Studio sound

THE PIERCE ROOM
London's latest prestige studio

FAME
Exclusive review

THE TUBE REVIVAL
Valve outboard roundup

3-D AUDIO
Surround for multimedia

The Bryan Carlstrom
Interview

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Recording in the bathroom maybe a joke to some, but it's taken seriously at The Pierce Room Turn to page 42

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CARLSTROM INTERVIEW

Metal, mayhem and microphones are the essential ingredients in Bryan Carlstrom's engineering. As revealed to **Studio Sound**

REVIEW SECTION

STUDIO SOUND PRODUCT OF THE MONTH

For some studios only one console is good enough.

"The SL 9000 sounds fantastic, and the number of studios to have installed them around the world proves that it is now an established international standard. Olympic is a world-class studio and we therefore have to install equipment that world-class artists want to use."

Ian Davidson, Director of Operations, Olympic Studios, London

OLYMPIC STVDIOS

72 Channel SL 9000 J Series in Studio 1 from September 1996

SL 9000

Solid State Logic

THIS EDITION'S CONTRIBUTORS

SUE SILLITOE has been involved in pro audio for so long that she suspects she must be related to Thomas Edison. Apart from children, her production credits include regular articles in *Music Week* and *Pro Sound News* as well as a review column for the Press Association

DAN DALY covers the machinations of the professional audio industry extensively world wide, as well as writing for publications like *USA Today* and *International Business*. His involvement in audio includes successes as a music composer, producer and studio owner.
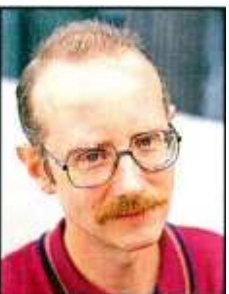
DAVE FOISTER runs the Guildhall School of Music and Drama's recording facilities in London. A *Tonmeister* involved in the music business for decades, Dave has been a *Studio Sound* contributor for 10 years, and reviews gear in the setting of a professional working environment

The hot seat

A funny thing happened to Digital as he was on his way to domination of the pro-audio industry: he met Valve. Digital hadn't seen Valve for a while–in fact, he didn't really know her very well. Recently, Digital had been making a lot of friends and Valve had been feeling her age and had adopted rather a low profile.

To begin with, the more Digital told Valve of his popularity and success, the more Valve felt old and tired. Digital's arrogance told her all she had suspected about herself was true: she was old-fashioned and unwanted, she belonged to a simpler, more graceful age when people had respected her subtlety and mystery. The world seemed to have moved on, and Digital had definitely moved in.

Valve found Digital odd; he was fair to a fault, but somehow Valve couldn't warm to him. She couldn't find anything to like about him–certainly he was efficient and reliable, but she couldn't attribute any particular character to him. He was almost invisible, transparent. In contrast, she regarded herself as being full of charisma, and able to be something different to almost everyone depending on their tastes.

She readily became involved with people, playing on their moods and emotions. But Digital, she decided, was a collector of souls–he kept a distance, didn't relate to people at all, he was like a mathematician trying to describe a woman in love. And hadn't she given her heart to wisdom, and to madness and folly, as the Bible had prescribed?

VALVE WENT HOME to reflect on her past and her apparent obscurity. She looked up a few old friends from whom she might draw a little comfort–and got the surprise of her life. Maybe she wasn't news any more, but quite a lot of people remembered her fondly. After all, in the early days of radio she had been the talk of civilised society the world over? And hadn't she been one of the forces' sweethearts as part of the renowned Radar show? Certainly people still talked about her, and some even missed her. Why, some people had even tried to persuade other technological circles–including Digital, and with very limited success–to do the things that had made Valve audio's belle all those years ago. Best of all, some of her friends had become celebrities in certain circles. Very prestigious circles.

Maybe it was time to stage something of a comeback–hadn't Dorothy Parker once said something about there being stories that every generation needed to tell for itself?

To her delight, Valve found that she hadn't really been forgotten at all. It was more a case of her not having been properly introduced to the younger generation. Fans all over the world revered her, even told stories about her talents and exploits that she wasn't sure were entirely true. Most of them were suspicious of Digital. Some of them reckoned he was too good to be true, and delighted in his failings while others, seduced by Valve's virtues, were left cold by his lack of character. Her fans desperately wanted Valve to make a full return to the entertainment scene, and to put Digital in his place.

But Valve was above having a public confrontation with Digital. She could see, even if some her fans couldn't, that Digital had virtues of his own and that time would see him become a mature and valued part of everyone's lives. And Valve had always prided herself in her ability to bring something special to a party...

She decided to befriend Digital.

editor

soundin

The IBC experience

AMSTERDAM: Expectation is a terrible thing, but in the case of IBC 96 we had good cause to expect a great deal. After last year's clash with the ITS in Montreux, and the staying at home of the so-called Big Five companies, this year's event had the broadcast field to itself, the huge conglomerates had returned and there were more exhibitors than before, with additional halls accommodating the extra capacity. So why did it all seem a little lacklustre?

It was big, there was no doubting that, and many of the companies had gone to town and the surrounding suburbs with their stands. The Sony display (always Sony) was like a mini hall in itself, with sections devoted to each of its areas of interest. These included the launch of Betacam SX, promoted with a rather crass ad tag along the lines of the joy of SX, which in itself showed how much the IBC is still largely a vision exhibition, despite the best efforts of the organisers.

As in previous years since the move to Amsterdam there was a dedicated audio hall, although there was co-habitation with visual peripherals, including camera supports and automation, subtitling devices and teleprompts. Although some audio companies chose to be out on their own, there were others who had wanted the camaraderie but found themselves in the heavily vision orientated halls.

Another factor for the lack of excitement could be that the European AES had taken place not that long before, and many audio specialists had chosen to make their major launches there rather than at IBC. One of the majors, AMS Neve, didn't exhibit at all, something that, perhaps, caused more bar talk than what the new audio direction was. (The official explanation was that IBC 95 was felt to have been 'quiet' and the company decided to do one show in each territory, in this case AES.)

AMS Neve is, of course, a leader in digital consoles for the broadcast and postproduction sectors, and this sector is seeing a distinct shift towards more recognisable user-interfaces. Notable among these are Amek's Digital Mixing System (DMS), the related, badged FAME for Fairlight, and the Amptec Stone-D 001. All three of these could be identified with Amptec's own blurb, that claims the desk 'changes the industry without changing the feel'.

The implementation of new technology in familiar and usable forms was a recurring theme. Still with consoles, surround sound is now becoming something of a standard feature. Soundcraft is already a leader in this field and added the B800 Surround to its range, while D&R offered the Cinemix and Yamaha added the SoundWare SP02 joy-stick controller to the 02R, which is making inroads into audio post and dubbing work.

Although not physically present, Dolby's influence on the world of surround is almost tangible and Digidesign announced a Dolby Surround plug-in for Pro Tools, which is fully digital, using I-O DSP. While the US Grand Alliance for HDTV has adopted Dolby AC-3 or Digital, the discrete 5.1-channel format, European broadcasters are hanging on for MPEG2 Musicam surround, which was finally demoed at Amsterdam.

Fronted by Philips and IRT, the system was linked with DAB and DVD, although, when I dropped by for a demo, the video disk player had been whisked off to Belgium for another demonstration, underlining the continuing delay problems this new format is currently experiencing.

Developers have been turning their attention to sound for computers of late, and another member of the expanding MPEG family appeared at IBC for this purpose. MPEG1/2 layer III is designed for low bit rates, and is being touted as 'hi-fi for computer networks', providing soundtracks for either H263 video or audio-on-demand and Internet services.

Initials and acronyms are everywhere, very often linked with codecs. Maycom displayed ISYS, an ISDN MPEG2 layer II device, while apt made a full entry into the broadcast market with the BCF 256, which can give dedicated lines of up to 15kHz stereo with ISDN backup.

The most interesting trend of the 1996 IBC was the number of SSC (Small, Sensible and Cute) products, units that may not necessarily be at the cutting edge of technology, but which are eminently useful and have been designed from experience of the application. Among these are ASC's Workstation Buddy Junior, an audio interface for DAWs that offers mic and line inputs; headphone outputs; and is ideal for radio reporters; the 360 Systems Short Cut editor; and motionworks' R2P2 remote controller that works on the Sony P2 protocol. This last one is so sweet you could put it on your mantelpiece.

KEVIN HILTON

ITALY: The latest Spanish post and mastering room to open is the Milan-based Nautilus facility. Equipped with a 32-channel AMS Neve Logic 3-AudioFile Spectra console and recording system, and and PMC-Bryston LB1-BB5XD monitoring, Nautilus' Blue Room also boasts Prism Sound DA1 24-bit reference convertors, Apogee Electronics AD1000 A–D conversion and UV1000 noise-shaping, Sontec MS432 parametric EQ, Focusrite Blue 330 compressor and Ampex ATR100 ½-inch machine. Facility design is courtesy of UK-based ADG. Nautilus, Milan. Tel: +39 2 469 2029. Web: http://www.planet.it/nautilus

AUSTRALIA: Following its temporary move to Sydney in 1995, the Australian Regional Convention of the AES returned home to Melbourne

gs

GERMANY: Ulm Cathedral's historic Statewide Trombone Day recently celebrated its 100th anniversary in style with two Schoeps CCM3 omni mics hung just 7.5m above the players involved in recording the proceedings for posterity. Since its inauguration in 1906, the festival has grown in stature to include nearly 10,000 wind players arraigned in a 100m circle under the direction of conductor Erhard Friess. The event attracted some 15,000 to 20,000 visitors. Schoeps, Germany. Tel: +49 7219 43200.

this year. So it was that the 6th Convention recently took place between 10th–12th September at the World Congress Centre.

The programme featured technical tours, papers, workshops and a small exhibition hall. Among the venues for the tours were national broadcasters SBS and ABC, local theatres and the Melbourne cricket ground—a notoriously problematic live and outside broadcast location. The six paper sessions attracted contributions from Australia, the US, Hong Kong, Malaysia and the UK, and embraced coding and transport of audio, amplifier technology, DSP, DAB, transducers, and the ever-contentious topic of education in audio. But perhaps most popular with attendees were the 21 workshops that covered areas as diverse as basic MIDI, location recording, audio postproduction and theatre design.

In contrast, the 60 commercial exhibitors were unanimous in their dissatisfaction regarding the number of attendees—which unconfirmed reports place at less than 600. Distributors and manufacturers criticised the AES' publicising of the event, while the AES Committee noted that few exhibitors had attempted to attract customers to the Convention. With this year's event showing a significant drop from 1993's attendance at the same

venue, and with the AES charter precluding co-participation with other audio events, many are now questioning whether the Australian market (population 18 million) can sustain an independent Convention. Nevertheless, anticipating a commercially viable solution, the Melbourne AES has already rebooked the venue for 1998.

GORDON REID

UK: The recent Plasa show was the event promoted by critics of the APRS show organisers' intended amalgamation with Vision for 1997. The choice of Vision—being a video-based show—as an APRS partner is both obvious and precedented in the format of other shows such as IBC. But the critics were having none of it; the audio showing at last year's Vision was not well visited and Plasa, although primarily a live show, was the one that had demonstrated its ability to grow at a remarkable rate. And hadn't Frankfurt's Musikmesse increasingly proven the viability of a joint live-audio show over recent years? As a result there was an 'audio floor' at Plaza where audio was invited to make a stand alongside the legions of live manufacturers and distributors.

◆ Tokyo's Omnibus Japan film and video facility has recently purchased a Daniel Weiss 32-channel digital console. The console features GML fader automation and 12 channels of dynamic and EQ processing. Daniel Weiss Engineering, Switzerland. Tel: +41 1 940 2006.

◆ London's Abbey Road and Townhouse studios have both added TL Audio EQ2 valve parametric EQs to accompany their C1 compressors. Other recent EQ2s have gone into Mute Records' London studio, Ocean Colour Scene's Birmingham studio and Dave Gilmour's Thames boat studio.
Abbey Road, UK.
Tel: +44 171 266 7000.
Townhouse, UK.
Tel: +44 181 932 3208.
TL Audio, UK. Tel: +44 1462 490600.

◆ LA-based Cherokee Studios has adopted BASF SM900 maxima as a house replacement for 3M 996. Cherokee specialises in music tracking to analogue machines. BASF, US. Tel: +1 617 271 4000.

◆ The London HQ of the national Radio 1 station has adopted Denon DN-990R MD and DN-951FA CD machines to cover a spread of broadcast functions. The CD machines employ Denon's Cart format for CD-R replay while the MD machines will provide jingles, inserts and so on. Denon, UK. Tel: +44 1753 888447.

◆ Kuwait's Al Nazear has taken a 64-input Harrison Series Twelve digitally controlled analogue console for music recording. The console features 48-track busing, 32 remote mic preamps and producer's desk. More Eastern action has seen The Korean Broadcasting System order two 36-input Harrison MR-20 consoles. Harrison, US. Tel: +1 615 370 9001.

◆ Atlanta's Synchronized Sound has installed a 32-channel Neotek Elan production console in its surround-capable Studio C. The Elan is equipped with Audiomate automation and will see service in sweetening, ADR, foreign language dubbing and mix to picture work. Synchronized Sound, US. Tel: +1 404 873 4477.
Neotek, US. Tel: +1 818 281 3555.

◆ BBC Cambridge has chosen the Micron MDR560 diversity receiver system for its VHF Mk5 radio car. The system will use a TX503 hand-held transmitter with omni capsule and have a range of up to 300m. Audio Engineering, UK. Tel: +44 171 254 5475.

◆ Stockholm's OAL Studio has ordered a 48-channel, 48-group Raindirk Symphony console and accompanying AD Systeme Optifile Tetra automation package.

Raindirk Audio, UK.
Tel: +44 1336 382165.
AD Systeme, France.
Tel: +33 1 42 53 3118.

◆ Rio de Janiero's new Studio Mega has been designed by South American Acoustic International Design and built around an SSL SL4000 G+ console. The facility currently commands Brazil's highest rates and chalked up its first Platinum selling project after just three weeks operation. SSL, UK. Tel: +44 1865 842300.

◆ Portugal's first Soundcraft Europa console has been purchased by Oporto-based PA hire company Aplauso. The 40-channel desk continues the tradition of Soundcraft consoles favoured by Aplauso's owner, Manuel Barbosa. The first Ghost console has been taken by Irish long-wave radio station Atlantic 252 where it will be used for in-house recording work. Soundcraft, UK. Tel: +44 1707 665000.

◆ Scotland's Perth College Faculty of Arts had added two MTA desks to its arsenal. A 32-input, 24-group split Series 980 and similarly spec'd in-line Series 990 will use Audiomation Audiomate automation and feed 48-track DA-88 MDM machines. Perth College, Australia. Tel: +44 1738 621171.
MTA, UK. Tel: +44 1252 318700.
Tascam, UK. Tel: +44 1923 819630.

◆ Germany's 25th Logic 2 console will go to the Technical University of Berlin's Studio 4 Electronic Music studio. The 20-fader desk will be run with DAWs for sound design and mixing contemporary electroacoustic compositions requiring 12-channel monitoring. Perpetuating the Neve history, California's Music Annex has installed a second vintage Neve console in the form of a restored 56-input, 48-bus, in-line 8128 with Flying Faders in Studio C. The new desk complement the 8036 in Studio A handling mixdown. Music Annex, US. Tel: +1 415 328 8338. AMS Neve, UK. Tel: +44 1282 457011.

◆ Big Ben, the notorious signature of UK BBC World Service broadcasting is facing a facelift in the form of the replacement of its aged 1967 vintage Electro-Voice 635A mics in favour of four new mics—E-V 635As. Although the existing mics are functioning, such a renovation would parallel the replacement of the UK telephone time service Revox tape machines over 10 years ago. Mark IV Audio, Deutschland. Tel: +49 9421 706 0. Shuttlesound, UK. Tel: +44 181 640 9600.

soundings

And some did. But while the likes of HHB and TL Audio tried, and largely failed, to take advantage of the vitality of Plasa, the hoards of the live sound business were more likely to chance across Sennheiser, beyerdynamic and Neutrik stands lurking among the dry ice, lights and general noise of the live equipment show taking place downstairs. It could be that the upper floor was badly signposted, it could be that the presence of audio had not been sufficiently promoted, or it could be that the pairing is a bad one.

In Plasa's own show postmortem the organisers claim to have entertained in excess of 15,000 visitors, hosted 300 exhibitors and overseen the launch of over 400 products. Included in these categories are assorted foreign shores including Australia, Japan and Portugal, making the show as international as any relevant even in the UK. Yet in the selection of endorses' effusive comments there is not one that relates to pro audio...

Perhaps the ultimate test will come with 1997's Vision and Sound Show is over, and a more balanced evaluation can be made. For now, however, the jury is still out on the future of the UK's show business.   TIM GOODYER

US: American-based communications systems company Telex scored a major installation of a number of audio matrices at the recent Atlanta Olympics. Involving a 400 x 400-point ADAM digital audio comms matrix with RTS TIF-951 telephone hybrids for American broadcaster NBC-TV, the systems were set up in the Georgia World Congress Center and served NBC, the Canadian Broadcasting Corporation (128 x 128 ADAM matrix), the Australian Channel 7 (80 x 80 ADAM matrix), Finland Broadcast (64 x 64 ADAM-CS matrix) , and the Korean Broadcast System and Manhwa Broadcast System of Korea.   NICK SMITH

EUROPE: Manor Mobiles became one of Europe's largest mobile recording operations recently, with the acquisition of a fourth mobile recording truck. The new addition to the Manor stable is the 48-track vehicle previously owned by Barcelona-based El Camion.

The purchase comes within weeks of the company's recent take-over of the Zipper Mobile —a compact 24-track unit that will specialise in live radio and television broadcasts, and which is now known as Manor Mobile 3—during August. The two new mobiles join the 48-track Manor Mobile 1 and Mobile 2. Of the move, Manor Mobile Operations Director Mike Oliver says: 'The continuing and growing demand for our services has made it imperative for us to expand fast.

'In the past year, company turnover has doubled due largely to the increasing number of major European live events on the calendar and TV shows like Chris Evans' *TFI Friday*, *Saturday Live*, *Hotel Babylon* and *The Big Big Talent Show* hosted by Jonathan Ross.'

The El Camion Mobile (renamed Manor Mobile 4) will be based in mainland Western Europe to reduce transit times to recording venues and events taking place on the continent.

'Both Manor 3 and 4 have stood the test of time,' says Mike Oliver, 'proving beyond doubt that they are made of the right stuff. They are now undergoing a thorough "fitness test" and upgrade to ensure that they meet our standards.'

Back in the UK, Manor Mobiles 1 and 2 have enjoyed a hectic summer that has included recording both the Phoenix and Reading festivals, *The Three Tenors* at Wembley and Oasis at the Royal Festival Hall.   TIM GOODYER

www.americanradiohistory.com

Indulgence and excellence

Just as different musical styles demand differing playing skills, they require different approaches to recording. But should this discrimination be extended beyond playing and recording to the recording equipment itself?

The typical pop record is an entirely artificial creation rather than a recording of a soundfield that once existed. Sound sources are manoeuvred by pan pot into apparent relationships that they have never really had. Equally, the relative levels set in the mix have no relation to the level in the session. Extensive signal processing is used so that the waveform of each sound source may be quite different from that the microphone captured. The reverberation is completely artificial and bears no relation to the acoustic of the recording space.

The record is a manufactured result of creative manipulation by the producer who will often have a particular 'sound' that becomes a carefully guarded trademark. Often the requirement is for novelty in order to appeal to a market that is transient. Pop songs have the life-span of a butterfly, and the great majority are consigned to oblivion once the novelty has worn off. Only a few achieve classic status to be played indefinitely after. Such is the skill of some producers that the band involved may be almost completely lacking in talent, yet a professional result can be obtained. No wonder some bands have such difficulty performing live. Nor is it surprising that many of that producer's records sound similar despite being 'recorded' by different artists.

None of the above is meant as criticism; rather an acknowledgement of the way it is. Pop music is part of our culture and pop-record production is an art form that has simply evolved over a very short period in comparison with other forms of creative art.

The recording of classical music represents a considerable contrast. The musicians are generally highly qualified in an art form that has evolved over an extended period, and become highly refined. For most of that period audio amplification and recording was unknown. Instruments that needed it had their own 'amplification' in the form of a resonant sound box or a horn which would give subtle characteristic sounds to each instrument.

In the absence of mixing consoles, classical musicians and singers evolved the ability to play at the correct level relative to one another. Essentially a classical performance is mixed by the composer through the interpretation of the conductor. In the absence of amplification, concert halls were designed to bring the sound to the audience, adding their own richness of reverberation. The various instruments of the orchestra were positioned according to their volume, with the louder contributions at the back in a stable layout. Music enthusiasts are highly familiar with the individual timbres of instruments and their traditional position. Many are familiar with the characteristics of individual halls.

Bearing in mind the huge contrast between the ways that classical and pop music have evolved, differences in approach are inevitable. Such contrasts are evident in other art forms. In painting the artistic skill resides in the painter and the subject is secondary. Everyone knows who Leonardo was, but hardly anyone knows anything about Mona. Spatial accuracy and perspective are seldom correct. The painter may put his subject in front of any background he chooses. The landscape painter might omit the gasworks. Seascapes may grow spurious boats and seagulls which were never present.

The photographer has less freedom because the photographic art resides more in recording what is actually there. Spatial accuracy and perspective are more accurate than paintings, except where painters like Canaletto used photographic techniques (camera obscura) to get the perspective correct.

So the pop producer is a painter who starts with a blank canvas and creates any picture he pleases for an audience who have no preconceived idea what it should look like.

The classical producer is a photographer who finds the masterpiece of the composer and conductor in the surroundings of the concert hall and has to realistically reproduce it elsewhere to an audience who may have a pretty good idea what the original was like. So the goal of the classical producer is to put as little as possible between the performance and the audience. It is to use a light hand that makes little improvements here and there without altering the nature of the piece—rather like the photographer who removes up the beer can that would other wise spoil his shot.

THE CONTRAST between the approaches precipitates a question: how do the respective requirements influence the design and performance of the recording hardware? Should a rock recorder differ from a classical recorder? Well, certainly, the rock tape recorder will need more channels because of the mix-pan approach, but the sound quality may also be different.
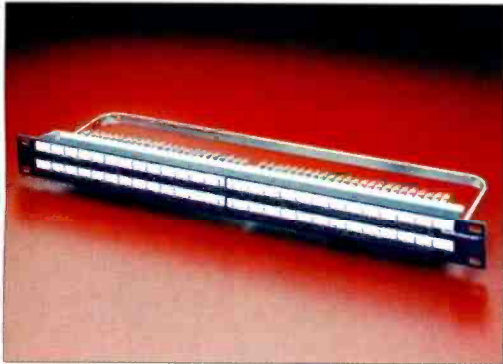
The criterion for classical recording is utter transparency in microphones, desks, and recorders, and orchestra, because the audience is critical and knows what the original sounds like. The monitoring must be neutral and run at an SPL similar to the original performance.

Transparency is not necessary in rock recording because no-one can compare the final recording with the original performance. Sometimes deficiencies in the equipment are used as effects; second harmonic distortion, for example, adds warmth, while an overdriven analogue tape acts as a crude compressor. Mixing is usually carried out for self-indulgence at frightening level using monitors which sacrifice neutrality in the interests of su-rvival. These agricultural implements sound pretty awful on classical material, but are all part of the rock idiom.

We don't regard painting as better or worse than photography, but as a different endeavour. Consequently we should not make any judgments about the relative merits of pop and classical technique. What we should do is to be quite sure which technique we are adopting and do it properly.

Take a chance on me
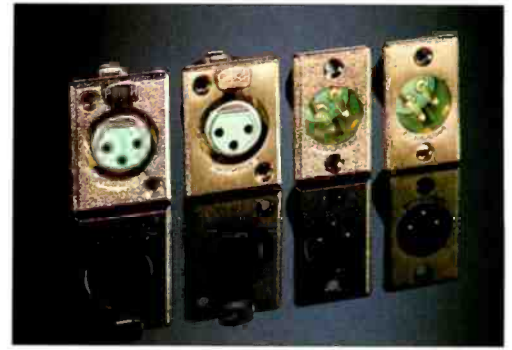
Having suffered the contraction of the recording market, studio designers are busy again. But the rules have changed writes **DAN DALEY**

It has been said, by myself and others, that studio designers are the most ideological bunch in the grand pantheon of pro-audio personages. They reside in a cosmos fringed and fraught with maths as obscure to most of us today as they were back in middle school. Designers can joust on a geometric plane high above most of us, and the innately grungy business of operating recording studios.

Well, to a degree, anyway. But, perhaps, to less and less of a degree. American designers were as ideological as the next guys. American studio design kind of hit its stride in the late 1970s and 1980s as the music business became the entertainment industry—and America established itself as the worldwide capital of that sometimes salubrious enterprise. Studio designs that had originally owed their heritage to a combination of radio-studio design, and the equally boxy and off-the-shelf geometries of early British studios, gave way to a striking diversity of design combinations. Some of the most innovative were places like Caribou Ranch in Colorado, with its vista-in-the-control room approach; and The Plant in Los Angeles; and Electric Lady in New York, which took ever-greater advantage of the use of odd configurations and surfaces.

American designers covered a broad swathe of approaches, from the ever-lower-frequency obsessiveness of Tom Hidley (he may be geographically peripatetic, but he is still an American) to the soaring ceilings of John Storyk, who probably did as much as anyone to bring a higher level of aesthetic to interior studio design, to Russ Berger's quintessentially American Gothic that ranges from elaborate home studios to radio networks.

But as the industry has changed, so has American design. Or, more to the point, the putative tastes of American studio owners have changed. America remains the routeing switcher of the entertainment industry's central nervous system, but the body politic of that culture is becoming more global on a daily basis. US film companies already budget productions based on losing money in the States, and making their profits on overseas rentals and sales. And the rootsy Americanisation of music that began its resurgence in the mid-1980s is running out of steam in the face of the musical Esperanto of dance favoured in Europe and Asia. Combine this with the massive proliferation of alternative studio environments—everything from warehouses to basements—and it's hard to see how the design community could remain insular.

What's been seen in the States in recent years is a handful of top-end designers increasingly surrounded by new entries into that field, most of whom are aiming at the so-called project studio market, which itself has matured to the point where it now at least dimly recognises the need for more than a room that holds a bunch of equipment.

Diversity is a good thing, a concept that, despite the never-ending debate in this country as to what it constitutes and how it should be implemented, is still the basis for the American culture. But diversity in entertainment, and in the technical trenches that produce its fodder, has created its own debates among designers. The upside is that the US is seeing a spate of new high-end room construction, one that may be partially attributable to the regular cycle of structural replacement that all industries go through, but which is definitely driven to a large extent by the need to compete more internationally. The downside for US-based designers is that they are feeling the competition from the new shingles being hung out on doors, and on the Internet, by a whole new crop of designers, with and without portfolio. They're also feeling increasing pressure to be more international in their approaches. For instance, the new Starstruck facility in Nashville (see last month's *Studio Sound* exclusive) chose UK-based Harris Grant Associates for its design, predicated largely on an international reputation, and an international design approach that fits with the studio's marketing plan—which was well into its formulation stage before the foundation concrete was dry—to attempt to not limit its appeal to American artistic tastes.

This combination of forces has exerted an effect on the US design community; there is a palpable sense of pragmatism pervading the ideology that once led the field. Technically speaking, some point out, the trend has been towards a diffusor-loaded, control-room rear end, with a front end that steers reflections towards it as a kind of catch-all approach. That could be considered a global response, as well, since the competitive pressures are not being felt here alone—all design firms mention that they are harder pressed than ever to cover a wider range of clients and spaces. But US designers are finding that more and more of their work is coming from below; the high-end studio renaissance is by its nature limited in volume—how many people can afford multi-million-dollar rooms?

Fran Manzella, owner of the FMRTS design company in New York, observes on both notions. 'I'm working on so many different strata of studios now,' he says. 'More than ever before. And it's not just me; that's the way the business has gone here. Everyone has had to become more practical in terms of what they can propose and do because everyone is having to look at smaller and smaller gigs mixed in with the large ones. Designers here are learning to cater to the low-end and middle-class of studios, because that's where the lion's share of work is coming from these days. The problem that US designers face is that this business is increasingly populated by people who read a story in a magazine, or a chapter in a book about design, and then think they know all there is to know.'

As for the sense of internationalism that appears to be creeping into American studio design, Manzella tips his hat to his British colleagues for that one. 'What's happening is that studio clients, here, are no longer just buying what they hear, they're buying into an aesthetic, particularly an international aesthetic, whatever that is. And in many instances, that means buying an international designer. There's still an American aesthetic—I think you can still tell a California recording studio form another one. And (US-designed) New York studios continue to look like what they always did: kind of rootsy and less sleek and finished-looking than those in other cities. But the ones that were designed by non-US firms, like the new Hit Factory, you can see the international influence at work. Certain acoustical approaches are becoming more commonly accepted and the emphasis is switching towards the aesthetic aspects of studios.'

John Storyk told me some time ago that more and more of his work is coming from overseas, a place, interestingly enough, where US designers hold as much allure for developing technical cultures as American music has had for their entertainment cultures. So American designers are having to look increasingly downward and outward to find the horizons of their futures these days. Downward is fine, since that's where the technology is leading the business, anyway. And outward? Well, at a time when the US is undergoing yet another debate on how high the walls should be around this continent, perhaps a very good thing after all. ◉

Shame about the sound

When the classy classical open-air productions that grace the English countryside fail to deliver sound to match their looks, it's time to learn a few lessons from other territories, and other musics writes **BARRY FOX**

Meteorologists assure us that Europe's weather is getting warmer. If they are right, then there will be more outdoor concerts, with larger crowds, and promoters will have to rethink their policy on amplification. Audiences will no longer tolerate a sound system that sounds like an amplified telephone. Promoters who cut corners will go to the wall.

I'm not talking about the mega-events that require the services of Clare Brothers, Britannia Row, Showco, and their like. I'm talking about organisations that promote open-air classics and equate a high-power system with deafening pop sound. Of course, once you have heard a high-rated system cruising with plenty of headroom, you can never again live with a low-rated system that is straining to cope and ends up sounding like a cardboard box. But if you have never heard high-power cruising, you book the cheapest system available. By cheap I mean barely into a 4-figure sum.

In California, where open-air concerts are a way of life, the audience would rebel. So it's ironic that one of the biggest disappointments is one of Europe's most glorious settings, which was modelled on the Hollywood Bowl.

In 1948, Frank Wright—music adviser to the Parks Department at the London County Council—visited America and went to a concert at the Hollywood Bowl. Fired with enthusiasm, he persuaded the LCC to start staging concerts in the bowl-shaped grounds of Kenwood House, a stately home in North London. The first Kenwood concert took place in July 1951, with the orchestra in a shell-shaped stage on the distant side of a lake.

After some experiments with tin whistles and balalaikas, it was decided that no amplification was necessary. The sound bounced off the water. By 1979, audience numbers had grown to over 5,000, and amplification was brought in. For years it was very mild, just a few speakers hidden in the trees near the shell to lift the quiet passages.

In 1986 the British government killed off London's council. English Heritage took over Kenwood and now manages it along with the many ancient sites (including Stonehenge) which EH does a good job of preserving. Kenwood audiences have grown and they have got much noisier. People who go to pop festivals now expect the music to be loud, and talk through it. At Kenwood the audience picnics and chats. 'Bad concert manners', complain the EH organisers.

British company RG Jones has provided the sound for as long as anyone can remember. They deserve a medal for doing what they do on a budget that would be laughable if it were not so sad for the audience (up to 8,000), who pay over £10 just to sit on the grass. I have been going there for years, have heard the situation deteriorate, and nagged English Heritage to take sound seriously. Nothing happened. On the first night of the 1996 season the orchestra sounded like a distant transistor radio. Some of the audience were hearing only the sound of boxy loudspeakers hanging in the trees.

Now music critics have joined the cry, branding the Kenwood sound 'tinny'. 'Save money and replace the orchestra with a trumpet gramophone', advised one reviewer. So it is no longer just a finickity hi-fi buff who is complaining. EH now sees revenue disappearing as sound quality becomes an issue and audiences start to dwindle.

The official line is that the stage shell has been made deeper to accommodate more musicians, and this makes it behave like a horn loudspeaker. I'd say the root cause is more simple; the sound system is under-powered.

After criticism snowballed, the sound system was upgraded to 6kW at each side of the stage, with four delay zones of 1kW each. Remember that this is for an open-air audience capacity of 8,000. The peanut budget does not run to cranes to fly the cabinets. So the left and right stacks are so widely spaced that the front seats cannot hope for a central image.

By comparison, when Tina Turner played Wembley this year the 350kW Clare system, trucked out of Basel, Switzerland, had a stack of 70 4-way cabinets at each side of the stage, plus a few more to fill in the sound gaps at the front. The cabinets are a mix of long and short-throw designs, to protect ears at the front while delivering a good level to those at the rear. The Who's reunion in Hyde Park, also a Clare gig, used 120 cabinets on stage, but with eight waves of delays round the park. But that's what pop people need, I hear the English Heritage advisers advising.

IF IT WERE NOT for the fact that, as a government-funded body, it would cost British taxpayers' the fare, I would suggest an EH educational trip to Concord in California.

The Concord Pavilion was built 20 years ago, thanks to local business-man and jazz enthusiast, Carl Jefferson. Jazz buffs will know Jefferson's name for the Concord record label, which was founded in 1972. Jefferson then began a local campaign to raise money for an open-air concert pavilion.

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In 1973 a local landowner donated over a hundred acres of barren farm territory, and the local city authority stumped up $4m building money.

The core was a bowl, hidden by the surrounding hills, much like Kenwood. The Pavilion opened in 1975 with a concert by Sarah Vaughan. Since then there has been at least one jazz festival every year, with other concerts by everyone from Russia's Red Army Chorus, through the Beach Boys to the San Francisco Opera and Symphony Orchestra.

The Pavilion is like a roofless building, supported by pillars and covering seats for 3,500 people. Outside the roof, and spreading up the hillside there is a picnic area for at least 5,000 more. So capacity is virtually the same as at Kenwood.

But the sound system, from JBL, is very different. It cost half a million dollars. Around 250 units are suspended from the roof over the stage, with a mix of 18-inch and 12-inch bass drivers, and 1-inch and 2-inch, mid and high-frequency compression horns, driven by 74 JBL-Urei 6290 power amplifiers, with a total rating of 88kW. There are four waves of delay. The overall effect is quite magical. Over 8,000 people hear intimate sound from the stage.

The lesson for Europe is that no-one in Concord expects good sound to grow free on trees. 🅂

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Fairlight FAME

The alliance between Fairlight and Amek enjoyed a high profile launch earlier this year, and the first offspring of the marriage is now shipping. **DAVE FOISTER** takes an exclusive look at the Fairlight Audio Mixer Editor—or FAME

THE OBJECT of the alliance between Amek and Fairlight was to bring Amek's expertise to the design of an integrated mixing and editing package built on Fairlight's established MFX3. The result is FAME, the Fairlight Audio Mixer Editor.

Integration is seen as vital by manufacturers and users alike, and has become both more necessary and more complex as digital technology has taken over. Generally all it comes down to is the ability for one piece of kit to talk to another, whether from the same manufacturer or not, so it has revolved around file formats, digital-interface protocols, and machine control and synchronisation systems. In the light of many other products on the market, it would be easy to assume that FAME was either a neat hook-up between the MFX3 and an existing Amek console, or an elaborate hardware controller for the Fairlight's existing functions, rather like a

Panel graphic detail of the channel strips

grown up CS10. Neither is the case; FAME is a complete integrated system, using Fairlight's audio-processing engine under the control of Amek's console technology to provide a working environment neither could achieve any other way.

The success the MFX3 has already achieved is quite remarkable, with several heavyweight facilities using it exclusively. When the likes of Warner Brothers and Todd-AO place such heavy reliance on a system—both have several MFX3s—then the system must have got something right. That something is generally cited as speed and ease of use; the MFX3 is very streamlined to enable it to do all the standard DAW editing tasks quickly and logically from a single screen, and can be learnt in a couple of hours. So ubiquitous has it become in Hollywood film facilities that one could be excused for assuming it was designed with that job specifically in mind, but in fact it is just as happy standing in for a multitrack recorder in a music studio—among others, ZZ Top record directly into their MFX3s.

Fairlight's track capabilities have long led the field, and a full-blown MFX3 has 24 inputs, 24 outputs, and 24 tracks each with up to 4,096 layers for multiple takes, although only one layer on each track can be used at a time. Audio on the tracks is cut up into clips for the usual array of copying, cutting, pasting and moving functions, and each clip can have its level independently set as well as fades in and out. Powerful 4-band EQ is available and is also clip based, and Fairlight's well-received TmeFX processing is standard, allowing clips to be stretched or contracted to fit a time slot and so on. That's about it: there's no automated mixing, no further effects, no screens full of mouse-controlled faders, and that is perhaps part of the appeal of the system in that it does the central job very powerfully and quickly without clutter. Its control panel, too, helps to make it very quick in use, and because it is designed with this in mind looks very functional and ordinary. It is little more than an enhanced computer keyboard, with additional keys for the principal editing functions, dedicated transport controls, a jog-shuttle wheel and a pair of small LCD screens for soft-key functions and timing information. Nothing is ever more than a couple of keystrokes away, and the things

needed most often are right there under the fingers. Many keys have LED indicators built in, making it easy to keep track of what's happening without cluttering the screen. This combination of a very nuts-and-bolts control surface and a screen designed for clarity rather than client-impressing potential makes the MFX3 look far less glamourous than many other DAWs, but for once operational considerations have triumphed over image in determining the product's success. Fairlight's clear aim of speed and ease of use has been achieved, and that is what is required above all else in the high-pressure world where the MFX3 has done so well.

With such a small range of processing

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possibilities provided within the system, the assumption is that the audio, in a choice of analogue, AES-EBU, or unusually Yamaha digital format, will be mixed through a separate console. Of course full synchronisation and control facilities are built in, making the MFX3's use with other equipment very easy, but at the same time there is a lot of unused DSP power in the MFX3's cards that could be doing the job on-board—hence the development of FAME.

The other half of the partnership is equally well established, although the product from which the FAME contribution is derived is a newcomer. June 1996 saw a *Studio Sound* overview of Amek's DMS Digital Mixing System, and it is the work

surface and control technology from the DMS that forms the basis of FAME's main console. Automation itself is nothing new to Amek, and many features of FAME's automation operate very much like the SuperTrue system familiar from Amek's analogue desks, but the new control surface was designed with assignable all-digital operation in mind, making it ideal for use with the Fairlight.

The point is that while the DMS uses Amek's own audio engine, FAME uses nothing but the standard MFX3 processing power, with software jointly written to enable the console control surface to provide comprehensive facilities from what is otherwise quite a simple machine. The console retains most

of the physical features of the DMS, with the addition of a section in the centre carrying a modified version of the MFX3's usual control panel. All the Fairlight's keys and functions are retained, including the small displays and the wheel, but the layout is slightly rearranged so that the panel takes up less space. Thus operation of the entire system is carried out from the console, with editing controls, mixing, processing and automation all equally to hand. It leads to the odd situation where the same function often appears to be duplicated; there are two sets of transport controls, two different sets of locators, and two different sets of EQ controls, but all have their separate uses depending on the particular job being worked on. For 🖙

FAME interface overview—literally

instance, the MFX3 can act as a master machine to the console, locked to other machines via 9-pin, word sync, video, and so on, if required, or the console can be the system master, in which case MIDI machine control is available. Using the console as master also provides interfaces to MicroLynx, Zeta3 and MotionWorker systems. In the first instance both sets of transport buttons are operative, the two surfaces talking to each other via the special 10Mbit serial link used to connect the two computers.

THE STANDARD FAME configurations have 8, 16 or 24 faders and 36 input paths, giving access to the Fairlight's 24 tracks plus 12 additional inputs for overdubbing, effects returns and so on. It is important to remember that

Faders can be assigned to MFX3 tracks, inputs, buses and most other things chiefly by means of 16 fader sets which can be instantly selected from a bank of switches near the screen–rotary controls for pan and other parameters are surrounded by LED rings

the MFX3's recorded signals never leave its own boards, so that its 24 inputs and outputs are available as output buses, inserts, aux sends, monitor feeds and talkback without compromising the flexibility of the central system. This is one of the most important points about FAME, and the concept that makes it a truly integrated system. Any DAW should be capable of being made to work with any console, and the two together should behave as a complete system, 'integrated' in the sense that their timings will follow one another and they can probably be jointly operated from a central point. It will still be necessary, however, to have a separate signal bus, be it digital or analogue, for every signal channel that

passes between DAW and console or vice versa. Thus even a high capacity system like an MFX3 would normally use all its 24 outputs to deliver all its 24 tracks into a console, and further mixing capabilities or processing power would be needed in the console, together with more inputs and outputs for auxes, inserts and final mixed outputs. FAME goes a stage further by creating its mixer inside the MFX3, leaving all its I-O capabilities available for talking to the outside world. This allows it a maximum of 12 insert sends and returns, 12 buses and 12 aux sends, although clearly not all at once; buses and sends are grabbed as required by the system without the need for convoluted user configurations. No outboard processing is required at all apart from the kind of specific effects, such as reverb, the MFX3 cannot produce itself.

Access in and out of the system is helped by a 256-way crosspoint matrix switcher that can route outboard devices in analogue and digital format to channels or external returns under automated control. One of these switchers is supplied as standard, partly to deal with monitoring and talkback, and the system can handle up to 17 of them. Routeing of external signals to the MFX3's record tracks is handled by the normal Fairlight routeing page via the conventional input ports.

The mixing side of FAME is operated globally by means of FAME's touchscreen and locally by moving faders, a small selection of controls on each physical channel strip and a large central assignable control panel. Faders can be assigned to MFX3 tracks, inputs, buses and most other things chiefly by means of 16 fader sets which can be instantly selected from a bank of switches near the screen. Electronic scribble strips are standard, and in conjunction with the moving faders this means that there is never any confusion about what is where. The rotary controls for pan and other parameters are surrounded by LED rings showing current positions, and, of course, being continuous rotary encoders are always live and ready for use without nulling. The same is true of all the controls in the assignable central panel.

This all combines to give the FAME system virtually the same functionality as the DMS itself. The console adds further EQ, again 4-band with a fully-parametric mode alongside a more straightforward setup with switched bandwidths and switched LF and HF. It must

be remembered that this is not achieved by the addition of any further processing power and that this is therefore not Amek EQ bolted on to the Fairlight; this is extra EQ created using the MFX3's existing DSP. It can be added to a track that already contains EQ'd clips, and adds further treatment rather than replacing the clips' EQ. Besides this, it is fully dynamically automated, and the control moves stay with the piece of audio to which they have been applied even if it is moved elsewhere in the piece. The sound of the EQ takes equal priority with the functionality, and is very musical indeed, combining flexibility and subtlety with precision and corrective power when needed. Because the processing is being carried out by the Fairlight DSP, there is no reason to suppose that the EQ sounds the same as that on Amek's DMS, although it is certainly of a very high quality. The only thing it does lose compared with the DMS is the choice of EQ flavours that Amek offers with the DMS' own engine. The MFX3 ☞

Panel detail—EQ, auxes and dynamics under assignable control

MFX3 users will recognise elements of FAME's heritage

already has an EQ curve display available on its screen, showing any existing clip-based EQ.

As standard the control of this EQ, like most other channel functions, is carried out from the central assignable panel. FAME's DMS can be specified with complete channel strips fitted if required, but the standard setup has the limited channel-strip facilities becoming familiar from a wide range of assignable consoles. Thus each

> Since the console comes from Amek there is the option of adding the Amek 9098 Rupert-Neve-designed remote controlled mic preamps, which can then be controlled from the desk. Also included is on-screen control of a range of familiar outboards

fader has pan, mute, solo and automation controls, an electronic scribble strip, and a few indicators attached to it. Everything else is controlled by calling the required channel on to the central panel, although there is also the possibility of globally reassigning the faders so that they all control some specific function across the board such as aux sends. The now-familiar advantages of the central panel are

clear; all adjustments can be made from the monitoring sweet spot, and the controls can all be given a bit more space as they don't have to fit on to one long narrow strip. Even so, FAME doubles up on the functions of some of the controls, with, for instance, a variable Q available as an alternative to the two default switched shapes and adjusted by the frequency control.

Also controlled from the central panel is the dynamic processing. This was not yet implemented on the system I saw, but the controls were in place and showed the range of functions on offer. The module includes a fully variable compressor, complete with gain-reduction metering, as well as a gate-expander, again with the usual set of adjustments. Both are available simultaneously, and the whole package is available to each channel in the system. Dynamics parameters are continuously automated as are EQ, aux, faders and pans.

Analogue inputs are line level, but since the console comes from Amek there is the option of adding the Amek 9098 Rupert-Neve-designed remote controlled mic preamps, which can then be controlled from the desk. Also included is Virtual FX which provides on-screen control of a range of familiar outboards including Lexicon, Yamaha and tc electronic devices by means of graphic representations of their control panels.

Comprehensive surround monitoring is provided, with six monitoring outputs controlled by the matrix switcher. Surround panning is fully catered for, with LCR, front-back and surround width adjustments all provided by the two pan controls on each channel. For more ☞

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Fairlight's MFX3 Digital Audio Workstation

elaborate surround mixes, a surround multiplexer option is available, allowing up to three additional predubs of up to eight channels each to be heard alongside the main mix.

FAME projects remain fully compatible with standard MFX3 project data, and mix data is automatically uploaded into the MFX3 computer as part of the project. Further work can then be carried out on the audio on a straight MFX3, with mix information remaining intact when the project is moved back on to FAME, although under these circumstances moving a clip does not take the associated mix moves with it.

THE IDEA behind FAME is very appealing, with its circumventing of all the problems a mix-and-match system can entail and its realisation of integration in its true sense. The danger of this kind of approach is that of putting all one's eggs in one basket, trusting all elements of the system to be as good at their jobs as the components you could have bought elsewhere. Fairlight have again avoided the

pitfalls, firstly by building FAME on probably the most successful workstation of its type, and secondly by joining forces with established experts in the mixing field rather than trying to reinvent the wheel. The many devotees of the MFX3 will find the self-sufficiency of FAME a very elegant proposition, and its potential to broaden the appeal of the existing system still further is shown by the orders already received from broadcast facilities. FAME is a bold move, but one that appears to have been carefully thought through. I felt more than comfortable with it in a very short space of time, true to the spirit of the MFX3 itself, and its flexibility and intuitive operation should see it forming the hub of many studios right across the spectrum. **S**

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XTA DP200

History is littered with examples of inventions that have found success in applications other than those originally intended for them. **DAVE FOISTER** examines XTA's DP200, and finds it has potential beyond that of a live sound processor

XTA IS A NAME best known in the world of live sound, appearing as it does on a range of graphic equalisers, spectrum analysers, and mic-line distribution systems. The DP200 processor belongs in this sphere as well, but its sheer power gives it a broader appeal—possibly more so than its manufacturer intended.

The DP200 sets out to provide a broad palette of treatments and processes of the kind required to set up a large PA system, all carried out in the digital domain. It contains enough processing power to operate as various combinations of cross-overs, delays, limiters and filters, and most spectacularly as a very large parametric equaliser. The different processing chains are selected from a menu

> The full-blown 16-band stereo parametric gives the DP200 more bands than some simple graphics, recalling the rarely used term 'paragraphic'—which could almost have been coined with this unit in mind

of configurations that allocates the DSP power accordingly, and make use of the two pairs of outputs in a variety of ways.

Not that you'd guess from the front panel how much power is contained within the box behind it. This is one of the most bland and unassuming panels you'll see, looking like a leftover from a 1980s delay line. It has three continuous rotary controls whose functions vary according to the operation in hand, two banks of three buttons for selecting those functions and choosing EQ bands, and a 2-line display to show what's happening. These may seem like meagre tools for controlling 32 bands of parametric EQ, but it's clear enough, and after all in its main application this is a set-and-forget unit.

But that is the scale of processing that

lurks within the DP200. It has two inputs and four outputs, and can either have all its processes in the same dual or stereo path or split both inputs in two, sharing the processes between the resulting four output channels. High-pass and low-pass filters are available in all channels, along with a maximum total of 32 fully parametric EQ bands followed by independent delays. Adding limiters to the paths steals enough DSP to reduce the available EQ bands to 14 per channel in dual-stereo modes and 5 per channel in quad mode, but this is still a pretty sizeable amount of EQ power. The full-blown 16-band stereo parametric gives more bands than some simple graphics, recalling the rarely-used term 'paragraphic', which could almost have been coined with this unit in mind.

The EQ bands are all identical, with a full sweep range from 20Hz to 20kHz, and gain adjustable from +15dB to -30dB. A large range of bandwidth control is provided, with around 100 Q values from 0.4 to 128, giving fine control coupled with the ability to create extremely narrow notches. Two of the bands can be switched to operate as shelving equalisers, with ranges that meet in the middle at 1kHz and 15dB of boost and cut available.

In Dual mode, all the bands are completely independent of each other, while in Stereo mode the control is ganged between the two channels as it is for all the other parameters. XTA is understandably keen to emphasise the

advantages of digital implementation in this context, giving exact matching between channels as well as accurate repeatability.

THE THREE front panel controls are labelled specifically with EQ in mind, giving dedicated knobs for FREQUENCY, Q and GAIN in the selected band, but they acquire different functions, shown clearly on the display, when other parameters are to be adjusted. Velocity sensitivity is built in to the encoders, so that turning them more quickly gives greatly accelerated rates of change, which is just as well since the available increments are very small —0.1dB in the case of the boost-cut.

The filters are unusually flexible, as they need to be if they are to be able to create complex active crossovers. Thus each filter has a choice of slopes, offering Butterworth and Bessel characteristics at 12dB or 24dB per octave, as well as Linkwitz-Riley at 24; the latest software update adds third-order slopes. These options are all available even when the filters are simply stuck on the ends of the equalisers for general use, but in the dedicated crossover configurations they really come into their own.

Four crossover modes are available: independent dual, and ganged stereo 2-way; single-channel 4-way; and 3-way plus sub; in which one input gets split three ways and the other gets its own signal path through to the fourth output. In all modes the filters are completely independently adjustable, with different slope characteristics if required, to allow the crossover to be set up exactly as the drivers demand. All outputs also have limiters available as well as adjustable delay. There is still enough processing left over for a fair amount of EQ besides, with even the most complex crossover mode, the 4-way, having six parametric bands before the crossover and two bands in ☞

XTA's DP200 offers enough power and precision to qualify it for mastering applications

each filtered output.

Whatever the configuration, delays on the outputs are independently adjustable up to 682ms each, in increments of 21µs; in this case, faster changes are made possible by a x10 key. The limiters also offer comprehensive control, with a choice of three release times, fully adjustable attack, and a threshold variable over a 30dB range related to the chosen input level setting. Individual limiter activity can be shown on the display as it is adjusted, and there is also a complete screen showing all four limiters' operation on gain reduction bar meters.

Flexible bypassing is provided, with separate switches for bypassing a complete channel or only the currently selected band or filter. These appear next to a SCROLL button which selects the channel to be worked on; individual elements or sections of the channel—parametric bands,

> The degree of control afforded by the DP200's front panel, never mind the remote software, makes it quite clear that this is a precision tool, and this is reflected in the sound it produces. The EQ in particular is about as neutral as it could be, in line with its intended function of correction and alignment rather than creative manipulation. To say that it has no distinctive sound of its own is not a criticism but a compliment

delays and so on—are selected with UP and DOWN keys that also navigate round the various options accessed by the MENU key. This is where the basic configuration of the unit is set, as well as operating levels, channel linking, and memory handling. In view of the complexity of the DP200 it is only sensible that it has 40 user-memories for recall either from the front panel or remotely, which can now be given names in the latest software. Security is managed from the menus, allowing a 4-digit number to be chosen as a lock-out password.

Further menu screens give away an interesting option, one that follows logically from the digital nature of the unit. It is possible to have it fitted with AES-EBU inputs and outputs, using the existing back panel XLRs under the control of recessed switches, and, unusually, each pair can be individually switched. This gives the possibility of having, for instance, a digital input, and all analogue outputs, or of having one pair of outputs analogue and the other digital.

REMOTE CONTROL as standard is via MIDI, but the board can be replaced with a serial control board, opening up the possibility of full remote operation. This in turn allows the use of XTA's dedicated Windows software package known as AudioCore, which offers all the control possibilities of the DP200's front panel on screen in a form that is rather easier to get round quickly. The full menu of configurations is available, and the chosen setup is shown graphically as a flowchart. Clicking on any of the elements in the chain opens up a screen to adjust that element, offering nudge buttons for all parameters and even more flexibility in some instances than the unit itself. For instance, delay times can be set in terms of distance—metric or Imperial—instead of milliseconds if required. As an adjunct to this, it is possible to incorporate the current temperature in order to allow the software to correct for variations in the speed of sound, although temperatures have to be entered manually. All inputs and outputs can be given names that then appear on screen in several of the displays to aid clarity.

The EQ and limiter screens both have graphic representations of the currently set parameters, the EQ in the form of a response curve responding almost in real time to the control adjustment, and the limiter as a transfer function graph where the threshold plateau can be dragged up and down with the mouse. The delay and gain screens score over the unit's front panel by offering all the channels, inputs and outputs at once, greatly speeding up the setup.

The AudioCore software allows multiple units to be controlled centrally by assigning each a device ID. Thirty-two such devices can be controlled, and can be chosen from the whole range of suitable XTA units, which currently also include the DP100 stereo 4-output delay line, and the DP202, a scaled-down 2-channel DP200 with just the parametrics and limiters.

The degree of control afforded by the DP200's front panel, never mind the remote software, makes it quite clear that this is a precision tool, and this is reflected in the sound it produces. The EQ in particular is about as neutral as it could be, in line with its intended function of correction and alignment rather than creative manipulation. To say that it has no distinctive sound of its own is not a criticism but a compliment. For more mainstream uses as well there is a place for neutral EQ; in this context neutral is not a synonym for characterless. In particular, I would think its combination of transparent control, precision, resolution and inter-channel matching would have a strong appeal in the mastering process, all the more so with digital I-Os. Perhaps XTA should consider applying this processing power to a specifically studio-orientated unit; until this happens, don't be put off by its apparent purpose if there's a chance this much power in one little box could be useful to you. **Ⓢ**

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BPM Studiotechnik CR-95

The German tradition of quality studio microphones is alive and well, and being continued by more than the obvious selection of manufacturers. **DAVE FOISTER** weighs up the latest FET microphone from BPM Studiotechnik

ITS STRONG ADVERTISING presence aside, relatively few British engineers are familiar with BPM Studiotechnik microphones. The ads do, however, promise various models very much in the tradition of classic German microphones—big, functional and metallic. Close inspection of the microphones in the flesh confirms the impression, showing a high standard of construction and an emphasis on mechanical and operational needs rather than cosmetic ones.

Current ads show the TB-95 with added glow making it quite clear that it is a valve mic, but it is otherwise extremely similar to the CR-95 looked at here. This is a conventional FET design, with a large side-firing diaphragm, a good selection of facilities, and clear high-performance aspirations.

First impressions are very striking, as the whole kit comes in a flight-case style aluminium carrying box with purpose-cut foam for the CR-95 and all its accessories. These consist of a substantial foam windshield, a well-constructed elastic suspension mount, and a rather upmarket XLR signal lead made from cable called The Voice by Equinox. There was no paperwork with the review mic other than a spec sheet (in German)—no frequency response curves or polar pattern plots.

The suspension mount is the only way of attaching the microphone to a stand, and it is commendably secure and positive, bearing a striking resemblance to that provided with the antipodean Rode NT2. The CR-95 is not light, but stays where it's put without drooping or wobbling about, and its controls remain fully accessible. There are two switches, providing the usual range of facilities in a slightly unorthodox way. The front one selects the polar pattern, with the usual choice of three—omni, figure-of-eight, and cardioid. The second is on the back, and in one direction inserts a 10dB pad while the other end adds low frequency roll-off, making it impossible to use both simultaneously. Both switches are flush with the body surface, out of the way of accidental nudges but still operable with fingers.

The dual-diaphragm capsule assembly is clearly visible through the open mesh of the grille, showing it to be a 1-inch capsule reminiscent of AKG's CK12. The obviousness of the large capsule and the size and styling of the body conspire to give a strong aura of old-fashioned quality almost calculated to inspire confidence in the musician it is placed in front of. It is more reassuring still to find that the sound of the CR-95 bears out the impression created by its appearance.

WITHOUT DOUBT this is a microphone to be reckoned with. It has a big open sound characterised by sharp focus and immediacy, without the hard edge that such descriptions might suggest. The top end is crisp and present but never harsh, and not at the expense of the rest of the spectrum; fullness, warmth and body are not lacking, and indeed flickers of inaudible low frequencies showed on meters more significantly than they did on my top-notch comparison microphone. The overall impression is of uncoloured naturalness, and this also extends to the off-axis pickup which comes over as particularly neutral, suggesting reasonably accurate polar patterns. The combination of close perspective and wide response make it a very interesting and rewarding vocal microphone, as well as an excellent all-rounder suitable for anything requiring high quality flat pickup, just as one would hope from the design.

There are several microphones around that think that looking the part is enough to gain acceptance, regardless of the sound they produce. The CR-95 invites such a response by virtue of its styling, but in fact is more than capable of standing on its own two feet in terms of its performance. If it's not already an option on the shopping list it should be.

Accompanying the review CR-95 but not specifically designed for it was a dual-channel microphone preamplifier intended to interface phantom-powered microphones with simple consumer-style DAT recorders. BPM's Roadstar is a straightforward portable box with a sturdy metal case carrying two end panels with the works attached. At the back are the unbalanced outputs on gold-plated phono sockets, together with trays for a pair of PP3 type batteries—rechargeables were provided as part of the package, along with a suitable charger, and the whole kit comes in a sturdy canvas carrying bag.

Appearing on the front is switchable phantom power, including full 48V, and a similar slide switch provides attenuation of either 30dB or 60dB, suggesting the box's use for line level balance sources as well. These are mounted below the XLR input connectors, between which appear LEDs for battery state indication. The final facility is a phase reverse switch for Channel 1, completing a comprehensive package which should overcome the limitations of many of the DAT machines in common use, which may have been chosen for portability and price rather than the quality (or even the presence) of microphone inputs.

These two items presumably represent the two extremes of the market BPM hope to address; both sit well with the competition and do an excellent job, encouraging the thought that this is a company one would like to hear more of. ◉

www.americanradiohistory.com

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Aardvaark AARDSYNC

The rise of digital-audio systems has given rise to more complex sync requirements than audio has previously known.
DAVE FOISTER evaluates an American synchroniser that will give the same sync standards as video
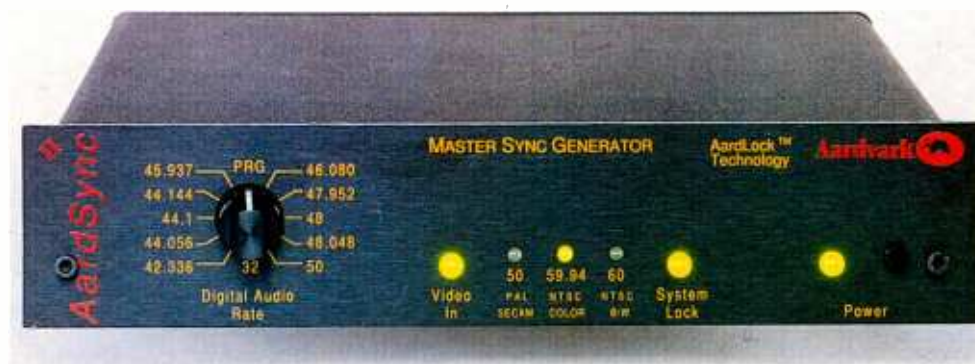
THE MORE COMPLICATED it gets, the more complicated it gets. Synchronisation in the analogue world meant nothing more than making two machines run together, with musical sense and lip-sync being the only requirements. Analogue signals, being continuous varying voltages, did not need to be synchronised to sub-millisecond accuracy in order to be mixed and processed, which was just as well as it would probably have been impossible. When we acquired our second digital recorder, or our first digital mixer/signal processor/DAW, the spectre of signal synchronisation began to loom large; suddenly digital words needed to be kept aligned with one another, otherwise even switching cleanly between them, never mind mixing them together, was impossible.

All this was already old hat to the video fraternity. All the video signals within a system have to be time aligned so that their frame edges occur simultaneously for exactly the same reasons—cuts, fades and wipes between them are otherwise impossible—and for everything to work properly it must be locked precisely to a stable and accurate reference. The signal from a VTR, in particular, is notoriously imprecise as a sync source for anything else. Early on TV facilities realised that lashing the kit together in different ways for different jobs, and trusting one bit to synchronise with the others was not a professional approach, neither was it convenient nor was it efficient. The idea of centralised house sync was soon adopted, where an accurate dedicated sync pulse generator provided synchronising signals for the whole facility, every other piece of equipment—cameras; VTRs; time-base correctors; switchers; mixers; monitors —being locked to it.

The same principle is the ideal for digital audio. Most digital equipment automatically locks its clock to any incoming audio signal, but relying on this alone has its limitations: a mixer with two sources cannot synchronise to both unless the sources are themselves in sync; the whole system becomes dependent on the accuracy and jitter performance of the clock in the master device; 2-way connections can lead to all sorts of problems with each clock trying to chase the other, and possibly ending up at the wrong sample rate. All of these problems are eliminated with centralised synchronisation, and if the central source can itself be locked to house picture sync then an entire studio can run seamlessly in sync, with the pitfalls of getting it wrong no longer an issue.

Aardvaark is an American company specialising in digital problem solvers, from A–D convertors to sample-rate convertors, to digital signal distribution. It also has the AardSync, a master digital audio sync generator designed to perform exactly the task described above in a simple set-and-forget, half-width, 1U-high box. Now in a Mark II version, it offers a highly stable low-jitter clock producing a Digital Audio Reference Signal (DARS) for distribution in various formats from its several rear-panel connectors.

IN ITS SIMPLEST FORM, the AardSync works as a stand-alone reference clock, providing two AES-EBU reference outputs on XLRs and four BNC connectors, two of which always deliver word-clock signals while the others are jumper-selectable between word clock and Avid-Digidesign 256 sync. SPDIF sync is available with an optional adaptor. The AardSync prides itself on the integrity of its clock, claiming 100dB of jitter reduction in the critically audible region where the jitter frequency exceeds 600Hz,

Aardsync II supports all sample rates and pull-up and pull-down rates for US and European operation

as well as 0.000% drift over time. All possible sample rates are delivered, with the Mk II adding pull-up and pull-down rates for both US and European work, as well as a user-settable rate. The growing demand for high sampling rates is supported with an Over Sample option giving 88.2kHz and 96kHz along with the same associated pull-ups and downs.

The internal clock can be locked to incoming video, sensing signal presence and automatically distinguishing between PAL and NTSC, colour and mono, showing the signal type on front-panel LEDs. It sets out to match the video frames long-term while ironing out the short-term discrepancies caused by VTR mechanics. I found it locked to decent video, such as a camera output, even faster than the spec claimed, but my S-VHS VTRs were outside its tolerance and had to be time-base corrected before providing an acceptable lock, indicated by a steady lock LED. No doubt more professional VTRs would produce better results.

The advantages of running all the digital gear in a facility from a reference like this, with or without video sync, are obvious, and for many small setups the outputs on the back will provide everything that is needed, further distribution, perhaps, being necessary for larger studios. The practical benefits of central sync are augmented by the spec of the AardSync's clock, which should outperform most onboard clocks comfortably in those areas which are known to have audible effects.

I suspect the installation of an item like this is something few small facilities have considered, despite their almost certainly having suffered from the very problems it solves. It's not expensive, it's not complicated, and it can only substitute a bit of professionalism for the *ad hoc* arrangements of daisy-chained jitter generators with which most places presently content themselves. ⑤

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Syntrillium COOL EDIT

You work in a business where the prospect of a free t-shirt can attract almost as much attention as the launch of a $500,000 console. The Internet's access to the recording studio has **CHRIS DENNING** on the trail of bargain software

IN A WORLD where the cost of both hardware and software seems to constantly tumble—but to simultaneously grow in both power and performance —one area of computer audio application that manages to hold some semblance of its price is that of high-quality, fully featured, nonlinear, digital-audio recording. Of course, nonlinear recording has benefited from the falling cost of computer hardware, but this has generally involved economies of power and quality—such as in number and quality of convertors.

The issue of software is a separate one—not that the hardware is of much use

The program has just about every editing feature you could reasonably want,

without it—in that the material costs in copying and distributing software bear no relation to those commonly associated with hardware. If you limit the function of the software—to encompass, say, audio editing rather than full-blown capture, editing, processing and reproduction —cheap software becomes a virtual inevitability. And, if you take into account the Internet as a means of cost-effective distribution, and the opportunities offered by the Net for evaluation software, entry-level software, and even vanity software, it will come as no great surprise that there is a high-quality, gratis, audio-editing program currently available via the Internet.

COOL EDIT from the American Syntrillium Software operation is a professional Windows-based sound editor with many exciting features. The shareware version is fully functional in every way, but to encourage users to register, there is a restriction on its usefulness—specifically, only two groups of features can be used in any one session. This means, for example, that you could record, mix and save freely-but, if you then wanted to process to add echo, flanging or distortion, you would have to quit and load the program again. This is not actually prohibitive for casual use, but in any professional environment, it would swiftly become tedious, and eventually be uneconomic. Registration, therefore, (most expediently addressed by credit card) is inexpensive at just $25 for a 'lite' version without several of the features, or $50 for the full program. There is also a $100 'preferred' registration that additionally

provides automatic software updates.

The program has just about every editing feature you could reasonably want, including compression and reverb. It can also generate DTMF tones. One of the most useful options is Stretching, which enables cuts to be lengthened or shortened to fit a given duration without affecting pitch, so that the 'perfect' take, that unfortunately is 32 seconds long, can now easily be made to fit the requisite 30-second slot.

The program supports many sound-file formats in both mono and stereo. At present, these include PCM, Microsoft ADPCM; and IMA/DVI PCM; .WAV; Sound Blaster; .VOC; raw PCM. SMP; ASCII text; AU; Apple AIFF and Amiga IFF. There is a clean Convert Sample Type option within the program, and the required audio format can also be selected when SAVing. Syntrillium claims that practically any file format can by loaded successfully by using the raw PCM file type, and then using the Adjust option to correct the sample rate, bit rate and number of channels. There is also a facility called Byte Swapping, which changes how Cool Edit interprets the actual waveform data by swapping high and low order bytes in 16-bit mode, or the signed-unsigned interpretation of bytes in 8-bit mode. You can try swapping the bytes if an unknown waveform does not sound correct. Effectively, these facilities mean that the user could actually invent a custom-file format that Cool Edit would be

Although Cool Edit is available as shareware, its value as a professional editor should not be underestimated

able to support.

More useful: the Batch Processor option enables any series of commands to be carried out on any given set of files, so that, for example, file-type conversion of hundreds of files could be carried out entirely automatically.

There are fully featured echo, reverb and delay facilities with all parameters such as attack time and fall-off ratio completely adjustable. Equalisation is available on the echo channel, and there are many subtle additional controls such as Tunnel (gives a tubular effect), and Spatial (tricks the brain into creating false stereo from mono by applying a 1µs delay to one or other of the channels). Invert, as the name suggests, inverts the waveform, and Mixing provides a control of how much, if at all, the original sound is combined with the returned delayed signal.

One particularly interesting facility in this area is called Echo Chamber. This function will calculate the echoes that would occur if the source audio (highlighted selection) and microphones (effectively the destination channels for the echoed wave) were in a room of any given size and with walls of any given dampening factors. The number of echoes to calculate is adjustable, up to about 25,000 echoes. The more echoes there are to calculate, the longer it will take the function to complete. Practically any ambience setting can be created using this function. ☞

REVIEW

One creative use for this function, like the Spatial feature, is to convert mono audio to stereo with the correct ambiance. Choosing a virtual left microphone that is 1ft–2ft away from the virtual right microphone will simulate the ears of a listener, and will give the effect of 'being there' when listened to with stereo headphones.

To give more control over the environment, damping factors can be applied to any of the walls, floor, and ceiling. If a wall has a damping factor of 1.0, it is totally reflective (like cement). If a wall has a very low damping factor, like 0.05, it will absorb most of the sound (like carpeting or sound proofing panels). You can also lower the damping factor of some of the walls to simulate the fact that other objects in the room are absorbing some of the audio. The length, width, and height of the room can be entered in feet, while room sizes can be as large as memory will allow.

Compression, expansion and limiting are controlled by the program's Compression routine, which also operates as an effective noise gate to silence the audio between words or sounds.

The Noise Reduction and Filter options make it possible to reduce broad-band noise up to 20dB, and constant noises—like mains hum or background air conditioning noise-over 70dB. You can filter out undesired frequencies, or you can keep certain desired frequencies by using the Passive mode.

A Logarithmic mode boosts or damps frequency components.

There is a Lock option for setting a continuous filter, or you can choose different Initial and Final filters. With this latter setting, filtering will gradually go from the initial state to the final in the manner determined by the Transition settings.

The Precision Factor determines how accurate the filtering is over time when separate initial and final settings are used. A low factor, means the filter settings will change roughly, or in chunks, from the initial to the final settings. With higher factors, the filter's transitions are much smoother. In any case, the higher the precision factor, the longer it will take to filter the selection, but the more pleasing it sounds. The FFT (Fast Fourier Transform) function takes a large group of samples, and filters them all at once. The precision factor determines how many samples from the entire group are actually saved in the final product.

As part of the Filter function, there are various 'Windowing' methods that determine the amount of transition width and ripple cancellation, from smallest width, and the greatest ripples, to the widest width, and the least ripples. The filters with the least ripples are also those that more precisely follow the drawn graph, and have the steepest slopes, even though they are wider, and pass more frequencies in a band-pass operation.

If the Morph option is checked, the transition from the initial filter settings to the final filter settings will 'morph' from one to the other. If this is not checked, the settings simply change linearly over time, which means if you have a spike at 10kHz for the initial filter, and a spike at 1kHz for the final filter, the spike at 10k will gradually decrease, and the spike at 1k will gradually increase over time. If morphing is on, then the spike will flow from 10k down to 1k, passing many of the frequencies in between.

Metering can be monitored from within the Cool Edit window, and although selection of the sound source and audio level must be controlled in Windows 95 by the sound card's own mixer facility, the program provides control over such aspects as the metering range (30dB–90dB) and whether peaks should be dynamic or static.

Cool Edit's Scripting feature automates repetitive tasks, by remembering, for example, everything done to arrive at a particular waveform or transformation, and the Music feature enables any sound to be set to music.

The program's Toolbar is highly customisable and, for ultimate convenience, The Cue List section enables the playing of any selected portions of the waveform in any order, as well as normal looping. It also allows the playing of specific parts of the waveform to be triggered by pressing user-assigned keyboard hot keys.

One particular benefit of the system is that multiple 'instances' of the program can be in operation simultaneously. This means that several Cool Edit windows can be open at once, each one loaded, for example, with different audio files which can be selected and pasted as required.

BUNDLED WITH Cool Edit are a number of *.XFM files, which are actually DLLs (Dynamic Link Libraries) that contain the code necessary to perform predetermined Effects. The format for these Transform and Generate effects modules is freely available from Syntrillium, so it is likely that, in time, further modules (whether from Syntrillium themselves or from third parties) will become available. If you are adventurous (and into Microsoft Visual C/C++) you can even write your own.

Additionally, there are CD player controls actually within the program so that, for example, sound effects can be loaded for editing direct from an audio CD. Included in the command set is a precise 'remember current location' marker, so that the CD can be started from an exact point.

An unusual feature of the program is the Brainwave Synchronisation function which generates various patterns of white noise type sounds. These patterns (according to Cool Edit's creator, David Johnston) provide—among other things—stress and insomnia relief, as well as reducing depression, migraines and headaches. No doubt useful after a heavy techno session.

Curiously, many of the sound patterns heard in this mode, such as helicopter, or 'washing' type noises, are not coming out of either channel but are actually artefacts psychoacoustically created inside the listener's head.

Since they discovered Cool Edit, the Commercial Production team at Manx Radio on the Isle of Man can't stop singing its praises. Producer Mike Reynolds describes it as 'phenomenal'. His colleague, copywriter Stephen Bosman adds 'Mike wasn't really into computers previously. But we installed it at 11am and by 2pm he was recording a session. It's so intuitive—I just sit in front of my PC and simply convert what I hear in my head into finished product'.

Cool Edit can handle files of any size, limited only by hard drive space and comes in two versions, either for Windows 3.x or Windows 95/NT. ⑤

New Technologies

Although much of the new equipment shown at the recent IBC Convention was video-based, there was also much of interest to professional audio people. **DAVE FOISTER** brings a roundup of significant releases from Amsterdam

Pitched at edit-intensive post and multimedia applications

Dolby Surround TDM plug-ins

Previewed at IBC were new TDM plug-ins from Dolby Laboratories allowing Pro Tools users to achieve Dolby Surround encoding and Pro Logic decoding without any additional hardware. Compatible with Pro Tools III, 4.0 and 4.1 for NuBus and PCI Macs, the •Dolby software runs on the Digidesign DSP Farm card entirely within the Pro Tools virtual mixing environment, allowing the digital production of Dolby Surround format material for video, TV, advertisements and multimedia. The system includes the facility for 4-2-4 monitoring, so that discrete 4-channel programme material can be matrix encoded and simultaneously decoded to show the effects of later surround encoding.
Dolby Laboratories, US.
Tel: +1 415 558 0200.
Dolby Laboratories, UK.
Tel: +44 1793 842100.

Harrison Series 12

Harrison featured the Series 12-MPC consoles at IBC, centring on the new ADTA (Analogue-Digital Transition Architecture) console with Release 3 software. ADTA allows the consoles to have analogue, digital or hybrid signal paths under the same control surface, and Release 3 adds SmartStart to the automation, giving seamless transition during interruption or

recovery of the automation mainframe avoiding sudden changes or breaks in the on-air signal. It also adds a motorised automated panning joystick.
Harrison, US. Tel: +1 615 370 9001.

Motionworks R2P2

New from Motionworks via HHB at IBC is a compact remote control for Sony P2 compatible machines, the R2P2. On being connected, the unit identifies the machine from a library of over 140 devices and displays its name in a 20 x 2 LCD window, automatically configuring function keys and menus according to the requirements of the machine. These would include, for instance, 8-track record arming for a Tascam DA-88, and ID writing and erasing functions for a DAT recorder. As more machines come on to the market, the relevant additional software can be downloaded from Motionworks' web site. Full transport controls are provided along with a jog-shuttle wheel, and the R2P2 can be operated from rechargeable batteries or a mains supply, both provided as standard.
HHB, UK. Tel: +44 181 962 5000.
HHB, US. Tel: +1 207 773 2424.

New Spectral software

Spectral's workstation range now has CD burning facilities added in the form of MediaMaker CD. This bolt-on takes

material assembled in one of Spectral's editing packages and controls the CD writing process; it can also be used for mixed-media and ISO9660 CD-ROM preparation. A new editor was also introduced at IBC. StudioTracks XP is intended to provide all the advanced tools for edit-intensive film-video postproduction, sound design, dialogue editing and multimedia authoring, and has a new Segment Editor page supporting a whole new series of DSP functions.
Spectral, US. Tel: +1 206 487 2931.

B&K miniature microphone

Hot on the heels of Brüel & Kjær's series of compact microphones comes the DPA4060 miniature microphone, launched at IBC. This is intended for use in radio-microphone applications such as theatre work and is therefore extremely small, and built to resist sweat and make-up. It uses a new 5.4mm prepolarised condenser cartridge, and can handle SPLs up to 134dB with an on-axis frequency response of 20Hz–20kHz within 2dB. Its two likely mounting positions, the head and the chest of the performer, are catered for by two different protection grids, the head version giving a soft boost at 8kHz–20kHz and the chest a larger boost at 12kHz. Factory-fitted connectors for most popular transmitter systems are available and 48V phantom powering is in the pipeline.
Danish Pro Audio, Denmark.
Tel: +45 48 14 2828 .

Soundscape additions

Soundscape's SSHDR1 workstation has received several additional features, including an 8-channel I-O option in the form of a TDIF port, complemented by an 8-channel analogue I-O rack. 20-bit convertors are used throughout and ADAT interfacing is also included. New signal-processing possibilities are added by the SSAC-1 accelerator card, giving improved mixing, routeing and EQ and enabling the use of third-party DSP plug-ins. One such is Wave Mechanics' Reverb plug-in, joined by Soundscape's own Time Module for ☞

Harrison Series 12 console is now at Release 3

time stretching-compression, pitch shifting and sample rate conversion. Further upgrades include an AVI file player for real-time video playback on screen synchronised to audio, and an EDL processor that, with the addition of an RS422 card, allows autoconforming from CMX, Sony and GFG EDLs. CDR mastering software is now available, offering compatibility with a wide range of audio CD recorders.
Soundscape, UK. Tel: +44 1222 450120.
Soundscape Digital Technology, US.
Tel: +1 805 373 1828.

Yamaha 02R additions
Yamaha has added two support products for the 02R digital console. The CD8-AES is a single-slot interface card carrying eight AES-EBU inputs and outputs on a 25-pin D connector, allowing the desk to handle up to 32 AES-EBU signals in and out. Panning facilities are extended by the SoundWare SP02 joystick controller, which allows any of the 02R's channels to be panned across multiple outputs in various surround formats, including LCRS, 5.1 and 7.1. The 02R's 32-bit DSP technology is now incorporated into a new reverb processor, the REV 500, sharing aspects of the flagship ProR3 processor, but at a remarkably low price.
Yamaha, US. Tel: +1 714 522 9011.
Yamaha-Kemble, UK. Tel: +44 1908 366700.

DNA Mikado
New to DNA's signal-processing range is the Mikado quad microphone preamplifier, designed as a high-quality front-end to a digital system. Four independent mic amps all feature switchable phantom power, phase and pad, a variable high-pass filter and level metering. DNA plans to add further features, mostly available as upgrades, including digital outputs, dual analogue outputs and limiting.
DNA, Netherlands. Tel: +31 527 620060.

Micron UHF diversity systems
Audio Engineering's Micron range of radio microphone systems debuted three additions at IBC. The SDR Diversity Receiver is a very compact UHF unit aimed at ENG and location sound recordists, weighing only 150g in its 120mm high stainless-steel case. It is supplied complete with antennae, and is available with camcorder mountings. The DDH-1 takes two SDRs and puts them in a half-rack 1U case with integral antenna

distribution system, specifically for stereo ENG and EFP applications. While the SQDC doubles this up to four receivers, in a leather shoulder case, for 4-channel EFP and film sound applications.
Audio Engineering, UK. Tel: +44 171 254 5475.

Sonifex Courier
Shown in prototype form at IBC was a new digital audio recorder for broadcast acquisition from Sonifex, known as the Courier. This uses industry standard PCMCIA hard disks to which it records linear signals, .WAV files or MPEG II format for a variety of uses, and it incorporates a built-in telephone, modem and ISDN communication software. A graphical display shows recording levels and time, and a scrub wheel allows editing in the field prior to transfer to the studio.
Sonifex, UK. Tel: +44 1933 650700.

Broadcast Electronics Predator
Broadcast Electronics used IBC to launch its new digital exciter-transmitter, the Predator. This is a modular system including a fully operational digital transmitter with power levels up to 250W. The modularity is intended to allow individual modules to be replaced with new technology when it becomes available, and can be purchased as an analogue unit with easy upgrade to digital when required.
Broadcast Electronics, US. Tel: +1 217 224 9600.

Millennia Mixing Suite
Millennia Media has licensed George Massenburg's GML HRT-9100 mixing bus technology, and used it as the core of the Mixing Suite, designed for maximum sonic integrity. The rackmount modular system offers mono and stereo input cards with four main buses and four aux buses, all fully balanced and with circuits based on minimum topology, high speed, high headroom, DC-coupled parameters. Facilities include swept high-pass filters, precision sine-cosine panning with bypass of pan and fader, true stereo solo and one-knob stereo aux sends. The 12-bay rack gives a maximum configuration of 20 x 4 x 4, and four of these can be used together to give an 80-input system. Master facilities include oscillator and master solo functions, and the main power-supply rack incorporates talkback, with the unusual facility of

The Motionworks R2P2 remote controller opened with a library of 140 machine profiles—with more to follow

complete power-off muting of the talkback power amplifier when not in use, and with special circuitry to allow it to power up within 0.25s when required without thumps or damage.
Millennia Media, US. Tel: +1 916 647 0750.

Sonic Solutions PCM9000 support
Sonic has announced plans to add support for Sony's PCM9000 M-O disc recorder to its SonicStudio premastering workstation. The PCM9000 M-O disc is becoming an acceptable format for mastering plants, and the Sonic enhancements will allow premastered audio to be transferred directly to the 9000's drive. Besides this, audio already recorded on to the Sony system can be transferred to SonicStudio at faster-than-real-time speeds for further editing.
Sonic Solutions, US. Tel: +1 415 893 8000.

Harris A2D2A
Harris Broadcast Division has introduced a combined A–D convertor and D–A convertor offering 20-bit performance both ways simultaneously at a low price. Sample-rate conversion is included to the three standard rates, and external clocks are accepted via AES-EBU.

Apogee Sound's amplifier control software will make its debut at the LA AES

Other features include LED level metering, low impedance, balanced and unbalanced, analogue outputs; and both AES-EBU and IEC format digital outputs.
Harris, US. Tel: +1 317 735 1704.

Apogee amplifier software control

Apogee Sound is to show its new power amplifier control software package at AES. featuring intelligent remote monitoring and control. Building on the facilities in the DA Series digitally controlled amplifiers, the new system allows a virtually limitless number of networked power amplifiers to be individually monitored, grouped and controlled from a personal computer. Any adjustments made either on the computer or at the amplifiers themselves will be displayed on both screen and amplifier front panel, and the amplifiers' built-in monitoring systems for measuring temperature, load impedance, line voltage and other parameters allow all this information to be monitored at the computer. The software allows various levels of password-protected access to a system, and will phone or fax to remote locations with preprogrammed warnings as well as providing information on the usage and performance of all the amplifiers in the network.
Apogee Sound, US. Tel: +1 707 778 8887.

Bel 7310

Designed to overcome the problems of differing delay paths for video and audio signals around a facility, the Bel 7310 from Michael Stevens has 1.3 seconds of audio delay and three modes of automatic delay adjustment: source and reference video syncs, active low TTL pulse, and RS232 interface for external control via a PC. All modes operate silently on programme in real time with a choice of algorithms, enabling different audio and vision transmission paths to be re-aligned without manual adjustment. Eight user-memories are provided for setup storage, and SMPTE delay is also included.
Michael Stevens & Partners, UK. Tel: +44 181 460 7299.

BASF reference cassettes

BASF has designed a completely new range of audio cassettes for professional use, with consistency and mechanical integrity the priorities. The Reference Maxima TPII uses a unique 3-piece shell design, able to withstand temperatures of up to 95°C, and contains a special control element and new azimuth pins to help minimise azimuth errors. The tape formulation itself is new, consisting of a high performance ferro-cobalt combination for high headroom and low distortion.
BASF, US. Tel: +1 617 271 6587.

Waves StereoMaker TDM plug-in

Waves' ever-expanding range of TDM plug-in processing now includes the PS22 StereoMaker, designed not only to enhance the spread of stereo sources, but to create a spacious stereo image from mono sources. Its interface provides controls allowing adjustment of specific sounds to chosen positions in the stereo image, and a graphical display shows position as a function of frequency, suggesting that this is a comb-filtering process as often found in mono-to-stereo enhancement systems. StereoMaker has, however, been designed to avoid the common problems with such techniques, claiming no unpleasant phasiness, low coloration, high tonal accuracy, and good mono compatibility.
Waves, Israel. Tel: +972 3 5107667.

CCS codecs

Two new codecs have been introduced by CCS, manufacturer of the established CDQPrima system. The first is the RoadRunner, a compact portable model with bidirectional mono capabilities and mixable mic and line-level inputs. This is complemented by the Olympian, a stereo unit with three mic inputs and one line. Both are compatible with G.722 and Musicam Layer II and Layer II codecs, and both incorporate G.711 for analogue transmission over standard telephone lines.
CCS, Germany. Tel: +49 811 55 160.

Room at

LCR DynaudioAcoustics M4 monitors and Neve
VR72 console dominate the Pierce control room

It took time and dedication to create what is destined to become one of London's most prestigious
studios, The Pierce Room. **SUE SILLITOE** discusses its conception with owner, Richard Pierce

THE OLD PROP STORAGE
warehouse behind London's Hammersmith
Apollo had been empty for so long that the
roof had caved in, and the interior was
completely derelict. But when Canadian
producer and artist Richard Pierce
discovered the building he knew instantly
that he had found what he was looking
for—the ideal place to build a state-of-the-
art studio to compete with the likes of Air,
Abbey Road and Whitfield Street. Better
still, by being in such close proximity to the
Hammersmith Apollo itself, there was the
potential to link the studio with the venue's
auditorium so that future concerts could be
recorded live using an established control

room rather than a mobile.
'It was a dream to find,' says Pierce with
little prompting. 'At first I was very quiet
about it because I didn't want anyone else
pouncing before I had a chance. I couldn't
get over how much potential it had. Even
though it was in a terrible state, I could
design the studio from scratch within a
huge space with 50-foot ceilings.'
Now, some three years after he first saw
the building, Pierce's dream has been
realised. The Pierce Room—as the studio is
called—is officially open and attracting
bookings from all of the major UK record
labels. Recent clients include The Long Pigs;
Genocide II; U2's label Mother Records;

Bjork; Simply Red; Kula Shaker; and BMG
artist Kent, whose last single gave The
Pierce Room it's first European No.1 hit. In
fact business has been so brisk that Richard
Pierce is barely finding time for his own
projects and is having to squeeze in sessions
at the weekends and during late-night
downtime.

TRANSFORMING a derelict warehouse
into a 48-track digital studio complete with
a Neve VR72 console with Flying Faders,
and a dedicated surround-sound monitoring
system to be based on DynaudioAcoustics
M4 monitors has not come cheap, but in
Pierce's opinion it is money well spent.

the top

mix or fool you into thinking something is good when it isn't.'

To achieve these 'standards', Pierce felt it was imperative that the choice of monitors and design of the room should go hand in hand. He had already come across the DynaudioAcoustics M4 monitors at Air studios and had liked what he'd heard, so he opted for them as the basis of his surround-sound system.

> 'A lot of dance projects have suffered because record companies haven't been prepared to invest money in their recording. But this seems to be changing now because dance artists are selling more albums'

'Monitoring and room acoustics are intrinsically linked,' he says, 'so I decided to use Andy Munro who designed the speakers to also design the room.'

Although it took three years for The Pierce Room to be completed, Munro Associates were involved in all stages of the project, helping Pierce put his ideas into practice, as he confirms: 'Even when the studio was completed, the team spent a month analysing it with computer measuring systems and tweaking the monitors and acoustics until we got the sound absolutely perfect.'

Most commercial studio owners would have reservations about spending so long on the fine tuning of control-room acoustics and the monitor systems, because they would be losing bookings. But as Pierce always intended to use his studio as well as rent it out he was prepared to take the time to get it right.

'Most studio owners are not aware they have a problem with their control-room monitoring because they are not the people who actually use the rooms,' he suggests. 'But from my perspective this studio was built as much for my own productions as it was for other people's and I wanted to ensure that I could get brilliant results from it.'

Richard Pierce is not the type of producer it's easy to pigeonhole—correspondingly his production credits cover a wide variety of musical styles. For this reason he wanted a studio that was equally attractive to all types of artist so that it would have the maximum commercial appeal, as he explains: 'The third-party bookings we are attracting cover all genres of music, including some big dance projects—which is interesting because that is one genre that really benefits from great technology.

'Until recently a lot of dance projects have suffered because record companies haven't been prepared to invest money in their recording. But this seems to be changing now because dance artists are selling more albums, and are thus able to recoup their royalties. This is a positive step for the genre because a really pumping mix is often what it takes to make a record happen. Dance records live and die on their ability to fill the floors and that depends so much on the quality of the mix.'

Few producers—especially those involved with dance music—would argue about the importance of good monitoring. But why surround sound?

'Surround sound is just such a great way to mix because it gives you so much room for experimentation,' Pierce explains. 'There's a massive new technology rush right now to get discreet 5-channel or 6-channel systems up and running that can be reproduced on a CD-type format . Once you have full bandwidth to play with you can really make use of surround sound without affecting your stereo mix.

'Aside from my own interest, surround sound is where the future lies. In 10 or 15 years I think everyone will be mixing in surround

Below: Richard Pierce at the VR72

'Basically, I built this studio because I was tired of dealing with rooms that I couldn't get good mixes from,' he explains. 'As a producer and artist I have worked in many UK studios, but even those that count themselves among the best often had such dodgy monitoring that I was constantly having to tweak my mixes in order to get them sounding good anywhere other than in the control room.

'Eventually I decided that the only way to deal with the problem was to build my own studio. From the outset I wanted a facility that would set new standards in monitoring and room acoustics. Accuracy of monitoring was my number one priority because you can't tackle the bottom end of a mix using Yamaha NS10s. You've got to have the big picture, and for that you need really good monitors that give you the entire sound and frequency spectrum, and don't flatter the

recorded for all media.

'We have installed sound tie-line and we will look at a permanent TV and video link at a later stage. It depends on client demand,' he explains. 'If there is sufficient demand, we will certainly do it.'

As the Pierce Room is both a commercial and a private studio, Pierce had to take into account the demands of the market as well as his own personal preferences—particularly where the equipment was concerned. Fortunately, the choice of the Neve desk was both personal and commercial—he loves it, and so do his clients.

'I had an SSL desk for years,' he recalls, 'and was reasonably happy with it because in the right hands you can get great mixes. But then I tried a couple of mixes on a Neve VR and I was blown away with the sound quality, and how easy it was to use. It's so simple that you don't even have to think about it—in fact you would almost have to be an idiot to make it sound bad.

'The way we have wired the Neve saves a lot of setup time. As I tend to shift my own work around to fit in with commercial projects, I don't have much time to set up the mixes on the desk. We have designed the wiring so that we can transfer my preproduction mixes on the Mackie to the Neve with a couple of EDAC patches. It's great because I can use the main studio for drum or brass sessions, then move back to the Mackie room so I'm not tying up expensive hardware for too long.

'Top studios are expected to have either SSLs or Neves, but in the end I decided to go with what I preferred. I was fortunate that there are not a million Neve VR consoles on the market, and that everyone seems to want to use one, so from that point of view it turned out to be a commercial decision as well.'

Along with the console, The Pierce Room is also equipped with two Studer A827 24-track analogue tape machines, with optional 32-track and 48-track digital machines, and a host of outboard equipment and microphones. The internal design draws its influence from the 1920s, and includes a number of unusual art deco details. Pierce has also insisted on a lot of clean air which is pumped into the entire studio complex through an advanced air conditioning system.

'Most air conditioning systems simply recycle stale air which is a major problem ☞

The Neve VR—'you would almost have to be an idiot to make it sound bad'

☜and I wanted us to be the pioneers in this field. As we were building from scratch I was in a position to do it.

'Control-room acoustics are so critical that we had to design the room for surround sound at this stage as installing it later would have meant completely redesigning the control room.'

In his quest, Richard Pierce has gone one step further than most studios by ensuring that the large recording area, and the three smaller recording rooms, have the same full-bandwidth acoustic treatment as the control room.

'Everyone acoustically treats the control room, but what about the recording area?', he questions. 'If you have a room that is enhancing or cutting one or more frequencies then all of your recordings are going to reflect this nonlinearity. So we decided to give the recording areas a similar, but more reflective treatment than we gave the control room.'

With the main recording area, Pierce decided not to go for a particularly live sound because then he would be faced with all the inherent problems of trying to dampen it when he didn't want it live: 'Instead we opted for a different route and went for a full-bandwidth neutral sound with no standing waves and no peaks at certain frequencies. To liven it up we installed a large, tiled reverb chamber which opens directly onto the main recording area and is shielded by 70dB sound-proofed doors. By opening and closing these doors we can increase or decrease the reverb time of the main recording room and create a lot of different effects.'

CLIENTS using The Pierce Room also have the option of recording in the Apollo auditorium which is tie-lined to the studio. At present no live concerts have been recorded because the Apollo is booked out for months with the Riverdance show, but Pierce is keen to capitalise on the live recording market because it will give the studio yet another source of income. This additional income stream was one of the reasons why he felt the warehouse would be ideal for his studio.

'I always wanted to link into a venue so that I could do live recordings, but I felt that it had to be the right venue,' he explains. 'The Apollo has a great-sounding auditorium —one of the best in the country in my opinion —and the Apollo's management were very happy to co-operate with my plans to link the studio with the auditorium.

'Once Riverdance finishes, and bands start using the Apollo again, I think that side of our business will pick up because it is so much better to record live music using a state-of-the-art studio rather than having to rely on a mobile.'

With the live recording market in mind, Pierce has recently installed a second studio that is equipped with a Mackie desk and a Pro Tools system.

'The second studio is designed to be a dual-purpose room—for writing and recording. The Mackie mic amps are surprisingly good, and it is an ideal desk for live recording projects where the band doesn't have a huge budget.'

For higher end projects where clients want the ultimate sound quality, they will be able to link up with the Neve room. The Pierce Room can link to remote TV and video units so that concerts at the Apollo can be

The glow of the reverb chamber and the 'bathroom' styling characterise the live area

Outboard rack slots into
the producer's console

for oxygen and pollutant levels,' he explains. 'This is because an acoustically sealed room will also be air-tight. As a result oxygen levels will decrease and carbon monoxide (from smoking) will increase so everyone starts to feel very tired. We pipe fresh air directly from the intakes on the roof, which get the Westerly flows off Richmond Park, through a series of filters and expel the smoky air. You wouldn't believe the difference this makes to our client's energy levels.'

PIERCE'S PHILOSOPHY requires that a top studio should have excellent acoustics and monitoring, and leads on to his other main bone of contention—mastering.

'Mastering rooms cost very little to build as they usually have very little acoustic treatment,' he asserts. 'There's no mastering facility in the UK that can handle surround mastering, and that is a concern. How can a mastering engineer hope to deal with a surround-sound mix when he can't even hear it properly?'

Pierce adds that once upon a time an experienced mastering engineer was vital for the effective manual transfer of masters to the vinyl format. But computer controlled lathes (for the vinyl club tracks) and compact discs have now made the art of cutting a record mostly a question of tone and compression—both of which Pierce feels should be determined at the mixing stage where the engineer has individual control of each track.

'We have incorporated a mastering facility into our mixing room because I was tired of mastering engineers attempting to alter excellent mixes simply to justify their substantial charges,' he adds. 'It seems that our views are shared by at least one major label which sent an album to one of the most famous mastering engineers in America, only to have it returned with a painful high frequency boost... Could it be overcompensation for the high frequency hearing loss that has resulted from 30 plus years of sonic abuse? Unfortunately, the nought to 30-something year-olds who we all try to flog music to suffer from no such hearing loss—yet.

'The point is that the major artist with experienced producers and engineers working in an acoustically accurate control room can certainly avoid the risk, and the cost, of having their mix mutilated by a mastering engineer. All of the equipment needed to master and edit straight off the mix is available in most quality mixing facilities.

'But what about the home-producer-mixed dance records recorded with engineers of limited experience who are working in acoustically challenged rooms with small

monitors that give no clue as to the level of bass? It's ironic that while these are the very records that most benefit from a "good cut", the majority of mastering rooms are ill-equipped to handle them. This is because these mastering rooms also lack effective acoustic design and full bandwidth monitoring. Even the most advanced mastering house will spend only a fraction of what a top mixing facility spends on acoustics and equipment.

'In fact, for the price of a "prestigious" cut you can hire a quality mixing facility and engineer, master straight off the mix and have a far greater chance of getting a great-sounding track than attempting to fix it in the cut. Or not as it may be.'

Without doubt, Richard Pierce has gone to great lengths to ensure that The Pierce Room lives up to his, and his clients', expectations.

'I want all my records to sound great, so I had to go for acoustic perfection,' he agrees. 'Although I'm not British, I intend to stay in the UK for a long time because

> 'We have incorporated a mastering facility into our mixing room because I was tired of mastering engineers attempting to alter excellent mixes simply to justify their substantial charges'

the UK is the one of the few places in the world where creatively one can experiment. Here, if you record a guitar through a Leslie cabinet people are more likely to say "that's interesting" rather than "you can't do that".

'On the other side of the Atlantic music is far more formulaic, and if you deviate from that formula you are not considered to be a professional. It seems that most of the major innovation in music and recording is coming from the UK, and that is what is special about the UK—that and the fact that you don't need enormous budgets to create a Top 10 hit.

'For an independent studio owner, producer and artist like myself this is an important point because I don't always have massive budgets to play with. I prefer to be innovative and have talented people and clever ideas work with. As far as music is concerned, there's no better place in the world to be than the UK; and that's why I'm here.

'Let's face it—it's certainly not for the weather!' ◉

Hair attitude and an SM57

Alice In Chains, White Zombie, PIL and Anthrax are just some of the fast and furious acts to have benefited from the engineering and production talents of Bryan Carlstrom. **DAN DALEY** investigates the science behind the noise that is 'metal'

METAL, the musical rubric that grew from plain old 'heavy' and has come to encompass other modifiers, such as 'buzz' and 'speed,' has been remarkably resilient over nearly 30 years on earth. Common wisdom has its genesis pegged to the release of the first Led Zeppelin album in 1969; from there the genre has wended its way through, and in some cases infiltrated into other forms of rock, many of which have come and gone.

But metal, the apocalyptic cockroach of modern music, moves ever onward, inspiring still new generations of 15-year-old boys with the notion that, while you don't have

Half of the Producer's Secret: between mics and outboard, great records are shaped

a car, a job, any money, your own flat or a presentable girlfriend, you can still be a badass dude.

Bryan Carlstrom fell into metal, so to speak. Eleven years ago, at the age of 24, he moved from the remoteness (cultural and geographical) of Sioux Falls, South Dakota, in the heart of the heartland that is the American West, and went, like thousands before him, to join a rock band in Los Angeles. He had approximately the same level of success that most of them had during that fallow period in SoCal rock —after the days of the Eagles but before Guns N' Roses reared their ugly but successful heads, in the days when quasi-metal operations like Winger roamed the world, consuming far more hairspray than was commensurate with the level of edginess they were hawking. Which is to say, 'Nice tape. We'll get back to you, babe'.

However, Carlstrom was far from bored. Instead of reading the stringslinger fanzines and learning more than one could possibly want to ever know about Kip Winger's tonsorials, he became a voracious reader of technical magazines such as *Studio Sound* in order to learn about what was going on on the other side of the glass. He chucked the guitar momentarily and enrolled in a 10-week recording class at the Los Angeles Recording Workshop. Halfway through the course, the school's owner and instructor recommended that he apply for an apprenticeship at Track Record Studios, a small 24-track studio on Melrose Boulevard in Hollywood. He got the gig, and two years later followed the studio to larger quarters in the Valley. Over the course of the next two-and-a-half years there, he got more lucky breaks than most aspiring engineers get, working with people like Keith Forsey on Billy Idol's *Charmed Life* album. His luck held when Mike Frondelli, head of Capitol Studios, used him on a recording and then hired him at Capitol as an assistant.
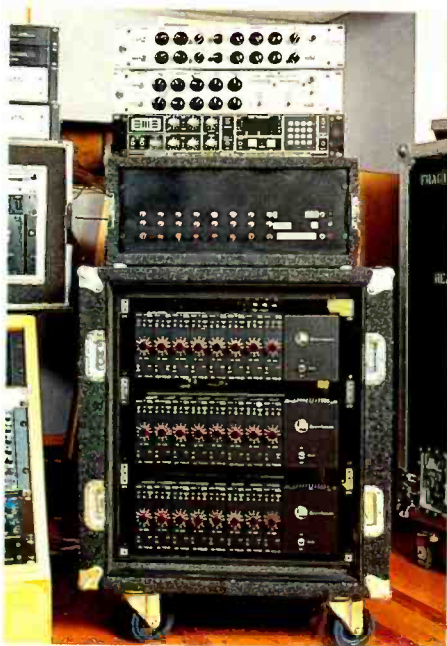
It was a long but fecund apprenticeship. Carlstrom did every type of session imaginable at Capitol, from small rock bands to 65-piece orchestras. He engineered award-nominated jazz records, including those for Eternal Wind and Yusef Lateef. Then came an association with producer Dave Jerden there, whose own oeuvre was considerably eclectic, with records for the Rolling Stones, Talking Heads and Jane's Addiction. In the six years since that relationship began, Carlstrom has made 25 records with Jerden. Interestingly though, he joined forces with him just as Jerden's career was heading decidedly towards the metal's edge. 'I don't know why it happened, but we were just getting call after call from metal acts,' recalls Carlstrom.

IN THOSE SIX YEARS, Carlstom engineered a literal metallic role call for Jerden-produced anvil acts, including Alice in Chains; Anthrax; Social Distortion; Armored Saint; Last Crack, Rob Halford; Sacred Reich; Public Image; Spinal Tap; Edna Swamp; Love Spit Love; Stavesacre; Orange 9mm; Rust; Biohazard; and The Offspring. Carlstrom is the first to admit that they tend to blur together a bit in his memory. But he distinctly remembers being contacted by the members of White Zombie in New York to become the engineer and producer of their destined-to-be-platinum record *I Am Hell*. It's been quite a ride.

There are those who cast metal as a 1-dimensional music and would question its ability to offer much in the way of variety. Contrarily, Carlstrom claims consistency of recording techniques and variety in the music itself as the key to its appeal.

'I had to develop new techniques for

'I don't put any effects on rhythm guitars, ever, even in a mix. I don't generally even use the ambient room sound, except occasionally on a solo sound. The whole thing is microphone placement'

recording,' Carlstom says, in contrasting metal with other musical genres. 'The thing is, those techniques do tend to be the same from record to record. What really makes one record stand apart from another is the songs, the vocals and the playing—because we tend to use the same amplifiers, same microphones and the same miking techniques with each act. White Zombie and Alice In

Chains were both similarly recorded, but they're very different records.

'Almost everything is recorded using close-mic techniques,' he elaborates. 'I put a Shure 57 right in on the speaker, and then run it back through all tube equipment—I have 28 Summit tube mic preamps—and I don't go through the console. For processing, I'll run it through a tube compressor, usually a Summit again, and then into the tape machine, which is running at 15ips with SR. I use analogue for recording drums and bass, and then we go to a Sony 3348 digital deck for guitars and vocals. I don't really hear a difference between analogue and digital for those instruments, and digital is better at giving you back what you put in. I only use the console for monitoring. It gives you the most direct signal-path run.

'I almost always use one Shure 57 on the guitar amp speaker cabinet, placed really close in. I place it dead centre, straight on. I've experimented with off-axis placement, but straight on really has worked the best. I'll listen to each of the speakers in a cabinet—in most of them there's usually four per cabinet. But most of the time it's the lower set of speakers that give you the best response, certainly in a slant cabinet. It might be that they're lower and closer to the floor, and they give you more bass response, but those are the ones that almost always sound best.

'As for the amplifiers themselves, the bands

use our amps, and we have a pretty wide selection of the ones that work best for metal. The main three are Matchless, Marshall and Bogner. The Bogners are the secret weapon of metal. We have some classic Vox AC30s, but those three are the ones everyone wants.'

Despite Carlstrom's claim that each project has its own sound, the consistency of equipment and technique beg further explanation.

'It's the songs, the playing and the vocals,' Carlstrom insists. 'It really is as simple as that.

'There's also the opportunity to customise the sound somewhat later, since almost all the guitars I record are overdubs. During basic tracking, the bands all play live while we get the bass and drums, which, as I said earlier, are recorded to analogue tape. But when I set up for basics, I setup for ☞

Just one product of the Dave Jerden & Bryan Carlstrom partnership: Love Spit Love

potentially having every track as a keeper; there are times when you use a little bit of the basic track guitars and mix them in a little with the individual guitars. But other-wise all the guitars are overdubs.

'I don't put any effects on rhythm guitars, ever, even in a mix. I don't generally even use the ambient room sound, except occasionally on a solo sound. The whole thing is microphone placement. The key to creating a sound is layering and panning, and the way the player plays. Everything is put hard left or hard right. The Alice In Chains record was really unique in that; it was a whole new approach in layering guitar tracks. On that record there are six individual tracks of guitars—three hard left and three hard right—all individually recorded using three different amplifiers and three different guitars. They all have different frequency ranges that they accent and highlight, so the combination of guitars and

The acceptable face of metal: Bryan Carlstrom at the helm

amps and panning makes the sound on that record. [Dirt.]

'On Armored Saint's *Symbol of Salvation* there is one stereo track, each side recorded individually, using vintage Marshall amps. What you're going for is to make the guitar amp sound as though it was in the living room of the kid who's listening to it. That's the kind of presence you're after, and the only way to get that is to mic the amp very closely and not put effects or a lot of processing in between the speaker in the studio and the speaker at home.'

Another important element in defining a recording is miking the drums.

'I use an AKG D112 on the kick inside the drum about halfway between the front

IT'S A TUBE THING

'MY KNOWLEDGE OF ELECTRONIC GEAR started at a young age. My father was an electronics engineer and he brought all sorts of gizmos home that I would take apart to see what was in them. At the age of eight I got my first reel-to-reel tape recorder—I was fascinated by the fact that I could record the sound of my own voice and play it back. I loved electronics and started building things from tube testers to shortwave radios. I was notorious for tearing apart everything around the house from our television to the old tube radios. Most of the time these things never worked again unless my father could figure out what I had done.

'By the age of 12 I was sure that I wanted to be in electronics; that was when I met my best friend Mickey Monger. He did something that would change my life forever—he began playing me the Beatles, Led Zeppelin II, Jimi Hendrix and Cream. Starting puberty and hearing this stuff for the first time created emotions in me I did not know I had. My passion became divided: in love with two things at once, music on one hand and electronics on the other. To me these seemed like oil and vinegar I never thought they could mix.

'By the end of high school I learned to play guitar. I went to college majoring in sciences following my strong love for electronics, but one day after studying the sex habits of rats for two weeks—which I found to be beyond boring—I changed my major to music. After college I played in a few bands and then decided to move to Hollywood with the band that I was put in but the day before we were to leave the rest of the band backed out and I went alone.

'After being in Hollywood about a month, I began looking at recording magazines and going to the Los Angeles Recording Workshop. After five weeks I got my first job assisting in a studio. I loved what I did and it seemed so natural, a marriage between music and electronics. After being there four years I got a break and began working on a Billy Idol record. It was then that I had opportunity to hear every type of gear imaginable API, Trident, GML, Neve that is when I heard my first piece of Summit gear, it really stood out from the crowd. From its sound, down to its design, I was very impressed.

'A year later producer Dave Jerden asked if I would be interested in engineering for him. I had always hoped to work with him because I was a big fan of the records he had done. We started by opening a new room on Sunset Boulevard, the Eldorado Recording Studio. We stocked the place with Summit gear—two TLA100 compressors, four Summit preamps, two Summit shelving EQs and two Summit parametric EQs. At the time I was only able to do overdubs through the Summit gear, it sounded amazing and so real, as if I could reach out and touch the sounds. And I noticed that Dave Jerden and the artists I was working with commented on the sounds as the best they had heard.

'I found myself recording straight through Summit gear and straight into the back of the tape machine, never going through the counsel unless I had to. The Summit gear made things so easy and the sounds they get sounded like those records that I loved and had listened to so many years before. In the past three years we have added three DCL 200 stereo compressors to our arsenal, which I use on the snare top, snare bottom, hi-hat, toms, room, bass, guitar and vocals.

'I also recently purchased a rack of 24 Summit M210 tube mic preamps so that every microphone and every instrument from the beginning of a record to its very end is recorded through tubes. After listening to every Mic Preamp on the market, I found the M210 to be the best I have ever heard. Summit Tube Technology is my secret weapon for high performance audio.'

> "The snare is always miked from top and bottom with Shure 57s. The hat gets an AKG 451 and a Shure 57 taped together on one stand"

and back,' Carlstrom reveals. 'The snare is always miked from top and bottom with Shure 57s. The hat gets an AKG 451 and a Shure 57 taped together on one stand so that the capsules are aligned and are the same distance from the cymbal—that gives it a very fat sound, but one with a lot of top end and presence.

'Rack toms get 57s across the bottom, and 421s across the top. I use a Neumann U87 set about 12 to 16 inches over each cymbal, and I put a pad on the mic. Finally, I use an AKG C24 stereo tube mic above the drum kit, setup between five and ten feet, and a pair of AKG C12s for a left-right room image. Oh, and one other important thing—I take a 251 and put it on a low stand and aim it directly at the kick drum from about ten feet away to get the kick from that angle.

'For processing, I compress the snare and the top and bottom rack tom mics, usually using a Summit compressor on them, which I also use on the room mics. I don't compress the kick drum mics.

'For bass, I take the layering approach again. ☞

I use two direct sounds and two amplifiers. One of the directs is clean off the amp, the other passes through a Tech 21 SansAmp to get a very growly DI sound, and both of those sounds are mixed together to one track. Then I'll use some combination of an Ampeg SVT, a Westminster or an old Vox bass amp for the live sound, with mic placement very similar to that of the guitars —I put a 421 on the SVT very close in, or a D112 on the Westminster because it has an 18-inch speaker and needs to be captured in a different way—it picks up the super lows from that. I run the mics in from the studio into four tube mic preamps, sum them to a pair of Summit parametric EQs, then through a pair of LA-2As, or a Summit DCL200 stereo compressor, and then straight into the tape machine.

'You need to define the bass a bit more to make it sit in the track the way you want. The guitars are mostly defined by the guitar player's own technique.'

Vocals are the third, and often the most identifiable, part of the trinity of defining elements. Here also, the recording process meets the most fragile artists' egos, and the approach to recording becomes as important as the choice of equipment used to execute it.

'Vocals can be different on every record,' Carlstrom agrees, 'but there are some common techniques to them, as well. For starters, we know that there are going to be a lot of tracks of vocals, all layered. As many as 12 to 14 tracks per song, many of them doubled. You have leads, harmonies, call-and-response tracks, all doubled. And sometimes I make a trigger track from the kick and snare to fire sampled kick-and-snare tracks during the mix.

'One trick I use is to run vocals through guitar amps. On Alice In Chains I had some of the vocals through a Pignose amp with

light distortion on it. When you mix an amplifier voice in with a straight track of vocals, it really beefs them up. On White Zombie's records, I had lead vocalist Rob [Zombie] sing 'Feed The Gods' through a Shure SM58 plugged into a Pignose, which was miked with a Shure 57 right on the small speaker. There were six tracks of lead vocal on that song, all sung through the Pignose. Sometimes I'll plug the mic right into some foot-pedal, guitar-type, effects boxes, like a reverb or a chorus, or a fuzzbox or a digital delay.

'I'm also processing the vocals during the mix by using band-pass filters. I use the EQ on an SSL console to chop off the very top and very bottom of the vocals, which is where you lose the junk sounds of these kinds of vocal recordings. I cut off the top at between 5kHz and 8kHz, and lop off the bottom below 300Hz. It cleans it up to the point where you can almost understand what he's saying. You know, whether you're listening to Rob through an amplifier or right next to you in the control room, I don't know if anyone can understand him.'

Carlstrom was working with Anthrax when he was contacted by White Zombie with the offer of production duties. After witnessing a few of their live performances in New York he felt comfortable enough to accept the challenge.

'I realised that there was little to change musically speaking,' he recalls. 'As producer I get to determine along with them which songs go on the record, and which takes are keepers, but otherwise it's an engineering approach to production. There's not that much artistic input. It's just like with the guitar sounds—it's the player that determines how it sounds by the way he plays.'

Logically, an engineer and producer who is allowing the songs and the playing to differentiate one song from another could run into limitations set by the band itself. But Carlstrom claims not to have encountered such a problem.

'The rules are knowing miking techniques and keeping the sound as in-your-face as possible,' he asserts. 'When kids put these records on they want to feel like they've just been run over by a lawnmower. A large, noisy lawnmower. I kind of fell into this niche of metal. But when I was a kid all I ever listened to was AC/DC; Jimi Hendrix; The Rolling Stones, The Beatles; and Led Zeppelin; so this is a logical extension of that music. Now it's become like second nature.'

And with 30 years already under its belt, we can expect to be hearing a lot more metal—some of it in studios, some of it on stage; and some of it through the walls. ⓢ

> 'The rules are knowing miking techniques and keeping the sound as in-your-face as possible,' he asserts. 'When kids put these records on they want to feel like they've just been run over by a lawnmower'
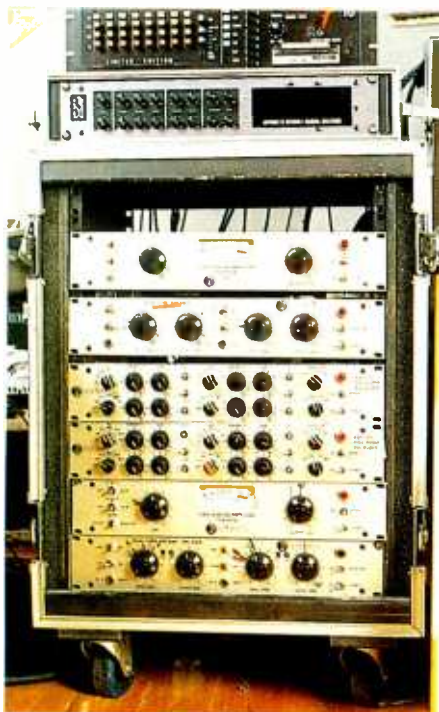
Carlstrom's choice of outboard processing is heavily orientated towards valve technology and Summit equipment

Dimensional travelling

Progressing from cinema-sound systems through up-market 'home theatre' systems to multimedia computers and consumer electronics, 3-D audio is establishing itself as the successor to stereo. **SIMON TRASK** takes in the surroundings...

SURROUND SOUND has been the preserve of multichannel cinema-sound systems since the 1950s. During the past decade it has slowly made headway in the home, courtesy of up-market 'home theatre' systems designed for viewing videos and TV programs encoded with 5-channel Dolby Surround. But it's only with advances in digital-processing power during the past couple of years that a new generation of inexpensive sound processors has arrived to

take 3-D audio into the mass consumer market.

At the same time, the consumer market itself is broadening with the development of new technologies like desktop multimedia and virtual reality, which are in turn opening up new applications for immersive sound. Computer games provide another consumer-level outlet for 3-D audio, while the growing popularity of 3-D graphic virtual worlds on the Internet is providing another medium that can benefit from 3-D sound cues. Technologies like Roland's RSS system, and Digidesign Pro Tools plug-ins from QSound, and Spatializer Audio Labs, provide musicians with tools to record in 3-D. However, much of the development in new 3-D audio technology is going into consumer-level playback technology that can be used to enhance prerecorded stereo signals.

Illustrating the diversity of possible applications for such technology, Kurzweil has just released the first digital piano to incorporate 3-D sound processing. The company's new Mark 12 piano comes fitted with 3-D sound technology licensed from Californian company SRS Labs, a leading player in consumer 3-D audio

development. Another SRS licensee from the musical field is Yamaha, which formed an alliance with SRS in May of this year.

Robert Starr, General Manager of Sales and Marketing at Yamaha's Systems Technology division comments: 'We believe the immersive effect enabled by SRS technology will be a key differentiator in the PC multimedia environment.'

The Division has produced the YSS225 Effects Processor, a multieffects DSP chip enabled with SRS technology, and the first completely digital SRS 3-D sound chip on the market. The company's product plans include integration of SRS into its complete family of PC audio chips this year. The YSS225 chip is also available to third parties, and has been used in the AudioTrix Pro DSP Effects module from Canadian company MediaTrix SRS (Sound Retrieval System) technology, and can be applied in real time to any stereo audio signal. Its effect is to remove the speakers as point sources and to make the overall sound seem more expansive in character.

The technology is based on psychoacoustic principles, making use of the phenomenon of Head Related Transfer Functions. This refers to our ability to position sounds in 360° space based on variations in spectral characteristics or frequency response due to the shape of the head and the outer ear (pinna), which act as frequency-selective baffles. SRS works out the spatial orientation

of sounds in the stereo signal from the sum and difference signals, extracts information that came from the sides and rear (generally, ambience information), and uses HRTF-based corrections to the difference signal to cause the ear to perceive these sounds in their original spatial relationships.

ONE RESULT of this process is that listeners aren't confined to a 'sweet spot' in order to hear the effect–they can move around the room and still perceive it. The SRS technology has an impressive pedigree. It was originally developed in the early 1980s by Arnold Klayman, senior audio scientist at the Hughes Aircraft Company, as a means to improve in-flight entertainment systems for wide-body aircraft. The company poured millions of dollars and numerous man-years into developing, refining and patenting the technology. Then, in 1992, with the end of the Cold War and reductions in defence spending, it decided to sell off some of its non-core aerospace ventures, after producing just one commercial SRS unit itself (the AK 100). In 1993 a startup Californian company purchased all rights and assets of the technology and formed SRS Labs Inc.

The company concentrates on research and development, and earns its revenue from licensing the technology to manufacturers –currently, over 60 companies–in five markets: consumer electronics, computer-based multimedia, electronic gaming, in-car audio and professional audio. Among the companies who have licensed SRS for their multimedia computers are Apple (for their Performa range), Packard Bell, Texas Instruments, Sony and NEC.

In a licensing deal with Motorola's DSP Division at the beginning of this year, SRS was integrated into Motorola's DSP56000 family of chips, already widely used by third parties for audio applications.

Paul Bundschuh, Manager Strategic Marketing and Product ☞

Planning for the DSP Division states: 'SRS compliments Motorola's high quality 24-bit DSP56000 audio algorithm portfolio. As an example, SRS could be added to our popular audio solutions for televisions and audio-video receivers at zero incremental component cost.'

In June of this year, SRS licensed their technology to Chromatic Research, the Californian company behind the new Mpact media processor chip and associated mediaware software modules, currently in the final stages of completion. Designed for use with PC motherboards and add-in cards, Mpact can handle seven multimedia functions: 3-D graphics, 2-D graphics, video (MPEG-1 and MPEG-2), telephony, fax-modem, videophony, and, of course, audio. As well as SRS

NUREALITY

3-D sound, the Mpact processor can handle Dolby Surround AC-3 and waveguide and wavetable synthesis. The chip, to be manufactured by Toshiba and the South Korean consumer electronics company LG Semicon, heralds a new generation of powerful multimedia computers, and is also expected to find its way into new consumer electronics devices. The message of Mpact is that 3-D audio is to be an integral part of future desktop PCs and consumer devices.

At first glance, SRS might seem to be in a different world from that of multispeaker surround-sound systems. However, it can be used in conjunction with a surround system as a preprocessor to the decoder, such as Dolby Pro Logic, and has the effect of removing the speakers as point sources as it does with stereo. NuReality make a budget stand-alone SRS unit, the Vivid 3D Plus, that sells for well under £100 (UK), making quality 3-D sound truly affordable. On the subject of affordability, Mitsubishi has just announced a new SRS chip for use in mini-component stereos; sample chips began shipping on September 30th at a cost of around $7 each, and the company plans to begin mass production of the chips at 50,000 units per month. Total shipments of SRS-equipped products from all licensees passed the five million mark at the end of 1995. Another company that makes use of the HRTF binaural approach to 3-D sound production is Spatializer Audio Labs. Unlike SRS, Spatializer makes products for the pro-audio and consumer markets as well as licensing its technology

to consumer electronics and computer companies. Its pro products are the PT3D TDM Pro Tools plug-in and the Pro Spatializer, the latter a 2U-high rack-mount unit with 64x oversampling, 16-bit A–D; 18-bit D–A convertors, and 56-bit internal DSP resolution, while on the consumer front it makes the stand-alone HTS-2510 unit. The company's 3-D audio processing technology was used in the production of the 1996 Centennial Olympic Games worldwide broadcast from Atlanta, and is incorporated in consumer electronics and computer multimedia products by such companies as Texas Instruments, Compaq, Hewlett-Packard, Samsung, Hitachi and Sharp.

THE GROWING USE of 3-D processing opens up the possibility of an existing 3-D signal being further 3-D-processed. Unlike SRS, Spatializer's system incorporates a feature (called Double Detect and Protect) that guards against excessive processing of the Difference signal. Chip manufacturer Intel has been busy working on its own multimedia chip technology, called MMX, along with its own 3-D audio system, RSX (Real Sound eXperience) and a new audio codec called AC '97.

RSX was developed with virtual reality applications in mind.

'The proliferation of multimedia software, growth of the Internet, and new storage technologies such as DVD allow the PC to continue as a broad-based platform for

home and business,' comments Intel Fellow Fred Pollack. 'Pentium processor-based systems were the springboard for innovative multimedia applications. Pentium processors with MMX technology will accelerate the rate of technology innovation.'

Intel's MMX technology will be integrated into the Intel architecture beginning with a Pentium processor code-named P55C, which the company expects to have in initial production in the fourth quarter of this year, with full production in 1997. MMX technology will also be integrated into succeeding generations of Intel processors. The purpose of AC '97 is to define a high-quality, digital-audio architecture for the PC that will advance the migration to digital audio, yet maintain support for the current analogue audio sources and interconnect buses.

The AC '97 architecture will support a wide range of high-quality audio applications, from a 2-channel mix of digital and analogue audio inside the PC to multichannel digital audio outside of the PC. In order to define the baseline feature set and optional capabilities of AC '97, the developers examined a wide range of applications categories, from arcade-quality 3-D games to consumer video and audio conferencing, DVD-ROM movie playback with AC-3 decoding to 3-D rendered audio for virtual worlds on the Internet.

One 3-D audio company already working on a product that exploits MMX is QSound Labs. The company, whose products include the QX/TDM 3-D audio plug-in for Digidesign Pro Tools III (reviewed *Studio Sound*, August 1996) and the QX/SDII plug-in for Sound Designer II, has also announced QMixer95 v2.0, a new software-only real-time 3-D positional audio mixer which exploits the multimedia capabilities of Intel's MMX technology to place multiple sounds in 3-D space around the listener. QSound expects that, with the benefit of MMX, QM95 will achieve a performance improvement of as much as 350% over current Pentium implemen-

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COMPANIES

Aureal Semiconductor: http://www.aureal.com/
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Crystal River Engineering: http://www.cre.com/cre/
Crystal Semiconductor: http://www.crystal.com/
Dolby Labs—Surround and Pro Logic: http://www.dolby.com/ht/ds&dspl.html
Intel Audio Codec '97: http://www.intel.com/pc-supp/platform/ac97/
Intel MMX: http://www.intel.com/pc-supp/multimed/mmx/
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SRS Labs: http://www.srslabs.com/
Yamaha Systems Technology: http://www.yamahayst.com/index.html

tations, allowing eight channels of audio to be positioned using less than 1% of the available CPU bandwidth.

WITH AFFORDABLE 3-D audio from 2-speaker systems becoming a reality, the pressure will be on manufacturers of

Aureal

multichannel home-theatre systems to lower prices. A development announced by chip manufacturers Crystal Semiconductor on September 4th should help to make this possible. The company has developed the first single-chip, surround-sound codec to support Dolby Digital Surround (AC-3) and Dolby Pro Logic applications. The CS4226, as the chip is called, integrates stereo A–D convertors, six D–A convertors (each with independent volume control), a mono A–D convertor and an SPDIF receiver. By replacing as many as nine discrete ICs and providing a 600% board-space reduction over existing solutions, the chip, which is priced at $32 (US) in quantities of 1,000, heralds significant cost reductions in AC-3-based home-theatre systems.

So what exactly can a single chip costing $32 offer? Can it provide a professional digital audio spec? The CS4226 incorporates stereo, 20-bit, sigma-delta A–D convertors

with volume control, a 20-bit mono, sigma-delta A–D convertor, and a digital-audio receiver compatible with SPDIF. The A–Ds feature 95dB dynamic range, and the D–As perform to 108dB signal-to-noise ratio, and 98dB dynamic range. A 3:1 multiplexer prior to the stereo A–D convertor provides selection for three different stereo audio sources. The stereo A–D convertors also contain a programmable gain of 0dB–9dB in 3dB step resolution. The on-chip SPDIF receiver supports reception of both stereo PCM audio data and compressed 5.1-channel AC-3 and MPEG audio data–which enables it to connect directly to both CD players and DVD players with digital outputs.

Another significant development on the Dolby surround-sound front comes from

Californian company Aureal Semiconductor, whose new A3D technology is expected to be available in Aureal-based chip solutions

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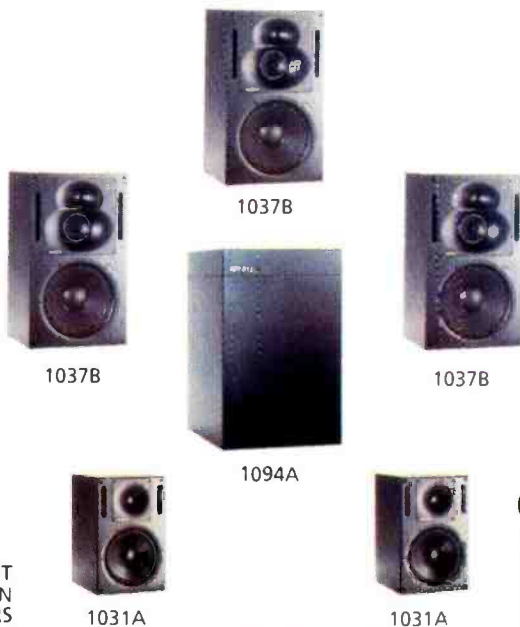
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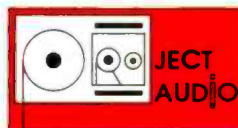
and products from licensees in the fourth quarter of this year, along with several A3D-enabled video games, and other software applications. A3D enables playback of popular surround-sound formats such as Dolby Pro Logic and Dolby Digital (AC-3) using only a single pair of ordinary stereo speakers or headphones, projecting five 'virtual speakers' into 3D space to create a surround experience called A3D Surround.

'We believe that A3D is the boldest and most realistic audio technology available in the market today, and secures Aureal's position as a leader in delivering the most advanced interactive audio technology,' states Aureal's CEO Kip Kokinakis. 'Our partnerships with PC hardware providers, OEMs and game and content developers are designed to bring A3D's audio realism to today's advanced computer and consumer electronics products.'

Dolby Lab's Technology Director Roger Dressler enthuses: 'Aureal 3-D sets the standard by which others will be judged. The technology will enable consumers to realise the benefits of Dolby Pro Logic and Dolby Digital (AC-3) surround soundtracks in multimedia PC's without the use of five speakers.'

In the PC market, A3D interactive is used to create an immersive 3-D listening experience for users of interactive software applications. Aureal, which earlier this year acquired Crystal River Engineering, a developer of positional 3-D audio technology used by NASA among others, plans to support upcoming 3-D audio APIs from Microsoft and Apple, as well as the VRML (Virtual Reality Markup Language) 2.0 standard for graphic virtual worlds on the Internet, to aid widespread adoption of its new technology. During its development, A3D has been used in virtual reality research labs, theme park rides, million-dollar flight and driving simulators—and in recording studios to create 3-D effects for movies such as *Twister* and *The Cable Guy*.

The landscape of 3-D audio, then, is changing fast, being reshaped by a mix of new technologies, processors and applications and the lure of mass market economics. How long will it be before stereo is as dead as the Flat World theory?

UNTANGLING THE WEB 3

RELATED INTEREST

3D Sound for Virtual Reality and Multimedia (Book information):
http://www.sdtb.com/books/BEGAUDS.html

3D Sound Reports:
http://www-engr.sjsu.edu/electeng/faculty/duda/Duda.Reports.html

'Cochlear Mechanics' by Jont B. Allen and Stephen T. Neely:
http://www.boystown.org/cel/cochmech.html

'The Auricle: Pinna/concha' by Roy F. Sullivan Ph.D:
http://www.li.net/~sullivan/pinna.html

CAD '96 (International Conference on Auditory Display):
http://www.santafe.edu/~kramer/icad/

Multidimensional and surround sound systems:
http://www.cs.tut.fi/~ypsilon/80545/SurroundSyst.html

Spatial audio work, Georgia Institute of Technology:
http://www.cc.gatech.edu/gvu/multimedia/spatsound/spatsound.html

Spatial Hearing (thesis): http://members.aol.com/philoneous/thesis.pdf

Real-Time 3-Dimensional Audio Rendering:
http://araz.www.media.mit.edu/people/araz/

Surround Sound FAQ: ftp://ftp.csn.org/Laserdisc/ld03

Three Dimensional Sound for Data Presentation in a Virtual Reality Environment (thesis): http://www-pablo.cs.uiuc.edu/Papers/taveraMS.html

Sursound Mailing List for discussion of surround sound audio and research:
send mail to majordomo darkwing.uoregon.edu with the words 'subscribe sursound <your email address here>' in the body of the message.

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Info

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Craft work

Soundcraft's impending digital initiative has prompted the appointment of a new Managing Director. **ZENON SCHOEPE** talks exclusively to Alison Brett about her conception of the console market and the company's goals

'We have for the last two years been investing in digital technology, and by that I mean a generic digital engine'

THE APPOINTMENT of Alison Brett as the new Managing Director of mixing-console manufacturer Soundcraft is being heralded as a significant one that occurs at a critical time in the Harman company's history. The reason is that Soundcraft is about to take its long expected dip in the pool of digital technology and Brett will oversee this shift in direction.

'What you have here is a company that has probably the broadest range of consoles in its portfolio. We are now building more than 400 mixers a day on three separate manufacturing floors. Each floor is dedicated to high volume, medium volume or low volume mixers so, for example, most of the Spirit products are on the bottom floor while customised desks such as Broadway are on the top. This tremendous diversity of output is one of the unique things about Soundcraft. It means we have broad appeal to a wide range of customers, and it also gives us a balance in our business—so, if some areas of the market are declining, we can focus on others and maintain a buoyant outlook,' Brett explains. 'We have for the last two years been investing in digital technology, and by that I mean a generic digital engine that can be applied to a number of different products. While you put a lot of your resources in to that you have to maintain the bread and butter business so you have the revenue coming in. Unless you keep all the balls in the air and pay attention to the old while developing the new then you go under.

'This is a critical moment because we are now at the point of introducing digital products, and it's time to re-evaluate what the Soundcraft name means,' she continues. 'When you introduce digital products the mechanics are invisible so the important thing is to understand how users use the products. The interface is what matters; you need to come back to basics. We have to ask what we are good at and what we want the world to think when they see the Soundcraft name on something.'

Brett reveals that the company will announce digital products next year. Although reluctant to be specific on their precise nature, she does say that, 'the first two will be aimed at specific areas of the broadcast and postproduction markets, while the third will adopt a more generic design with broad appeal at the lower end of the market'.

These are exciting times for the new MD who joined Soundcraft in 1987 as marketing manager after gaining an MBA from Warwick Business School. Most recently she held the position of Marketing and Business Development Director, and spearheaded the launches of the Delta, Spirit, Europa, the DC2020 consoles, and the digitally-controlled theatre console Broadway among others in that time.

Brett thinks one of her strengths is that she has grown up in the industry; 'I have cut my teeth at Soundcraft, working my way up right through the company from marketing into sales and latterly business development. It has been a great training ground to learn how a dynamic company works and how to succeed in the pro-audio industry'. She regards the industry as fascinating and markedly different from any other.

'It's the cottage-industry background that makes this industry unique because you can't apply clinical business models to pro-audio,' she states. 'Soundcraft has a combination of customers ranging from a man in a suit who is making a purely business decision with layers of purchase decision beneath him. and at the other end you have the rock 'n' roll boy-racer who is buying something because it is the hip thing to buy. Not many industries cope with that diversity.

'There is so much emotion involved with the way people buy things, even with the broadcasters, and I can't think of another industry that's like it.

'IT'S A GREAT COMPANY. We have more than 600 people here and many of them have been with us a long time. That is a huge advantage because it brings with it a real understanding of our roots. We have been in business for more than 23 years and have 🗦

products for lots of different markets. I think some of the things we knew when we were younger, we have to relearn.

'Recording is a good example. We have had a string of great recording products—the 2400 and 6000 series were fantastically popular in their day, and latterly we have had great success with the DC2020. But apart from the 3200, one of my favourite Soundcraft desks, we have not really addressed the very top end of the market. Basically, that area has not been very attractive. The market was shrinking, companies such as SSL and Neve were doing very nicely there, and there wasn't room for another console company. Now things are different; that market is coming back, demand for really good-sounding consoles with masses of inputs and loads of of creative features is definitely there, and, of course, with the shift in postproduction, project studios and the whole area of broadcast recording the recording market is definitely much more vibrant than it was. And that's happened in the last two years.' The upshot is that a serious move by Soundcraft into music recording should be expected in the the near future.

For many the company name is strongly linked to live sound and it is no coincidence that its most significant recent technological drive has been into performance sound with the Broadway desk. This digitally controlled analogue board has been regarded as a sensible halfway house towards a fully digital console, but some have been surprised by the lack of publicity since its launch at last year's US AES Convention.

Brett believes that there may not be much visible in the marketplace, but that there is a lot going on. 'For example, Broadway is now in production; there are four units on the line in the factory, and one unit is currently being used in parallel with another system in a London theatre and is being used as an extended beta site. People in the theatre business have been really passionate about the product, one or two particularly have

> '**With the shift in postproduction, project studios, and the whole area of broadcast recording, the recording market is definitely much more vibrant than it was**'

been working closely with us to get the feature set and functionality exactly right. We could not have done it without them.

'Of course, it is frustrating not to have the system out there already but when we launched it in New York we wanted people to look at it, and tell us if they wanted any part of it improved. We made a commitment then to listen to what people wanted, and incorporate their requests so that the finished product is exactly right. The theatre market has specific needs and we have spent a lot of time making sure Broadway responds in the way people want, and gives them precisely the right amount of control when and where they need it.

Brett is clearly excited by the Broadway system claiming that, 'It is the most important product for Soundcraft for a long time, not only because it re-establishes the Soundcraft name at the very top of the market but because the Broadway architecture offers lots of possibilities for use in other areas. Broadway's technology is tied into the company's digital development for example.

'One of the main things that makes the Broadway unique is the Harman Communication Architecture. We will definitely be using this as an internal communication mechanism inside smaller digital desks, and for some of the larger digital products because it will provide the medium for sending messages to control external boxes.'

The intention is evidently that it will be a new and dynamic Soundcraft that faces the next millennium. Brett gives some clues about what she believes the brand name will mean to people by then. That depends on whether you think that mixing in the year 2000 will be the same as is now,' she observes. 'I don't think it will. I think it will still be about combining, enhancing and adding creativity to audio signals, but the physical thing you'll do it with may be quite different. Soundcraft will still be synonymous with high quality, "available" rather than affordable or cost-effective, mixing. What we're about is making high technology available to an ever wider group of customers. Price comes into that, of course, but price on its own is meaningless unless you add the other things that matter: pedigree, clever design, and great sound.' **S**

Station to Station

Renowned around the world for its innovative contribution to TV themes and incidental music, the BBC's Radiophonic Workshop is now 40 years old. **RICHARD CLEWES** assesses it past and future in a commercial world

THE BBC RADIOPHONICS department –formerly known the world over as the Radiophonic Workshop–is one of the most notorious soundtrack facilities around. Rechristening such a distinctive and high-profile facility is part of an effort to shake off the 'boffins in white coats' image the Workshop suffered since its boffins in white coats origins –although there is no desire to obscure past achievements. Radiophonics has consistently pioneered the use of new instruments and recording techniques, and its composers have been producing innovative and memorable music for the last 40 years. Highlights include the classic science fiction TV series' *Doctor Who* and *The Hitchhiker's Guide To The Galaxy*, and natural history programmes such as *The Living Planet* that have gained worldwide popularity

Located in the BBC's labyrinthine Maida Vale studios in London, Radiophonics is home to composers Elizabeth Parker, Peter Howell and Richard Attree, multimedia expert Anthony Morson and sound restoration specialist John Hunt. I met with them to discuss their approach to soundtrack work and studio design, and to find out how they would like Radiophonics to build on its achievements.

Peter Howell has played a pivotal role in changes at Radiophonics. In 1987, he initiated a major overhaul of the studios, beginning with the redesign of his room at Maida Vale. Completed in December 1988, the new 'circular' control room allowed Howell to switch from a conventional mixing desk to seven Yamaha DMP7s.

'There's an enormous benefit in having smaller units, because you're concentrating on what you're working on,' he explains. 'You've got a keyboard, computer and two speakers all in the same plane. The reason why I developed this studio was I got fed up with having cricks in my neck. I actually do a lot of my balancing with pedals, so it's important to hear what the music sounds like as I'm playing it.'

Initially there were reservations about the new consoles' sonic performance. 'When we bought the DMP7s we were quite aware they had their drawbacks,' Howell recalls. 'Sound quality down the bottom end of the faders

> 'When we bought the DMP7s we were quite aware they had their drawbacks. Sound quality down the bottom end of the faders was not brilliant, but we were looking for a total recall setup'
> – Peter Howell

was not brilliant, and noise levels were slightly higher, but we were looking for a total recall setup.'

After six months' evaluation, Radiophonics bought the DMPs and integrated them with an Akai DP3200 digital routeing system and ☞

Composer Peter Howell in his 'circular' studio

RECORDING

THE RADIOPHONIC WORKSHOP began its life during the mid-1950s, when Daphne Oram and Desmond Briscoe experimented with unusual sound sources in an attempt to provide an alternative to orchestral incidental music—and to produce such classic sound effects such as Major Bloodnok's rumbling stomach for *The Goon Show*. This experiment took place years before multitrack recorders and synthesisers became available, making soundtrack work a hands-on experience: Della Derbyshire's famous arrangement of the *Doctor Who* theme had to be recorded one note at a time using manually-tuned oscillators.

In 1964, Phillips professional tape machines were brought in to replace the Ferrographs which had been used until this point. At the time, The Radiophonic Workshop was also keeping pace with synthesiser technology, and installed a massive EMS Synthi 100 in 1971. The Delaware, as it became known, boasted such wonders as a 256-event digital sequencer, but the monster was best known for the lead line sound on the theme to the prestigious BBC series, *The Ascent of Man*.

Polyphonic instruments such as the Yamaha CS80 and Roland Jupiter 4 followed in the late 1970s, and it was around this time that Peter Howell joined the team. He had no musical training but his soundtrack work for amateur dramatics groups gave him the most important skills: 'The main requirement was that you were 100% imaginative and innovative. Having musical qualifications was regarded as a drawback when I started, because a lot of the people coming out of music colleges frowned on what we were doing.'

In 1979 Howell was given the daunting job of rerecording the *Doctor Who* theme. He ended up spending six weeks on a 3-minute piece of music. Extensive use of backwards tapes and a vocoder gave the new theme some unique sounds, although Peter says he is forever being harangued by fans of the show who claim to know how it was done.

In October 1981 the Radiophonic

Workshop bought a Fairlight CMI, which combined sampling and computer sequencing for the first time. As Peter recalls, 'Richard Attree and I had a great love affair with the music composition language [MCL] on the Fairlight. It was totally, unashamedly 'computer'—it had no resemblance to music whatsoever!' Because of the Fairlight's considerable price, a cheaper sampler was needed for use by the other composers. The PPG Wave 2.2 provided most of the features needed, so in early 1982 one was purchased and immediately put to work by Elizabeth Parker for *The Living Planet*.

In 1983, the Yamaha DX7 brought the MIDI age to the Radiophonic Workshop. Two years later, the Yamaha QX1 sequencer helped the composers get more out of the new keyboards, and in 1986 Apple Macintosh computers were installed, providing a much friendlier means of composition.

Today, the Macs are continually updated, and the latest versions are used to run Opcode's Studio Vision software, and MIDI and hard-disk recording technology has replaced the 24-track analogue tape machines that used to be the facility's workhorses. On the synthesiser front, the composers are excited by the possibilities of Yamaha's virtual acoustics technology, but are waiting to see how this develops before buying more keyboards.

Peter Howell remains unsure whether the progress over the years has always been for the better: 'As time goes on, there's a sort of 'two steps forward, one step back.' For instance, only recently have I seen the words 'ring mod' come back on adverts, whereas that disappeared without trace when the DX stuff came along.'

Elizabeth Parker's set-up includes Yamaha KX88, DMP7s, TX802 and TX816, Roland S770 and Tascam DAT

of having similar equipment in each studio is that it made things much easier if a unit breaks down. This common-sense approach is carried through to synthesisers and samplers as well as mixers. Each composer uses a Yamaha KX88 master keyboard, connected to synthesiser modules and rack-mounted samplers. Among the favourites at Radiophonics are Yamaha's mighty 'eight DX7s in a box', the TX816 and the cleaner-sounding TX802. Richard Attree's enthusiasm for the 816 is readily demonstrated to visitors —the complex FM overtones being well suited to the creation of, and offer a welcome counterpoint to the analogue sounds currently in fashion.

Another favourite sound source is the Roland S770 sampler, an instrument that many people considered far too expensive on its initial release, but has since earned respect for its excellent user-interface and comprehensive filter section. The samples from the Roland are complemented by E-mu Proteus modules, which are useful for creating arrangements quickly.

In Peter Howell's room are the only two 'vintage' synthesisers in evidence at Radiophonics—an Elka Synthex and ARP Odyssey. Both of these keyboards have unique sounds that cannot be reproduced by any of the modern synthesisers, and while there is space to accommodate them, they will continue to be used on new projects. Most of the synthesisers are played through the built-in effects in the Yamaha DMP7s, rather than external units. This helps to ward off the curse of many studios—cables running everywhere—and keeps the noise floor at an acceptable level. However, there are some external effects units in use, including a Lexicon 200 and PCM70, Alesis Quadraverb and Roland Dimension D.

MANY CHANGES at Radiophonics have been prompted by the people working there rather than financial considerations—a relatively unusual situation in today's business environment. The business is also changing: most of Radiophonics' output is for the BBC, but some work is now accepted from independent companies. The increase in customers has coincided with the growth of 'producer choice,' meaning Radiophonics now operates within the commercial soundtrack world.

Apple's HyperCard program for the Macintosh. In Peter Howell's studio, two of the DMP7s are the DMP7D model—one is used as the 'front end' for an Akai DD1000, and the other is used as the main mixer. There is also a custom hardware master fader that can be routed anywhere in the audio chain, and a digital signal routeing box made for Radiophonics by Yamaha. A variety of tape machines are available for mastering purposes, including Tascam DAT and Alesis ADAT recorders—U-matic machines are brought in if needed. Monitoring is handled by Rogers LS5-8 speakers, which give the BBC-specified response necessary for precise mixing.

In the last few years most of the studios have been redesigned in the style of Peter

Howell's room, although Richard Attree uses a Soundcraft 24-channel mixer in his studio. The DMPs have given excellent service, but there are plans to update to Yamaha ProMix 01s. These are preferred to the more recent 02R, simply because very little work is carried out with multitrack tape recorders at Radiophonics. Instead, what is needed is a better sounding version of the DMP7, with more inputs. The increased amount of inputs on the ProMix will reduce the number of mixers in each studio from seven to three.

Howell points out that one of the advantages

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Anthony Morson's equipment racks, including Alesis ADAT and Quadraverb, E-mu Proformance, Proteus, Oberheim Matrix 1000, Roland D550, S770

✍❚ Attree is confident that Radiophonics will hold onto its reputation despite the competition: 'It has made it financially tougher because there are producers around who will buy the cheapest possible solution to their problems. What we're selling can be summed up in the word 'quality', as opposed to the minimum that can be used to fulfil a certain job.'

The composers point to their relationship with producers as one area where they differ from independent musicians. Richard describes the extremes this can take: 'One producer I work with doesn't actually want to hear the music. We've got to the point of trust where I can finish the music, send it to the dub, and he puts it on the film, and that's it. The opposite extreme is where three people walk in the room, you play them something, and they all disagree about how they want it changed.'

Working at Radiophonics gives the composers more of a coordinating role than many independents would be allowed.

'We feel like subcontracted producers,' says Peter Howell. 'It's almost as if there's a bit of the producers' job that they're not able to do, that they subcontract to us. In a

way that's why we feel different to the average person just working from home, because we take on a lot more responsibility for other things.'

Howell 's work on the children's drama serial *Spywatch* illustrates this argument. In addition to incidental music, this required writing songs and finding singers.

'The producer asked me if I could deal with everything to do with the music,' he recalls. 'So I auditioned the singers, arranged the rehearsal pianist and venue. I then rehearsed the singers, organised the recording session, worked out a way of multitracking one singer so she could sing 3-part harmony, and then I collaborated with the graphics guy so we could post-sync the vocals. All of this is extra to composing, writing all the songs and 130 cues of incidental music. But that side of it almost gets taken for granted, because to the producer the most important thing is that it doesn't go wrong.'

The bottom line for Radiophonics is meeting its customers' requirements. To broaden the services on offer, the department has started working with multimedia and sound restoration. As one example, Anthony Morson has a showreel currently in development for Radiophonics on a laptop Macintosh and is eager to explain that multimedia is an extension of the composers' film and

> 'It's almost as if there's a bit of the producers' job that they're not able to do, that they subcontract to us. In a way that's why we feel different to the average person just working from home'–Peter Howell

television work–the disciplines remain essentially the same. The ability to run graphics packages and video on computers, alongside sequencers, also gives the composers more scope to influence the visual side of a project.

John Hunt is also helping Radiophonics to branch out, with the CD mastering and audio restoration room at Maida Vale. A Sonic Solutions system with NoNoise is used for most of the restoration work, plus 👉

John Hunt in the sound restoration room. Equipment, includes the Direct 20–bit A–D convertor, Sony DAT, and Roland Digital EQ

Composer Richard Attree's setup includes a Soundcraft mixer, KX88, TX816, and Lexicon PCM70

CEDAR equipment when necessary, and CD mastering is carried out with the Direct 20-bit A–D convertor. The vast BBC tape library has proved a rich source of material for sonic repairs, including the first concert given in the Royal Festival Hall in the early 1950s.

Although he finds digital equipment time-saving and effective, John Hunt has yet to be convinced it is superior to analogue: 'It is difficult to get digital technology to do justice to old analogue tapes. There's a lot of information on them, and it's a shame to start losing that. I'm not terribly happy with 16-bit digital audio. I'm a lot happier working with 20-bit. I find it much more difficult to be creative with digital equipment than with analogue, because analogue has a wonderful hands-on approach to everything, and out of that comes the creativity.'

With CD now the major format, John believes understanding analogue technology is more important than ever: 'Working with digital, I think it's so important that I've had my grounding in analogue, because you know why you're using digital technology in the way you are. It really worries me when people come

> 'I find it much more difficult to be creative with digital equipment than with analogue, because analogue has a wonderful hands-on approach to everything, and out of that comes the creativity'–John Hunt

here and have probably never seen a reel-to-reel in their lives, and start trying to edit on a digital workstation. It worries me because they're not using their ears, not listening to what's coming back.' The ability to listen and be creative has always been the basis of Radiophonics. The facility does not buy new synthesisers and samplers every year, because it is more rewarding to get the best from the equipment in place. Elizabeth Parker, who has seen Radiophonics progress through the MIDI revolution and beyond, believes hard-disk systems bring tape-style experimentation back to the MIDI studio: 'Studio Vision is the obvious step forward, because it's always been frustrating to not be able to play around with sound in the computer. You can cut it up but you can't transpose it as you could with a sampler.'

Memories of running tape loops down the endless corridors in Maida Vale brings the composers round to discussing Radiophonics' future direction, and Elizabeth Parker summed up the qualities that will no doubt help it continue leading the field—whether in composing, multimedia or sound restoration: 'Being a community, having a group of five highly motivated and talented people is an enormous strength. We want to be at the forefront of technology, and this is the ideal place to develop and try things out.' ⊙

Logo loco

The line between power play and power mad continues to elude those in authority. The future of broadcasting is in the hands of the powerful: as forecast by the likes of HG Wells and William Gibson writes **KEVIN HILTON**

There is a school of literature and cinema that maintains technology is 'a bad thing'. The grandfather of science fiction, HG Wells, warned of the danger of scientists playing God in such works as *The Invisible Man* and *The Island of Dr Moreau.*

From there we have seen a string of books and films that act as little more than campaign slogans for the Luddites: don't trust the little boxes with flashing lights, you don't know what they're up to, they seem to say. Remember HAL the loopy computer in *2001: A Space Odyssey* who gradually loses the plot and puts the whole mission in jeopardy? Television got it in the neck in the now little seen French black comedy *Kamikaze*, where a deranged inventor (natch) managed to turn TV sets into weapons, zapping people through their screens.

It's the fear of technology taking over that gives people the biggest nightmares. Look at the dystopian view of James Cameron's *The Terminator* and *T2: Judgement Day* (with more than a nod to the mad vision of author Harlan Ellison), where an artificial intelligence (AI) has waged war on humans, laying waste to cities and plotting to kill its opponents by sending cyborgs back through time.

All pretty scary stuff, enough to make one look at the microwave oven in a new light, especially late at night when you're on your own in the kitchen... But then there's a more recent oeuvre that presents technology as being the gateway to countless opportunities, but which is often exploited by ruthless or deranged people. The classic example is William Gibson's *Neuromancer* trilogy (completed by *Count Zero* and *Mona Lisa Overdrive*), where an AI also becomes all powerful but it's the corrupt humans who are the real threat, either because of their madness or their greed.

Of course, we're still many years away from true artificial intelligence and virtual reality, although I'd be far more impressed if we could develop artificial cunning (AC) and virtual fantasy. Even so, there is still a resistance to technology, particularly with the continuing march of digital. Recently I was asked to contribute to a

local radio programme that was covering a speech on DAB by BBC Director-General John Birt at the Edinburgh Television Festival; to say that the presenter had an agenda is an understatement.

Now while I have often pointed out that the word 'digital' is being used as a rather meaningless marketing word by broadcasters who sometimes run ahead of themselves, beguiled by the cleverness of the technology, I can see the benefits of what is being planned. While DAB and DTT won't necessarily bring an overwhelming improvement in quality, it will stabilise signals and offer increased capacity, not just for extra channels, but for additional data and complementary services.

The trouble is that both the broadcasters and the equipment manufacturers are charging ahead, not considering the views of the consumer, who is trailing behind with the prospect of forking out for a new TV set and a new radio receiver. Which has understandably caused a backlash. A month or so back I was working on the show guide for a big London-based consumer electronics and computing show and was trying to get hold of photographs to illustrate the features. To say that I met with some resistance is like saying the *Titanic* shipped a bit of water.

First, to be circumspect, was a large multinational of Japanese origins that works in both the professional and consumer markets. When its press relations person asked me to send a fax saying exactly what I wanted the pictures for; how they would be used; and what the text was saying? I, not unnaturally, asked why. He replied that recently a number of publications had carried articles knocking digital technology and illustrated them by using shots of the company's products.

Not long after this I contacted Sky Television to get hold of its new logos for another feature in the show guide. I was met with pretty much the same thing. It appears that Uncle Rupert's channel is now a little sensitive about the criticism it's been getting, and so thinks that by denying publications access to its logo, it can stop all this nastiness.

Rather than run for cover, the

manufacturers and broadcasters who are pushing the technological evolution should stand up and either defend their stance or explain it properly. Much of the resistance that is being encountered often stems from uncertainty as to exactly what benefits all this change is going to bring.

WE'VE LIVED with innovation for a long time, but just when things start to settle down, it all changes again. If the big corporations think that people are just going to roll over and accept everything without a murmur, then they're been reading too much science fiction. Sky in particular should realise that people are going to be wary of Big Bad Murdoch, particularly if they've read of the machinations of Sense-Net in Gibson's cyberpunk trilogy.

And he's been at it again. It was

> The trouble is that both the broadcasters and the equipment manufacturers are charging ahead, not considering the views of the consumer

announced at IBC that DMV, the MPEG2 DVB compatible compression system manufacturer owned by Murdoch's News Corporation, has entered into a strategic alliance with Drake Automation, while also linking up with Standard Communications Corp and Motorola to produce integrated receiver decoders and chip sets respectively. This, of course, strengthens Sky's influence on what can be used, and where, so it's no surprise that writers want to highlight this.

Rather than get paranoid about how a logo is being used, perhaps the company ought to explain what is going on rather than bringing on the spin doctors, most of whom seem incapable of convincing people that technology is not 'a bad thing'.

Hitting the bottle

The world of valve-based equipment is plagued with the jargon of subjectivity and black magic. **ZENON SCHOEPE** leads a guided tour of currently available equipment around and what it has to offer

THE DEVELOPMENT of the valve (tube) as the core electronic element in outboard processing is an interesting one. Once it was all that there was—but it was to become unfashionable in the face of newer, alternative technologies and all but forgotten once the solid-state brought reliability beyond the means of a glass bottle. Curiously, it was the advent of digital technology that kick-started the valve's resurrection, with the biggest argument in favour of the glowing tube being that it could restore some of the warmth that the digital process took away. This was a peculiar justification because when digital arrived, while many remarked on its 'cold' nature, few actually praised it or its ability to 'chill off' analogue.

The mainstream resurgence of valve equipment is a relatively recent phenomenon and part of this has to be attributed to decreases in price, achieved in many instances by the use of hybrid valve/solid-state electronics, and the sheer amount of choice. Due to their scarcity, original, 'classic' units tend to be expensive while new units need to prove they are as good or better in order to gain popularity—and a good many are.

Ultimately, the manufacturers are on to winner here because it is clear that a lot of people like to use valve gear. To some extent it's a feel-good thing and in the same way that certain non-valve bits of outboard inspire confidence by their appearance and demeanour so the military-style presentation of much valve equipment makes you want to believe that it's making a difference. This is why any processor with valves in it that looks dull and ordinary is missing the point.

One of the confusions surrounding valve equipment that no one is really prepared to try to quantify is what the inclusion of valves means sonically to the user. Visually they mean big knobs and a reassuringly hot top panel but how they contribute to the sound is a personal thing. Valves are emotive; some remember the original old stuff but many more base their judgment on experiences with guitar amps where tubes are nearly always a better option than solid-state alternatives. Others associate 'that sound' with valve compressors running at full tilt while still others fall back on the observation that a valve preamp can approximately do for an ordinary mic what mysterious wonders an old valve mic can perform on its own.

This situation is all the more curious when you consider that the majority of gear mentioned here is capable of being extremely clean and almost non-clichéd-tube in character. Dirt and thickness are not the default results with this gear, merely aspects of it.

Price has been the biggest contributing factor to the acceptance of valve gear. Some of the units here are cheap enough to be real alternatives to other analogue boxes—or at least close enough to entice the buyer to go the extra distance for the pleasure of owning vacuum tubes. Valve outboard now takes care of all the demands of stand-alone processing from the predictable preamps and DIs, through to compressors with that peculiar ability to retain spectral balance and retain top and bottom end under extreme settings, and EQs that either mimic true retro passive designs or extend them to 'tube load' bands in a signal in a manner that cannot be approximated easily by other means. There are also combinations of these themes and you can even buy a preamp now with an A–D convertor built in.

We're clearly on to something here and it's likely that the humble tube will remain in favour for very much longer than it did the last time around. Each one is a veritable work of art.

What follows is a roundup of currently available tube-based outboard equipment. Although the descriptions are necessarily brief, the sheer volume of units is testament to the current value in audio circles.

ANTHONY DEMARIA LABS
Mono compressor with stereo linking capability.

Stereo compressor with two independent stereo linkable channels.

Stereo DI box with two independent channels. Switchable outputs (buffered or direct), ground lifts, transformer out, rackmountable or stand-alone.

APHEX
7. Thermionic mic preamp.

V • Single-channel compressor-limiter with Aphex Easyrider, Expressor, Spectral Phase Refractor, and HFX processes and a valve. Stereo linkable plus a side chain with low cut filter. Auto and manual compression modes, switchable slow and fast release times, and hard and soft knee curves.

DEMETER
2. Dual-channel stereo linkable compressor-limiter. Fully variable pots for input gain, attack, threshold, release, and output gain. Vu meters.

Dual-channel mic preamp with large fine gain rotary faders. Each channel has switched gain, peak LED, high-pass filter, phantom power, 20dB pad and phase reverse. Front panel instrument level inputs.

DIGITECH
Stereo mic preamp-DI and 18-bit A–D convertor with 4-band EQ, analogue vu meters and effects loop. Features balanced and unbalanced I-O on XLR or jacks.

DRAWMER
Dual channel compressor-preamp with auxiliary front panel input with EQ. Variable gain, high-pass filter, threshold and switchable attack and release times. Vu meters.

Dual channel EQ with sweepable and high and low-pass filters. Four fully parametric ±18dB switched frequency bands with bandwidth variable from 0.3 to 3 octaves.

Dual-channel preamp with 24-bit analogue to digital convertor. It includes

Drawmer 1961 EQ

sweepable high-pass and low-pass filters, zero overshoot limiter, 3-band fixed frequency EQ, dynamic enhancer, variable tube drive and a mix output section with pan pots and output level. AES-EBU, SPDIF, and TDIF I-Os and word sync. Selectable sampling rate and word length.

FAIRMAN
Single channel mic-line preamp with EQ and compressor. Phantom power and variable gain. Four frequency low-cut and high-cut filters. Six-frequency low boost band, ten-frequency variable bandwidth mid dip band and 10-frequency variable bandwidth high boost band. The compressor has variable threshold, 4-position attack time, 4-position release time, and variable switched output level. Pre an post EQ compressor routeing plus frequency dependent compression via the EQ. Vu meter.

DW FEARN
Single channel mic preamp with low impedance setting for transformerless mics, phase reverse, phantom power 20dB pad and vu meter.

Stereo version of the VT1.

GROOVE TUBES
Single-channel valve-based DI preamp; self-powered version available as DP1P.

• Single-channel mic preamp-DI; alternative version with lighted meter available as MP1M.

Half-rack, single-channel compressor-limiter with switchable ratio (2:1, 4:1, 8:1, 12:1), switchable +4dB/-10dB input ☞

Groove Tubes CL1 compressor-limiter

OUT BOARD

SPL Tube Vitalizer

sensitivity and vu metering. Units may be paired and linked as CL1s with stereo-dual mono operation.

EQ1 Half-rack, single-channel passive equaliser. Five bands with switchable crossover points determining Q and detented cut-boost controls, unbalanced input, balanced output.

INWARD CONNECTIONS

TLM-1 Single-channel Vac Rac limiter module employing an opto-isolator circuit. Stereo linkable to other modules. Line input, vu meter. Fully variable gain and gain reduction pots.

DEQ- Single-channel Vac Rac module continuously variable 3-band ±12dB EQ. Individually bypassable and continuously variable high-pass and low-pass filters. Low band with shelving switch, high-band with shelving switch and bandwidth control, mid band with two frequency ranges and bandwidth control. Phase reverse.

MIC EAM Single-channel mic-line preamp with insert send and return. Vu output meter, phantom power, 20dB pad, phase reverse, 8-position gain trim and level control.

TEQ1 Single-channel 3-band step equaliser Vac Rac module. Offers 12dB cut-boost at 15 frequency points, two band-pass (50Hz and 50kHz) filters, and high-low peak-shelf filtering.

Ti-1 Single-channel Vac Rac instrument interface with control of input and output levels plus bass and treble controls. A 3-way selector taps the output at various stages for altering the amount of tube gain.

LA AUDIO

V8 Eight-channel signal conditioner. Line inputs with two front panel instrument level inputs. Each channel has a drive and level pot and a +6dB overdrive and bypass switch. LED clip and soft clipping indicators.

MANLEY

MIC PREAMP Available in single and dual mono configurations with 40dB and 60dB versions. Phantom power, phase reverse, high impedance input, input attenuator and switched gain.

EQ500 Single-channel differential equaliser with HF and LF switched frequency control and line gain.

MIC PREAMP EQ500 COMB Single-channel combined preamp and EQ500 equaliser with vu meter.

QP A Enhanced 'Pultec' EQ: Single-channel EQ with 5-position switched frequency switch in LF with fully variable cut and boost pots, 11-frequency HF with fully variable bandwidth and boost, and fully variable cut on five switched high frequencies. Rear panel mounted input phase reverse.

M F Single-channel 3-band EQ with switched 5-frequency, 11-frequency and five frequency switched mid bands and fully variable peak, dip and peak respectively.

D RE 1 Available in mono and dual mono versions with ground

lift and console boost output switches plus 5-position EQ switch for tonal shaping of the input signal.

VARIA M
MITER-COM SSORS Available in single channel and dual mono stereo linkable versions. Fully variable input, threshold, attack time and output with 5-position recovery time switch. Vu meters and limit-compress switch.

Gi FRE Y DE-ESS
LIMI ER Single-channel de-esser working over the frequency ranges of 7–16kHz and 5–9kHz with side chain monitoring, variable threshold and re-ess function for adding back top end extension after de-essing. Vu meter.

LEC O AL
M R Dual-channel stereo linkable levelling amplifier with stereo balance control, vu meters, and individual continuously variable gain and reduction pots.

MARTECH MODULAR STUDIO SERIES

M E OD Mic preamp with direct out and dual signal paths for mic and line inputs—serial for external equip-ment patching while maintaining full input and output level control and parallel for mic preamp direct out and line level input with output level control. Vu meter. Transformers.

E IZE O Single-channel 4-band fully-parametric equaliser with sweepable high-pass and low-pass filters, individually bypassed bands and filters.

RETROSPEC

EEZE Single-channel compressor limiter presented as a floor 'stomp box'. It handles instrument and -10dB line levels with balanced XLR output. Variable threshold, compression ratio, EQ contour and output level.

Single-channel transformerless DI with ground lift and variable gain.

RIDGE FARM

GA ER Two-channel DI with 20dB pad, buffered unbalanced output (pre gain control pot) and unbalanced and balanced post gain pot outputs. Ground lift. Free standing box.

SPL

CHARISMA Eight-channel tube processor for insertion between modular multitrack systems and console. Controls include Drive for valve input level, Output for level matching and Bypass for process defeat.

E Twin-channel hybrid design incorporating Vitalizer functions and a tube output section. Vu metering per channel.

D E E Twin-channel valve-based de-esser offering control over S-reduction, Auto Threshold for live applications, Male-Female for gender selection of processing, and Active defeat.

STUDER

9 VALVE Twin-channel and 8-channel combined hybrid valve and solid-state mic preamp and 20-bit A–D ☞

TL Audio Classic EQ

convertor systems. Uses AES-EBU in a choice of formats including 20-bit and 16-bit, 44.1kHz and 48kHz conversion. Controls include Bass Warmth enhancement and Angel Zoom equalisation. LED ladder metering.

SUMMIT

D L20 : Dual channel stereo linkable compressor-limiter with variable attack and release times, threshold, slope and gain. Dual meters.

EQF10 . Single-channel full-range passive EQ with four overlapping bands each of which has seven switchable centre frequencies and adjustable bandwidth. Three-position high-pass and low-pass filters.

** F20 :** Stereo valve EQ with passive EQ section. Switchable high and low EQ frequencies with ±16dB and a separate HF shelf attenuator switchable between 5kHz, 10kHz and 15kHz.

TLA 00 . Hybrid tube/solid-state levelling amp with stereo coupling and side chain access.

PA20 B: Stereo preamp with balanced mic, line and high impedance inputs for continuously variable clean to overdriven characteristics. Phantom power, channel overload and overdrive indicators.

MPC 00 Single-channel mic preamp compressor with fast attack times. Preamp has -15dB and -25dB pads, loading control, clip indicator and stepped attenuator control. Compressor/limiter has variable threshold and slope controls and 3-position switches for input, output and gain reduction.

TL AUDIO

Dual-channel stereo linkable soft knee compressor with variable ratio. Mic, line and instrument level inputs. Vu meters. Fully variable input gain, threshold, attack, release and gain make up. Phantom power.

Dual-channel stereo

Summit MCP100A compressor-limiter

linkable 4-band EQ with mic, line and instrument level inputs. Four switched frequencies per ±12dB band with two peaking (Q 0.5) mids and shelving LF and HF. Phantom power.

Dual-channel mic preamp with instrument level inputs. Four- position switched gain and continuously variable trim pot, 3- position high-pass and low-pass filters, LED metering and level pots. Phantom power and single channel phase reverse.

Dual-channel mic preamp with instrument level inputs. Phantom power, peak LED, fully variable gain, and single channel phase reverse.

Dual-channel stereo linkable 4-band fully-parametric EQ with individually bypassable bands and variable high-pass and low-pass filters. Mic and line inputs, phantom power and instrument level inputs. Fully variable gain, output level and frequency, band-width and ±15dB pots.

Eight channel valve interface with balanced I-Os for inserting in signal chains. Operating levels can be switched for -10dB /+4dB operation.

Four-channel mic preamp each with phantom power, variable input gain, variable output level, peak LED, 90Hz high-pass filter and phase reverse.

Dual-channel 4-band EQ with four switched frequency peaking mids, and two frequency HF and LF shelves. Fully variable gain. Line inputs and front panel instrument inputs.

Dual-channel 2-band fully-parametric EQ which can be linked for single channel 4-band operation. Fully variable gain, bandwidth, cut-boost and frequency pots with divide by 10 and multiply by 10 switches on bands 2 and 3. Line inputs and front panel instrument inputs.

Dual-channel stereo linkable compressor with line and instrument level inputs. Fully variable gain, threshold, ratio and gain make up pots. Two-position attack and release times. LED metering.

Dual-channel overdrive unit with 3-band fixed frequency EQ operating over both channels with individual channel bypass. Variable gain, boost level, low-pass filter and output level. Line and instrument level inputs.

TUBE TECH

Stereo mic preamp with DI amp. Phantom power, 70dB of switched gain, 60dB on DI inputs.

Single-channel equaliser with three passive bands of rotary switches for frequency selection and amp sections to restore gain. 10dB of LF boost on five frequencies, 10dB of mid range attenuation over 11 frequencies and 8dB of HF boost over five frequencies. 🖙

NEWS FROM TUBE-TECH EQ 1A

The new TUBE-TECH EQ 1A is a state of the art full range parametric equalizer. Featuring one channel of low and high cut, low and high shelving and three overlapping bands

LYDKRAFT

Lydkraft Aps • Ved Damhussoen 38
DK 2720 Vanlose • DENMARK

Single-channel passive equaliser with gain restoring stage. Shelving 3-frequency HF and 4-frequency LF,

Manley's extensive range of tube outboard

10-frequency mid range.

Single-channel 5-band EQ with 6-frequency high-cut and low-cut filters adjustable for 6dB/octave or 12dB/octave operation. Six- frequency high and low shelving bands and three 12-frequency peaking bands with variable bandwidth. All bands bypassable. Mastering version also available.

Single-channel compressor controlled by side chain amp with fixed and variable attack release settings. Up to 30dB of gain. Unit can be interconnected for stereo or multiple operation.

Dual-channel stereo linkable compressor with separate and independent limiters in each. Attack and release times can be adjusted manually on pots or can be substituted by six attack-release presets. Variable gain, ratio, compressor threshold and limiter threshold. Bar graph metering. ⑤

Poor protection against radio interference can compromise the best audio installations—and many are now being found to be vulnerable. **BEN DUNCAN** tunes in to the sound of a good ground

The new age of radio defence

IN THE PAST few years, the level of RF 'in the air' has sharply increased. Much of this pollution has to do with the use of mobile phones—even in deepest Tibet, Buddhist monks are now seen carrying them —but there are many other radio services including pirate music broadcasters contributing to the situation. Unfortunately, it has emerged that a large amount of professional as well as domestic audio equipment is wrongly wired, making it unduly (or more) susceptible to RF reception. The damage this can cause to recordings and real-time performance alike shouldn't need spelling out—see sidebar 'DIY Analysis'.

When RF problems arise, there are a host of possible causes that extend well beyond plug wiring into cable shield earthing, balancing techniques and building wiring design. The significance of 'wrong' or non-ideal wiring (alias shielding, routeing, grounding and balancing) procedures is that where a system suffering from them may be 'cured' so that interference levels are tolerably low, the resulting noise will never be eliminated unless the 'right' techniques are employed—and if there is much that is non-ideal, this probably requires a wholesale reworking of the entire system.

The 'wrong' wiring will not stop the music getting through. Rather the opposite in fact, since the outcome of the wrong XLR Pin 1 connections is that all works normally until radio K-RAP breaks-in over your recording. Different wiring conventions revolving around Pin 1 are also the centre of regular connection and grounding incompatibilities between equipment. Alas, unless you know

—or can identify—the optimum or 'correct' baseline, and the mixing console is wired accordingly, then the *ad hoc* measures which 'solve' hum, buzz and radio interference are likely to be palliative. Add one more item of equipment (with some other 'wrong kind of' grounding), and a system that did work tolerably, may well go into a buzzing relapse.

Last year an entire issue of the AES Journal was given over to a logical approach to kinds of RF protection and related topics [1,2,3,4,5]. An emerging keynote cause of RF and related noise problems, is the industry's pandemic, subtly wrong connection of XLRs' Pin 1. For balanced (and quasi-balanced) connections, Pin 1 in all professional audio equipment should be solely used to connect the cable shield to some suitable ground point. The first and foremost problem is that the chosen ground is often the wrong one: but the consequences remain invisible until some strong enough RF turns up (see **Fig.1**).

To avoid creating loops at audio frequencies, a hard connection should be made at one end only of balanced connecting cables, not both. Usually—with two main exceptions—the hard connection should preferably occur at the preceding source equipment. The main exceptions, for foolproof operational purposes, are mic amps (or whatever accepts a mic), and power amps. In both cases, the cable shield must be tied to these devices, as well as the mic but not the source feeding a power amplifier. The reasons—to do with the risk to speakers being fed what might appear on long, shieldless wires—should be obvious enough. The second problem, arising from the need to connect shield at

one end only (may be abbreviated '1eo'), is that whereas Pin 1 is a terminal unto itself on XLRs, with jack plugs it is the body shell. One consequence is that only XLR connections offer positive control over where and which end Pin 1 is connected and while enabling standard pin-to-pin leads to be used.

> **It has emerged that a large amount of professional as well as domestic audio equipment is wrongly wired, making it unduly susceptible to RF reception. The damage this can cause to recordings and real-time performance alike shouldn't need spelling out**

Other than ¼-inch jack plugs (phone jacks in US) being strictly unsuited to installation due to their unstable contact surfaces, the jack sockets (as found on lower quality equipment and even some expensive power amps) would require dedicated cables with the shield detached at one end, to avoid forming hum loops at 50Hz or 60Hz, or harmonic frequencies. Further, the detached end must be assured of being placed at one or other end of the signal flow, for optimum results. Meanwhile, the equipment's socket body should be isolated from chassis —and everything else—at inputs, and connected to chassis only, at outputs. This rarely happens. With XLRs by contrast, the '1eo' rule is safely covered, by arranging that the equipment chassis connector at one end (usually the destination) should be left open at DC, while being tied down through a typically 10nF capacitor (see **Fig.2**). This order of value ☞
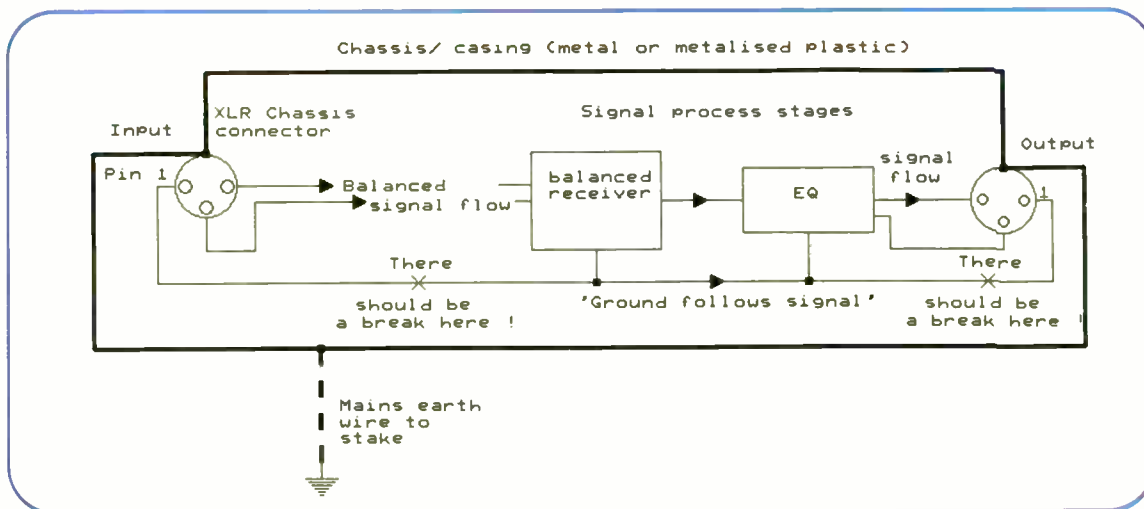
Fig.1: Showing the foremost pin 1 problem—connection of cable shield via pin 1, to 0V (signal zero volts), most particularly at the input end of circuitry or PCB—based on 'ground follows signal' philosophy. This works admirably until the shield is called upon to carry away significant RF current. Then the RF intermodulates audio currents in the 0V path, or at least sets up potential differences along the way.

presents a high impedance at audio and power line frequencies, but increasing looks like a short circuit at high RF, say above 1MHz. Thus the shield is tied down at both ends at RF (where a loop is less likely to occur, or be a problem), but remains 'leo' at audio frequencies. Also, if the shield's hard ground is ever absent (because of damage or because alien equipments' wiring conventions differ), it will at least remain safely 'tied down' at RF, and this will help prevent the worst of potential damage

The extent to which RF shielding ultimately depends on earthing (or 'earth-grounding') to the planet's water table and-or bedrock is an, as yet, unexplored question. If recording studios and concert halls were readily cocooned within perfectly hermetic, highly equipotential (stable, highly conductive, metal) surfaces with no sharp edges, rather like some aeroplane fuselages, then connection to earth is not needed to secure efficient RF protection. But few studios can meet or justify the extra layer of requirements and cost for a Faraday cage, with its ideal absence of sharp edges or curvatures, and with EM wave leakage then needing to be juggled alongside sound containment and isolation, in the already bulging budget for doors, windows, air conditioning, and service passages.

THE ALTERNATIVE is earthing. Most radio signals propagate with some reference to earth, either implicitly (as with fixed-site transmitters), or by the aerial (antenna) being positioned effectively against the earth and using it as a groundplane—as with most hand-held mobile phones and mobile CB radios. It follows that RF is ultimately drained away by being returned to this point. Even if the problem is radiating from miles away, this is achievable through connecting to the planet's crust locally. At the same time, the 'efficient' RF filtering, both of signals and power lines that good design and also EMC procedures may require, put increasing demands on the quality of the connection to the earth (see **Fig.3**). This is analogous to
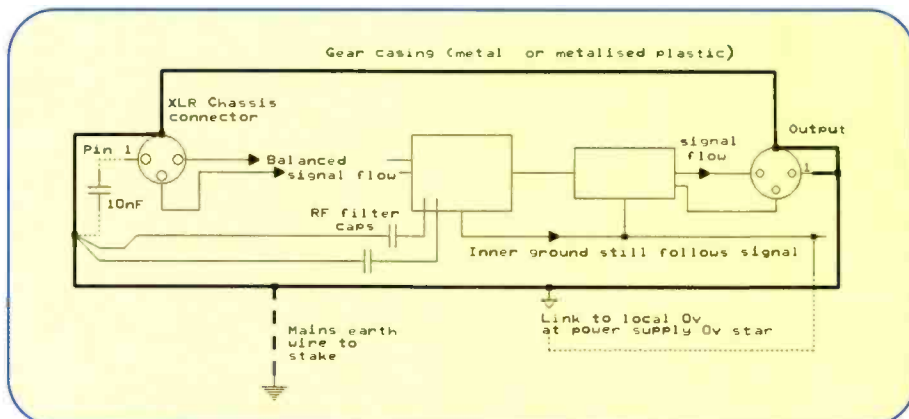
Fig.2 showing the correct scheme, where all output pin 1's terminate directly to the case or chassis that is the natural extension of the cable shield, and also ideally an equipotential surface at RF. The chassis must still refer to 0v, but independently and directly to the power supply's central 0v star point. Without mingling with audio currents there is then no RF ingress by superimposition.

the way that the improved drainage of land and towns makes river flooding more likely, downstream...

In ordinary commercial and domestic buildings, earthing is principally provided (at least in the UK) by either the outer, protective metal sheath around the feeder cable (if from underground), or if from overhead lines, an earth stake adjacent to the supply entry point, and-or that provided by the PME system, used in both the UK and US, where the neutral conductor is tied to earth repeatedly (for example, at every other pole) between the nearest transformer substation and the supply entry point. Second level earthing is provided in many cases—but not always or in perpetuity—by 'equipotential bond' wires. These are the green-yellow earth wires that you should see strapped (for safety's sake) to all metal plumbing in 'wet rooms', under sinks, baths, to immersion heaters, and even (in newly wired or rewired rooms) fitted radiators, and ideally, all other exposed fitted or structural metalwork. If the plumbing is

solidly metal down into the soil (Be sure to check that no sections of copper pipe have been replaced with plastic!) then these connections provide added earthing.

The effect of several adjacent earths in parallel naturally reduces the ground impedance, and can also reduce the worst case value occurring when the water table sinks in a drought.

The upshot of having at least two and possibly a dozen earths in or adjacent to a building is solid earthing—even if some wires

> If recording studios and concert halls were readily cocooned within perfectly hermetic, highly equipotential surfaces with no sharp edges, rather like some aeroplane fuselages, then connection to earth is not needed to secure efficient RF protection

become disconnected. The trouble is that none or few are likely where the audio equipment is. For these, a highly local technical earth is needed. Ideally, reliable contact with substrate (wet soil, sand or clay), or otherwise, major building steels embedded in the substrate, should be achievable within 5m of major audio equipment earthing demands—consoles, tape machines, FX & amp racks and so on. The reason for this is that ordinary earth wire has significant resistance, inductance and skin effect. The skin also quickly turns into a thicker and thicker layer of copper oxide, which is not a good RF conductor, and increases the effective inductance. The outcome is that conventional mains earth wires, however chunky, act as progressively better aerials with higher frequencies, and the effect is exacerbated with ageing. Another 🖙

DIY ANALYSIS

THE STATUS of Pin 1 connections in equipment you own, are about to purchase, or are already using, is easily determined if you can get under the hood to take a close look at the input and output connectors. Using a strong torch on the input XLRs, look for Pin 1. In optimally wired systems, Pin 1 on input sockets (excepting inputs accepting microphone and instrument levels; and power amp inputs) should be wholly unconnected, or at most, connected directly to the chassis or other principal metalwork or shielding material only, via a small (<470nF) capacitor, and possibly a small-valued (<120Ω) series resistor as well.

If the XLRs are PCB mounted, where the tracking goes will need to be studied. The trace from Pin 1 (usually positively verifiable in a dark and fiddly space with a simple 'bleep' continuity tester) should be maximally wide and lead immediately to the XLR's chassis connection tie pin, or other solid chassis connection. The XLR's chassis must be securely bonded to the chassis with screws made of a like metal (brass, if the XLR shells are likewise fabricated), for this to take useful effect. Particular washers which help break through surface crud and make gas-tight contact, and maintain the torque also, make sense. Rivetting (which involves aluminium) is preferred for practical XLR fastening in some road-going quarters but the dubious aluminium-cupro class of electrochemistry will likely cause RF detection problems after a time.

If the Pin 1 connection does not go to chassis, but only disappears into the mass of PCB track, you can be sure it is wired wrongly. On the other hand, it is quite possible with inputs only, for the correct connection to also be associated with a PCB track or wiring from Pin 1 that goes into the circuitry. But this should only connect to any input RF filtering capacitors (Fig.2).

Fig.3: The EMC-EMI industry also has no clothes, since potentially high current is passed through earth (earth-ground) system by capacitors performing RF and power-line filtering in industry-standard filters. The filtered-out hash is then imposed on the audio equipment by superimposition in the usual common earth wire, often of near antennae status. There is also an RF ground loop formed by mains filter capacitors in PME systems. The problem is exacerbated by ineptly drafted EMC legislation which is blind to downstream pollution at higher system levels...

outcome is that satisfactory RF earthing is less and less likely, the higher the equipment room is above ground level.

WHETHER THE EARTH is 'technical' or not, the connections to it must be either short, or have short apparent length. This is another name for cables that exhibit low inductance and skin effect, while also having low resistance. Such cables may be broad tapes or foils, wide PCB traces, wide braids, multiple braids, or Litz types, where multiple insulated strands are plaited to largely cancel the internal eddy currents that cause skin effect. Cable designed by US maker Kimber, now long used by audiophiles for sound-enhancing

CPC: the Circuit Protective Conductor —the green (US) or green-yellow (rest of world) wire associated with building and flexible power cables or cords.
EARTH, EARTHS, EARTHING: May refer to the CPC or to the physical 'stake or mat in the bedrock'. The latter is distinguished as 'Earth Ground' in US parlance.
EARTH WIRE: The CPC if associated with power cabling. Else, if leading to a technical earth, it is known by electricians as supplementary earthing. Or if it connects baths and sinks, it's an Equipotential Bond Wire.
EARTH STAKE: A pointed metal rod, typically 10–30-foot long, driven into the ground far enough to reach into a reliably wet or damp or otherwise conductive substrate layer, that makes a bulk, low resistance contact with the greater mass of rocks that are the earth's crust.

speaker and mains wiring, is a rugged example of this. Unfortunately, none of these forms are legal for use as earthing conductors in buildings unless they are appropriately rugged (for example. Minimum 10mm² for a equipotential bonding conductor for a building in the UK, wired to BS 7671, alias the 'IEE regs'), and are clad or sleeved throughout with the familiar green-yellow insulation.

In ideal conditions, every significant item of audio equipment would have its own short and straight earth connection to its own earth stake, using such conductors. **Fig.3** shows a little discussed problem that is solvable by providing all mains RF filter capacitors with their own, 'dirty' technical earth stake, with a choke-isolated connection onto the existing safety earthing conductor. The possibility of interaction between clean technical earth stakes and the 50-60Hz power system's safety earthing may be avoided—as described elsewhere[6]—by having a 0.1µF high current, >1kV rated RF-grade capacitor in line with each technical earth stake. This and use of earth chokes might require special dispensation from your local electrical safety authority.

In RF-rich environments, it may not be enough to earth the audio system. In live show production, even quite subtle, delicately negotiated changes in the earthing and earth continuity of the lighting system could significantly reduce PA and recording noise levels. It may also be beneficial to check the earthing quality of significant areas of metalwork. Metal claddings, and similar constructional metalwork—even structural steelwork—may not be properly earthed, if they were ever, and can act as unwitting RF reflectors or directors, at the 'wrong kind of' frequencies, until given robust RF earthing treatment.

References and Further Reading
Texts that cover grounding:
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2. Atkinson, Charles & Giddings, Philip, *Grounding Systems & their implementation*, June 1995.
3. Fause, Ken. R, 'Fundamentals of grounding, shielding & interconnection', *JAES*, July 1995.
6. Duncan, Ben, 'Black Box' (technical column), *Hi-Fi News & Record Review*, April, May 1996.

Texts that cover Pin 1 connections:
4. Muncy, Neil. A, 'Noise susceptibility in analog & digital signal processing systems', *JAES*, June 1995.
5. Macatee, Stephen R, 'Considerations in grounding & shielding audio devices', *JAES*, June 1995.

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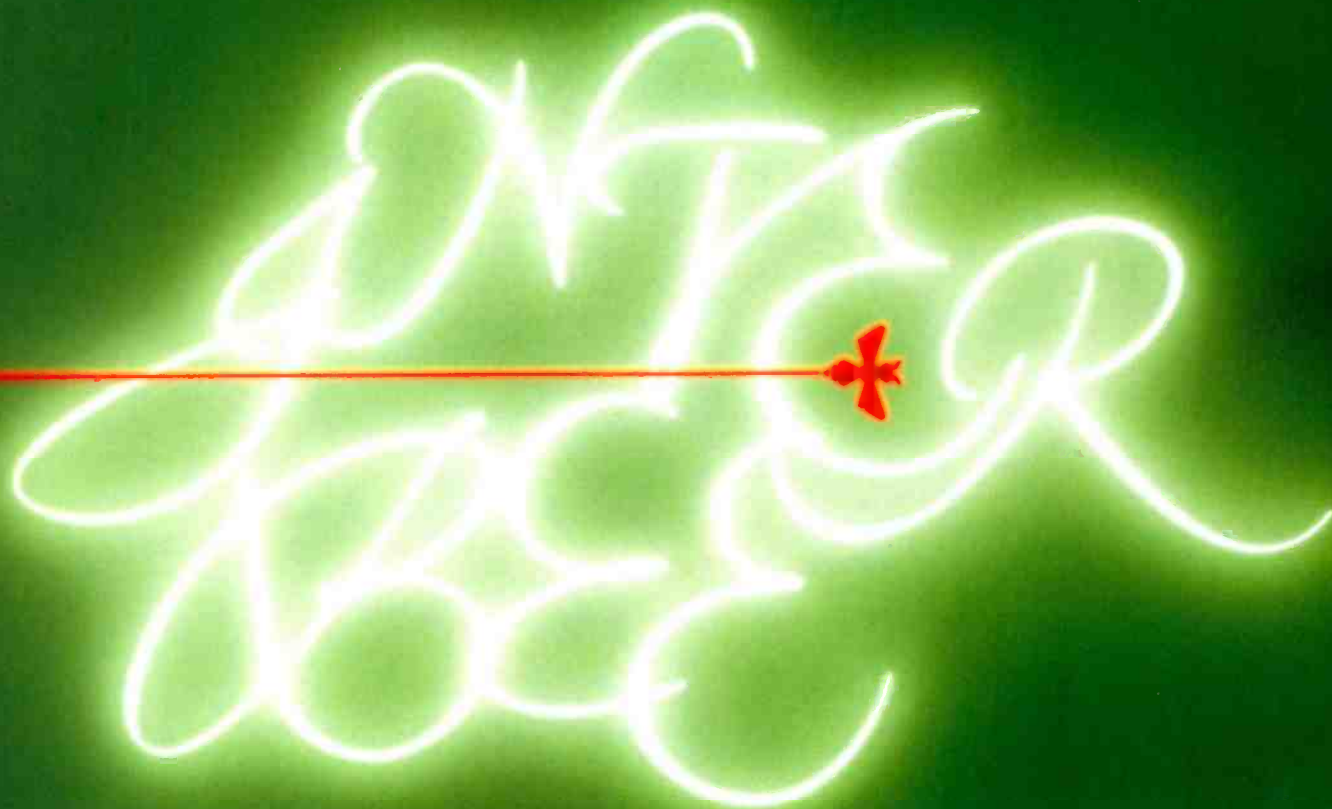
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◆ **October 9th–12th**
NAB RAdio Show
LA Convention Centre, Los Angeles, California, US. Tel: +1 202 429 5350.

◆ **October 10th—12th**
World Media Expo 96
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◆ **October 16th—19th**
The 2nd Pan-European conference of Community Radio Broadcasters
Copenhagen, Denmark. Tel: +44 114 279 5219.

◆ **October 24th–26th**
Broadcast India 96 Exhibition & Symposium
World Trade Centre, Bombay, India. Tel: +91 22 215 1396.

◆ **October 30th -November 1st**
WirelessWorld Expo
Orange County Convention Centre, Florida, US. Tel:+1 301 986 7800.

◆ **October 31st -November 1st**
ITS 8th Annual Magnetic & Optical Media Seminar
Mark Hopkins Hotel, San Francisco, US. Tel: +1 609 279 1700.

November 1996

◆ **November 1st–3rd**
The Seventh BKSTS International Special Effects Seminar
Pinewood, UK. Tel: +44 171 242 8400. Fax: +44 171 405 3560.

◆ **November 5th–7th**
Broadcast Engineering Society Expo 96
Hotel Taj Palace, New Delhi. India. Tel: 91 11 371 9978.

◆ **November 5th–9th**
PT/Expo Comm China
China International Exhibition Centre, Beijing, Peoples Republic of China. Tel: +52 525 592 3257; US Tel: +1 301 986 7800.

◆ **November 6th–9th**
Apple Expo 96
Olympia, London, UK. Tel: +44 181 390 0203. email:gautamp@computing. emap.co.uk Web: http://www.apple-expo. com/apple

◆ **November 6th–9th**
AV & Broadcast China 96
Beijing Exhibition Centre, Beijing, China. Tel: +852 2862 3460.

◆ **November 7th**
21st Sound Broadcasting Equipment Show (SBES)
The Metropole Hotel, NEC Birmingham, UK. Tel: +44 1491 838575.

◆ **November 8th–11th**
101st AES Convention
LA Convention Centre, Los Angeles, California, US. Tel: +1 213 258 6741.

◆ **November 12th–14th**
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◆ **February 22nd–25th**
Middle East Broadcast 97
Bahrain International Exhibition Centre. Tel: +973 550033; UK. Tel: +44 171 486 1951.

March 1997

◆ **March 13th–15th**
REPLitech Europe
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◆ **March 22nd–25th**
102nd AES Convention
MOC Centre, Munich Germany. Tel: +49 8142 14167. Fax: +49 89 32399 351. email: 102nd-chairman @aes.org. Web: http://www.aes.org.

April 1997

◆ **April**
Entech 97
Melbourne Exhibition Centre, Australia. Tel: +61 2 876 3530.

◆ **April 21st–23rd**
Cable & Sattelite 1997
Earls Court 2, London, UK. Tel: +44 181 910 7884. email: catherine.barrett@reedex .co.uk. Web: http://www.cabsat.co.uk/

May 1997

◆ **May**
Pro Audio Light and Music China 97
Beijing Exhibition Centre, Beijing, China. Tel: +852 2861 3331.

◆ **May**
Expo Comm Wireless Korea 97
Korea Exhibition Centre, Seoul, Korea. Tel: +82 2 555 7153. Fax: +82 2 556 9474.

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◆ **June 25th–27th**
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◆ **July 1st–4th**
SMPTE 97
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August 1997

◆ **August 25th–28th**
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September 1997

◆ **September 12th–16th**
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Amsterdam. Tel: +44 171 240 3839. Fax: +44 171 240 3724. email: show@ibc.org.uk Web: http://www.ibc.org.uk/ibc/

◆ **September 26th–29th**
AES 103rd Convention
Jacob K Javits Convention Centre, New York, US. Tel: +1 212 661 2355. Fax: +1 212 682 0477. Web: http://www.aes.org

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◆ **October 16th–20th**
International Audio, Video, Broadcasting and Telecommunications Show (IBTS 97)
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◆ **October 29th**
22nd Sound Broadcasting Equipment Show (SBES)
National Exhibition Centre. Birmingham, UK. Tel: +44 1491 838575.

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◆ **November 4th–6th**
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WE'RE SO USED TO some form of editing being integral to most of what we do that it is easy to forget the controversy it still causes in some circles. When so many jobs are impossible without overdubs, drop-ins, or even complete cut-and-paste editing, the attitude of some of the classical specialists —such as Nimbus, who avoid editing wher-ever possible—can seem quaintly archaic. Yet there is still a substantial demand for recordings of musical performances in the pure sense, and the arguments over the ethics of editing haven't gone away just because our DAWs make it easier to do and easier to get away with.

While the arguments in favour are essen-tially practical ones, based on the saving of time, effort and money; and in some cases the feasibility of getting the job done at all, the arguments against cut much deeper into our understanding of what a performance is, and how it comes into being. There are those who will assert that no two performances are ever the same and neither should they be, and that therefore editing between takes can never truly work. Strictly, perhaps, this is true, but most practical musicians do in fact duplicate performances remarkably

closely. There are those who will say that if an artist can't play a piece from start to finish to the satisfaction of himself and his audience he shouldn't be playing it at all; this overlooks the fact that a mistake on the concert platform is soon gone whereas on a recording it lives forever.

But where do you draw the line? It would be a hard taskmaster who would make someone play a whole 10-minute movement again for the sake of a cock-up in the coda —yet at the other end of the scale lies bar-by-bar, note-by-note, editing for reasons varying from incompetence to prima donna perfectionism, with a small grey area of true artistic sensitivity somewhere between. In theory it sounds both impossible to pull off and fraudulent, yet many top classical musicians have indulged in very heavy editing without anyone thinking the less of them, assuming they even knew.

There are, however, other considerations. Musicians who know the power of modern editing techniques risk losing the edge in their performance, simply because there is no pressure to get more than a couple of bars right at a time. Many truly great performances have an air of brinkmanship,

sweat and danger about them, and that is easily lost when there is such a reassuring safety net; it follows that rather than creating great performances, editing may be depriving us of them. Yet few would disagree that a recording for CD should be as flawless as we can make it, as long as the final result is not a distortion of the performer's intentions and abilities, which in a way still begs the question.

BUT I HAVE a special reason for worrying about such things. Often I have to record demo tapes for classical students to send away as an application for an overseas course of study, or as an initial entry to an inter-national competition. Should I edit these

> Many truly great performances have an air of brinkmanship, sweat and danger about them, and that is easily lost when there is such a reassuring safety net; it follows that rather than creating great performances, editing may be depriving us of them

recordings? There is no doubt that with the exercise of a little skill and patience I can make most of them sound more competent than they really are; but if another candidate has sent in—for reasons of honesty or expedi-ency—an unedited tape, my student may get through at the other's expense despite being less deserving. And that's not fair.

In the worst case, these hopeful performers risk making prats of themselves at the next live stage by being unable to live up to their recorded 'image'. On the other hand, if I impose my moral stance on my students and refuse to edit, I put them in the same position as the other candidates, disadvantaged compared with someone else's carefully edited tape. And that's not fair either.

It would be reassuring to think that it wasn't just the obvious slips on a recording that gave away a musician's ability or lack of it, that a sensitive listener could detect musical quality in more subtle details so that the editing wouldn't matter; but I've seen panels listening to these tapes, and it ain't that deep. The glib answer is that one should use one's judgement, but that's a cop-out; blanket guidelines can't apply, particularly when you don't know what the competition is doing.

I can't be the only one making these tapes; anyone for a code of practice? ⓢ

NOT JUST ANOTHER AMPLIFIER.

Just as our mixers combine high performance, great value and extra features, our new Fast Recovery Series™ M•1200 sets a new benchmark for amplifier value. It can help your speaker cabinets put out tighter bass and enhanced high-end audibility. It can play all day into big, ugly reactive loudspeaker loads. And it can survive sizzling ambient temperatures and brown-out voltage drops that croak other amps. The M•1200 was created by a talented design team, many of whose members have over 20 years of power amplifier circuit experience. And it's made in a state-of-the-art, automated manufacturing facility that lets us keep quality up and prices down.

THE FAST RECOVERY DESIGN STORY.

Thanks to exotic technology borrowed from high-speed digital components — and sparing use of negative feedback — the M•1200 keeps sounding good when driven to the max. If most conventional amplifiers are driven into clipping, "corrective" negative feedback

HIGH WATTS. LOW PRICE.
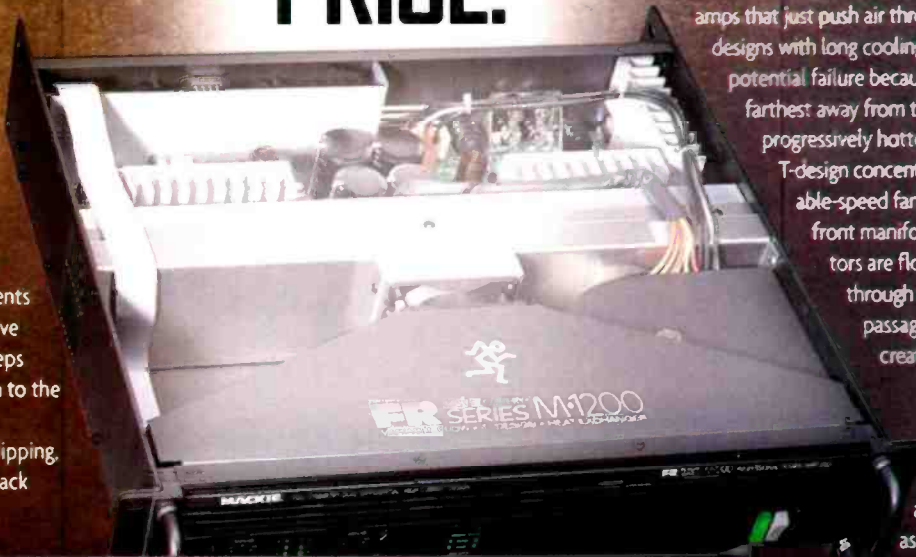
actually causes internal saturation that keeps them "latched" in a prolonged — and audible — state of clipping. The M•1200 uses a high-speed, latch-proof design with extremely low negative feedback that eliminates high-frequency sticking and gives the amp enhanced stability. Until now, this solid, proven circuit principle has only been found on very expensive designs. The M•1200 achieves efficiency just 3.5% under the theoretical maximum possible, with an output stage that delivers in excess of 60 amps of current. It is capable of 4000 watts of power dissipation.

T-DESIGN CONSTANT GRADIENT COOLING

Amplifier output transistors generate intense heat. If it isn't conducted away, reliability drops. Cheaply-built amps that just push air through the chassis or amp designs with long cooling tunnels court potential failure because the transistors farthest away from the fan get bathed in progressively hotter air. The M•1200's T-design concentrates air for the variable-speed fan through an oversize front manifold. All output transistors are flooded with cool air through the short, dual air passages (see drawing below), creating a constant temperature gradient and vastly increased reliability that lets the M•1200 run all day at ambient temperatures as high as 113° F!

M•1200 T-Design cooling
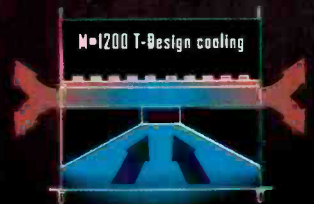
BUILT-IN FEATURES INSTEAD OF OUT-BOARD BOXES & PLUG-IN CARDS

A **low pass subwoofer crossover** eliminates the need for an external electronic crossover or plug-in amp modules. **Variable low cut filters** let you dial in the tuned bass cut-off point of your speaker cabinets or stage monitors for dramatically tighter low end.

Variable high frequency compensation eliminates the need for hard-to-adjust crossover modules or graphic EQ tweaking to achieve the treble compensation required by compression drivers mounted on Constant Directivity horns. **Subsonic stabilization** stops visible, random woofer cone movement that robs amp power and causes intermodulation distortion.

M•1200

1200 WATTS INTO 4 OHMS, BRIDGED WITH LESS THAN 0.05% THD

600+600 WATTS INTO 2 OHMS, STEREO

BUILT-IN SWITCHABLE 63Hz/120Hz ELECTRONIC SUBWOOFER CROSSOVER

SWEEPABLE 5Hz-170Hz LOW CUT FILTERS

SWEEPABLE 2kHz-6kHz CONSTANT DIRECTIVITY HORN COMPENSATION & "AIR" EQUALIZATION

SUBSONIC STABILIZER

LOADED WITH EXTRAS FRONT, BACK & INSIDE.

...Such as full LED instrumentation including Temperature & Short Circuit indicators, multi-step detented Level controls calibrated in dB and volts, three inputs per channel to make signal pass-through & signal splitting easy, elaborate short circuit/overload/thermal protection & automatic turn-on delay. Visit your Mackie Dealer today for more information on the M•1200!

MACKIE.

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