakers dotted around the room, as in home cinema or HDTV applications, the additional loudspeakers provide the possibility of controlling low frequency room colouration by using the principles of 'active absorption', where the room response is used to provide electronically assisted absorption by the speakers of room resonances. However, active absorption techniques are not used in the first generation of room and speaker equalisation products, and in practice are unlikely to be very efficient with just two loudspeakers, since these do not give very great absorption of resonances.

Conclusions

The whole area of digital room and speaker equalisation is an exciting development that will permit some of the traditional problems of analogue speaker technology, such as phase response anomalies, especially in the bass, to be solved, and will allow a degree of control of low frequency room resonances and reverberation across a useful listening area. While room equalisation cannot perform miracles, it is another

step towards getting more accurate and repeatable results. The improvements obtained are generally not spectacular, but give a less muddled and more accurate sound, important for high quality domestic and monitoring applications.

It is a difficult technology to master, relying on a considerable amount of psychoacoustic knowhow and experience, just like the traditional arts of speaker and room acoustic design. Digital equalisation provides another very useful tool complementing these more traditional skills, capable of giving better overall results than was hitherto possible using just analogue technology. ■

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The time looks right for a format update. This winter the entertainment industry is all set to tear itself apart with a standards battle. The first round is between Sony's Mini Disc, and Philips' Digital Compact Cassette. But the winner could end up challenging recordable CD. The prize is to become the *de facto* standard for digital sound recording in the home.

At the same time there is another battle on the consumer front, to create a *de facto* standard for multimedia interactive CD. Meanwhile the film industry has to choose a digital sound format for release prints. Then, the home video and satellite broadcasters have to make a decision on which way to jump.

Philips delayed the sale of DCC from April to September, claiming that worldwide interest in the new tape format ruled out a launch on just one continent. There would not have been enough hardware and software to satisfy demand in Europe, the US and Japan. So throughout the summer Philips have been 'creating awareness' of DCC, by demonstrating players, recorders and compression coders at tradeshows, public shows and to selected reviewers and 'opinion formers'.

Sony has been continuing its own awareness campaign for Mini Disc, showing prototype players and dummy packaged 2.5 in discs. Right through to Spring 1992, Sony's Mini Disc demonstrations were like a conjuring trick. You did not realise what you were not seeing.

The standard Mini Disc demonstration has been very impressive. The demonstrator shakes the player while it plays music. The demonstrator then takes out the disc, proving that the anti-jog memory buffer has enough capacity to store several seconds of music.

But no one other than Sony's official demonstrators has been allowed to get their hands on the player prototype. This may well be because the buffer system only works if the player is jogged for a few seconds and then given around 10 seconds to restabilise. And this is not a real world situation. What matters is how the Mini Disc player will cope with repeated, continual jogging as it would suffer in a car on a bumpy road, or when worn by a runner.

Rumours have hardened that the Mini Disc prototype shown by Sony was, in fact, a modified Data Discman electronic book player, playing a write-once disc containing only a short sequence of sound recorded with conventional ADPCM coding. Certainly no one heard the MD prototype play the claimed hour or more of music; certainly no one saw the prototype make a recording.

All the pocket MD players have been nonworking mock-ups.

In May Sony engineers were touring the studio world with an ATRAC compression coder-decoder, demonstrating how music sounds before and after encoding. It impressed, but is still a breadboard the size of a microwave oven. A demonstration of Mini Disc recording, on a magneto-optical blank, was promised for late May or June. This timescale raises important questions. Can Sony really be ready for the world sales launch now promised by Alan Phillips, for November? Will sound quality from the chips match the breadboard? Remember that DCC has for many months been demonstrated from finished chip sets.

Sony says MD chip sets and record-playback optics will be sampling by June and in mass production by August.

How will Sony tackle the patent situation?

European firms, led by Thomson, have pooled all

Barry Fox

Confused about the new audio formats?

Barry Fox elucidates

patents on the compression coding system developed for digital audio broadcasting, subsequently modified by Philips for DCC. Dolby Labs has its own patents on digital compression coding, already used for satellite and landline transmission, and now for film recording. Although Sony will doubtless file patents of its own on the compression coding used for Mini Disc, it is hard to see how anything Sony develops can steer clear of existing patent rights.

Sony says it now has a licence from Philips and Thomson and will take one from Dolby if necessary.

So Mini Disc looks less like the anti-DCC spoiler that the trade has suspected. But if MD now misses the promised November launch, Sony risks being very seriously discredited.

Whether DCC or Mini Disc, or both, survive, both may well have to contest with write-once recordable 5 in CD.

There is an obvious attraction for consumers in the idea of having a disc recorder, which can make dubs and compilations of CDs, onto a blank CD, which then plays back on any domestic CD player. Studios already know the system works, but they also know the price of blank discs (£20 to £30 each) which would be prohibitive for domestic use. Here Kodak enters the picture.

Kodak's Photo CD system lets photographers pay a Photo CD Centre for the transfer of up to one hundred high quality still photographs onto a blank CD. The disc then plays on a Photo CD player, through a TV set. The system is due for launch later this year.

The blanks which Photo CD Centres will use are write-once discs of exactly the same type as those used by existing CD-R audio recorders. Kodak both buys in discs from Taiyo Yuden of Japan and makes its own in Rochester, New York.

And Kodak has now confirmed the price which a Photo CD Centre will charge a customer for transfer of photographs.

The customer will pay just under £5 for a blank disc and then 40p or 50p per picture transferred (depending on whether it is individually selected, or just one of an entire roll). Once Kodak starts selling Photo CD blanks for under a fiver, it will be hard for the audio industry to justify the current absurdly high prices.

And at under a fiver a blank, CD-R becomes a viable alternative to DCC and MD.

Commodore was first on the market with an interactive CD system, called CDTV. But by rushing onto the market ahead of Philips' *CD-I* system, Commodore ended up selling something which was clearly not ready for sale. Also, Commodore is a computer company which knew little of the consumer electronics market, and most importantly, showed no interest in learning about it.

As a result CDTV has been a disaster as a domestic product and is likely only to survive as a CD-ROM peripheral for the Amiga computer. As such it will let people play exotic video games,

without changing floppy disks halfway through and soaking up valuable hard disk space.

Although there are several other domestic multimedia CD-ROM formats on offer, or in the pipeline, it now seems a foregone conclusion that Philips will win out with *CD-I*. This format is compatible with Photo CD and is already on sale in the US and UK, with Europe to follow later this year. Performance, and software, are streets ahead of CDTV.

The first demonstrations of Full Screen, Full Motion Video were very impressive. Whereas DCC and MD reduce audio data rate by a factor of four or five, FMV compresses moving video by a factor of over hundred, down to a data rate of around 1.2 Mbit / s. This lets the 5 in disc store over an hour of moving video, of quality similar to VHS tape, along with stereo sound compressed in the manner of DCC.

Dolby Laboratories have already quietly tested the SR-D digital cinema sound system, releasing two films (*Star Trek VI* and *Newsies*) to cinemas with digital soundtrack. Audiences have not known whether they are hearing digital or analogue sound. The big advantage of *SR-D*, of course, is that it is backwards compatible. The film carries both analogue and digital soundtracks so one film will play on old or new projectors.

Kodak's partner in its rival digital soundtrack format, the Optical Radiation Corporation, has dropped out. Kodak talks of continuing the work on CDS (Cinema Digital Sound), but with *SR-D* now available it is hard to take the system too seriously. It is hard, too, to take too seriously Sony's promise of a rival system. Likewise there seems little future for the several companies which have at some time or other proposed syncing a film projector with a CD or Laser Disc which carries a digital soundtrack. In the real world, distributors and cinema managers do not want to be bothered with films that come in two parts, celluloid and disc. They also do not want to be bothered with double inventory releases, analogue and digital. In each case there is extra cost and too much potential for sending the wrong track to a cinema.

This summer Dolby will make a splash with publicised *SR-D* releases. Bearing in mind the sound quality of *SR-D*, its backwards compatibility and Dolby's close working relationship with the film industry, there seems little doubt now that *SR-D* will become the future standard for the movie industry.

Already Dolby is demonstrating a domestic version of *SR-D*, called *AC-3*, to firms licensed to make domestic Dolby equipment. *AC-3* is a six-track digital sound system, matching *SR-D* in track format and performance, which can be squeezed into a narrow data space in any domestic tape, disc or broadcast format.

Dolby is demonstrating *AC-3* to licensees by way of advance notice. If they leave space in their data stream for the *AC-3* code, then the format can later be upgraded. Dolby sees *AC-3* as applicable to HDTV, satellite broadcasting, home video tape and Laser Disc. I've heard the system working, and have technical details. More of *AC-3* next month.

Finally, watch out for bet hedging. Already Sanyo and Sony have developed conventional CD players which use the memory buffering technology on which Mini Disc relies, to provide a jog proof CD player. The disc runs at twice normal speed, continually feeding data into a buffer memory. If the player is jogged, it plays music from the buffer for several seconds while the laser gets back on track. ■

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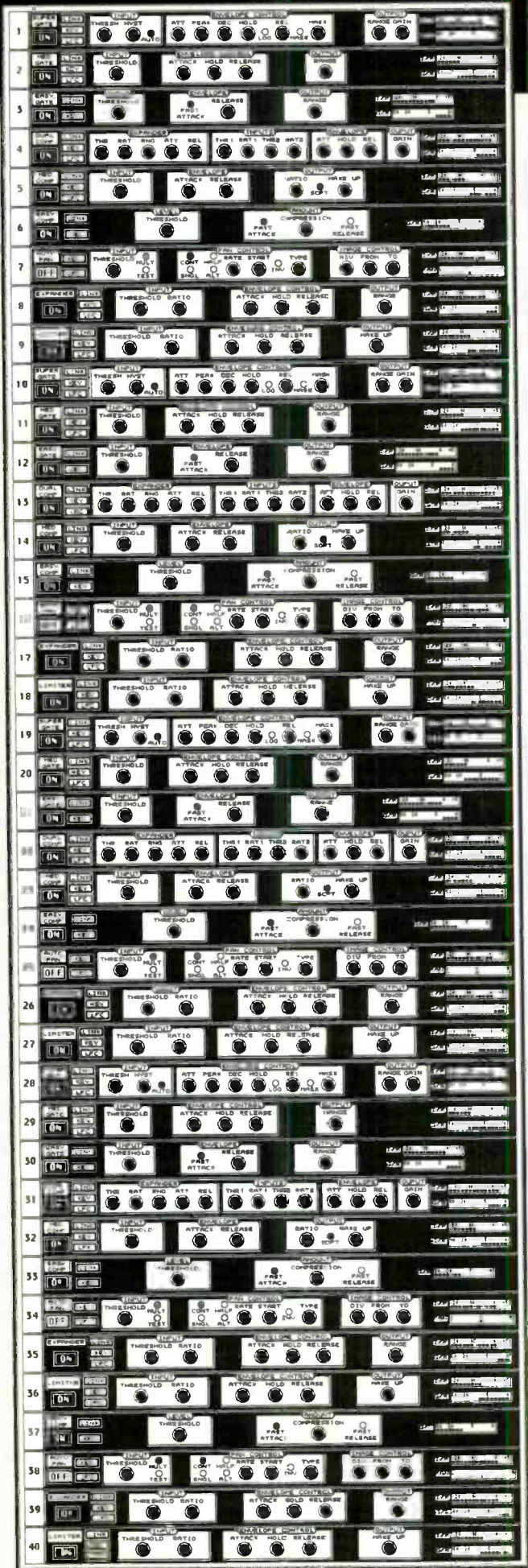
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Martin Polon

Our US columnist makes a plea for the Eastern bloc and the setting of audio standards

Nobody asked me again; but isn't it time we stopped yelling and screaming about the newly freed Eastern bloc of Europe becoming the 'afterburning thrusters on the economic rocket engine of the European Communities' future? To many, it appears the East promises to behave more like a 100,000 Kilogram reinforced concrete anchor to the EC's valiantly chugging economic ship of state. It is time to make a realistic appraisal of just what the East means to the West in the early 1990's. To hear many in the audio business, the East will be the new 'gold rush' for the West.

1. Despite the fact that pollution issues are somewhat removed from the wonderful world of audio, it is virtually impossible to do business in the East and not be somehow impacted with pollution concerns. To quote one US expert, 'the totality of pollution in Eastern Europe is so impossible to comprehend that it makes the problems in Western Europe pale by comparison. In fact, the total pollution of the ten worst sites in France, Germany and Italy together can be matched by just one site in the Russian Federation and that is just one of many'. It is inevitable, that the concept of 'inheriting' responsibility for sites previously contaminated by toxins will be a reality of doing business in the East. Locate a venture on an existing manufacturing site, and you have a good possibility of becoming embroiled in issues involving the damage caused by previous use to former employees and-or adjacent households plus the liability for still present contaminants to current employees and-or adjacent households. With a scarcity of clean sites, the downside of being able to locate either hardware manufacturing or software production plants without future liability is a real conundrum.

Secondly, the lack of experience in handling the toxic elements of current activity such as CD pressing provides a second layer of difficulty to deal with. In joint ventures, the nagging concern of what the Eastern partners will do with toxic by-products has bothered many potential Western partners. And the lack of suitable new 'safe' waste disposal sites suggests that this problem is not just an intellectual one.

2. A far more dangerous issue is the worldwide concern over the status of Russian and Eastern bloc nuclear power plants. Consider that there are facilities scattered almost equidistant throughout the European East. Plants are located at Lubmin and Rheinsberg in the eastern portion of unified Germany; at Dokovany and Jaslovske Bohunice in Czechoslovakia; at Paks in Hungary; at Kosloduy in Bulgaria; at Krsko in Slovenia; at Ignalina in Lithuania; at Lukomi in Belarus; at Chernobyl, Rovno and Zaporozh'ye in the Ukraine and at Balakova, Kalanin, Kursk, Leningradskaya and Novovoronezhskiy in the Russian Federation. In fact, more than 50 Soviet designed and built power plant reactors operate in the East, with fifteen units being of the Chernobyl-type design — considered the most dangerous type used anywhere in the world today, due to it's past history and the lack of any kind of containment vessel.

Now the problem concerning any kind of major investment is the difficulty of citing a pressing

plant or audio hardware manufacturing facility away from any potential 'fallout' zone (no pun intended) and the problem of 'blacklisting' any products from a new plant due to a nearby nuclear 'accident'. One cannot even think about the possibility of a Soviet-made nuclear power facility blowing it's top with the end result of a contamination greater than Chernobyl or worse. The scenario is too horrific to even contemplate, except by the most morbid among us. On the plus side, however, we can recall the results of a US Defense Department study released to the public some years ago. The study pointed out that after a nuclear holocaust, two entities would still exist and function. One was cockroaches and the other was recordings made on tape and the magnetic tape itself. So, perhaps an EC Eureka study is due to verify whether the newer formats of CD and MD would also survive such an event and whether the roaches would prefer these to cassettes for musical entertainment. It is a virtual certainty that there is precious little else we can do to cope with the Eastern nuclear power hazard, at least in the short term!

3. A particularly virulent style of recorded musical software piracy has developed in the East. It consists of two modes of operation. In one, counterfeit records, discs and tapes are funneled into the East in competition with legitimate products from the West. Such efforts might involve the Soviet tour album of Billy Joel, since both the artist and the concerts are well known. A second mode is to use the record, cassette and disc production facilities located in the East to create counterfeit recordings for sales in the West — especially in Europe.

What makes all of this so difficult is the calibre of individual one might find in the pirate trade plus the existing infrastructure for bootlegging and distributing vinyl and cassettes. A retired economic affairs attache of some experience in the former Soviet Union explained, 'you are not dealing with the usual riff-raff found in piracy in the West. Even the much feared Mafia on both sides of the Atlantic cannot measure up to the black marketeers now involved in Eastern bloc piracy. You have to remember that these people have outwitted by definition the paramilitary police minions of the Kah Geh Beh (KGB) — the Militia — not to mention the KGB itself.

'So if these folks are still alive, then they have triumphed in the past over the very hounds of Hell. In addition, the network to produce and distribute music on cassette, on vinyl record and of

late on compact disc has evolved into a 'basement' industry of extreme durability. You must remember that the regime in power wanted to suppress not just the product but the music as well. So the penalty for getting caught was not the confiscation of this or that and a cash fine plus the occasional short prison sentence handed out in the West. The penalty in the East for piracy was usually the firing squad with the possibility of life in Siberia for those who were really unfortunate! So the calibre of these pirates makes anybody counterfeiting music in the West small potatoes indeed!

In addition, both East Germany and Bulgaria had earned substantial reputations for being safe havens for third party piracy. The Palestinian movement relocated its not insubstantial audio and video piracy operations from the Lebanon to the relative safety of the Eastern bloc after the major armoured incursions of the Israeli Army in the mid-1980's. Other renegade movements could operate with relative impunity in the East and usually did!

4. The trials and tribulations of running a recording studio in the former Soviet empire today are truly staggering. Aside from problems of doing business with the essentially unconvertible rouble, the basic combination of anarchy and crime have made it very difficult for the handful of recording studios trying to operate in Moscow and elsewhere. Western recording equipment is at such a premium that theft at all costs is almost impossible to thwart. While it is true that the running joke of the 'Bulgarian one-channel mixer', really does not exist — the indigenous equipment industry is producing little of perceived value today. The major premium is for Western gear and with it the status of tempting target.

None of this means that the Russian Federation and the former satellites do not hold enormous promise for the world audio industry — at some point in the future. For right now, the issue should not be 'what can we take out?' — but rather 'what can we put in?'. If the audio industry in the West looks towards a different kind of investment now — then real returns will follow in the future!

A. Consider the option of an internship. Find a place for an audio practitioner from the East, who can live and learn for six months or a year in UK or US recording facilities.

B. Set up a similar programme in record companies and pressing facilities in the West.

C. Establish a scholarship programme to allow Eastern bloc students to attend Western schools and universities offering courses of instruction in recording, music technologies and music business.

D. Offer a grant to selected individual audio practitioners from the East so that they might attend the most significant trade meetings in the West such as AES, NAB, IBC, APRS and CES.

E. Corporations in the world audio industry should create teaching and administrative sabbaticals for their top administrators to serve at Eastern companies and schools in an advisory role.

Now all of this will cost money. But it can certainly be considered as seed money. There is nothing more important to the future of the recording and record industry worldwide than providing this kind of support for the East now! ►

Nobody asked me again; but it surely seems that the business of setting and adhering to standards in the audio industry has become a double-edged sword. Standards — called by some 'a universal agreement to disagree', represent the acceptance of a standard technological base by the various manufacturers within an industrial grouping; in this case — audio. Within the industry, the flexibility of each manufacturer to elaborate and articulate product from the base goes without saying. The classic example of a successful standard is the Philips analogue cassette. Originated as a rather lo-fi product in the 1970's, it has evolved today into a versatile family of products by which every single equipment maker profits. The standard is adhered to as the basis of innovation which allows each manufacturer to customise and configure product based on that manufacturer's perception of the marketplace. Today, the Philips cassette exists as the most popular medium for the sale of music in millions of units sold worldwide. It is the basis for a forthcoming digital audio system, can provide virtual digital quality with Dolby S, is used by professional radio stations all over the world and is the basis of the portable audio revolution in the car and via the ubiquitous *Walkman*. A real success story — a win-win situation for all concerned.

Yet for all the financial and creative positives of the cassette and, of course, the compact disc, one cannot easily point to any other such 'success' stories within the audio industry. Especially of late, there has been 'a disagreement to agree' on virtually everything. Each camp of hardware manufacturers and in most cases their software allies have refused to consider the benefits of cooperation. Clearly, all of the various entities involved in the numerous 'turf battles' over technology in audio today do believe in standards. Each one wants their system to triumph over the competition and become the standard. Such a victory would, they argue, bring such a windfall of profits and licensing that it justifies the conundrum in consumer confidence, product compatibility, creative interchangeability and pricing advantages that inevitably suffer from a lack of standardisation. Let us observe the flip side, however, where a successful current standard has been compromised in the several sectors of the audio industry and how the repercussions have impact far beyond the original sphere of influence.

In the world of the law in the United States, there is a phrase that describes the unfortunate effects of illegally seized evidence by law enforcement officials. It is known as 'the fruit of the poison tree.' The principle of this criminal evidence dictum is that there is a ripple effect in

which any information gained illegally such as via an unauthorized wiretap, negates the effect of, say, a raid on a drug warehouse because the original information was tainted. Now, you ask, what has this to do with the audio business? Quite a bit really, because as we look at the burgeoning business of stereo television worldwide we see the impact of a 'poison tree' affecting first the US stereo television marketplace and, secondarily, the world stereo television production community. Here indeed, is the story of a standard gone bad or at the least badly compromised!

It seems that early in the 1980's, the United States conducted an inquiry to ascertain an appropriate standard for the inclusion of stereophonic sound into the broadcast signals of television programming. Since this was a mandated addition to the technical standards for TV stations licensed by the US Federal Communications Commission, a decision was made by executive fiat rather than falling to the wayside due to electronic industry inertia. A standard was chosen for multichannel television sound (MTS) that utilised the dbx noise reduction system. By 1984, the first broadcasts went on-the-air and the first stereo-compatible TV sets were offered to consumers. From 1984 until the end of 1991, in excess of 40 millions television sets equipped with the MTS stereo system and the dbx noise reduction scheme were sold — achieving an

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in-home penetration of about 35% — or were they?

It seems that late in 1991, the licensing entity for the dbx technology informed the audio industry in particular and the consumer press in general that the TV set manufacturers for five of the large brand names serving the United States of America had been using pseudostereo chips and technology to provide a stereo-like image for the home viewer. It was suggested that financial considerations may have driven the decision-making process for these TV set makers, since true MTS and dbx stereo circuitry was used in the TV sets larger than 25 inches, produced for these brand names. All of these sets were advertised as having some kind of stereo capability. Advertising slogans such as 'Stereo Broadcast', 'Stereo TV', and 'MTS Stereo', were used by retailers selling these alternative stereo sets.

It is estimated that at least 50 different models have been sold configured with the alternative stereo technology. Some analysts have expressed concern that the practice has gone on for some time and that as many as a hundred different models from the five brand names could be involved. It is difficult to estimate how many tens or hundreds of thousands of sets have been sold with the alternative sound systems since actual sales figures are proprietary to each manufacturer, but the figure is estimated by several observers to be in the millions. Several of the brand names

involved have indicated that they will continue with the practice of selling the alternative stereo sets and argue that their technology serves the public at least as well as the adopted standard. The other makers have stated that they will adhere to the standard as soon as manufacturing changes allow.

Although no other set manufacturers or product brand names were so identified as having deviated from the standard — the entire process has become the 'poison fruit' of the US stereo TV standard 'tree'. The fact that the charges have not been completely refuted by all of the set makers in question, is seen in some quarters of the industry as validating the cries of 'foul!' The presence of consumer law suits against at least one of the brands for fraud and deception have further muddied the waters. Purposefully, the specific names of the indicated companies have not been used — first to avoid entering the controversy directly and throwing more editorial 'gasoline' on the flames and, second, because the damage is already done. The stereo TV community has had its 'dirty linen' aired in public. The real damage, just as with the so-called 'poison fruit' concept, is the indirect impact on the rest of the stereo television community.

The audio community in the United States has been fighting a pitched battle with the television networks, independent programme syndicators

and Hollywood studios over TV programme stereo audio quality for years. The argument always raised by the TV production and distribution communities is that nobody at home can hear the difference anyway. Frequently, to save money most usually, the mix to stereo for a programme is done minimally or sometimes in monaural sound with a stereo synthesizer used to create an 'image'. The programmers' justification involves a panoply of 3 inch speakers, milliwatt audio chips and generally mediocre sound systems perceived as associated with stereo TV sets. Now, at production meetings in Hollywood, a new chord is struck. 'Most of the TV sets produced during the past several years aren't even equipped for real stereo sound', is a phrase that has popped up in Tinseltown at more than one meeting.

So, the damage is done. The poison has been spread. And the impact has ramifications far beyond the TV production community. The consumer, of course, has heard of all of this and clearly the slow acceptance of the stereo video home theatre has not been enhanced. And the world production community pays a price since the position taken by the Hollywood partner in most co-production deals — necessary in today's economic climate — is one of reluctance to pay for first class stereo production. The motivation is a foregone conclusion. A curious piece of work indeed. ■

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TIME LINE



Tracks for the Converted

A Technical Review by Sam Wise of the Yamaha DRU8 8-Track Digital Recorder with AD8X and DA8X Convertor Systems

The Yamaha DRU8 8-track digital recorder under review forms a part of Yamaha's thrust into professional digital audio production tools which began with their first digital mixer DMP7 several years ago. Since then over 500 DMP7 consoles have been sold in various forms in the UK alone, making Yamaha market leader in terms of product volume shipped. Indeed, they say that this total exceeds the sum sales of all other companies' digital mixer products.

According to Alan Martin (Sales and Marketing Manager), there are two main thrusts to the current product line. The first is to the high-end private user studio where the DMR8 Digital Recorder provides a cost-effective digital mixing plus recording solution. The DMR8 uses what I would describe as a highly condensed user interface. For example, the linear fader which is used as the Channel 1 audio level control, might also be used to control the level of Aux Send 1, the boost cut of the LF equaliser section, monitor level adjustment, and in mix-down to control to these functions on Channel 9 and 17 as well. This reuse of control function takes some learning, but is very appropriate to users who do one thing at a time virtually single-handed. Memory cards can be used to store setups for various stages of the production

process, instantly switching the whole console setup to a state which suits the process. DMR8 includes a rerord section identical to the DRU8. In systems using DMR8 as their hub, the DRU8 might be added to extend track capacity up to 16 or even 24 tracks. It would also provide a tape copying capability, for back up perhaps.

The second main market area is broadcast, with the DMC1000 Digital Mixing Console as a system hub. DMC1000 is more conventional than the DMR8, providing simplified channel strips for each of the eight input channels with some centralised assignable controls such as equalisation and auxiliary send control. DMC1000 is a similar cost to DMR8, but does not include the digital recorder on-board, adding additional facilities like an extra band of equalisation. In these systems, the DRU8 will be the recorder of choice, though protocol conversion boxes allow for easy interfacing to other manufacturer's products.

But the DRU8 can also be used as a stand-alone 8-track digital recorder, with 20-bit recording resolution. In this case, outboard converters are used, and in this review the Yamaha AD8X provides eight channels of input A/D conversion with 19-bit resolution, and the DA8X provides eight channels of output D/A conversion with 20-bit resolution. Again, other manufacturers'

products may be used for this purpose. For tape monitoring, the DRU8 contains a pair of onboard 20-bit D/A converters. The RC8 remote controller-locator acts as a remote for the system, but was not available for review.

To produce larger multitrack configurations, DRU8 units can be combined. At present, three units can easily be joined to form a 24-track recording system, which can be controlled with the RC24 remote (also not reviewed). The RC24 provides a conventional-style record enable interface and machine remote with 100 locate memories.

Initial impressions

Outwardly, all of these units appear well made and professional. Lifting the cover of the AD8X reveals a tidy, easy to access set of PCBs which also meet acceptable standards. Inside of the DRU8 recorder itself is more complex, with the self-contained transport mechanism on the right and layers of PCBs on the left. All wiring is, however, plug connected for easy dismantling.

The transport uses Yamaha's own cassette format, loaded with a special Video 8 type of tape. Apparently, to protect their interests, Yamaha produce the cassette enclosures themselves, while tape loading is done by their suppliers. In Pause mode, the tape is loaded to the fixed heads, ready for a quick start when Play or Record is activated. The tape is driven by a pair of servo controlled capstans and pinch rollers, as in a conventional analogue machine like flat tape path. The transport is small, but more robustly made than rotary head DAT transports. This should be a reliable mechanical design.

Most of the PCBs are of fibreglass, with clear

legending of all components. Sockets are provided for firmware ROMs, which are easily accessible for software updates. A service manual is available, giving sufficient information for disassembly and first line servicing. It does not contain full circuit information on every circuit. Minute detail is given about subjects such as MIDI exclusive data. Test procedures and adjustment information are included for clock timings, levels, convertor trimming and similar items.

AD8X and DA8X

These 8-channel convertor boxes are very simple. The D/A unit has no operational controls at all, appearing only as a 2 U rackmount assembly. The output connectors are electronically balanced XLR compatible types. These may be connected for 'semipro' use with gain reduced by 6 dB. The front panel of the DA8X has LED displays of Power, PLL (clock locking), and Emphasis.

The A/D unit contains level controls to set input gain, level meters which read the convertor digital output levels, and peak overload indicators. The gain control follows the input amplifier, preventing impedance change with level. Further rear panel selectors allow 48 kHz or 44.1 kHz sampling to be selected, emphasis on-off, and two clock master control functions. Unlike all other digital recording products we have measured, the Yamaha DRU8 does not force output emphasis selection to follow the input or tape signal emphasis status. Discussion with Yamaha staff reveals that the Yamaha digital format does not contain any flags. This means that emphasis will not be automatically selected if necessary on tape replay. Therefore, it is up to the user to mark every tape with the emphasis status used. The emphasis on-off function of the DA8X is set by a system control menu function, not directly by a front panel switch. As with analogue tape, it is probably a good idea to put some reference tones on the tape header which will make it obvious if emphasis is used or not. On the tape labels, Yamaha provide tick boxes for sampling frequency, time code frame rate, etc. They should add one for emphasis status as a reminder to the user.

A further switch on the rear of the AD8X selects DMP7D mode, allowing its use as an input convertor for their inexpensive digital mixer.

Other connectors on the rear are a D-type multipin carrying all eight digital channels, and a BNC clock input-output.

The AD8X has clock Mode, Sampling Frequency and Emphasis LED indicators and level meters. The meters are relatively coarse, with 3 dB or 6 dB

MANUFACTURER'S SPECIFICATION	
DRU8 DIGITAL RECORDER	
Track Configuration	8 PCM Digital, 1 time code, 1 control, 2 analogue
Sampling Frequencies	48 kHz, 44.1 kHz, 32 kHz
Frequency Response	+1/-3 db, 20 Hz to 20 kHz (fs=48 kHz, 44.1 kHz)
	+1, -3 db, 20 Hz to 14 kHz (fs=32 kHz)
Dynamic Range	120 dB (digital recorder section)
Quantisation	20-bit linear
AUX Tracks	
Frequency Response	+1/-3 dB 100 Hz to 2 kHz
Maximum I/O Level	-1 dB
Hum and noise	-45 dB
Wow and Flutter	unmeasurable
Variable Pitch	±10%
REC and PLAY time	20 min (48 kHz), 21 min (44.1 kHz), 30 min (32 kHz)
DSP Quantisation	28-bit (monitor mixer) 24-bit Digital I/O
Monitor Analogue Output	20-bit D/A, 8 times oversampling
Distortion	<0.007% @ +17 dB, 1 kHz
Hum and noise	-87 dB (average reading, 12.7 kHz bw)
REW and FF time	100 sec
Power	120 v/220 — 240 V/75 W
Dimensions	480 (W) × 144.3 (H) × 396.8 (D) mm 19 × 5 7/8 × 15 1/2 in
Weight	11.5 kg (25.3 lbs)
AD8X ANALOGUE TO DIGITAL CONVERTOR	
Frequency Response	20 Hz to 20 kHz, +0.5/-1.0 dB at 48 kHz sampling 20 Hz to 19.5 kHz, +0.5/-1.0 dB at 44.1 kHz
Total Harmonic Distortion	<0.0018% @ +23 dB, 1kHz (Harmonics only, no noise) <0.005% @ +4 dB, 1kHz with input control at nominal, level -20 dB re full scale
Dynamic Range	>110 dB (Emphasis off)
Hum and Noise	<-90 dB
A/D Conversion	19-bit (digital floating)
Sampling Frequencies	44.1 kHz and 48 kHz
Power	120 v/220 — 240 V/60 W
Dimensions	480 (W) × 99.8 (H) × 364.1 (D) mm 18 7/8 × 3 7/8 × 14 1/2 in
Weight	8.0 kg (17 lbs)
DA8X DIGITAL TO ANALOGUE CONVERTOR	
Frequency Response	As AD8X
Total Harmonic Distortion	<0.0025% @ +24 dB, 1 kHz, fs=48 kHz 0.0018% typical (excluding noise)
Dynamic Range	>106 dB (measured to EIAJ CP-307) 108 dB typical
S/N Ratio	>110 dB, 113 dB typical
D/A Conversion	20-bit with 8 times oversampling
Power	120 V/220 — 240 v/40 W
Dimensions	480 (W) × 99.8 (H) × 364.1 (D) mm 18 7/8 × 3 7/8 × 14 1/2 in
Weight	7.30 kg (16 lbs)

TABLE 1: WIDEBAND NOISE MEASUREMENTS

NOISE TYPE	RMS 22 Hz-22 kHz	RMS A Weighted	Q-Peak Unweighted	Q-Peak CCIR 468-3 Wtd.
Channel 1	-104.8	-108.5	-100.8	-94.0
Channel 2	-107.5	-109.0	-101.3	-95.5

steps between displayed settings. This is more than adequate for maintaining safe levels, particularly with the wide 108 dB dynamic range of the system, but not adequate for level alignment purposes. The meters are peak responding with a fairly rapid decay rate. The eight input level controls are 32 detent devices with no calibration markings and reasonable control laws.

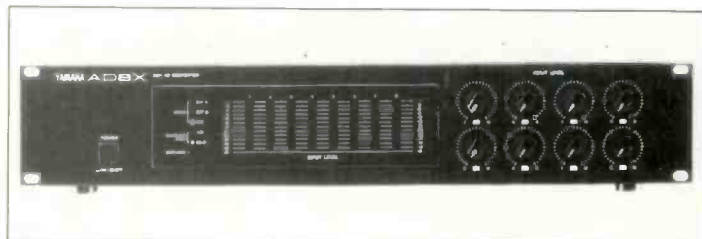
DRU8

In the usual way, the DRU8 was initially operated without reference to the manual to test the user

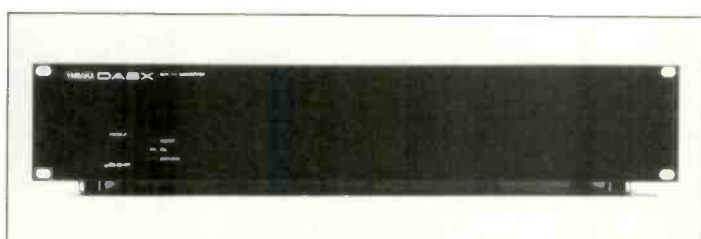
interface. Most functions behave exactly as expected, including flashing Record Ready status LEDs which is steady in Record mode. The front panel of the DRU8 is functionally 'clean', being set out in a logical sequence, with collections of related controls being grouped together. It is therefore convenient to describe these in the order they appear on the panel.

Monitoring

At the left of the front panel, are the POWER switch, BALANCE control and MONITOR Level. As mentioned



Yamaha AD8X



Yamaha DA8X

earlier, the DRU8 contains a monitor mixer which digitally sums selected channels and outputs the result to a front panel PHONES 6.35 mm stereo jack socket, and to rear panel balanced XLR compatible Monitor connectors. Adjacent MONITOR SELECT buttons allow a pair, or all eight digital tracks to be selected for monitoring, or alternatively, the two AUX analogue tracks. MONITOR LEVEL and BALANCE behave in the manner expected, like a hi-fi system.

Metering

The next set of controls select the level meter source. REC-IN/TAPE is a toggle function, and with this stationary head machine allows switching between the input signal and off tape monitoring. MIX is below, and when selected disables meters 1 to 6, selecting left and right monitor outputs on 7 and 8. These levels are those resulting from digital summing, and are not affected by the front panel monitor level control. Obviously, summing up to eight tracks has a high likelihood of generating an increased overall level on the monitor digital buses, producing the potential for overload if track levels are high. This allows the user to quickly check the source of an audible overload. LED indicators at the top of the meter panel indicate the selected function.

Beneath these is PEAK HOLD, with internal LED indication. The meters do briefly retain peak level under normal circumstances. With PEAK HOLD active, the highest peak level is retained indefinitely, useful for determining the source of an audible nasty.

System Control Functions

The SYSTEM CTRL key provides access to three submenus — INIT, MON and UTIL. These are

displayed in the 2 lines by 16 characters LCD display to the right of the metering section. Cursor control keys allow submenu selection. Menu selection and cursor controls are further to the right. The SHIFT NEXT confirms the choice and activates the next menu level. SHIFT BACK reverses the process, moving back up the menu tree. Right and left cursor keys select the parameter to be changed, and the up and down keys increment and decrement these.

The first main menu selection — the INIT function — provides for adjustment of basic working settings. The first of these allows selection of sampling frequencies of 48 kHz, 44.1 kHz or 32 kHz. Automatic selection of sampling rate on replay does not occur, but the system warns if the tape being played has been formatted or recorded to a different sampling rate to that selected. This warning is a pretty obvious flashing LED, accompanied by a text warning in the display. On the review sample, we had problems getting system clock synchronisation at any sampling rate except 44.1 kHz. This, we expect is a software fault which Yamaha will address.

INT Clock Mode permits the choice of a highly accurate crystal controlled clock, or PLL (phase locked loop) which allows vari-speed operation. If PLL is selected, the option of fixed or variable speed is provided next, speed being controlled by the cursor keys incrementing in 0.05% steps. In use, the speed can be increased or decreased at about 1% per second, and this is click-free and audibly smooth. We did find that changing speed while recording causes glitches when decreasing from maximum speed, otherwise performance is excellent.

The next menu option allows individual input tracks to be selected to come from either of the two 8-channel digital input ports. After this, a choice of remote control protocol is available, allowing selection between Yamaha protocol, A-type (Sony),

or B-type (CMX-Ampex). This versatile system allows for easy integration of the DRU8 into a large number of video edit suite environments.

The MON main menu selection activates a menu for setting up the monitoring system. Unlike other multitracks, the DRU8 includes not only output switching to facilitate overdubbing, but even contains an 8 into 2 digital mixer and on-board D/A converters. Menu selections allow tracks to be switched in or out of the monitor. Each track may also be panned and level balanced with the other tracks, and finally the digital level of the monitor may be adjusted to match the type of programme, preventing digital overload.

Lastly, the UTIL menu can be selected. First, this allows adjustment of input delay on each channel up to 999 samples. Next, is emphasis on-off, which only affects the D/A output, leaving the A/D as set on its rear panel. However, the status of the A/D is indicated. Next digital Cascade In can be switched on-off, allowing extension of the 8-track monitoring system when machines are linked together. C2 warning is also selectable, allowing PCM audio faults to appear on the LCD display.

MIDI interfacing

The next key on the panel selects MIDI control menus, providing a full range of control over MIDI channel selection, sending or receiving Control or System Exclusive information, powerful Parameter assignments which allow MIDI control over practically any menu setting monitor function. Finally, a Bulk Dump can be initiated.

Record mode

ALL REC places all tracks into record ready condition, necessary for an initial run to stripe the tape and lay the first tracks. SYNC DUB allows selected tracks to be placed into play or record ▶

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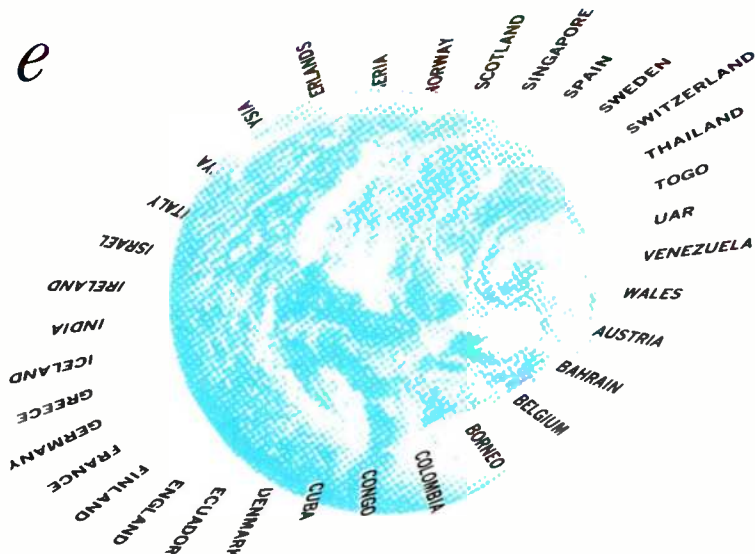
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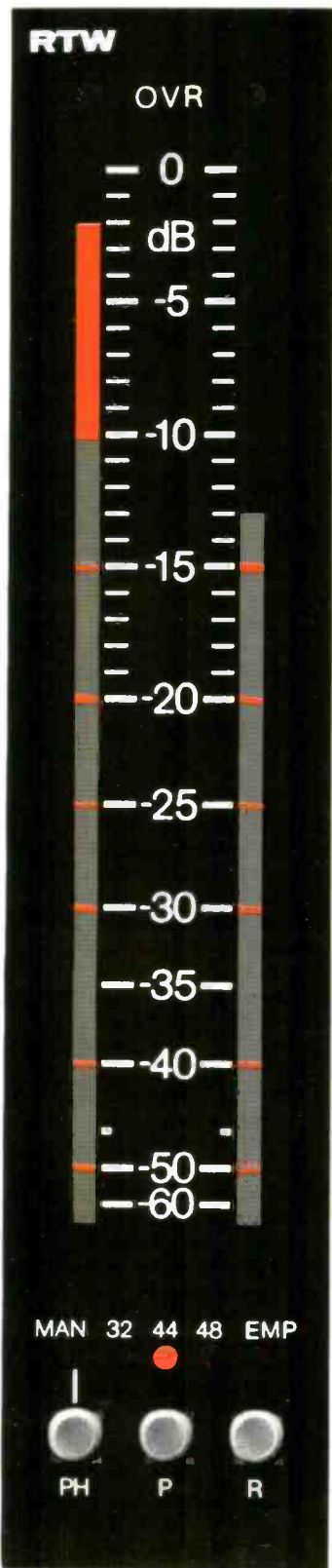
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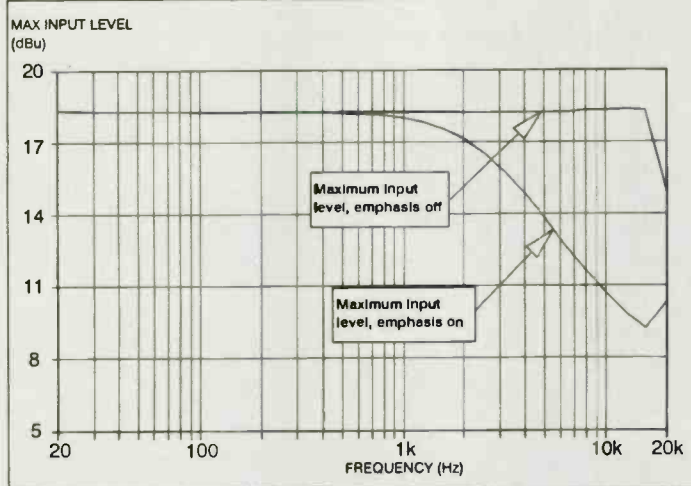


Fig 1: Yamaha ADX8 and DA8X, maximum input level (dBu) for 1% THD+N. Input gain at maximum

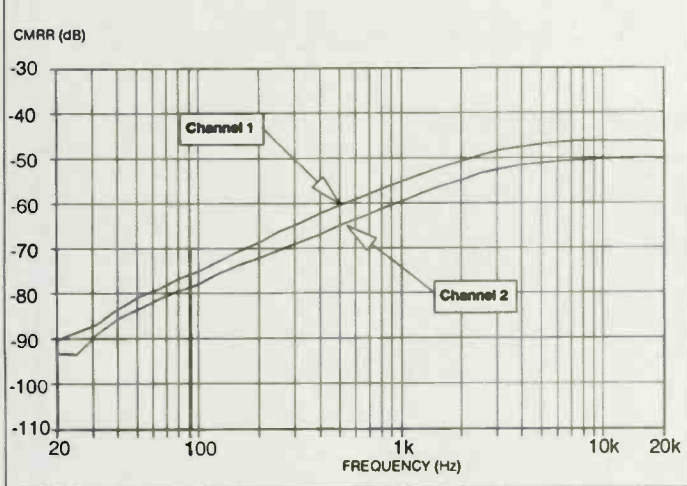


Fig 2: Yamaha DRU8 Digital recorder, input common mode rejection ratio

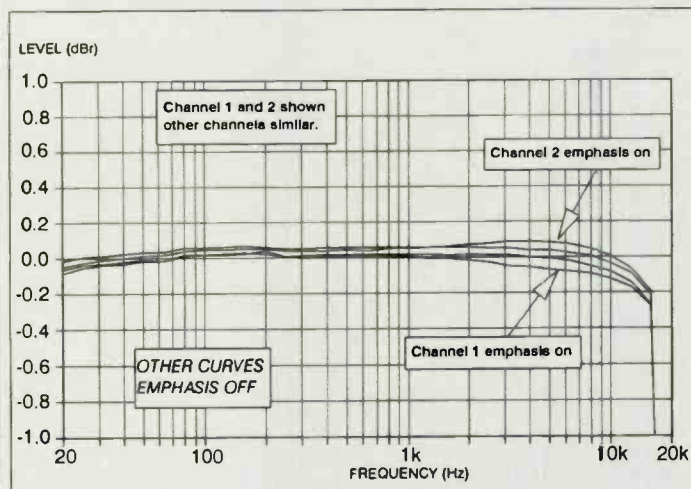


Fig 3: Yamaha DRU8 digital recorder, analogue to analogue frequency response. Input level at +4 dBu, input gain at max

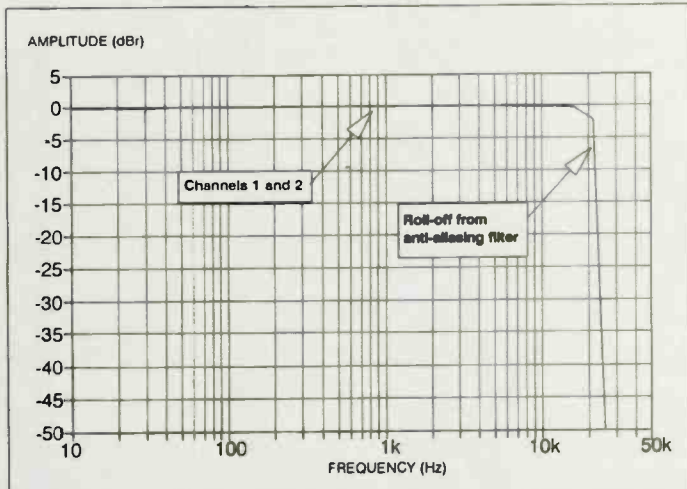


Fig 4: DRU8 recorder, wideband frequency response. Input level from AP at +4 dBu, Input gain at max. Emphasis off

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ready mode for sync over dubbing. Tracks can be also be selectively set to monitor either input or playback. Dubs can be rehearsed by using the transport PLAY and REHE together rather than the conventional PLAY and REC which are used to enter record mode. The time code displays shows the current time and the time code where rehearsal started. This information is used to automatically rewind to the start position for either a take or further rehearsal.

PUNCH IN also allows tracks to be selected to record ready status and monitoring requirements to be set. The difference is that PUNCH IN allows for an automatic crossfade between the previously recorded material and the current audio input. Crossfade time can be set from about 1.5 ms to 3 ms. In addition, time code references can be set for punch-in and punch-out times, along with pre and post-roll time. Rehearsal mode works similarly to SYNC DUB. Once time codes are ideally set,

pressing PLAY and REC performs an automatic punch-in or out with crossfade and monitor identical to the rehearsal. Alternatively, punch-in can be done manually.

PARKING seems a strange title for a button, but basically speeds up control selection by switching in the last used menu for any of the recording modes. ALL SAFE disarms all tracks, while OTHER allows track recording selection of the Aux tracks and ALL ERASE to clear the tape for reuse.

In practice, while most things worked well, we did have some small problems. For example, we recorded a first pass from the beginning of the tape. Subsequently, in ALL REC mode, we were left with about half a second of this initial recording which would not go away. Eventually, we got rid of it by entering PUNCH-IN mode commencing at the beginning. We suspect that this is a software problem relating to the initial reading of data or time code at the beginning of a tape.

Also we had Sync Error twice, when trying to put the machine on record. At least the system warned us, preventing continuation in these circumstances.

Locator Functions

The time code synchronisation of the *DMR8* — *DRU8* system has been recently tested by Audio

Kinetics and stated to be excellent. They have far more knowledge on this subject than I do, so am prepared to accept their comment. The only problem occurred when using the system in chase mode. An instantaneous 10% speed change, caused the digitally generated time code output to be thrown into shock. AK perform a complicated comparison of the machine's output time code with the input time code to push the machine into lock as rapidly as possible. Disappearance of the correct output code therefore caused a problem — otherwise use of this output worked well. Replugging time code to the auxiliary tracks produced no such difficulty, enabling 'perfect' control to take place. Yamaha are investigating the other problem, which is software related.

The TIME CODE-OFFSET key restores time code into the LCD display, and also provides access to an offsetting capability, enabling the machine to be nudged into exact synchronisation with an external source — that is removing all tolerances. MENU provides access to various selections of time code type (24,25,30,30D frames/second). internal-external, chase or master, jamsync, video-sync, etc. Things are pretty comprehensive. MIDI time code can also be used if required.

MEMORY-ADDRESS IN is the means of entering time code positions for the Locate function. Pressing this twice will automatically enter the current ▶

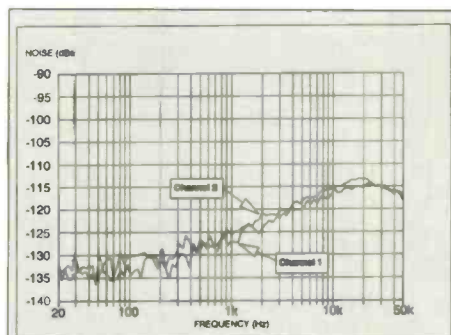


Fig 5: Yamaha AD8X and DA8X 1/3-octave noise versus frequency. Analogue input to analogue output, 44.1 kHz sampling frequency

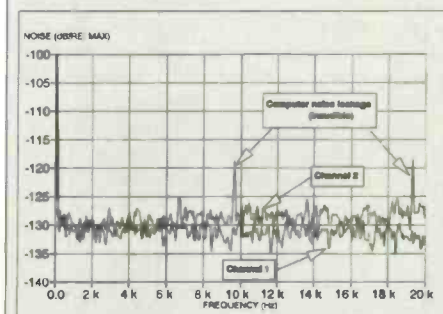


Fig 6: Yamaha AD8X and DA8X, noise spectrum (dBr). 16 times averaged FFT

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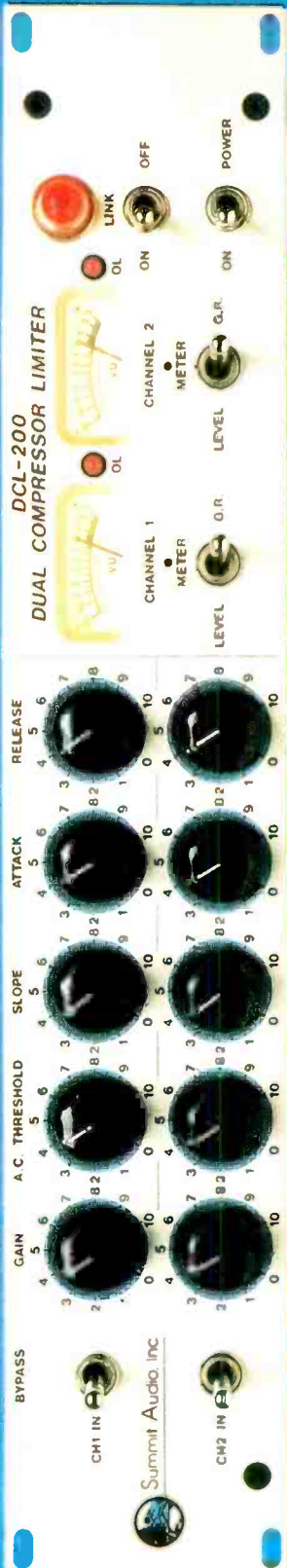
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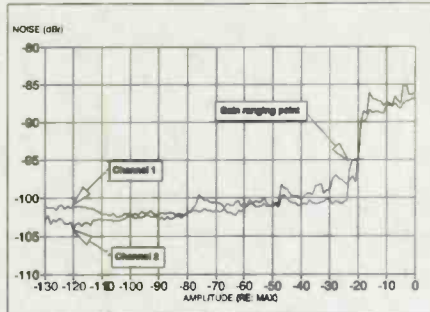


Fig 7: Yamaha DRU8 digital recorder, quantisation distortion 22 Hz to 22 kHz, referenced to full scale. Analogue input to analogue output, 44.1 kHz sampling frequency

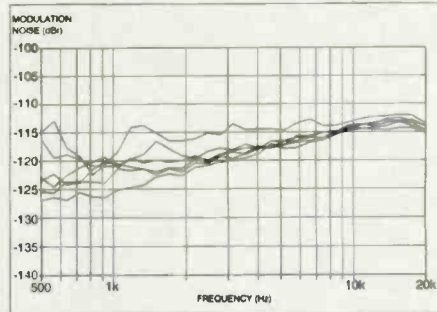


Fig 8: Yamaha AD8X and DA8X, modulation noise — 1/3-octave spectrum. The noise in the 500 Hz to 20 kHz region is generated by a 60 Hz tone between -100 dBr and -40 dBr exercising the D/A and A/D

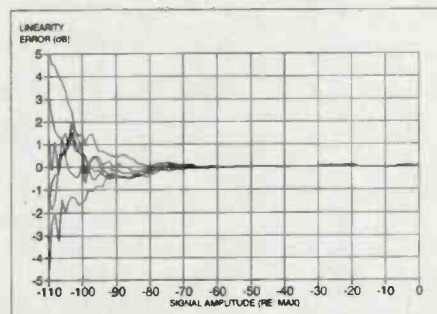


Fig 9: Yamaha AD8X and DA8X, analogue to analogue linearity test

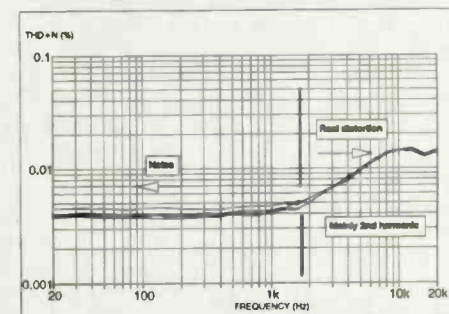


Fig 10: Yamaha DRU8 digital recorder, THD + Noise versus frequency, 80 kHz bandwidth. Analogue input to analogue output at 44.1 kHz sampling rate. Distortion is mostly 2nd harmonic, that is musically pleasant if you hear it at all

time code position. Otherwise, one press followed by use of the cursor keys allows manual entry. Pressing LOCATE returns to this position. Thus, the basic machine has one Locate memory, with a further 100 available when the RC8 or RC24 are in use.

START-STOP when pressed while the transport is stopped, outputs free-running time code to the time code output connector to use elsewhere in the system.

CHASE leads the machine into a system check, followed by entry into time code Chase mode if operations are successful.

Transport Controls

Describing these requires a complete table within the manual. Suffice to say that if you want to do something reasonable, Yamaha have provided a way, and there is some logic to the choice. Most normal procedures behave as one would expect.

Performance

In a digital recorder, given that the error correction and basic recording properties are well designed so that there are very few digital errors generated, there will be no effect on audio quality produced by the recorder itself. Therefore, aside from the DRU8 monitor chain, these measurements really relate to the AD8X and DA8X convertor systems.

Some trouble was had with the machine in trying to get things running properly in 48 kHz sampling rate mode. This may be finger trouble with us, but we could not discern the cause in the limited time available with the machine. Therefore, all of the following tests are in 44.1 kHz

mode. The real differences at 48 kHz should be small.

We generally like to examine the A/D and D/A systems separately to pinpoint the cause of any performance limitations. However, on the review system, there is no digital port which is easily accessible to our test equipment, except for AES/EBU output via the monitor. Therefore, all tests also reflect the complete A/D, record, D/A chain. We hope to follow in a short time with further tests where the signals can be intercepted in the digital domain, and with the 48 kHz sampling problem overcome.

When measurements get to the 19 or 20-bit level, comparisons with the analogue recording really do disappear — even with the Dolby SR in use. A real dynamic range of 107 to 108 dB means that the performance is limited no longer by the recording medium, but rather by the acoustics of the environment, and the physics of microphones and air. Summing up 24 tracks at this level reduces the dynamic range by about 14 dB, still leaving more than 90 dB available where our typical room only allows 50 to 60 dB. Noise is not there anymore. At little more than £1000 per track, this has to be good value for this performance level — so long as it works. The only limitation presently is the 22-minute recording time, which is something Yamaha are currently trying to increase.

Inputs and outputs: the maximum input level is limited by the input stage at about +24 dBu, certainly high enough. With input controls on the AD8X set to maximum gain, the resulting maximum levels, with and without emphasis are shown in Fig.1. This machine is clearly intended for a professional environment, since full professional levels are required to fully drive the ▶

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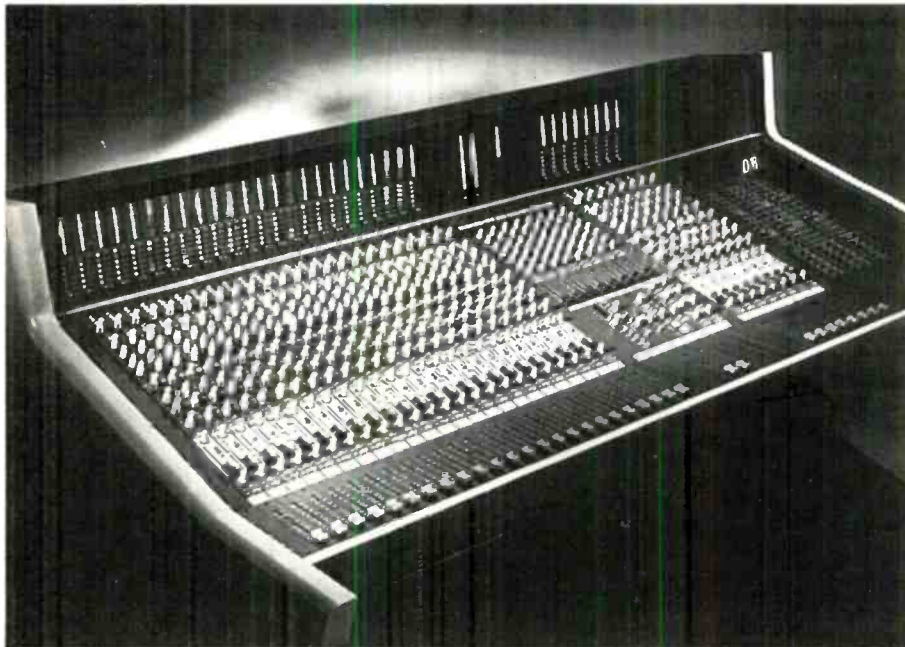
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Input common mode rejection is shown in **Fig.2**, achieving an average, but acceptable sort of performance level.

Frequency response: **Fig.3** shows the superb frequency response with and without emphasis. I would say that a worst case level variation of -0.2 dB was acceptable, wouldn't you? The effects of the anti-aliasing filters is shown in **Fig.4**. These are very steep, creating almost no additional delay at high audio frequencies. There was no measurable phase error between channels.

Noise and distortion: one of the main features of this product is the achievement of superb noise levels, making the recording process less critical while improving the resulting output. The wideband noise levels are shown in **Table 1**, indicating an operating dynamic range of 107 dB using conventional noise measurement techniques. This is less than the published specification of 110 dB, but the measurement techniques are different, and in our limited time we were unable to replicate Yamaha's methods.

Fig.5 shows the 1/3-octave noise spectrum, while **Fig.6** shows the same measurement accomplished by FFT (Fast Fourier Transform) techniques. Note that in **Fig.5** the frequency scale is logarithmic. The rising characteristic is a result of using a 1/3-octave filter — the bandwidth of this filter in Hz increases as the centre frequency increases, letting more noise into the measurement. The **Fig.6**, the frequency scale is linear, and spectrum is divided into equal sized bins. If the noise is 'white', then each bin will have the same contents, giving a flat response as seen. The FFT measurement is also more sensitive to individual tones, making the very low level computer noise breakthrough visible. We will be using the FFT type of measurement in all future reviews. Yamaha's technique is to take the FFT results and sum up the contents of the bins — we could do this, but just ran out of time in setting up new tests.

In **Fig.7**, the quantisation distortion is shown. Notice the 'kink' in distortion level at an input level of about -24 dB. The A/D convertor system gives a 19-bit resolution. It does this by placing a standard 16-bit convertor within a 3-bit convertor gain loop. At high signal levels, gain into the 16-bit convertor is reduced to minimum — giving 16-bit resolution, but at high levels the signal masks noise and distortion due to the convertor. As signal level drops, gain is switched in ahead of the convertor, maintaining the full usefulness of all 16 bits down to an input level of 24 dB below maximum. This gain ranging creates a 'conversion window' of 16 bits, to which is added the range value itself, giving a total of 19 bits resolution. Without access to the digital output of the A/D, we are unable to be conclusive, but there is a large gain change at about -24 dB, which might be the only gain change. That is what is shown in the graph. Note that quantisation distortion level drops about 15 dB when the gain is switched. A further 3 dB seems to disappear at about -76 dB, reaching the theoretical 18 dB benefit resulting from an additional 3 bits of resolution. Additionally, this graph is rougher than others we have measured, indicating some

deficiency in basic convertor performance.

Modulation noise is shown in **Fig.8**, and seems to indicate that either the convertors have some problems at low levels, or that dithering is not handled correctly. In a 16-bit system, this would be likely to be audible, but with 19 bits of resolution, the resulting modulation noise is still quite low. The best product we have seen so far in modulation noise terms is the BSS *TCS804* digital delay.

Fig.9 displays the linearity of the complete A/D/A process, which is almost error free down to -70 dB, remaining within about ±1 dB up to at -100 dB, then rising rapidly. The deviations occurring at between -80 and -100 are in part the

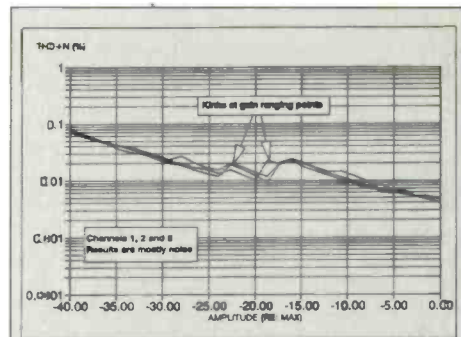


Fig 11: Yamaha DRU8 digital recorder, THD + Noise versus signal amplitude at 1 kHz. Analogue input or analogue output on channels 1, 5 and 6. 44.1 kHz sampling frequency, input level re: 1dB below full scale

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cause of the modulation noise described above. If the results are mentally shifted upward in level by 20 dB, the basic 16-bit convertor can be compared to other systems, in which case it is shown to be somewhat poor compared to other products.

All of these 'negative' comments about the convertor systems should be considered within the product's overall performance. First, the additional 3 bits at the top end will tend to hide performance errors at low levels, therefore they will not often be audible, and then only to special people; secondly, the product is in a competitive price bracket, directly hitting analogue prices with much improved performance; and thirdly, the recorder can be used with other manufacturer's convertor

systems where the additional performance is required.

THD+N versus frequency is shown in Fig.10 and versus level in Fig.11. The unit does just about meet the manufacturer's specifications. Noise is the limiting factor at lower frequencies, with some second harmonic distortion appearing above 1.5 kHz. Most people will not be able to hear this distortion, and if they do it is likely to be pleasant rather than unpleasant. Fig.10 mostly shows noise, which has kinks at several points. As before, these kinks are likely to relate to the 'floating window' nature of the A/D convertor system.

Intermodulation distortion measurements also give low readings — for example the SMPTE twin-

tone measurement using 60 Hz and 7 kHz at maximum level gives 0.008% IMD.

Delay: overall delay, input to output, is just over 2 ms, but can be increased independently on each channel up to 999 samples or about 20 ms.

Drop In - crossfade: crossfade time is very adjustable — up to about 2 ms. Audibly it is also very smooth, and when examined on a sampled waveform seems flawless. All of the record, drop-in and overdub features work well in practical terms. Purchase of the RC8 or RC24 remote controls provide an easier means to control track selection and monitor sourcing than the cursor keys and the LCD window. On these units, there are separate Tape and Locate time code displays, and individual record ready and track monitoring selectors. Additionally, 100 locate positions are provided. But, for a one or two-man musical team producing recordings by overdubbing, only a few settings are changed each pass — making the system useful in practice. Besides, MIDI allows rapid reselection of regularly used setups.

It says in the manual that it is possible to bounce a track onto itself, at least within the DMR8 system, enabling track bounce down without a spare track. We were unable to verify this though the block diagram also indicates that it is possible.

Auxiliary track performance: basic measurements were made of the auxiliary track performance, largely confirming the manufacturer's specifications. These tracks are intended to be used as subordinate control tracks, and Audio Kinetics have verified their suitability for that purpose. These inputs are unbalanced semiprofessional, using phono (Cinch) type connectors.

Summary

As a value for money product, able to be linked easily into MIDI-based music production systems, or operated in a video editing and production environment, the DRU8 system is excellent. In time code-based environments also, the system seems to excel. In other areas, the product shows its youth. Software on such products is getting more and more complex, while manufacturers try to retain a low cost hardware package. Complex software implies bugs, and the DRU8 has some. Nicely though, manufacturer commitment enables users to get value immediately out of an, as yet, imperfect product while allowing software to be upgraded in about 10 minutes when it is available.

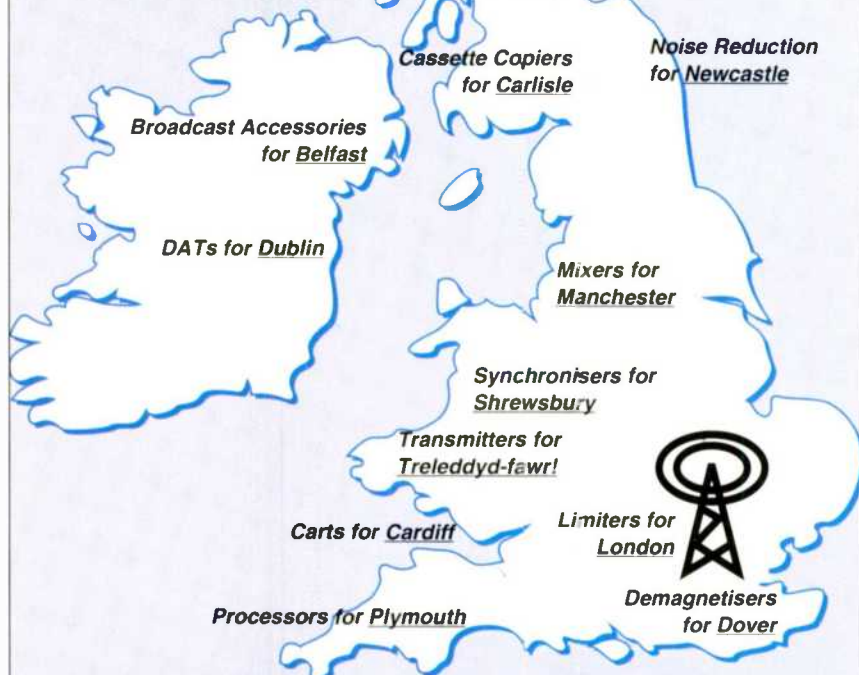
The control flexibility of the system is most impressive. A lot of thought has been devoted to this area, and important it is too, for no longer does a recorder stand alone. Most of the time it is part of a highly integrated production process.

Yamaha seem serious about the professional market. I was impressed by Marketing Manager, Allen Martin, who was both helpful and honest. If Yamaha follow his recommendations, then customers can expect to be well supported.

We look forward to spending more time with the product over coming months, and using the DMR8 and DMC1000 console versions as well, for DRU8 is only part of a larger system of products which are most interesting. ■

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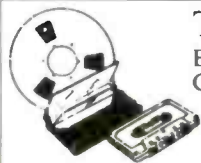
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I do not think I was ever much of a tape-op. I could never apply myself to learning the tape as thoroughly as some. Nor did I have the quality of memory for the sandwich, tea and coffee run. I was preoccupied with watching the engineer and client.

It must have been sometime early in my first studio post that the chief engineer parried my endless questions by describing the engineer's job as 20% skill, 80% PR. In fact, it was probably the worst thing that I could have been told because while I could understand the concept of skill, PR sounded decidedly out of place. There was definitely something else going on and I had to watch the engineer even more closely if I was to crack this PR thing.

A few days later another question and answer attack on the chief engineer generated the phrase that he could 'train me to engineer in a week but to be an engineer takes years'. That tape machine was going to have to look after itself while I tried to watch the engineer even harder.

Over the next few weeks it gradually became clear. The gear was only a tool and there were only so many ways of using it — the real variables were the musicians and the client. That was the difficult part. They didn't use the same phrases as those inscribed on the console controls. They used these seemingly vague descriptive terms to guide the engineer and I would watch with great interest which variation of EQ would equate with which emotive term. This was the skill of engineering — translating the communication.

Now we have all quite likely met the $\frac{1}{2}$ dB at 12.3 kHz types who I would find it easier to work with if I believed them; as well as those who get so carried away with EQ that they request 'more of all frequencies' — which is actually very easy to do — with the fader. But my interest has really remained with how emotive and descriptive phrases translate into console knob positions.

Some are, of course, easy — high end, mid and lows are fairly clear. The only difficulty may be where the mids begin and end, particularly as many mid-band EQs are now being designed as wide bandwidth EQ controls. How about *super* and *ultra highs* (Those frequencies that tickle your ears). Where are they on the dial?



Keith Spencer-Allen puts in a good word for terminology

If you're still with me, how about *sparkle*, *sizzle* and *sheen*. A recent ad in the consumer press suggested that *fizz* was a desirable commodity in tonality (less bucks, more fizz). They obviously did not mean what I mean by that. So which phrases infer what and are they desirable or not?

If a sound is *hard*, *solid*, *loud* (but not in level terms) or *dominant*, how do you *thin* it, let it *breathe* or *open* it up? It would suggest that having *impact*, being *punchy* or *attacking* may be related to being *dynamic* but then in what way does this differ from having *balls* or being *gutsy*. If you go too far, when are you *harsh*, *raspy*, *peaky* or *spiky*. Perhaps we need to get *sweeter*, *rounded* or *warmer*. If we are *flat*, how do we add more *life*, *kick* or *cut* while not getting too *busy*, *overpowering* or *intimidating*. That solo might need to *sing* but not *ring*, have *spread* and *brilliance*. Logic would suggest a connection between making something *warmer* and *hotter* but we are talking another language here. How would you *thin* out the *emotion* while keeping the *whoomf* (or *whoomph*)? We need it *squeaky clean*

but certainly not just *squeaky*! And I could go on.

In general this magazine has avoided the use of descriptive terms for sound because they are imprecise and have different meanings to different folks. Sometimes we do need to express sound or desired changes in sounds in words, and words are lacking when we use them to describe sound as they make so few direct connections. The story goes that the Eskimo language has 107 words for snow. Snow and its variations seems about as difficult to describe as sound so maybe it is not impossible to develop a way of achieving direct communication — sound subtleties in words. You may say 'Just why is he bothering — as long as we are all happy with the finished result, who cares?' In many ways you would be right. The problem comes with the printed word.

It would be immensely useful to know that the words of sonic description mean the same (or very close to it) on the printed page as that meant by the author. It would open up a whole new approach to

reviews and subjective assessment. In the not too distant past there was the case of a product being referred to as sounding *hard*. The manufacturer took exception and said that their product was not *hard* although they could live with the term *bright*. My analytical sounding monitor could be either *accurate*, *clinical* or *unmusical* to you dependent upon your perception. You see the problem?

To be realistic I do not think it will ever be possible to overcome this problem. It just has to be put down to that part of learning to be an engineer that takes the time — mastering the vocabulary and how to act on it.

Since writing last month's column I realised I should have mentioned a device that will help overcome the difficulty of keying the talkback button for someone else to speak, or if you are the someone else, getting someone to key the talkback for you. For several years Brainstorm Electronics have been making an infrared remote switch called *The Communicator* that interfaces with the console talkback system. It is hand-held and will operate the talkback from almost anywhere in the control room. The angle of infrared transmission is wide and does not need aiming at the receiver which would normally be placed on top of the console. The REMOTE switch is momentarily like the standard TALKBACK button and can be switched on the remote to one of four talkback groups. A useful appliance.

I was also interested to read a quote from Peter Gabriel in the 'Sleevenotes' supplement within the June issue of *Q* Magazine in which he put the musicians' insecurity very well.

'I always do that (sing in the control room) because it means you are never working against the mood of the producer and engineer. If you are out in the studio, you see them laughing behind the glass and you do not know what it is about. Call it singer's paranoia, but you want everyone there on your side . . .'

References

1. Brainstorm Electronics Inc, 1515 Manning Avenue, #4, Los Angeles, CA 90024, USA. Tel: +1 213 475 7570. No UK distribution at present.
2. *Q* magazine describes itself as 'The modern guide to music' and is published monthly by EMAP Metro, 42 Great Portland Street, London W1N 5AH. It has an international distribution.

Your genius needs a control surface

EINSTEIN

FROM THE MAKERS OF HENDRIX

AUXILIARY SENDS

8 auxiliary sends, 4 mono and 2 stereo, can be selected between the two input paths. Moreover, sends 7 & 8 access the 24 routing busses giving you a massive 32 aux buss capability. Using the buss level controls on the E30 module, full control of all aux output levels is in your hands.

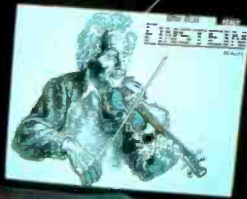
COMPREHENSIVE METERING

Each channel has its own LED meter with individual overload LEDs in each path. The meter can be selected to follow Bus/Tape signals or input channel pre-fader. Complete stereo signal metering is provided with follow monitor input section.

AUDIO GROUP MASTERS

Each buss output has its own level control with variable and fixed (calibrated) positions. Each output is also internally selectable for either 0dB or -10dB operation, providing full compatibility with all types of recording equipment.

...AND AUTOMATION
AMEK SUPERTRUE, a standard in console automation with over 200 professional installations worldwide, is much sought-after for its speed, accuracy and ease of operation. SUPERTRUE automates faders and mutes on both EINSTEIN signal paths, giving you 64 inputs with full automation. EINSTEIN SUPERTRUE is fully compatible with all AMEK MOZART and HENDRIX automation data.



TWIN EQUALIZERS - A NEW FIRST

Two powerful and richly-detailed 4-band AMEK Equalizers give total control of each audio path, going beyond split Eq to give absolute flexibility and independence on both paths. The standard 32-input chassis has, in fact, 64 inputs with full Eq.

VIRTUAL DYNAMICS

EINSTEIN can be retrofitted with AMEK VIRTUAL DYNAMICS, giving you a choice from 9 software-controlled screen-based Dynamics devices on each main fader input.

EASE OF OPERATION

The EINSTEIN control surface is extremely easy to use and has been designed to avoid the need for constant repatching of signals. Every signal always stays in the place, is always controlled by a fader, and always has full Eq. The advantages of this system will be immediately apparent to anyone who has worked in situations where the engineer is also musician and producer.

COMPREHENSIVE MONITORING

Three monitor outputs may be sourced from 5 external stereo inputs and 5 internal sources, with Mono and individual Left and Right Mutes. The copy function allows DAT to cassette copying without disturbing normal mixing. Oscillator, talkback and in-place solo are standard.

PRICE
Once again, this dynamite package is delivered for the right price. As usual, our promise is more computing and quality knobs for your money than you ever thought possible.



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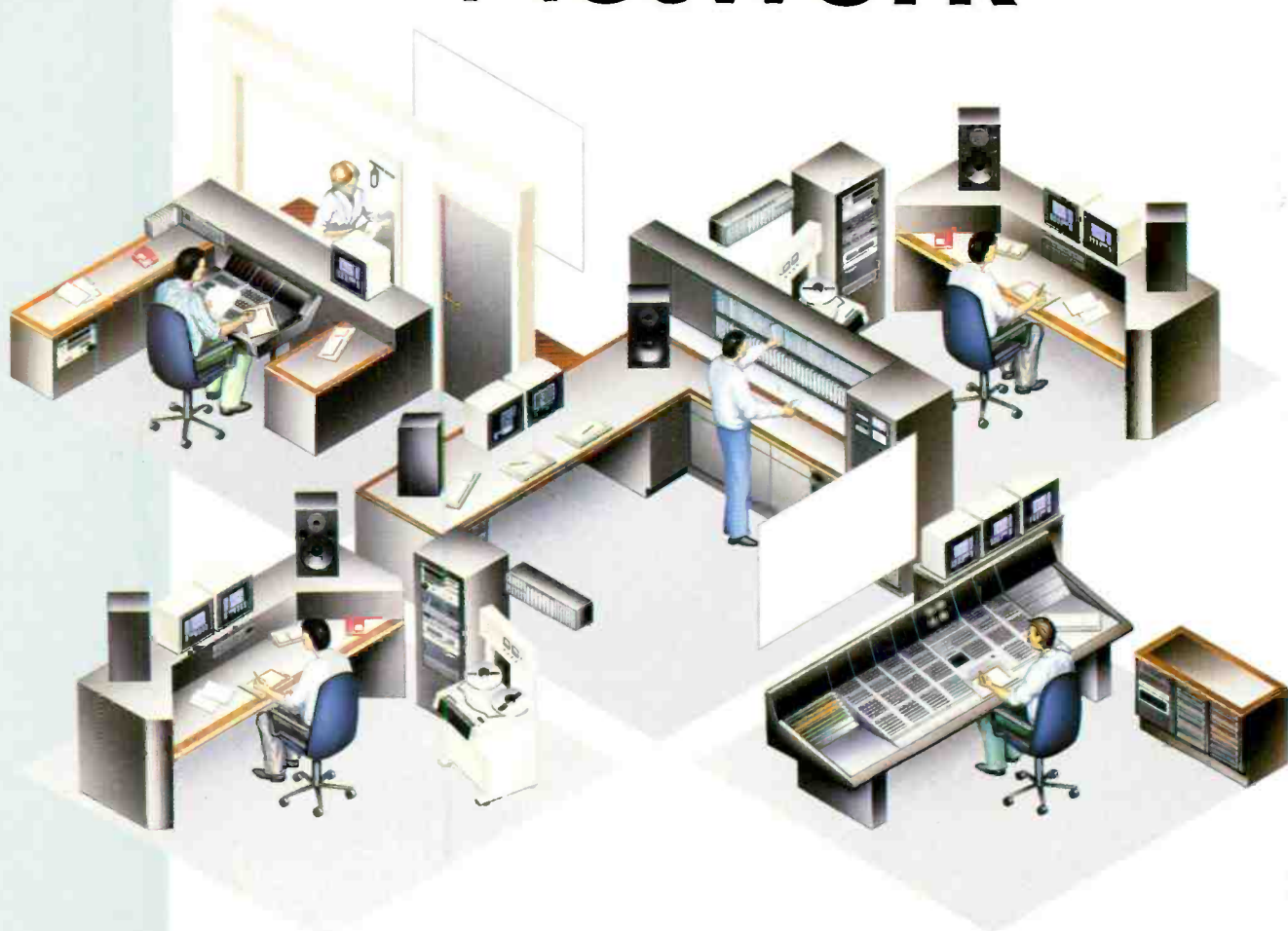
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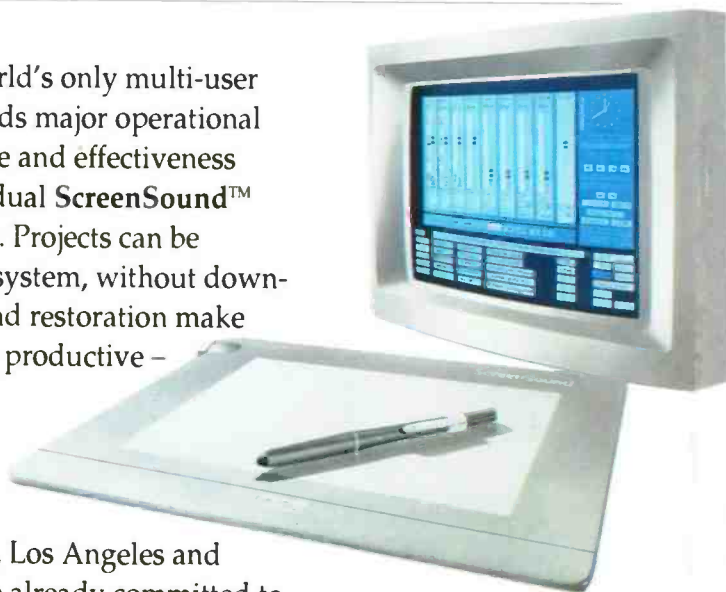
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