Studio sound

THE INTERNATIONAL
TECHNICAL MAGAZINE FOR
PRO AUDIO, POSTPRODUCTION
& BROADCAST

EXCLUSIVES

Neumann M149 Tube
Amek RCMA
P&G PP10
Yamaha 02R
bench test

BROADCAST

NAB preview
Audio compression
GP D-ESAM 200 console

The
**John
Leckie**
Interview

04

contents

Willi Studer's contribution to the evolution of audio is a matter of record; his influence over its future is assured.
Obituary on page 8

REVIEW SECTION

PRODUCT OF THE MONTH

STUDIO SOUND

START PAGE 18 👉

LECKIE INTERVIEW

From Marc Bolan to Radiohead, from Pink Floyd to The Stone Roses: John Leckie gives an exclusive interview to **Studio Sound**

For some studio owners, only one console is good enough.

"For our large music studio, I was looking for a world class console with a 6 channel monitoring system, that was familiar, easy to learn and capable of integrating with our post production studios. The SL 9000 J Series is the only console able to deliver our high standards, and sounds 'magnifique'."

Thierry Rogen, Owner, Mega West, Paris

SL 9000

Solid State Logic

editorial

THIS EDITION'S CONTRIBUTORS

RICHARD HULSE is a senior recording engineer for New Zealand Public Radio where he specialises in classical music recording and editing. He also acts as an audio processing consultant (a title he dislikes, but it pays) for several commercial FM stations. He also juggles a lot.

ALEX van SOMEREN's latest UK-based business venture has developed secure data transmission technology based on cryptography for use over the Internet. Despite being a total techno junkie he can still produce quarter-inch tape splices of which he is justifiably proud.

SAM WISE is a technical consultant and regular contributor to Studio Sound. He runs a UK-based consultancy practice specialising in systems engineering for the performing arts including audio, acoustics, audio-visual, video, datacoms and control systems.

Caught in the net

We've made a mistake. A big mistake. Again. We made it when we graced information with a capital 'I' and built new businesses around it. We made it when we tabled terms like Information Technology and Information Security. We made it when we unconditionally associated the term Information with advantage.

We made it when we failed to discriminate between information and interference.

Some years ago, a well-respected friend told me that 'information without organisation was useless'. And he was quite right—but he'd only seen half the picture we can now see. He was talking about the reams of paper that we tend to acquire only to discover that we can't find the critical page when it counts most. Since then, the 'paperless office' has proven its ability to see of more trees than its paper-dependent predecessor. Recently, the virtual world of the Net has arisen and with it comes information in variety and quantity never before presented to the likes of you and me.

How could we resist the invitation to avail ourselves of information concerning everything from audio to anime? How could we know that we were exposing ourselves to volumes of information that outstrip nightmare levels of junk mail?

The information overload that accompanies electronic means of communication (and particularly publishing) is potentially suffocating. Never mind those hours lost to computer games a few years back, this is the real thing—you can set out with every intention of accruing essential information on the subject of your choice only to come away deluged by downloads from web sites and usegroups that challenge your resources of time and tenacity. The Net is talked up everywhere; it promises everything, yet it is still liable to reward you with the most disappointing and distracting results.

YOU SEE, the Net has no blueprint. It has evolved over the last ten years or so largely supported by academic institutions but crucially shaped by the whims and passions of those prepared to devote their time to it. And you must choose to deal with it on its own terms or be prepared to help shape it yourself.

Like Archimedes—formerly Ludwig—Plutonium. A mathematician friend tells me that Archimedes is believed to be a self-educated member of the maintenance staff of an American university who has ready access to the Internet and uses it to challenge the ideas promoted by the likes of Stephen Hawking and his eminent predecessors. I don't know how serious he is, but you can choose to be enraged or entertained by his activities. Either way, you can be assured that the privilege is eating up your time and running up your connection fee. Weren't you meant to be downloading a software update for your studio's hard disk editor?

If you were, you may have recognised another of the Net's shortcomings. The ability to distribute software over the net is attractive in principle but can be more than frustrating in practice. A 2-hour download that is prematurely terminated moments before completion by a failed comms link should carry the Surgeon General's health warning. But then, it's cheaper to send programs over around 750k (half a diskette) long within the UK by post in any case. The same mathematician observed that a comms link is all that's likely to go down on you when downloading pornography. He also reckons magazines are better value for money...

Looking to the future, the Net is currently threatening to rediscover its academic roots with the development of a second 'tier'. The argument is that providing another level of service with such refinements as a higher transmission bandwidth would clear the anorak element from the path of the academic institutions. Whether this will set the Net back on track, or simply give Archimedes a better platform for his insanity remains to be seen.

editor

best-selling workstation, Pro Tools III, with mountains of record tracks, I/O, and real-time effects and mixing. Each step of the way, you'll learn and grow with Pro Tools software, share work

between systems, and take an upgrade path that lets you build on the value of your investment. Audio professionals now have more choices and flexibility than ever before. And getting started couldn't be easier. To learn more about the clearest path to the best in digital audio, call **01753 653322.**

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soundin

Another fine Messe...

THE FRANKFURT MUSIKMESSE which took place from 13th to 17th March at the Messe Frankfurt, proved to be the largest ever, with the affiliated Pro Light & Sound exhibition doubling in size after its introduction last year.

Featuring over 450 sound and lighting exhibitors over 40,000 sq metres of hall space, Pro Light & Sound was the major success story of the whole show which featured over 1,600 exhibitors from 45 countries being seen by over 100,000 visitors over the five days. The MI sector halls also saw an increase of over 200 exhibitors due to the expansion in overall exhibition space.

Surprise of the show as Focusrite, who previewed their new Green range of signal processors to dealers and the press prior to their official launch at AES in May. Designed as affordable products for the musician and project studio, the first three models in what will become a 12-product series are the Green 1 dual mic preamp, the Green 2 Focus EQ multi-input direct recording module with six stages of Focusrite EQ and the Green 3 Voicebox, which is a single-channel mic pre with three stages of Focusrite EQ and multi-function dynamics section featuring compressor, de-esser and noise reducing expander.

Introducing the new product, which are all 1U-high 19-inch rackmounting with a projected price-point of sub £1000 (UK), Focusrite managing director Phil Dudderidge said, 'For every Red Range owner there are probably a hundred other people who aspire to owning a Focusrite but who are deterred by the relatively high price. It is for this greater recording and performing community that the Focusrite design team has conceived and designed the Green range.'

Whereas the industry in general will have to wait until May to see the Green range on show, there was still plenty to whet the appetite with Sony launching the rack-mounting MDS-E3 MiniDisc recorder, DTC-A6 DAT deck and CDP-L3 CD player alongside the SMS-1P powered close-field monitor. New monitors also appeared from Spirit, who has now added a further three models to the Absolute range with the Absolute Zero close field, Absolute 4P internally bi-amplified close field and Absolute 1P active powered monitor subwoofers which feature two 100W amplifiers.

Elsewhere, Soundtracs was showing its new Virtua digital 48-channel mixer which at £15,000 for the 48-channel analogue input base model, looks set for success in the serious project studio, commercial music studio and video post production markets. Prepare also for Pro Tools 4, a beta version of which was being previewed by Digidesign.

Despite its somewhat confusing Pro Light & Sound tag, with an overall increase in studio and audio production product on show, this years Messe may well be seen as the first part of a trend-setting reorganisation of the European exhibition calendar—and its certainly worth adding to any exhibition visit list for 1997. Next year's show will take place from 26th February until the 2nd March. **ANDY WOOD**

DVD AUTHORING will become a viability for the first time when Sonic Solutions unveils its authoring-premastering system at NAB. Called DVD Creator, the system incorporates all the elements of DVD production: audio, video and authoring.

Sonic has put together a suite of DVD applications—video compression and decoding (MPEG2), audio compression and decoding (AC3, MPEG2 and PCM), and disc authoring (Scenarist-2) and formatting. Applications are networked using Sonic's high-speed MediaNet.

The system was developed in collaboration with the Japanese graphics company Daikin Industries and will laterally dictate how DVD contents will be created, both for player manufacturers and programme developers.

Until now, the industry has been waiting to see which element of DVD would be launched first. With the introduction of the Creator, the programme owners (labels) will be given a set of production tools such that they can produce discs by the Christmas period. Hardware manufacturers will be able to manufacture players to handle the software in time for a simultaneous introduction.

Daikin's Scenarist, a custom created White Book 2.0 authoring tool for video CD is the basis of Scenarist-2 which is poised to become the de facto standard for DVD. Sonic used the original White Book authoring tool for an earlier system called Sonic Cinema which placed MPEG1 video on CD.

'That's where we started,' said Mark Ely, Sonic's DVD Marketing Manager. 'What we've done since is adapt our tools and production methods for Digital Video Disc. When it became apparent that DVD was emerging, we knew that we were ready to modify our creative production tools specifically for the DVD production process.'
DEBRA PAGAN

WILLI STUDER, one of the true pioneers of the audio industry, passed away on 4th March 1996 at the age of 84 years.

The original 'self-educated' man, Herr Studer found his first company in a basement, making test equipment. However, the move into audio itself was not long coming. Having finally got his hands on the ubiquitous Telefunken Magerophon Willi Studer embodied the time-honoured innovator's phrase 'there has to be a better way' and ensured the invention of the first Revox tape recorder was born.

In fact, the whole of Studer's career reflected his two parallel loves: profession excellence, as typified by the Studer range, and the finest in reproduction as represented by the Revox hi-fi equipment.

UK: The control room of London's legendary Abbey Road Studio 2 has reopened after a complete refit. Designed by Sam Toyoshima and Abbey Road's in-house team, the studio showcases a new 60-channel Neve VRP console with Flying Faders. Monitoring is via a custom-designed Quested installation and B&W 801s (which can be removed to accommodate other main monitors). The refit was completed in eight weeks during which time structural work— which was limited to the control room, with the live area being restored to its original condition after plaster had begun to fall on noisier clients—had to take place between 8am and 10am to avoid compromising the other studios. The Studio 2 echo chamber has also been permanently reinstated at its original site and with two KM56 mics as originally used. **TIM GOODYER**

gs

News in brief

DAB TRIALS conducted by British independent radio broadcasting body NTL began last month. A guinea pig group of seven national stations constitute a full stereo DAB multiplex whose transmissions will bring state-of-the-art transmission to mobile receivers beset by multi-path reception problems and offer the first radio text services. NTL Radio Business Manager Mike Thorne described DAB as 'being surrounded by questions—this should be the start of the answers'. There is no doubt that the FM band is overcrowded, and those critics of any advance in audio quality offered by DAB should draw solace from the relief the format will offer. DAB is now approaching a world standard with wide support throughout Europe (only France having conflicting interests in frequency allocations) although the US is still deliberating over Eureka 147 and the position of the Japanese uncertain. Given momentum by the British Broadcasting Bill, the trials place the UK at the leading edge of the field, according to NTL's Andrew Sukawaty. Latest broadcast news reports that on 28th March, NTL was bought out by International CableTel Inc for approximately £200m with a further payment of £35m to NTL shareholders one year after completion. Also, American-based Orban has introduced a low-cost Optimod unit, the FM2200, aimed at smaller broadcasters seeking the facilities associated with established Optimod units.   **TIM GOODYER**

Studer and Revox were closely associated for much of the way. Parallel milestones were the C37 and G36 recorders with valve circuitry, the A80 series and A77 recorders, the A800 and B77. The professional range was continued with the A820 and A827 analogues machines and D840 and D827 digital multitracks, not forgetting the range of broadcast consoles and peripheral equipment.

Dr Studer (as he became) enjoyed an almost obsessive attention to detail—which could be considered a having had positive and negative effects, as it tended to slow initiative when everything had to be submitted for approval. It also meant that he had little time for social niceties and the problem of deciding what to wear was solved by ordering five identical grey suits.

With the changing face of audio,

the early 1990s saw Studer at the crossroads with the problems of Willie Studer's impending retirement and succession.

A stint with Motor Columbus followed by severe streamlining and restructuring and the culmination of the 'strategic alliance' with Harman, now sees Studer AG as 'a company for the 1990s', but to many this is a company with a concept removed the principles of its founder.

Willi Studer leaves a legacy of innovation and engineering excellence to the audio industry, where the result took precedence over cost. A little old fashioned today, perhaps, but its a memorable heritage.

Studer's epitaph could be his favourite saying: 'We may not be the first, but we will be the best'.

TERRY NELSON

♦ Norwegian NRK and Finnish YLE national broadcasters have taken the first large European sites for Avid's NetStation 14.0 newsroom automation system. The system facilitates networking of PC workstations running Win95 and NT and hosts 250 stations at NRK and 500 at YLE. The YLE installation is a pilot for the broadcaster's entire operation.
Avid Technology, US.
Tel: +1 508 640 3158.
Avid Technology, Europe.
Tel: +44 1753 655999.

♦ Taipei's Creator A-V studios has replaced its SSL G-series console with a 72-fader Euphonix CS2000. The studio has hosted a variety of Eastern hit acts alongside various post duties.
Euphonix, US.
Tel: +1 415 855 0400.
Euphonix, Europe.
Tel: +44 171 602 4575.

♦ California's Skywalker Sound has purchased four TimeLine Studioframe workstations to bring its total up to 14. The prestigious post facility is running v6.20 software on all systems.
TimeLine Vista, US.
Tel: +1 619 727 3300.

♦ British rock legends The Kinks have bought tow 6-track Pro Tools 3 systems. One has gone into Ray Davies' writing suite along with an 882 interface, the second with two 882s, an ADAT interface and a D-Verb plug-in has gone into Konk studios.
Digidesign, US.
Tel: +1 415 688 0600.
Digidesign UK.
Tel: +44 1753 653322.

♦ Dutch commercial TV network NOB has installed two Graham-Patten Systems D-ESAM 400 digital edit suite mixers. One of these is in use with an Accom Axial 2010 edit controller on a virtual reality production stage; the other is in a conventional video edit suite. NOB has four further D-ESAM consoles on order.
Graham-Patten Systems, US.
Tel: +1 916 2370 8412.

♦ Independent UK broadcast service Black & White has ordered a 48-channel S-series Calrec console for its Unit 1 OB vehicle. B&W specialise in sports OB work for Sky, HTV, Granada and the BBC among others.The installation will complete the truck's refurbishment.
Calrec, UK. Tel: +44 1422 842159.

♦ Bulgarian National Radio's Sofia-based facility is to see a major refurbishment centred around the country's first SSL console. The SL4040 G+ will go into BNR's Studio One where the broadcaster concentrates its music production effort. Four further SSL consoles have been

delivered to Parisian private radio network NRJ. The 32-channel 4032 G+ desks all have Total Recall and will be installed at the broadcaster's HQ in Paris XVI. In the US, Washington's KCPQ-13 Fox has ordered an Axiom console for live on-air production work and the state's National Mobile Television has installed an SL8000 GB console in its digital remote truck.
SSL, UK. Tel: +44 1865 842300.
SSL, US. Tel: +1 212 315 1111;
+1 213 463 4444.

♦ Cincinnati-based Bootzilla Re-Hab studios—home to Bootsy Collins—has installed a Meyer HD-2 main monitoring system. The HD-2s share acoustic space with an Amek Big 44-channel and two 32-channel Mackie desks, 32 tracks each of ADAT and DA-88, and both 4-tack and 8-track Akai hard-disk recorders.
Meyer Sound, US.
Tel: +1 510 486 1166.
Amek Systems & Controls, UK.
Tel: +44 161 834 6747.

♦ Slovenia's TV network has awarded the contract for three transmitters: a Platinum Series HT EL 10kW Band III and two UltraVision 2kW UHF transmitters.
Harris, US. Tel: +1 217 222 8200.

♦ London's Tele-Cine multimedia post house is to install the city's first MPEG2 coding system. The Minerva Systems Compressionist 200 works at between 3Mb/s and 10Mb/s and will provide for Tele-Cine's broadcast and new DVD requirements.
Tele-Cine, UK.
Tel: +44 171 208 2200.

♦ The American ABC Television Network has installed two pairs of Bag End Studio System monitors. The monitors have been chosen to meet the requirement for extended bass and spl in a confined listening space.
Bag End, US.
Tel: +1 708 529 6231.

♦ The Japanese Tokyo Broadcasting System's Midoriyama studio has ordered a 40-channel Neve VR Legend to replace its Neve 5315 in TV programme recording. Also in Tokyo, Fuji Television has purchased a 60-channel VR Legend for music and general programming. Both facilities already have Capricorns.
AMS Neve, UK.
Tel: +44 1282 457011.

@American studios, Presence, based in Westport, has taken delivery of a pair of ATC SCM10 monitors and The Boston Symphony has taken delivery of a pair of SCM20 monitors. Both orders were placed after extensive evaluation tests.
ATC, UK. Tel: +44 1285 760561.

Studios of the future

The falling cost decent of audio equipment means that almost anyone can afford to own a studio of sorts. Where does this leave the traditional studio?

The last few years have seen a dramatic alteration in the way that audio equipment is built. The swing to digital inevitably meant that things would get smaller and cheaper, bringing equipment, which previously only a well-backed organisation could afford, within the reach of the individual.

Probably the first such item was the much loved PCM F-1 which produced sound quality out of all proportion to its price. While unashamedly a consumer device, with all of the consequent restrictions, it served as a toe in the digital water for a lot of people who couldn't afford the professional digital kit of the day.

Next came DAT which was killed off as a consumer product but found a home as a semiprofessional recorder. The obvious next step was to combine a video deck with DAT technology to build today's popular 8-track digital machines.

Mixing consoles and effects units have kept up with recorders in the economy department, with project consoles appearing having a staggering list of features with a silly price tag.

On the face of it, these developments don't paint a bright future for the traditional studio. What is to stop absolutely anybody from setting up their own studio?

In fact, things could be worse. It is always instructive to see what technology has done in other areas to see if there are any lessons to be learned. Take photography; traditionally photography meant lugging around a ton of equipment including your own darkroom. The process required a lot of skill. Today's electronic camera requires no skill at all to operate, many even load the film automatically in addition to automatically removing the lens cap, setting the exposure and turning on the flash. Such cameras have brought fuss-free photography to any of us who want to get involved in it—all we have to do is compose the picture. Unfortunately most camera users know nothing about composition and the results are predictably dire.

Exactly the same thing has happened in home movies, first on film, later to be replaced by video cassettes. Anyone could make a movie, and of course, anyone often did, to the extent that watching people's home movies has about as much appeal as waking up in a Turkish prison with Colonel Quadhafi.

Word processors and, later, desktop publishing changed the preparation of the typed word dramatically. Now anyone who wants to can publish a book. So where are all these books?

RETURNING to the would-be (that's what an old fart like me calls a wannabe) home studio owner, how does the low cost of recording, mixing and effects change things? Apart from reducing the capital cost of those devices, the frank answer is that nothing else changes. The cost of a reasonable selection of professional quality microphones will still be substantial. The cost of suitable premises for making a recording will be unaffected. A decent pair of monitoring loudspeakers will still be necessary. But the most expensive item which any studio needs is the one that can't be bought over the counter. It's the creativity, knowledge, skill and experience of the engineers and producers.

This is the thread running through all of my examples. Certainly, reducing the cost of a product brings it to a wider market, but that very popularity reduces the chances of the product being used by anyone having the requisite skills. Now, I wouldn't dream of standing in the way of any of this. It's a free country and people can indulge in any pastimes they wish. If this involves buying a home studio, that's fine. While a small number of home studios today demonstrate the necessary skills producing good work on a tight budget, my suspicion is that even if the hardware was free with breakfast cereal that number would not increase significantly. Equipment cost has never stopped the audio enthusiast before. The very keen will build their own equipment or make do with patched up second-hand stuff. To go that way you have to know what you're doing and it's an excellent way of learning about audio.

There is only so much creative energy around in any field. Spread it wider and it gets thinner. We've seen that with television; more and more channels, less and less merit. Consequently, the traditional studio which has its act together has nothing to fear from inexpensive production equipment.

Where the traditional studio scores is that it has (or should have) properly treated recording spaces and a good selection of quality microphones. The equipment will be lined up so there are no level hassles. There will be a good set of monitoring speakers in a suitable control room. There will be a big-name mixing console and as many recording tracks as you could want. But the biggest strength of the large studio is the staff. There are people who know how to select and position microphones, who know how to use the equipment to best effect.

Like most enterprises, the large studio is only as good as the people. With skilled and motivated staff, the product

> These developments don't paint a bright future for the traditional studio. What is to stop absolutely anybody from setting up their own studio?

will be consistently good and that's what clients are looking for. Consequently the large studio that is going to survive will be the one that invests in its people. Working with today's high tech audio equipment requires a good grounding in the theory as well as operational experience. While taking on a tea boy and expecting him to learn by watching might have worked when equipment was less sophisticated, today it's a bit of a joke. What can you learn about ISDN by watching LEDs blink on a modem? The studio that wants to survive will be the one that recruits people with the right potential and then invests in staff training. That training programme will cost more than a home mixing console.

Feel the need in me

The conflict between personal and commercial facilities is causing feelings to run high from coast to coast writes **DAN DALEY**

In a industry of bits, bytes, dBs and other specs, the emotional components that runs like an underground stream through the landscape of technology are often overlooked. I suppose if this were the Italian Column instead of the US Column, a discussion of feelings would be implicitly understood. But Americans are doers, not feelers.

Yet the feelings of traditional, commercial recording facility operators have ridden an emotional wave since the arrival of personal (home, project or whatever they're called this week) studios. Initial feelings of resentment quickly boiled over into outright loathing. After mostly fruitless attempts at stemming their tide, there was a wave first of resignation, then of apparent conciliation as commercial facilities began to find and implement ways in which to co-exist with personal studios. For a while it looked like white smoke was issuing from the Vatican chimneys; the conclave had come to a consensus. Or maybe it was more like a white flag.

Post houses and recording studios alike bought into ADATs and DA-88s to interface with the proliferation of personals but the acrimony remains, subtly but decidedly, at the recording studios. They have not gone easily into that night. Post facilities have developed new applications for MDMs—for instance, the DA-88 has established a new niche for itself as a replacement for the mag film dubber. But recording studios have yet to evidence any public pride in their down-market acquisitions; while in most homes a Mackie 32 is the equivalent of a Mercedes in Moscow, in a recording studio it's considered a concession.

It's ironic that many audio post houses, which are often affiliated with larger film and video studios and are thus more capital-capable than most audio studios, are able to command far higher hourly rates for equipment complements, that include MDMs as an integral part of their offerings than audio recording studios, for which the MDM is a bitter pill and an icon for an evolutionary turn in technology that has conspired to keep their rates frozen in place since the first Reagan administration. I was at the inaugural meeting of a new association of recording studios in Nashville in February. In the off-the-record discussions I had with several of the early principals of the group months earlier, the issue of home studios was a regular topic, and the level of vehemence and venom regarding them generally increased as the evening's bar tab piled up. Unlike the LA version of this phenomenon, HARP, in which the home studio issue was at the top of an otherwise thinly veiled agenda, the Nashville studios are approaching it gingerly, at least in public, with conciliatory rhetoric, if there is any rhetoric at all.

But it's useful to note that the only last-minute changes to the group's official manifesto involved qualifications for voting membership, and that was finally defined as being limited to those recording studios that '...can evidence their ability to transact business by [having] a valid business license'. There are many ways to define entities, using technical, bureaucratic, economic or scientific parameters and in choosing the business license as an arbiter of exclusion or inclusion, Nashville's studios have chosen an emotive parameter as much as a bureaucratic one. It remains, by definition, an emotional issue for recording studios. So while a cordial front has now developed between the traditional, commercial recording studio on one side and the project

studio on the other, beneath is a roiled cauldron of feelings that, other than in back room and bar room railings, have little or no opportunity to vent themselves.

NOBODY HAS inveighed against competition, which would be out of character with the quintessentially American nature of US studio owners, who are as deserving of the sobriquet 'cowboy' as the original range drivers were. What they have consistently groused about is what they called 'unfair' competition (an equally quintessential—and often vague—modern American complaint in an age of professional victimisation), the idea that they need to file papers, meet codes, pay specialised taxes and accommodate the other radars of bureaucracy that home studios manage to fly under. Whether it is unfair or not—and no one really has a lock on the truth in this department—it still builds up a substantial head of steam in the pressure cooker that is the recording studio business in America today. There are release valves built into that metaphor; other US industries have undergone significant down-sizing and restructuring—equally as painful

Spinning a line—would you buy a place on a Detroit audio course for these men?

as that which recording studios are undergoing—adapting to current economic and technological climates.

Many recording studios are doing the same, adding new services such as multimedia, or creating alliances with record labels, or starting their own labels. Others are changing focus altogether, such as Seattle anchor facility Bad Animals, which announced in February that it was essentially getting out of the music recording business (co-owners Ann and Nancy Wilson of the band Heart will still use the facility for music, as will other select customers) in favour of audio postproduction.

Not everyone can make such radical shifts of focus, though, due to financial or other constraints. For them, new ways of approaching marketing their existing technology, expertise and client bases will be the answer. In the long run, the change will be good for the overall industry. In the meantime, however, it's useful to acknowledge the fact that below the surface of the commercial studio-project studio detente lies a Poe-like uneasiness. As one anonymous studio owner in LA put it to me recently, 'I know I have dealt with them [home studios], but that doesn't mean I have to like it'. And as with any major sea change in the evolution of a culture, from racial integration to the fall of Communism, changes in the emotional component will inevitably lag behind the march of events.

But these components will, just as inevitably, work out the way they're supposed to. In the end, the greatest asset any studio can have in this environment is not a digital one. It's plain, old-fashioned, analogue patience. ●

Breaking cover

Pressure from the press has brought Sony's latest recording system into the open—along with the accompanying debates it writes **BARRY FOX**

The European press are first to get a nonconfidential briefing on Sony's Direct Stream Digital system. Sony took over Studio 3 at Abbey Road for three days in February, staged an excellent live versus recorded sound demonstration and answered questions frankly. But this only happened after Sony's European and US offices warned Tokyo that the situation was fast turning sour.

Previous briefings in New York, LA and London's West Side had been open only to studio engineers and producers. Any of these who also wore a press hat were either barred or asked to sign a document promising not to write about the event. Inevitably the story leaked out and the Acoustic Renaissance for Audio started lobbying against DSD.

The first prong of the ARA's argument is that because DSD samples at 64x the CD rate, the data streams at over 2.8Mb/s and swallows up too much land for even a high density disc to store a meaningful playing time for multichannel surround. The second prong is that Digital Signal Processing techniques cannot be used on the DSD data.

Sony Music in the UK set up the Abbey Road sessions to clear the air. Sony Tokyo then saw the first press reports on the DSD leaks and pulled the plug on press invites. I was told I could attend only if I committed not to write about the event. I refused. Sony's sanity prevailed and the event was reopened to the press.

So we are at last able to report on the thinking behind DSD—or, perhaps more accurately, the way Sony's thinking is being presented following the record industry's concern over the high-density digital video disc and the hi-fi industry's enthusiasm for a super-fi audio-only version.

As David Kawakami, Sony's Director of New Technologies, puts it: 'Record labels are not exactly jumping up and down at the idea of a new product. They are more interested in the continuing growth of CD.

'First press reports in the debate on DVD and high-density disc put the cart before the horse' continues Kawakami. Perhaps this is so, but Sony must shoulder a lot of the blame for this by demonstrating DSD to industry engineers and expecting them to keep quiet.

Sony's line is that DSD was developed as a method of archiving the CBS vaults. 'There are 300,000 analogue masters, and around 30,000 or 40,000 digital masters. That's what we paid CBS $2bn for. And some are in perilous condition' says Kawakami.

DSD accommodates a bandwidth of up to 100kHz and a dynamic range of 120dB. The master recordings can be stored on 60Gb cassettes, in silos, for archiving. The archived signal can then be down-converted to 88.2kHz or 44.1kHz format, for possible future release.

Continually playing down the HD-DVD application Kawakami talks only in the long term about 'starting to prepare the industry for a new distribution format'. He stresses the value of 'scalable access to match future distribution options in efforts to improve the quality of CD'. The prime aim he repeats is for 'archiving our life's blood to allow future mining of the catalogue'.

Sony has also demonstrated DSD to the Advanced Digital Audio committee in Japan, 'but primarily to the professional group', stresses Kawakami. 'We are trying to promote discussion. We are looking at the long term and how you get there'.

AT THE PRESENTATION, David Smith, Director of Audio Operations at Sony Music Studios in New York, introduced a 'bare all' test. A group of five top UK jazz musicians (sax, piano, guitar, acoustic string bass and drums) played live while the output of the mixing desk fed studio monitors through four chains, controlled by a passive switch box. One feed was live sound. The next was through DSD A–D and D–A convertors and recorder. The third feed was through a dCS 900-950 convertors to Sony's PCM9000 M-O recorder. The fourth feed was to a half-inch analogue recorder running Ampex 499 tape at 30ips with no noise reduction.

The band played three extended tunes, with guests free to switch between the four settings. The same music was then played back from the three recorders.

I am happy to admit that I could hear no repeatable difference between the live, DSD and 20-bit feeds. Perhaps my ears are too old; perhaps I was enjoying the music too much. Perhaps not. Some industry golden ears had previously thought the DSD image wider, while others thought it narrower.

'Digital is getting very analogue in character', says Kawakami.

There is another way of looking at this. At Abbey Road the analogue recording showed perhaps slightly less harmonic edge to the string bass. Otherwise the tests proved just how good analogue recording can be if the machine is accurately aligned. In the Q&A discussion afterwards, David Smith admitted that he is currently archiving old masters onto half inch Ampex 456 analogue Grand Master tape at 30ips if 'frightened by their condition'.

Smith is reluctant to say whether Sony Music has yet made a positive decision to use DSD, and if so when he will 'throw the switch' to start full-scale DSD archiving.

'We want to digitise as soon as possible', says Smith. 'We know it should not be 16-bit, and not 20-bit. We think it should be DSD but there are ten people involved in the decision and evaluation will last nine months to a year. My advice is not to commit to 24-bit, 88.2kHz. My firm opinion is that DSD is the best we've seen.'

LAST MONTH's column revealed details of the BBN project which the RIAA sponsored after the Copycode debacle. BBN smears a layer of coded noise over the audio signal to watermark it. The RIAA has confirmed that BBN is 'managed by our association'.

David Kawakami confirms that Sony is currently proposing DSD to all the record industry trade bodies, including the RIAA. This is necessary because record companies will not commit to an archive format unless there is industry-wide acceptance to guarantee long-term hardware support.

David Smith confirms that the BBN hardware stayed with Sony in New York after David Stebbings left the company to head up the RIAA's New Technology Division. The RIAA says it 'makes perfect sense' for Stebbings now to co-ordinate the examination of all identification systems.

The buck now stops with the RIAA. Can it really back DSD as a industry standard for archiving and future releasing, while at the same time backing BBN as an industry standard for watermarking releases by polluting them with noise? And how long will Sony continue to show implicit support for the BBN noise pollution system by holding the hardware? ⓢ

Broadcasting begins at home

Often dismissed as national protectionism, there is a strong case for positive discrimination in developing territories' broadcasting policies writes **NICK MASTERS**, Editor of *Asia Image*

Whether your taste in films errs towards the heaving cleavage of the lavish English costume drama or the comic book violence of shoot-'em-up action adventures, it is quite likely that you have recently witnessed the work of Asia's hottest film directors.

No strangers to regular attendees of fashionable London art house cinemas—where anything without sub-titles is dismissed as a 'movie'—Asian directors' work is now regularly seen in mainstream Canon and Odeon theatres around the Western world. And it regularly involves productions featuring established Western actors and actresses.

At present the two highest profile examples of this cross-continental film phenomenon are John Woo's *Broken Arrow* (with John Travolta) and Emma Thompson's *Sense and Sensibility*, directed by Taiwan's Ang Lee.

A long-time director of hit cult movies, Woo recently achieved a high profile after receiving generous praise from American Director Quentin Tarantino for his low-life, Hong-Kong-based gangster films. Meanwhile, Ang Lee played to a large international audience with his breakthrough movies *The Wedding Banquet* and *Eat, Drink, Man, Woman.*

It must be acknowledged that both directors are far removed from the brooding melodrama and literary angst of celebrated Eastern directors like Zhang Yimou (*Raise the Red Lantern*), but they are nonetheless the first Asian directors to cross over the Pacific to Hollywood and successfully ply their trade. And what makes their achievements particularly noteworthy is that the pair of them were individually invited to Hollywood over the heads of the hordes of wannabe directors that throng Beverley Hills, desperately searching for that first big break.

Where Malaysia differs from some other Asian nations, is that it has started to produce artists that are gathering international recognition in their own right, rather than for their point origin

THE ELEVATION of Woo and Lee to the status of international entertainment celebrities looks to have signalled the initial step down the path of cultural exchange between East and West in the entertainment industries —better still, it is unlikely that this exchange will remain restricted to the movies. While it is a change that has long been anticipated and widely debated, the early reactions to both movies have been favourable enough to suggest that both sides have plenty to learn from each other and that Eastern governments' fears of 'cultural invasion' from the West are as exaggerated as Western claims of Eastern hegemony.

Similarly—in the television arena—overseas producers are beginning to pool financial and creative resources in a bid to enter the Asian market without alienating viewers back at home. Japan's state broadcaster, NHK, recently coproduced *Escape From Jupiter* with ABC and Hong Kong's United Media is currently involved in the production of a series entitled *Bonds of Blood*. Both ventures illustrate the East-West connection: *Escape From Jupiter* features a multiracial cast while *Bonds of Blood* is a miniseries based on a book about a famous Chinese family, and is destined for the US prime-time market and is backed by LA-based MTM Productions.

WHILE IT WOULD be wrong to overstate the technical importance of the East-West style of production, it is probably fair to say that the moving-image industries are notably ahead of their audio cousins. What makes this all the more surprising, and therefore worthy of note, is that the Chinese-language, popular-music (Cantopop) science is already a well established and lucrative local industry. Stars like Andy Lau, Sandy Lam, Sally Yeh and Faye Wong are well-established singing stars in their own right. In this they are not unlike Anita Mui, who in traditional music-industry style was performing to packed auditoria across the region as recently as last month on her comeback tour, five years after retiring.

Despite the local appeal of these artists it is hard to find a single Westerner who could name even a couple of popular Eastern artists besides *Dadawa* (a musical equivalent of the aforementioned *Raise the Red Lantern*) and Vanessa Mae, the prepubescent Singaporean violinist who is rehashing old Sky albums. And this failure to break out of their traditional markets is reflected in the lack of significant growth in the studios market in Asia. The majority of audio equipment currently finding a home in Asia is destined for broadcast usage, which is on the whole fairly straightforward in terms of dubbing and voice-over applications.

Take a look at Singapore: while it has seen the opening of postproduction houses like Powerhaus, DigiPost and Asia Broadcast Centre during the past year, nothing since 4MC Asia (which provides MTV Asia's facilities) has enjoyed a significant audio aspect to its specification. Now compare this with neighbouring Malaysia where a very different and more optimistic scenario unfolding: SynchroSound in Kuching is awaiting its relaunch after a significant refitting, down in metropolitan Kuala Lumpur (KL), the MeaSat broadcast centre will feature a state-of-the-art audio complex including facilities for Digital Audio Broadcasting and surround-sound production. Last year also saw the expansion of the AddAudio studio in KL.

Where Malaysia differs in audio production terms from some other Asian nations, is that it has started to produce artists that are gathering international recognition in their own right, rather than for their point origin: 1995 was a breakthrough year for two young Malaysian singers, Ning Balzura (*In Another World*) and Deanna Yusoff. Part of this success has been attributed to the fact that their government operates a 'Made in Malaysia' broadcast policy that encourages movies, music and television programming to be originated from within the country.

Although this kind of success is dismissed by many in the West (and by some locals) as another example of Asian protectionism, it has proved that developing markets are just as capable of producing talent as mature ones, given the right conditions, it is no surprise therefore that some other developing South-East Asian nations—like the Phillipines and Indonesia—are considering introducing similar legislation.

And while it would be difficult to substantiate a similar argument in the sophisticated environs of Hong Kong or Taiwan (which threw up the like of Woo and Lee, respectively), it proves that a government really can work in everyone's interests while seemingly only trying to protect its own. ◉

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Yamaha 02R

We've heard the hype and the eulogies, now it's time to make a serious technical assessment of the 02R.
SAM WISE presents the first results of a bench test which will expose its strengths and weaknesses

AT ITS LAUNCH in June last year, the 02R caused the biggest buzz a small console had caused in a long time. Could Yamaha really produce a 40-input, fully digital desk for hardly more than the cost of an equivalent Soundcraft Spirit? Had they cut corners or had they achieved a major breakthrough? We needed to know.

Certainly, there are no cushioned armrests or scribble strips; no channel meters or group meters (though there is an option for these). Wisely, Yamaha has realised that in trying to create and own a new market, price is a key issue. Paying for elegance becomes a customer choice separated from achieving the required performance, features and functionality.

Each 02R has eight mic-line channels with extra features, eight basic mic-line channels, four stereo line inputs and a few other inputs as well. Four interface slots accommodate modules allowing connection of your chosen multitrack recorders. The mixer supplied for test was fitted with one 8-channel AES-EBU I-O module (two slots required at present) and one T-DIF Tascam DA-88 interface module (one slot required). The remaining slot can be used for another T-DIF or ADAT module, providing 24 digital I-Os combining three formats into the one mixer.

Other options include a future 8-channel analogue I-O (two slots) and a cascade module (one slot) which allows multiple 02Rs to be connected together into one system. Depending on the installed options, up to 32 multitrack channels can be connected to each 02R. In practice, installation of I-O cards requires practically no customer setting up.

My Audio Precision System One digital I-O was plugged up to the AES-EBU multitrack Buses 1 and 2, the analogue I-O from the test set was plugged into Channels 1 and 2 XLR inputs and stereo outputs, with additional DAT playback into AES-EBU Channels 3 and 4. The 02R motorised faders are normally associated with the analogue-input, channel-level controls, with individual rotary encoders in the next row above which are normally used to adjust tape-return gains. FLIP reverses these functions. Other settings can also be brought under motorised fader

APPLICATIONS

YAMAHA HAS BEEN at pains to point out that the 02R is relevant to a wider user-base than a first glance would suggest—and they have a point. The nature and in-built flexibility of this board really does make it suitable for a host of uses in addition to the obvious project-studio sector. Much of this adaptability has to be attributed to the price tag as the console tempts experimentation in areas where more traditionally priced digital desks are prohibited.

This type of all-things-to-all-men stance is actively resented by prospective buyers of large-scale digital consoles, because for the sums of money involved users expect well-honed near-custom configurations for specific applications. When you're talking about a lot less than £10,000 people are quicker to forgive generality in favour of a Swiss-Army-knife approach rather than a dedicated tool for a dedicated function.

The 02R is intended to be broad-based in its appeal and largely succeeds, and with a bit of imagination crosses into more markets than any other desk in recent times.

The fact that the 02R is digital is only part of the story and is most relevant when connecting it to the outside world, but the real selling point is the high density of inputs, fully dynamic automation, onboard effects and dynamics.

There is an argument that says that buying the board simply to access its effects and dynamics processing makes economic sense when held up against traditional alternatives. However,

'traditional' is not a word that can be easily applied to the 02R because, while its operation is familiar, its total package is precedent setting.

The 02R is cheap enough for any, even moderately successful, facility to buy just to play with, and discover the uses to which it can be put.

MUSIC RECORDING: Project and home recording has been earmarked, perhaps most predictably, as the market to embrace the 02R. By the same token the physical presence of the desk is unlikely to impress day-rate-paying clients in a commercial studio and, without external mic preamps, it is restricted in the size of live recording it can handle. Project studios are likely to be working within these limitations and direct interfacing to 8-track digital machines plus the automation and processing make it a class leader—there is currently no other console offering this level of sophistication for this sort of money.

Automation is refreshingly straightforward in principle, and considerably easier to grasp than many low-end VCA fader-only automation packages—a considerable achievement given that it is total and dynamic and that the majority of project users are likely to be automation first-timers.

Assignability is presented well and represents a quantum leap from the ProMix 01. While there are no 'floating' assignable 'strip' controls, the structure of the console means that hard control is never more than a single button press

away. Real bonuses of the arrangement include such things as fader-based access to aux-send levels and that the LCD switches to track the movement of whatever controller is accessed.

The desk also has spin-off applications for live work and the fact that snapshots can be fired via MIDI makes it a very smart keyboard mixer while manual snapshot recalls should endear it to theatre-sound installations or even front-of-house mixing for the adventurous.

POSTPRO AND BROADCAST: For the post sector the 02R meets the demand for a low-cost, grown-up, digital-desk accompaniment to DAWs that extends its use beyond simple tracking to disk, and into the realms of automated (and therefore repeatable) premixing. The added facilities it offers happily coincide with the gradual gravitation of sound editors towards sound design.

One of the earliest placements of the 02R is currently being lined up for SDDS and other format mixing of a major feature film.

Dynamically automated EQ—the desk is particularly strong in this area with four fully parametric bands—is a real boon for sweetening work to picture as is the convenience of having extensive dynamics processing to hand. For bulk-buying broadcasters the 02R is unlikely to replace the hordes of small rudimentary analogue desks but it must be an attractive replacement for any medium-sized desk currently involved in a, at least, part-digital production chain.

control. For example, Aux 1 for all channels can be selected and the faders will drive to the currently set Aux 1 level for each input, providing an instant display of balance, and allowing quick adjustment.

Alternatively, the central LCD panel allows all functions of a particular channel to be seen at once and adjusted by dedicated encoders and switches in the Selected Channel controls section of the desk. There are a large variety of other modes of control and display which are too numerous to include in this review, but become quite helpful as familiarity with the desk grows.

INTEGRATING THE 02R into a hybrid analogue-digital system requires digital synchronisation. The console does not provide sample-rate conversion on the inputs, so either the whole studio must be set up to the same sample rate and locked together, or sample-rate convertors can be added externally where required—at present a costly solution. Any differences in clock rate will result in lost samples, noisy glitches, or complete failure of the system to work.

You can select which of the digital sources is to be the main timing source. These include the digital inputs from practically anything attached to the system, a specific word clock, or the internal clock generator. This is simple to operate but complicates bringing the 02R into an existing studio. If your studio format changes from day-to-day I would recommend that everything you own with digital outputs is locked to a separate external source with a very stable low

noise and jitter clock.

Twenty-bit resolution makes a digital mixer competitive with analogue desks in terms of dynamic range and distortion, so it seems appropriate to examine the other aspects of the 02R's technical performance to see if it measures up to the analogue competition.

THE FIRST 16 CHANNELS are analogue mic-line inputs. The analogue section of this is the minimum essential —a microphone preamp-line driver and insert-return buffer amplifier. Inserts are provided between the mic amp and A–D conversion stage. After that everything happens in the digital domain.

Maximum gain from a 150Ω source at the mic input measures 78.22dB from mic input to stereo output, with input gain, channel fader, and stereo master full up and panned as appropriate. An input signal of -80dBu at maximum gain, routed to an AES-EBU output via the Direct Out provides a digital output level of -30.42dBFS which, when related to the corresponding analogue level gives a gain of about 75dB. To digital bus (group) outputs, the level is about -27dBFS, giving a gain of about 78dB, with a maximum gain difference between outputs of about 0.25dB. This last figure relates to a full left or right pan where the PAN control is calibrated -3dB in the centre position with reference to full left or right pan. Without an analogue I-O card in one of the slots it is not possible to measure the bus output levels in analogue form, but assuming continued consistent calibration of levels throughout the desk, these to would also

Vying for a place in broadcast, post and music tracking—the 02R

have 78dB maximum gain. A level of 0dBu on the analogue output corresponds to -25.68dBFS on the equivalent digital stereo outputs.

Channel input to insert output has a maximum gain of 60dB, with an insert out clipping level of +21dBu. The insert outputs are unbalanced and are designed to feed a relatively high impedance load, so you should not passively split these across a number of different devices without a distribution amplifier in between.

Fig.1 shows the frequency responses of the whole system, from analogue input to stereo output, and from multitrack return to stereo output. These are comfortably flat over most of the audio band, falling away a little at the extremes. Note that the microphone preamp rolls off a little more below 100Hz at maximum gain compared to the minimum gain condition. In Fig.2, mic input to digital mix bus and multitrack return to digital stereo output are shown. It is clear that apart from the small roll-off at the microphone stage, the mixer is virtually totally flat during all mixdown processes.

Microphone input common mode rejection is good, as revealed in Fig.3. In addition, harmonic distortion from the input preamp at minimum gain is low, as shown in Fig.4 where the output is taken from the insertion send. At higher gain levels, the distortion is hidden in the noise, which itself is low.

In summary, this looks a good microphone preamp, forming an excellent entry point to the console.

WHEN THE CONVERTORS are in the chain, the distortion level remains ☞

acceptable (Fig.4) below the published spec of 0.2% maximum, but no longer competing with better analogue mixers at the higher frequencies.
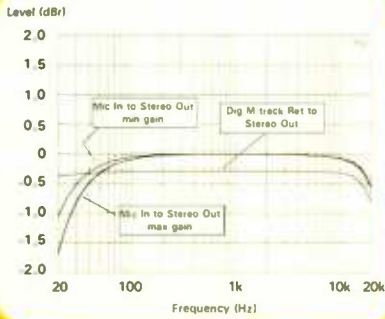
Fig.1: Frequency response during mixdown
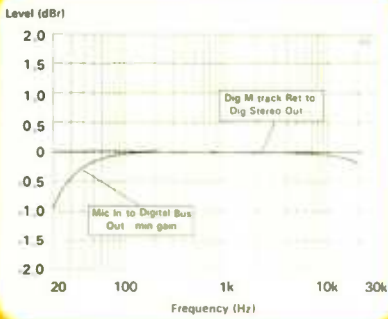
Fig.2: Frequency response of different input and output arrangements
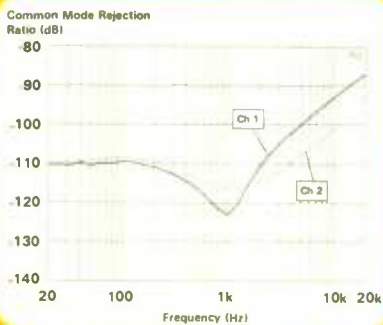
Fig.3: Mic input common mode rejection ratio (dB) versus frequency (Hz). Input level 0dBu, mic gain at minimum. Both channels shown
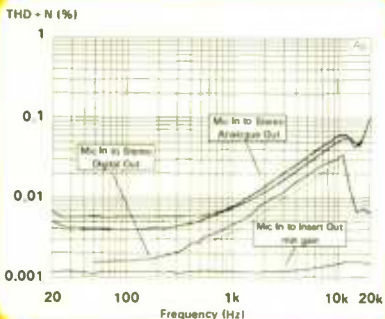
Fig.4: THD+N versus Frequency at various input-output configurations. Mic preamp behaviour is excellent with majority of distortion created in the A-D convertor stage

As will be seen, most of this is caused by the external clock-input circuitry and the D–A convertors used in the analogue stereo output chain, so will not appear in digital mixdown to digital stereo recorders.

The analogue clip-point was detected by observing the insertion output signal. At minimum gain, a level of about 1dBu ±0.5dB at the mic input illuminate the clip light and at +5.5dBu reaches the input-amplifier input clipping-point. Inserting the 20dB pad results in a gain reduction of 19.8dB, giving a clipping point of +25.3dBu—more than adequate for any professional audio application.

At maximum gain, the input-clip lights first illuminate at an input level of -43dBu from a 150Ω source. At -37.9dBu the analogue clip point is reached. This means that the clip light gives about a 5dB warning before clipping. It should also be noted that the clip light is part of the analogue circuit only. Digital clipping is shown, but only on the related LCD meter displays which do not remain visible during most mixing operations. Yamaha informs me that the optional meter bridge can display levels continually from various points in the 32 input or 32 output signal chains.

With the channel fader fully up, the input convertors begin to overload at a channel input level of -48.15dBu. This point can be reached with the channel faders right open without any clip warning, so bear in mind that the overload indication is calibrated for the faders in the '0' position which is 10dB off the top. Stereo output clipping (0.1% distortion level) occurs at 25.74dBu into a 100kΩ load, or 24.68dBu into a 600Ω load. This corresponds to +0.05dBFS on the digital output.

With a 150Ω source connected to the microphone input, the RMS noise level over 22Hz–22kHz is -49.3dBu, with a gain of 78.4dB. The equivalent input noise is therefore about 127.7dB, just below the specified 128dB (though there was no rectifier defined for this specification). This is a good performance.

With the stereo output fader at maximum, two channels routed and all input faders closed, the noise level over a 22Hz–22kHz bandwidth is -88.8dBu RMS, which, when considered with the 25.7dBu clip point, gives a 114.5dB potential dynamic range. Therefore, under no-signal conditions, we are indeed getting a dynamic range to rival a good conventional analogue desk—here is the benefit of Yamaha's 20-bit convertors. However, to achieve this in the studio listening chain will usually require some adjustment to monitor amplifier gains, since most amplifiers will reach clipping with a signal level of only +4dBu to +6dBu under full gain setting and under our conditions need to peak at input levels over +25dBu.

Corresponding noise measurements for those readers in other countries is -78.3dBu weighted and -84.7dBu unweighted CCIR to standards, and -92.5dBu A-weighted with an average-reading rectifier.

Having determined output-convertor dynamic range at about 114dB (nearly 20 bits), the next performance characteristic to consider is the linearity of the convertors, along with any other potentially audible side-effects they may have.

Performance of the overall mixer analogue-to-digital-to-analogue linearity (Fig.5) is better than most 16-bit convertors but shows signs of errors at the -90dB level, which increase down to the theoretical limit of -120dB for a 20-bit convertor. With dither it is possible to maintain linearity to levels below the convertor noise floor. Since we
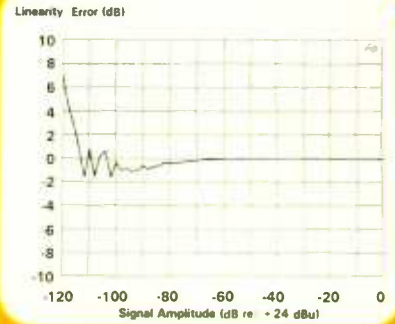
Fig.5: Linearity error in dB. Signal into mic input, signal out of stereo analogue output
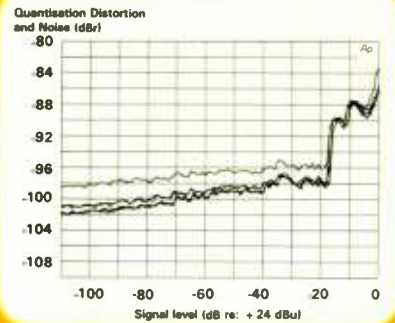
Fig.6: Quantisation distortion and noise versus signal level. Input to insertion return, output from stereo analogue output. Results from 5 channels are shown. The jump on the right of the plot is likely to be from D-A convertors as it is common to all plots. These plots are made with an external clock source and are noisier than when the internal clock is in use

are now looking at levels which are very low, exercising the noise limits of analogue circuitry, there can be other noise effects which corrupt these measurements. But noise is a random phenomenon and my measurements were repeatable, indicating that the convertors are not really linear over their full 20-bit theoretical range.

QUANTISATION DISTORTION for the full analogue-to-analogue path is shown in **Fig.6**. There is a significant increase with rising signal level from about -18dBr which could be due to either the A–D or D–A systems. The fact that this repeated consistently from channel to channel over the five channels tested indicates that it is most likely a D–A issue. This was confirmed by later measurements and is also related to the selected clock reference.

D–A linearity (**Fig.7**) has a smoother characteristic than **Fig.5**. The various plots correspond to various digital signal resolutions from the test set, ranging from 16 to 22 bits, all with TPF dither on the last bit. In theory, the 02R should cope with any of these, and 24 bits as well, dealing with the word length automatically. However, it can be seen that D–A linearity is best with a 16-bit digital input. Fortunately, this is the norm in the majority of potential applications and matches the resolution of affordable digital multitracks.

The quantisation distortion versus amplitude plots (**Fig.8**) show a significant increase with
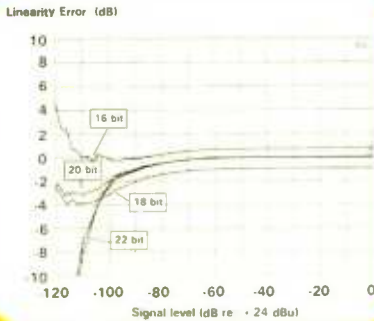
Fig.7: Linearity error of D-A convertors. Inputs to multitrack return 3 and 4, output from stereo analogue outputs. The varying curves result from varying bit widths from the source, all with triangular probability density dither. The errors are similar down to -90dB. Interestingly a 16-bit source gives most linear result down to -110dB
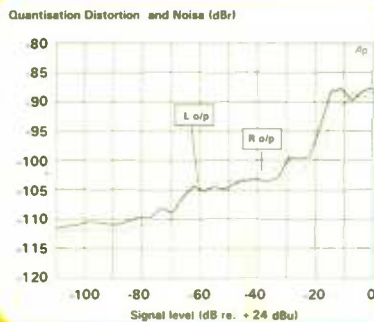
Fig.8: D-A quantisation distortion and noise versus signal level. Input to digital multitrack return, output from stereo analogue outputs. Jumps on the right of the plot confirm the source of the bump shown in Fig.6 as the output convertors. This is another view of the increased noise with increasing programme level generated in the 02R with some external clock sources

level above -18dB, confirming the source of the bumps in **Fig.6** as the D–A convertors on the stereo outputs.

Getting this far in convertor measurement proved troublesome (see sidebar). An investigation with a Prism DScope followed, examining noise and distortion in the presence of tone to pin down the causes of the shortcomings seen in **Figs.5–8**. Most of the problems relate to poor internal digital-clock quality when locked to external clock sources. As this setup is probably the norm in most studios, the deficiency had to be addressed. In fact, when these shortcomings were shown to Yamaha prior to publication of this review, their engineers were asked to investigate and correct the problem immediately. This they did (in part, and arguably adequately well) and sent another mixer which left Japan on Christmas day. The before and after effects are shown, to make the point of the importance of these matters for audio quality. The modified clock circuitry has been the production standard for the 02R since late December.

Fig.9 is an FFT of the output of Bus 3 via the AES-EBU interface. The input is an 11kHz tone fed into Channel 1. **Figs 9–13** relate to this pathway which shows us the detail of what is

going on in the A–D convertor section of the mixer. A perfect result would show only the single line at 11kHz, with narrow skirts. **Fig.9** shows the effect of changing the clock source on apparent noise in the presence of tone on the original mixer. These results display the information from the quantisation distortion plots in **Fig.6** in a different form, making the cause of the distortion clear. Selecting the 02R internal clock produces the lowest noise, using the word-clock input with a DA-88 as the reference source produces the worst apparent noise, while selecting the DA-30 AES output gives a result somewhere in the middle. No really pure external clock-generator was available at the time which could be connected to the external-clock input of the mixer, but aside from not cleaning the clocks coming in from elsewhere, the original 02R external clock circuit generated significant noise of its own.

Fig.9a displays the results for the same sources on the revised mixer design for production. It can be seen that the situation is much improved, the self-noise of the external-clock circuitry has been improved substantially. and with it changes in selected clock source have a much reduced effect. Will these effects be audible? Those of you who remember tape recording back in the 1970s will recall the clearly audible effects of modulation noise when the tape, being less than perfectly smooth, would vibrate against the heads, producing spurious tones in the presence of a signal—no signal on the tape, and no modulation noise was audible. Given the dynamic range of tape in those days when Dolby A was new, this was a more seriously audible effect than we can expect from these convertors. But there are similarities. A lot of the time, these signal-induced side effects will be masked by the signals themselves. At other times, the desired signal will be somewhat pure in sound—a flute perhaps—and beneath it will be a little bit of rasping or breathing noise, perhaps not clearly audible. As a result, you might prefer the sound of one machine over another, not really knowing why. That is the level at which this will be audible. For 90% or even 98% of applications and tracks, nothing will be heard. But Yamaha considered the problem serious enough for an immediate response, removing the risk of audible artefacts of this type from the production 02R, and so should other manufacturers.

To properly appreciate the effect, a few more plots are in order. **Fig.10** shows how the noise varies with channel fader position. Note that the noise drops as fader level drops and the noise spectrum remains the same shape. This we do expect and get in every mixer; turn up the fader, and up comes the noise. But, we do have the option, also, of altering the input gain to improve things where possible. At least when the fader is fixed. the noise will be constant, not varying with signal level.

Fig.11 shows the original mixer; here we alter the signal level into the channel, leaving the mixer gain fixed and observe the effect on convertor noise with the external clock selected. The signal levels are 0dB, -10dB, -20dB and -30dB, with the greatest noise occurring at the 0dB level, where it is mostly inaudible. Some readers may suspect the signal generator of being noisy, but read on. The same test on the modified production 02R is shown in **Fig.11a** There are several observations: first, the 'hairy' effects just adjacent to the wanted tone have been substantially reduced in level, secondly, the spikes at about
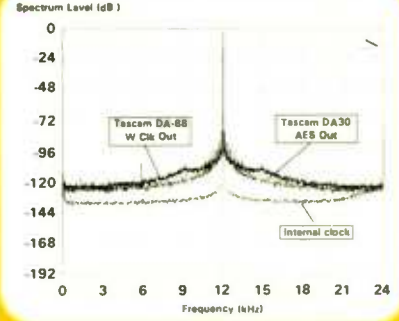
Fig.9: Spectrum of 12kHz probe tone plus noise and distortion. Nothing changes in setup except the selected clock source on original 02R. 0dBu into mic input, minimum gain. Output from bus 3 via AES interface. Note how the noise skirts and general noise level alter with the selected clock source
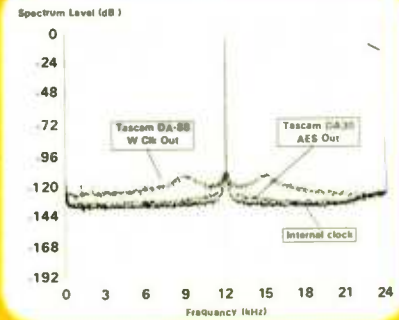
Fig.9a: This is the same setup as in Fig.9, only using the new 02R. The improvement when the DA-30 is selected is large, while the apparently more jittery DA-88 still degrades the convertor performance, though not as much as before

1.2kHz and 6kHz have been reduced by about 12dB, and lastly, the wide-band noise at the highest signal level has also been reduced by about 12dB. Bear in mind that the external-clock source—a Tascam DA-30—has been shown to be a low-jitter source. **Fig.11b** shows the same audio-signal path, but with the word-clock output from a Tascam DA-88 as its reference. The ☞
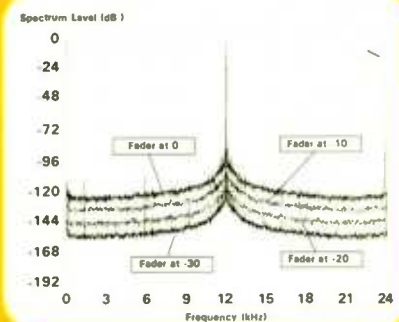
Fig.10: Spectrum of 12kHz probe tone plus noise and distortion. Here we are locked to the external clock and are adjusting the level with the channel fader of the 02R. As expected, the noise and signal now fade down together as the fader level is reduced

signal-induced noise is now significantly worse, though still better than on the original mixer. Fig.12 shows us the same setup as Fig.11 with the internal-clock source is selected. Note that the noise levels are generally lower, and variation with input level is much reduced. This demonstrates

Fig.11: Spectrum of 12kHz probe tone plus noise and distortion. In this plot the original 02R is selected to an external clock—in this case the word clock from the Tascam DA30. Note that as the level from the generator changes, the noise and distortion level from the convertors changes as well. There are no adjustments to mixer settings

Fig.11a: This plot is identical to Fig.11, except it shows the improved performance of the new 02R. Note that the level of noise and width of shoulders about the probe tone are significantly less. The DA-30 has a relatively low noise and jitter clock

Fig.11b: Again, this is identical to Fig.11, except on the new 02R. However, in this case the external clock is generated by a Tascam DA-88 word clock output. Here the new 02R behaves better than before, but still shows a susceptibility to jitter and noise from external clock sources

that the effect is created within the convertor systems, not by the signal generator and makes painfully clear the need for good clock-recovery circuitry. The results here are the same for both versions of the mixer.

Fig.13 shows the equivalent measurements on a Tascam DA-30 when locked to an external clock. Note that the overall noise level is a little higher due to its 16-bit convertors. However, noise changes very little with signal level compared to the 02R. The point to note is that when a low-jitter source is used as an external reference, the results will be best. The production version of the 02R can now achieve excellent results when a good reference clock source is used, so that is what we recommend for high quality studio installations. Purchase a good clock generator, drive everything from it, then forget most of the problems you have seen pointed out in this review.

To give Yamaha due credit, the 02R will track the varispeed range of the DA-88 (±6%) without difficulty, which is the advantage of their wide-range clock-recovery circuit. Although if you are to do this in your studio, there may be other devices which will not remain locked and so a stereo sample-rate convertor may be good to have around.

The Prism-Sound-defined Jitter or J-Test signal was applied to the 02R in External Sync mode —this test is based on the fact that when a digital signal is sent down a cable, jitter occurs due to the cable capacitance and inductance properties acting on the normal data patterns. The test helps to ensure that a digital input will respond properly to digital signals when longer cables are used and as an inexpensive approach to jitter-susceptibility testing. The test is supposed to be run using 100m of typical microphone cable. Fig.14 shows the results of tone with and without the J-Test signal.

With J-Test, there are a number of additional spectral lines visible from 6kHz upward. Ideally, the two plots would be identical. The 02R tolerated 40m of Canford starquad microphone cable before failing to synchronise. I use starquad exclusively for loose cables for analogue audio in live applications due to the uncontrolled proximity of dimmer cables. But starquad gives horrendous problems when used for digital audio, which work best using something like CAT5 data cable. The lesson here is that cables cause jitter by themselves, so selection of appropriate cable for a digital interconnect is important. In addition, some products handle this jitter better than others. Most applications of the 02R will use nearby digital sources, so cable choice and jitter rejection from cable-induced jitter should rarely be an issue.

Most of the inputs on the 02R cannot be activated unless the signals on them are synchronised to the reference clock. However, the control-room monitor chain has an independent clock and will lock to whatever source is selected on the control-room panel. In this case, the 2TR inputs D1, 2 or 3 can operate at their own and possibly different sampling frequency. However, when they are selected as additional stereo inputs, either by routeing through the stereo line inputs (D1 and D2) or directly to the stereo bus (D2 only), they, too, must be locked to the reference clock.

THE 02R'S FADERS feel a little notchy in operation in Automation mode but they are quite acceptable. Since they are digital faders, they will not be entirely smooth in electronic operation, though they should sound smooth to the listener. At the '0' fader point, 10dB from the top of travel, step size is about 0.35dB, increasing to about 0.7dB at the -40dB position and further to 2.5dB at the

CLEAN CLOCKS AND CONVERTOR PERFORMANCE

TESTING THE CONVERTORS in this mixer proved to be one of the biggest measurement challenges I have faced recently. This was mostly due to what at first appeared to be a somewhat random performance ranging from something which resembled a 20-bit convertor, to something less bitty depending on how measurements were made. Further exploration of the original 02R using a Prism DScope highlighted the cause of apparently poor convertor performance seen in earlier measurements. This proved to be related to the choice of clock system driving the mixer, and resulted in improvements to the system by Yamaha.

Switching from the DA-88's word clock to an internal reference caused the noise surrounding my 12kHz probe tone to drop by about 20dB. In much audio equipment, and probably on the 02R, switching to the internal reference usually involves substituting an entirely separate circuit for the external work-clock circuitry. The internal clock offers a crystal oscillator—these are relatively cheap and exhibit fairly low noise and jitter. However, clock-recovery circuits which have both a wide capture-range (to deal with jittery sources and varispeed) and low jitter and noise is

another issue entirely. In fact, the revised Yamaha design also indicates the importance of low self-jitter in the recovery circuit. Their present solution retains a wide capture-range and is capable of excellent performance when locked to a good source—the original mixer was not.

Convertors are only as good as the clock circuits which drive them—noise on your clock will result in noise on your audio if it gets through to the convertor stage. A good clock-recovery circuit may be sufficiently complex and costly to be an excessive overhead in a low-cost unit with only two inputs and two outputs like a domestic DAT recorder. However, the 02R, or any other multichannel device, needs only one good clock-recovery and cleaning circuit to serve its multiple input and output convertors. It appears, therefore, to be worth spending some money or attention on this element to get it right. This is especially important when one is starting to construct a more-or-less all-digital studio where an external-clock reference is the norm. Why doesn't Yamaha, Sony or another large player develop a special-purpose IC which would make excellent clock recovery affordable to all?

Fig.12: Spectrum of 12kHz probe tone plus noise and distortion. This plot shows the same input level changes as fig.11, but using the internal clock. Performance is the same for both original and new versions of the mixer

-70dB position. Above the '0' position, resolution also decreases a little to about 0.45dB/step. The fader range is approximately 112dB, and then off. This is more than ample for the purpose. With the variable step size, it would appear that the 256 steps of an 8-bit fader resolution have been distributed for most beneficial audible effect. The fader is silent in operation under the conditions in the test lab, and this is confirmed in Fig.15 which shows the that the excess noise generated by fader operation is minimal.

IN SUMMARY a number of observations can be made. The absence of dedicated controls for most mixer functions is certainly outweighed by the compact size, cost and features of the console. However, there are controls which particular users like to get their hands on frequently. For me, an example is the need to clear peak-meter readings (overloads) during level setting for a mix. It would be handy to have programmable function keys that would allow me to access this requirement easily. Can we please have nondedicated buttons which can be used for any purpose and assigned very simply? Of course, these should be able to be stored along with other parts of a mixer preset.

With correction of the primary weakness in the clock recovery zone, the 02R is impressive. Were I to buy one, I might reserve some funds for a really good stereo A–D for those special sources, and one really good D–A for monitoring. I would also put some money aside for a 20-bit adaptor for a few channels of my DA-88 so that I could record those 20-bit conversions. Then, I might need a stereo sample rate convertor...

The 02R is a significant step in digital console development for those of us with real-world budgets. And at the asking price, there will be several thousands of us who will be willing live with the few remaining limitations. Ⓢ

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Fig.13: For comparison, the spectrum of 12kHz probe tone plus noise and distortion in a Tascam DA-30 is shown. In this example, we are looking at the analogue output of a Tascam DA-30 with an AES digital input signal. Note that though the basic noise floor is a little higher than the 02R, there is almost no change in noise with input signal level, though the DA-30 is locked to an external source.
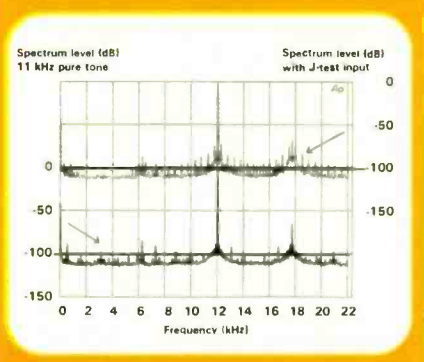
Fig.14: FFT of D–A convertor output. The left scale is for the lower pure 12kHz tone plot, while the right upper scale is for the upper J-Test plot. This J-Test, devised by Prism Audio, has as its main purpose investigating the effect of cable properties and length on jitter generated by the data itself in travelling down the cable. With perfect jitter rejection, the upper and lower plots would be identical

Fig.15: FFT of stereo output signal with and without fader movement. The purpose of the test was to measure excess digital noise created by the fader DSP process. This proved to be very low, creating only just visible skirts about the probe tone

Neumann M149 MICROPHONE

No other new valve microphone could hope to stir as much interest as Neumann's M149 Tube. **DAVE FOISTER** visits Neumann's Berlin HQ to assemble this exclusive report on its development and performance

Old and new, all Neumann mics returned for servicing are assured individual attention

THERE CAN BE no doubt that vintage equipment is a live issue which continues to get liver. The market in revivals, reissues, replicas and straightforward secondhand equipment has never been busier, and brave indeed is the engineer who professes no interest. The intriguing thing is that when you get past the misty eyes and cut through the unquestioning nostalgia there is only a very small hard core of old equipment that is genuinely viable today and which the consensus would accept as such, either in the flesh or in the form of a modern replica. There will always be the enthusiasts prepared to expend the time and money necessary to keep a 30-year-old console or tape machine running, but they are few and far between; rarer still are those who would extol the virtues of old loudspeakers—was the recent Beatles work monitored on 1960s Altec 604s? I doubt it. No, the real hankering after the landmarks of the past tends to focus on just two areas: signal processing and microphones. Hence the kudos attached to Pultec, Fairchild and a few other manufacturers of processors (and their more recent clones) and to the classic early condenser microphones from a small nucleus of companies. This continued reverence for vintage (sometimes veteran would be a better word) microphones is the more remarkable in view of the nature of the objects themselves, inherently fragile and more likely to have been dropped, knocked, thrown into cupboards, generally moved about and otherwise mistreated than almost any other part of the chain. The fact that a 40-year-old microphone can have survived at all is to its credit; for it still to be more highly regarded than its more recent counterparts is a extraordinary achievement.

OLD MICROPHONES, of course certainly are treasured items, with their revivals, however authentic, subject to more scrutiny and criticism than any other reproductions of old equipment. Those revivals will almost always have come from the original manufacturers, which draws attention to another curious microphone phenomenon: almost no other area of our industry can point to such continuity of manufacture as the microphone sector. No-one is better placed to participate in the vintage forum, as no-one else has survived since the days when today's classics were their contemporary everyday equipment. And of those manufacturers, no-one has a better claim to originality than George Neumann.

When I started in this business over 20 years ago, the original valve Neumann U47 was already a highly-prized collector's item—'like gold dust', I remember hearing it described. Of course, by then the company was already nothing short of venerable, holding patents going back to before World War II when the Neumann Bottle was the first practicable commercial condenser microphone. The company was founded in 1928 and is acutely aware of its history, which is detailed on its latest creation, a CD-ROM catalogue. Besides offering the most detailed Neumann product catalogue ever produced, the CD-ROM provides a multimedia scrapbook of the main developments and their place in history, all beautifully produced and representing the perfect bridge between Neumann the revered icon and Neumann the contemporary innovator.

In a sense the new Neumann microphone represents a similar bridge. Neumann is as aware as anybody else of

Prompted partly by outside demand and partly by a desire to apply modern design possibilities to vintage technology, Neumann's designers, led by Head of Microphone Development Stephan Peus, have produced the M149 Tube

the huge market which still exists for the legendary models of 30 and 40 years ago, as well as the current resurgence of interest in microphones in general. Prompted partly by outside demand and partly by a desire to apply modern design possibilities to vintage technology, Neumann's designers, led by Head of Microphone Development Stephan Peus, have produced the M149 Tube, representing the best that classic Neumann design has to offer alongside modern advantages made possible by more recent developments.

THE M149 is a multipattern dual-diaphragm valve condenser microphone, using a capsule assembly descended directly from the original U47 with new valve circuitry and, for the first time with a valve, a transformerless output stage. The
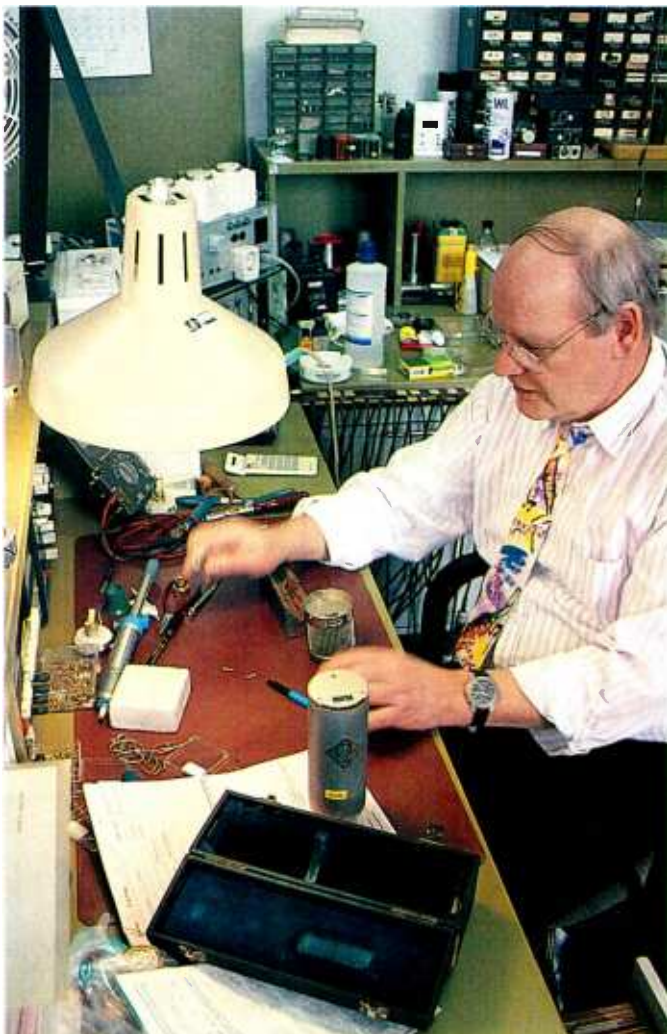
outward design is consciously modelled on the old M49 and its related M50, which also used the forerunner of the capsule found in the M149.

The capsule itself, crucial to the sound and survival of these microphones, has a surprisingly simple history. The original M7 was introduced 50 years ago, and despite its undoubted success it was only ten years before Neumann realised its shortcomings, borne of the technological limitations of its time. The membrane material was not stable over time, leading to increased stiffness and consequent HF lift with age. It was also originally necessary to glue the membrane to the backplate assembly, precluding any fine adjustment of tension across its surface. Both problems were addressed in the K49 capsule, which used a new polyester for the membrane and a screwed ring to clamp the membrane in place. This version of the capsule was used in later U47s and M49/50s, and has remained in production, virtually unchanged, ever since, giving the new model a direct line of descent from its forebears.

THE CAPSULE IS protected by a familiarly shaped wire mesh basket, and it is no illusion that the capsule seems more clearly visible through it than expected. This is not for the sake of appearance, however; measurements showed that the original three-layer mesh basket was applying a subtle acoustic compression to the sound entering the microphone, giving the effect of slightly separating the high and low frequencies. The removal of the interleaving fine mesh eliminated this effect, and in the process revealed the internal construction more clearly. This change is not believed to have a significant effect on the capabilities of the basket as a wind shield, and ,of course, for close vocal work a separate screen will probably be used anyway.

It has been common knowledge for some time that Neumann's original microphones could never be truly replicated, because the valve used in their design, the VF14, is no longer available. Neumann's designers point out, however, that contrary to popular belief the type of valve is far less important than the circuit built around it—in fact they seem slightly impatient with the obsession with knowing what type of valve they have chosen to use. The new microphone uses a miniature valve (an AC701 if you must know) often found in high frequency applications such as Radar; the team have found that valves with this ultrasonic capability tend to have a better noise performance at audio frequencies.

THE M149 HAS, uniquely for a valve microphone, a transformerless output in the usual interests of surviving long cable runs with fewer problems, among other things. The designers were prepared to sacrifice the all-valve nature in order to achieve this, as the output driver is built round an op amp. They clearly anticipate a slightly sniffy reaction

to the presence of an op-amp in a microphone with the M149's aspirations, but defend it vigorously, pointing out that with careful design there is no reason such a circuit should not be able to deliver the required levels of performance. Nonetheless, this is one of several elements conspiring to make the inside of the microphone an interesting sight.

A valve microphone could reasonably be assumed to have most of its substantial body cavity filled with electronics, even if one no longer expected to find the boards made of Paxolin and the wires covered with fabric. In fact the inside of the M149 contains not much more than 'a lot of Berlin air,' as Stephan Peus put it. A small circular PCB sits in the base of the microphone and carries the op-amp surrounded by surface

Neumann suggests that the M149's AC 701 valve could be changed to alter the microphone's character

mount components. Plugged into this board with an edge connector is a sub-board carrying the valve itself, giving rise to the thought that it could be exchanged for another design of amplification stage if required just by changing this plug-in board. According to Neumann, this may indeed be a possibility at some time in the future. This treatment of the valve element of the circuit as a separate module simply adds to the incongruity of seeing it working alongside modern surface mount technology.

At the top of the cavity is another circular board upon which the capsule is mounted, and this also carries the microphone's two control switches. Of course, being a valve design, the M149 has a dedicated outboard power supply, and it is often the case that the controls for 🖘

polar patterns and the like are mounted remotely on such a supply. This is not so here; the power supply is a simple rugged box with a 7-pin input connector for the microphone, an XLR output socket, an ON LED and a connection for the simple mains adaptor which provides primary power. The advantage of this arrangement to my mind is that the power supply can be left on the floor in a corner (as it often is anyway) and forgotten about, as everything that can be adjusted is on the microphone itself. On the subject of the power supply, it is worth mentioning that great care has been taken to ensure the microphone is always operating optimally, with the inclusion of sensing wires in the main microphone cable to maintain the correct operating voltage for the valve.

THE MICROPHONE controls are remarkably comprehensive. This is not a microphone that is going to trade solely on its sound on a take-it-or-leave-it basis; it sets out to be a useful all-round tool in its own right. Neumann decided early on in the design stages that the M149's chief competition was going to be old Neumanns, and that as long as its price remained competitive with them there was no point cutting corners. The idea of an infinitely-variable polar pattern was toyed with briefly but dismissed because it is not accurately reproducible from one session to another, unlike switched settings. The decision arrived at was to provide the expected five polar patterns—omni, subcardioid, cardioid, hypercardioid and figure-of-eight—together with intermediate positions between those settings, giving nine possibilities altogether. This is the first Neumann microphone to feature this arrangement, which while not unique is still fairly rare. Those who know the difference it can make to spill and focus to have such a palette of polar patterns available will appreciate the facility, and those who don't should try it for themselves and perhaps be surprised.

The switch on the back controls the high pass filter cut-off frequency, and again offers a far wider range than normally expected. Generally no more than two settings for LF roll-off are offered; the M149 has seven, with cut-off frequencies from 20Hz to 160Hz in sensible small steps. Again this may sound like excessive subtlety—overkill
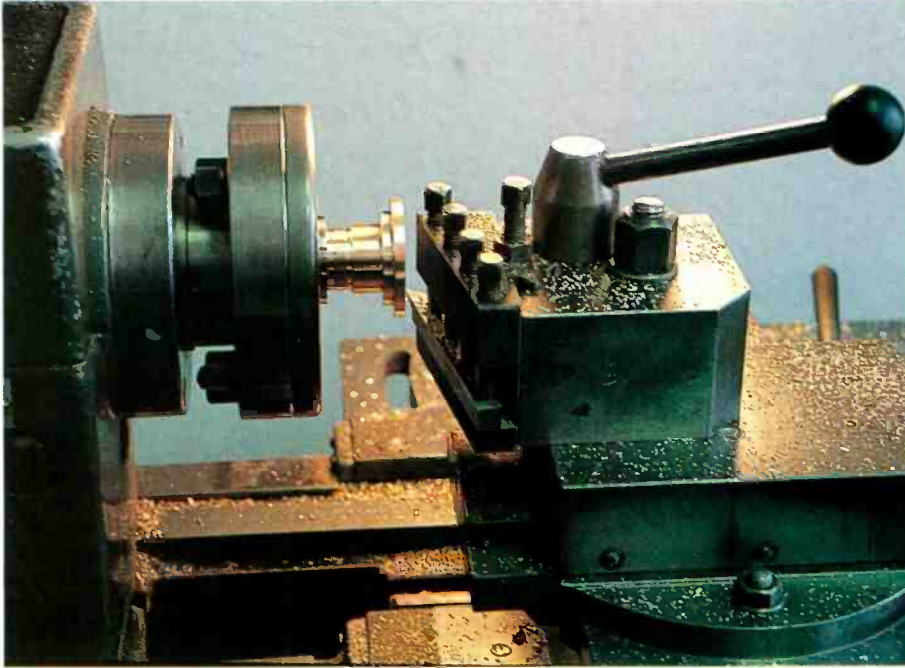
Elements of the M149 are hand-made on lathes at the Berlin premises

even—but in practice the fine tuning it offers is unquestionably very useful. There is no pad switch, but nothing I was able to throw at it gave it any trouble, no matter how loud.

The power supplies are, of course, provided with the microphone as part of a large and comprehensive kit, within which the microphone itself is packed in an elegant and protective wooden box. Also supplied as standard is a cat's-cradle suspension mount, which is apparently the only way of attaching the M149 to a stand. The microphone is held in its mount by means of a threaded locking ring at its base, and in the best tradition of Neumann suspensions the body is held firmly and rigidly at any angle with no tendency to sag or droop—assuming of course that the microphone stand is locked off tightly enough to support the substantial weight, as despite all the air inside the M149 still weighs 730g without the suspension. The suspension not only isolates the microphone from stand-borne shocks, but is large and all-enveloping enough to protect the body from knocks.

The M149 will be seized upon by vintage Neumann aficionados as an ideal vocal microphone, and Neumann sensibly mentions this in the literature as an especially suitable application. Besides this, however, the manual also highlights its suitability for use in classical recording, claiming as it does an unusually low self-noise for such a design. Indeed I have it on good authority that for a long time Decca used the M49 and 50 almost exclusively for classical recording work, often in their distinctive tree configuration of three spaced omnis. Neumann's explanation of the philosophy represented by the

Sound alone would warrant its price and its inclusion as the primary microphone in most applications, but the extra flexibility of the range of polar patterns and LF cut positions means that there is very little it can't tackle

microphone is cleverly ambivalent; it is specifically stated that the M149 is not an attempt to recreate a classic design, and yet the sound the world expects from those old microphones clearly had to be incorporated in the new model. This is not a replica in any sense, and yet for all its new technology, new ideas and low noise Neumann intends it to be seen as having a direct line of descent from the collector's items that might have been the subject of a replica had one been produced. The appeal therefore covers those enthusiasts who treasure the old models and those who might regard any such attempts to recreate the past as not particularly constructive.

WITH ALL OF this in mind, I used the M149 on everything that I thought it might possibly be usable on, and not once did it disappoint me, either in terms of what one expects of the classic Neumann sound or in terms of a dispassionately

viewed new model—although frankly it is hard to be entirely dispassionate about such an imposing microphone. Its vocal sound is, of course, extraordinary, coupling warmth and depth with a free and open top end, all unstrained and musical. The same qualities, together with its attention to detail and undeniably low noise, made it ideal for a huge range of solo instruments, from violin to cello and from trumpet to tenor sax. The sound alone would, in fact, warrant its price and its inclusion as the primary microphone in most applications, but the extra flexibility added by the range of polar patterns and LF cut positions means that there is very little it can't usefully tackle. If I had never heard a vintage Neumann (and to be honest I haven't been in the business long enough to be as familiar with them as some) I would be highly impressed by the M149; those who have worked extensively with M49s, M50s and U47s and know and love them tell me the M149 is indeed a worthy successor. However you look at it, this is a highly desirable microphone, likely to become as much of a legend as its illustrious ancestors. Ⓢ

Amek SYSTEM 9098 RCMA

Combining high-quality mic preamps with a system of remote control is becoming the order of the day for many digitally based recording sessions. **DAVE FOISTER** distances himself from Rupert Neve's preamps

ONE OF THE JOYS of the audio business in recent years has been the 'return' of the microphone. Interest in microphone techniques and the available hardware has revived dramatically, and in the process some of the skills which were in danger of extinction have been brought back from the brink.

This has brought with it a new appreciation of the microphone preamplifier, which has seen its role and importance reappraised. The realisation that the quality of the preamplifier depends upon the entire subsequent recording process is a healthy one, and has brought home the fact that some console preamps are perhaps not up to this responsibility. There has, therefore, been an explosion of outboard preamp designs, from the pragmatic to the esoteric and incorporating every kind of audio technology known to man. These in turn have contributed to the move towards siting preamps locally to microphones, minimising low-level cable runs and allowing the big hop from musician to console to be handled by proper line drivers.

The logical next step is to provide remote control of those preamps, and this is becoming a familiar concept, central to the DG 4D approach, the AGM digital microphone system, SSL's Axiom, the Harrison Series 12 and many others. It is, therefore, hardly surprising to see a system of this type appear from the drawing board of Rupert Neve, long revered for his unique approach to analogue circuit design.

Neve is no stranger to the idea of his console building blocks being repackaged as outboards—whether it's vintage Neve 1066 EQ modules being racked up together or Focusrite console EQ appearing as a stand-alone. Nowadays, of course, he is with Amek, and the flagship 9098 console is rapidly becoming the centre of a whole system of components derived from it. So it is with the RCMA microphone preamps and associated RCU controller, both of which bear the legend System 9098 along with a simple hand-written 'Rupert'.

THE RCMA PREAMPS are an interesting combination of simplicity and versatility, and can in fact be used on their own without any form of remote control. The rack contains four preamps, all of whose expected facilities and more besides are operated from a common set of controls. Each channel has a SELECT button alongside a row of status LEDs, and selecting the channel allows all its functions to be adjusted and displayed on the LCD screen next to the common controls.

Each preamp has individually switchable phantom power, phase reverse, earth lift and mute, and of course

An interesting combination of versatility and simplicity, the RCMA preamps can be used without remote control

adjustable gain. No fine control is thought necessary, so the gain is incremented in 6dB steps over a range of 0–66dB by a continuous rotary encoder or a pair of nudge buttons. Phantom power switching is accompanied by a 3s mute to avoid the usual thumps, and the associated LED flashes during this process as a reminder.

The screen can show vertical bar meters for the output levels of all four channels together with their currently set gains, and highlights the active channel; alternatively it can show an individual preamp's settings in more detail, with provision for naming the channel. The software additionally allows Channels 1 & 2 or 3 & 4 (or both) to be configured as ganged stereo pairs, in which case the chosen pair's level meters and settings are shown on the screen together, and any adjustments to channel settings affects both identically. A headphone socket allows local monitoring of any or all of the microphone signals, and this monitor feed also appears on the rear as a line level output.

Despite the fact that the microphone input XLRs are on the front panel, the back is remarkably full of connectors. Each channel has three output XLRs, which can have three separate sets of functions depending on the chosen mode of operation and the options fitted. Output A is always the primary line output, and in normal use Outputs B and C are fed in ☞

parallel at a slightly reduced level. Each channel can be individually switched to distribution amp mode, however, which gives three identical transformer isolated outputs, the only drawback being a reduction in headroom.

A further option will shortly be available which will bring the unit even more closely into line with current trends by providing digital outputs direct from the preamps. With this installed, the DA mode is lost, and the C output connectors are dedicated to the digital convertors, giving two 2-channel AES-EBU outputs and synchronisation in and out.

A SYSTEM LIKE this really comes into its own when under remote control. The System 9098 RCU can control up to 16 RCMAs, all daisy-chained together using a new proprietary protocol from Amek called Exlink. This data link operates via standard balanced audio cable—its terminations are XLRs—allowing it to be connected easily via existing tie lines. The pulse shape used for data transmission has been specially designed to minimise data crosstalk into adjacent audio lines, and I checked this in a ridiculous worst-case situation by running both it and the preamp outputs in the same single-overall-screen multicore; the data was audible during silences, but at a commendably low level. Using decent multicore or separate tie lines it disappeared completely, and the process of listening for it showed just how low the noise floor of the preamps themselves is.

The right-hand side of the RCU's front panel is identical to that of the preamps themselves, although the available screen displays are slightly different—it is not possible, for example, to view four channels simultaneously. From here all the parameters available on the RCMAs themselves can be adjusted and viewed, and several global system functions set. Besides this, all the individual channels' information is present in the incoming data so that the screen can show levels (including a variable-time peak hold) from the selected preamp. The left hand half of the panel carries the 16 selection buttons, each of which will cycle through the four channels of the associated preamp unit when pressed repeatedly. This gives quick access to up to 64 channels of preamplifier, all down one audio tie line. The monitor feed already referred to on the RCMA can also be daisy-chained together through multiple units back to the RCU and controlled by it, so that any

chosen channel can be remotely selected on to this feed and monitored either on the RCU's headphone socket or on its own line-level output on the back. An example of the thoughtful design is the ability to choose between two systems of numbering multiple preamps, either as consecutively numbered channels all the way up to 64 or as numbered units each with Channels 1–4. This, along with the channel names, makes it easy to keep track of large setups in whichever way suits the job.

An unusual feature for a microphone preamp, although less surprising in view of the nature of this system, is a preset library—16 nameable memories for complete system configurations. These will store stereo groupings, gain, phase, phantom and mute settings and channel names for the whole system of 64 channels, and can be recalled virtually instantly. This degree of recall clearly makes the system ideal as a high-quality front end for the growing number of fully recallable, resettable consoles. In case there are times when the RCU's memory and that in the preamps themselves does not tally, because the RCMAs have been adjusted locally, the RCU can send a complete setup to restore synchronisation—worth doing almost as a matter of course as otherwise strange things can happen.

Exlink is set to become widespread throughout the Amek product range, allowing control of the RCMAs from other devices besides the RCU. A PC software package called MicTalk is already available, providing full control over the preamps from one expansion card. Soon to come is control via the Visual Effects option within Amek's Supertrue console automation system. Visual Effects provides integrated control of a range of MIDI processors, and will also be able to control the preamps, again via an Exlink card in the PC.

IT CAN BE taken as read that despite all the features and controllability, the presence of Rupert Neve's name indicates that this system is not a toy putting technology before sonic excellence. For the record, the preamps are, as one would hope, extremely fine preamps in their own right, surely the equal of any other outboard

preamps available. Neve's overview of the design in the manual gives his thoughts as to the requirements of such a device, which in this case uses quasi class-A circuits, intended to give the benefits of class-A in terms of eliminating crossover distortion with the efficiency savings of class-B. Naturally his Transformer Like Amplifiers are used for the input, and in line with his stated approach to HF performance, which he maintains should extend well beyond the accepted range of human hearing, the response of the units is quoted as 3dB down at 110kHz at 0dB gain and 60kHz at 66dB gain. Together with corresponding 20Hz responses of -0.1dB and -1.2dB and similarly impressive distortion, noise and crosstalk figures, it would be surprising indeed if the preamps did not sound as superb as they do.

The RCMA system is, for a lot of applications, a highly elegant solution. Its audio performance is impeccable, and it succeeds in being extremely helpful in the way it operates, being in many ways more informative about what is going on than a traditional simple mic preamp. The only trouble I had with it was a tendency for the shaft encoder to jump erratically around the range of values, but in view of the overall feel of the system I am happy to put that down as a one-off.

Few systems are available to do what the RCMAs do outside dedicated console configurations, and the opportunity to follow this remote preamp path with whatever other equipment you choose, and with the confidence that comes with the Neve name, makes the 9098 system's appeal very wide indeed. Ⓢ

> It succeeds in being extremely helpful in the way it operates, being more informative about what is going on than a traditional simple mic preamp

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Digidesign PRO TOOLS 3

Now in its third revision, Digidesign's Pro Tools system remains a serious contender in the nonlinear recorder-editor field.

DAVE FOISTER runs through the latest developments and its compatiblity with the Power Mac

Pro Tools screen display from the Macintosh-based variant

IT'S HARD ENOUGH to keep up to date with what's on offer from the computer hardware companies and to incorporate new developments into our systems; how much harder must it be then for a software company to anticipate all this, and deliver the maximum possible performance on the available hardware. Add to this a fiercely competitive market and a hard-bitten customer base who believe that software upgrades equal bug fixes and that they're being used as Beta testers, and the job becomes unenviable.

But software moves onwards and upwards to maintain its position, and so we have a new upgrade of Pro Tools from Digidesign-Avid. Pro Tools comes from a line of Mac packages, each of which has built on the capabilities of the last and helped establish Digidesign as one of the market leaders in the field. Building a network of Development Partners and creating a signal bus within the system for which third parties can produce plug-in processing has made Pro Tools one of the most versatile systems available.

Just as the Macintosh began to emerge as the winner in the competition to become the machine of choice for musical and audio use, Apple's changes of direction came close to upsetting the apple cart. The abandonment of NuBus and the concentration on the Power Macintosh were seen by many at the time as a retrograde step as far as audio users were concerned; it is the job of a company like Digidesign to see further than that and

find advantages in change, and that is what has happened. The most important new feature in v3.2 is an ability to work with the Power Mac audio facilities without additional Digidesign hardware.

This requires the PowerMix extension to the Digital Audio Engine, Digidesign's DAE which is central to the operation of Pro Tools and other software including third party sequencer-recorder packages. DAE is in fact an application in itself, launched automatically by Pro Tools, and used by other products which make use of the Digidesign DSP card for their audio component, such as StudioVision, Logic Audio, Digital Performer and Cubase Audio. PowerMix will allow stereo recording and playback of up to 16 tracks using the Mac's own PPC processor as a playback engine, with the actual capabilities set by the processor speed of the computer. Thus a Power Macintosh CPU running at 100MHz or faster will support the full 16 tracks, 80-100MHz allows 12 tracks, and speeds lower than 80MHz drop the capability to 8 tracks. (Power Macintosh Power Book portables run the processor at too slow a speed for PowerMix.) This is not to rule out the use of a full-blown Digidesign hardware setup with the Power Macs; the standard range of interfaces and cards can be used, but the option is there, given the power available on the Mac, to dispense with it.

OTHER ASPECTS new to Pro Tools 3.2 centre largely on support for other

packages and streamlining of operation from the internal working point of view rather than anything obvious to the user. Other programs now supported are Pro Tools Project, Session 8 Macintosh and Audiomedia II, and there is a direct upgrade path from Pro Tools Project to Pro Tools 3, with its expandability, real time processing power and machine control. Importing of mono and stereo audio files from elsewhere can now be performed using drag-and-drop in the Mac's Finder.

The Bounce plug-in has been streamlined so that it no longer requires the use of its own DSP chip for bouncing operations, which in turn means that more effects processing and mixing functions are available for performing bounces. The SCSI setup has also changed, with larger Pro Tools 3 systems no longer needing valid drives on Disk I-Os B and C, freeing up the I-Os for other uses.

For the operator, a convenient addition is a Get Info window for storing notes about a Pro Tools Session. Otherwise, little has changed on screen, but the program remains as fast and intuitive as ever.

There are a few bug fixes; it seems, for instance, that the VTR control functions inadvertently disappeared from PostView during the last upgrade, and these have now been restored, giving machine control via keyboard shortcuts. Others relate to large systems, fixing their occasional TDM timing and oversized menu problems, and the ADAT interface sample accuracy is now correct, although the new XT firmware is not yet supported.

This upgrade consists largely of detail improvements, with the major exception of the Power Mac PPC compatibility, which offers Pro Tools power to owners of suitable machines with a substantial saving in hardware costs. This can only help Digidesign widen further acceptance of an already successful package. **S**

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Graham Patten D-ESAM 200

The gap between video and audio broadcasting presents a valuable opportunity to the handful of manufacturers prepared to treat both mediums on equal terms. **DAVE FOISTER** finds Graham Patten up to the challenge

Unorthodox to dedicated audio engineers the D-ESAM 200 makes perfect sense in a video studio

YOU'VE GOT TO integrate to accumulate. Equipment for video-editing suites not only has to work well, it has to work with existing equipment and in a way that follows established practices—in fact, it sometimes seems that where sound is concerned, integration and convenience take precedence over quality.

Graham Patten Systems' awareness of this, and its determination to elevate audio to its deserved status go without saying. As developer of the ESAM II protocol for controlling audio mixers from video editors, GPS has identified, and cornered, a market in dedicated mixing control for pictures in a way that accords audio the same importance as video. The D-ESAM range will be familiar to anyone in the video-editing business and is almost completely unknown outside it, as its specialised way of working fits only into that environment and its whole concept is foreign to that of conventional music or broadcast mixing.

D-ESAM stands for Digital Edit Suite Audio Mixer, and the 200 represents a scaling down of the existing models —which can have up to 56 channels—into an affordable version for smaller scale facilities, but without sacrificing functionality or quality. The job of an edit-suite mixer, rather than mixing continuous streams of audio, is making transitions to match the picture cuts, taking audio from a succession of source machines and placing cuts or crossfades between them to match the associated picture transitions. Like everything else in the suite, it must also be driven from the central-editing controller, in the same way that a vision mixer has its cuts, wipes and dissolves programmed remotely from the same system that controls the VTRs. As a consequence, the D-ESAM range looks and functions very much like a vision mixer.

Assignability and Virtual Machines are central to the concept. The mixer has 16 inputs, which can be configured as various combinations of 48kHz AES-EBU and analogue, with optional sample-rate conversion. The inputs are identified as source machines—with one, two or four tracks—and these can then be assigned as virtual machines anywhere on the panel's 12 faders. This assignment can be changed on-the-fly by the editor, removing the apparent physical limitations of the small control panel, and any channel settings associated with a machine stay with the machine rather than the physical channel, so that they will remain unchanged when the machine is recalled to the panel even if it is on a different set of faders.

Four output buses appear in both analogue and digital form, and the vision mixer analogy means that any combination of channels can be routed to the buses as a preset, taking the place of the existing audio at the edit point. The transition can be a straightforward cut or a variable-time crossfade, which is measured like everything else on the D-ESAM in frames not seconds, and can even be different for individual buses.

THE NATURE of the unit means that buttons far outnumber pots and sliders, and button illumination is very helpful in keeping track of what's happening. For instance, the routeing buttons light red when live and green when preset, and have two brightness levels showing which function is currently available for selection. The other main group of buttons doubles as a numeric entry keypad and a means of calling other functions to the display screen for adjustment.

The large, clear LCD screen can show the various aspects of six channels at a time, and is scrolled sideways to see the rest. Soft keys beneath it are invoked by the function selected on the main keys, allowing adjustment by two rotary encoders of such parameters as EQ, panning and delay. The EQ is surprisingly elaborate; its three bands offer combinations of shelf, peak and deep notch characteristics, with a good range of adjustment and a very impressive sound, capable of both gentle sweetening and more extreme correction and effects. EQ can be adjusted simultaneously across groups of channels and settings can be copied elsewhere. The delay function is not an effect, but a means of correcting for processing delays elsewhere in the system, so its maximum delay is only a few frames.

All of this is easily accessed from the front panel, but the point is that it can also be controlled remotely via Sony 9-pin by any editor with a suitable driver. It is curious to note that fader levels, elsewhere the first thing to get automated, are the only function on the D-ESAM not controlled from the editor, giving away once again, the very different nature of the job it does. Fader levels are, however, stored with virtual machine tracks, and adjustment takes the same forms as found on conventional automation, with auto-nulling, trim and manual control.

Despite my relative unfamiliarity with the job the D-ESAM does, I found its concept and operation surprisingly easy to grasp. Having seen video editors struggling with basic, inappropriate mixers while everything else in the room flies by wire, it is clear why a mixer like the D-ESAM is needed, and GPS' implementation leaves little to be desired. ⓢ

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Dennis Charles & Ronnie Wilson - 1st Avenue (Producers - Eternal, MN8, Michelle Gayle, Dana Dawson, Louise) - "The EQ-1 and C-1 units helped us put the power into the EMI album "Power of a Woman" by Eternal."

Alex Marcou - Abbey Road Studios (House recording engineer) - "The VI-1 makes hard sounding digital sound like cosy, rounded analogue - a joy to listen to. The control that the EQs, Pre Amps & Compressors give is excellent."

Chris Porter (Producer - Take That) - "I bought one of the first EQ-1s and I've enjoyed using it immensely. It gives a unique quality to the vocals in particular - Take That's 'Back For Good' is a typical example of the EQ-1 adding depth and presence to a vocal track."

Penny & Giles PP10

In the crowded world of outboard digital-signal processing devices, innovation is a precious commodity.

DAVE FOISTER gets an exclusive look at P&G's PP10—which is sufficiently innovative to make itself unique

THIS IS REAL hold-the-front-page stuff. Everybody knows Penny & Giles, and knows that P&G faders are the best in the business, components that no self-respecting console should be without. We know, too, that the product range goes beyond mere pots and faders and includes the MM16 endless-belt MIDI controller, but even that does not make P&G a company from whom to expect excitement. Solid, dependable, rooted in the finest British engineering traditions perhaps, but not the company most likely to branch out into something so radical, and so different, that it has no real competition.

Yet this is what Penny & Giles has done. Out of nowhere P&G presents us with the PP10, probably the most open-ended, flexible, digital signal processor the business has yet seen. More than any other platform on the market, the PP10 can become anything the software tells it to become, with new developments easily loaded from the front-panel diskette drive. Just as surprising is the fact that the initial range of software covers not the expected reverbs and delays (they are to follow in time), but EQ and dynamic processing.

THE BASIC UNIT comes with a single, digital, I-O card supporting AES-EBU or IEC at up to 24 bits with 32-bit internal processing. What happens between the input and the output is enormously flexible and powerful, but at the same time extremely easy to manage thanks to thoughtfully integrated control across the

software (produced so far) and a helpful screen. The initial suite of software packages is called Pythagoras, and comprises a set of EQ configurations and a set of dynamics processors. A third set, the Studio package, should be available by the time the PP10 is launched, and will contain a mix-and-match selection of EQ, dynamics, panning and mixing modules —highly suggestive of the expandability to come. It should be noted that this mixture is necessary because modules from different packages, or different diskettes, cannot be used together.

The PP10 is physically unassuming, although its styling is eye-catching and different from the run of the mill. Its contoured, beige, 2U-high front panel carries a remarkably small number of chunky convex buttons, a couple of rotary controllers and a large yellowish-green-on-black display screen. It soon becomes apparent that Penny & Giles' expertise in ergonomics and tactile control has been applied meticulously to the user-interface of the PP10, as everything on the screen is so clear and so well related to the controls that the manual, for once, really does seem like an optional extra, despite the machine's huge capabilities. It is also immediately obvious that the smooth reassuring feel that is part of the appeal of P&G faders is present in all the PP10's controls, with a positive action and just the right weight and solidity.

There are almost no labels for the controls; instead each carries an icon which is duplicated on screen where

The P&G audio multiprocessor group showing the PP10, slave PP20 and remote PP20R units

relevant. Thus, if a process has three pages of parameters, the pages icon is shown, indicating that the similarly-marked button will access the other pages. The screen is in two sections, a text-based portion on the left and a more graphic screen on the right, and each can be independently cycled through its available options. Thus, for example, one side can be showing the currently selected EQ band's settings while the other is switched between simple metering, a graphic representation of the EQ curve as set, and a real-time spectrum analyser—no Klark Teknik this but a useful guide nonetheless.

Adjustment is made by selecting a parameter with the up and down keys —there are always several available on any one screen—and turning the big smooth dial alongside, and level adjustment to compensate for the potentially radical effects of the processing is handled by a smaller wheel the other side.

The chosen software can be either installed into the PP10's flash memory, where it will remain resident until deliberately replaced, or run direct from the disk. Either way, once loaded, the full range of processing modules is immediately accessible (although initial loading off the disk takes a while), and can be arranged in the signal path in any order and any combination up to the limits imposed by the processing power. Setting up a signal path is simplicity itself once the basic idea of configuring it from the output backwards is grasped. To set up a chain of processing modules, the output must have a source defined, which can be any available processing block; that block in turn has its source selected, and so on back to the input. Each block can have its whole function altered at any time—the order and nature of the chain is never restricted—and can be individually bypassed. Available blocks in the EQ package include straight, 3-band, low-mid-high EQ, three bands of parametric EQ, combinations of the two, notch filters and some novel ideas of P&G's own. For instance, a Warm module adds a broad sweepable peak and lifts all frequencies below it by an amount set by a GLOW control, while a Broad EQ gives a flat-topped boost of variable width and frequency and with variable slopes either side. These are genuinely new EQ setups which will find a wide range of uses, and like all the blocks can be used in combination with other modules or even further versions of themselves. This means, for instance, that it is possible to ☞

The PP10's front panel

set up a chain of 12, 3-band, stereo parametrics, or several notches and broad boosts or cuts elsewhere—the possibilities are enormous.

More to the point, all the EQ sounds superb. Digital EQ remains a controversial issue, with many regarding it as the Achilles' Heel of digital processing, but the signs are that it is now coming of age, and P&G's algorithms are further evidence of this. With huge ranges of control, including cuts of 40dB available on most settings, it has undeniable power but a musical, controllable sound. Zipper effects are almost nonexistent however fast the wheel is spun, noise is not a problem and subtlety and delicate sweetening are as accessible as brutal correction.

It is a similar story with the dynamics modules, where familiar compressors (with soft or 'firm' knees), gates and expanders are augmented by a kneeless compressor and a 'hyper compressor' which actually has a negative compression ratio above threshold, making loud peaks quieter than the lower-level signals. This bizarre concept, it was suggested, could be used to lift dialogue above an explosive effects background by making the explosions quieter than the speech. Gating and expansion are also included, although these were not yet working the preproduction unit I had.

Again, modules can be stacked up in series, and this palette of processing possibilities gives both serious manipulative power and gentle unobtrusive control as appropriate. Where the screen's EQ display shows curves, the dynamics shows transfer functions—the gain slope as set by the various parameters and changing in real time (almost) as controls are adjusted. The resolution is not great, but sufficient for the job, even showing a gradual softening of the knee. Input and output metering with peak hold is shown on the display, but not gain reduction or any other indication of when the process is actually doing something or how much, which I would have found reassuring.

With both the EQ and dynamics software, complete configurations, with complex routeing and parameters for every processing block, can be saved to diskette for future use.

THE FORTHCOMING Studio software package, with its mixing, panning and routeing modules, gives away the fact that the PP10 can be much more than a straight stereo processor. Slots in the back allow additional digital I-O modules to be added, up to a maximum of eight stereo channels, which the PP10 can treat as 16 individual sources internally. Alternative cards will allow analogue in and out with a choice of 18-bit or 20-bit conversion, but with a sacrifice in the number of channels purely dictated by the number of connectors needed on the limited back-panel space. Also planned for the future are direct interfaces for ADAT, DA-88 and TDM.

With more channels fitted, the PP10 will be able to become a complete all-digital mixer, with EQ, dynamics, multibus routeing and whatever else may be in the pipeline all running on board. The internal memory has room for expansion as demands grow, and even the existing 33MHz TMS32OC31, 32-bit, DSP, core processor will eventually be upgradable to a 66MHz version.

Managing a mixer configured like this would be awkward from the PP10's front panel, so a remote controller will be available, following on from P&G's MM16 MIDI controller and using the same endless belts to drive the system. This in turn allows the use of an expander version of the PP10 itself, with an almost blank front panel but the same processing capabilities. Either version can be controlled via MIDI, and software is under development for control from a PC or a Mac using off-the-shelf MIDI cards.

The PP10 represents an extraordinarily bold move even for a company as well known as Penny & Giles; in fact P&G's very reputation means it has got to get this right. All the signs are, with just weeks to go before final production units appear, that this has been achieved—and quite an achievement it is. The potential of a completely open processing platform which places no restrictions on what it can be programmed to do, coupled with software which goes way beyond the expected functions, superbly high quality, and an amazingly intuitive access to what could so easily have been an intimidating amount of power, must surely make the PP10 a winner. Ⓢ

New technologies

Due to the prominence of the Las Vegas NAB Show, much of this month's new equipment focus is given over to launches scheduled to take place at the show. **DAVE FOISTER** previews the Broadcasters' Product Paradise

NAB will take place at the Las Vegas Convention Centre and the Sands Expo and Convention Centre in Las Vegas, Nevada between 14th–18th April 1996. It is anticipated that there will be in excess of 1,000 exhibitors, 250 seminars and sessions, 11 conference programmes and some 87,000 visitors from all over the world.

Announced in the run-up to the Las Vegas show was the NAB 4th European Radio Operations Seminar, scheduled for 17th–19th November at the Rome Cavalieri Hilton, Italy. Programme details will be announced shortly, and information is available from the US and French offices.

◆ *NAB, US. Tel: +1 202 429 5426.*
◆ *NAB, France. Tel: +33 1 46 92 12 79.*

Merging Technologies Pyramix workstation
Pyramix is a digital-audio workstation system running under Windows 95 and providing a wide range of familiar and novel features. A modular interface allows inputs from analogue and digital sources including ADAT and TDIF, and a communication interface provides machine control, MIDI, SMPTE sync and VITC. Mixer configurations and routeings can be defined by the user, including aux and cue sends, parametric EQ and multiband dynamics processing. OMF is supported, and various data compression techniques can be used, including Merging Technologies' own Lossless Real-time Compression (LRC) providing data-saving ratios of more than 3:1 without compromise.

◆ *Merging Technologies, US.*
  Tel: +1 619 675 9703.

D-Vision video editing software
The first professional, nonlinear, video-editing system for Windows NT will be showcased by D-Vision at NAB. Real-time digital video-effects will be demonstrated, as well as nonlinear editing over a local area network.

◆ *D-Vision, UK. Tel: +44 181 540 0515.*

TimeLine MMR-8
Announced a few months ago, TimeLine's MMR-8 modular multitrack recorder will be making its first appearance at NAB. The disk-based system is designed to replace the current generation of magnetic dubbers used throughout film postproduction, and while initially it will only support the TimeLine Studioframe workstation it is engineered to support OMF. Also on show will be software enhancements for the Studioframe system itself, as well as expansions to the control possibilities for both Micro Lynx and Lynx-2 packages.

◆ *TimeLine, US. Tel: +1 619 727 3300.*
◆ *HHB, UK. Tel: +44 181 962 5000.*

Ciprico RAID introductions
New developments in high performance RAID products will be on show at NAB from Ciprico. The 6900 UltraSCSI array is the first disk-array to support the new 40MB/s UltraSCSI interface, allowing real-time playback of uncompressed video. The Spectra version of the 6900 is compatible with Silicon Graphics Indigo² workstations, and demonstrations will be with Avid-Parallax's Advance and Communicacion Integral's Jaleo. Faster still is the 7000 Fibre Channel array, exploiting Fibre Channel's 100MB/s capability. The 7000 is claimed to have the bandwidth to support multiple uncompressed streams of video or several dozen streams of compressed video.

◆ *Ciprico, UK. Tel: +44 1635 873666.*

HHB media
New from HHB at NAB will be an upgraded version of the already successful DAT cassettes. Improvements include longer tape lengths, a new antistatic lid, an improved tape formulation giving a quoted archive security of 30 years, and reusable shatterproof PP cases. Alongside the new tape will be the promised high-capacity M-O discs for disk-based recorders such as the Genex GX2000, available in 1.3Gb and 2.6Gb versions.

HHB recently expanded and aligned its US operation (formerly Independent Audio) under HB Inc to assist promotion of the company's own products, as well as support of distributed equipment.

◆ *HHB, UK. Tel: +44 181 962 5000.*
◆ *HHB, US. Tel: +1 207 773 2424.*

Pro-Bel XD routers
Pro-Bel has several new products and ranges on show at NAB, including the new XD Series routers. This is a new generation of large scale, digital-video and digital-audio, routeing switchers, with total control compatibility with all current Pro-Bel routers and control systems, and featuring unrivalled field expansion capability and advanced system facilities.

◆ *Pro-Bel, UK. Tel: +44 1734 866123.*

AMS Neve software updates
New software for both Logic and Capricorn consoles will be shown for the first time at NAB. Version 1.7 for Logic 2 allows three-man operation, and can show timings in feet and frames. Other enhancements include automated insert-point switching and enhanced control of dynamics. Capricorn's new features centre largely on mix management, with compression of mix tree branches into a single mix, merging of different sections of mixes, and conforming of mixes to film edit points.

◆ *AMS Neve, UK. Tel: +44 1282 457011.*

Barco broadcast products
Debuting at NAB are two new products from Barco. The first is the new upgraded LUXOR ☞

The AMS Neve Capricorn digital mixing console gets its new software for NAB

optical-fibre link system, which offers a means of linking two production sites up to 30km apart using both analogue and digital signals. It can handle all commonly used video signals, with four audio channels per fibre, and is fully modular allowing it to be configured to users' specific requirements. The other new offering is the DVM 1637 digital viewing-monitor, a general-purpose monitor for all digital systems with the same footprint as the CVM 3637 Series 14-inch monitors.

◆ Barco, UK. Tel: +44 1734 664611.

Apogee FC-8 Format Convertor

Seen for the first time on Apogee's NAB stand will be the FC-8 Digital Format Convertor, catering for transfers between Alesis ADAT and Tascam DA-88 (TDIF) formats. The unit is bidirectional, with 25-pin D-connectors at one end, and optical interfaces at the other, for the two machine types, plus word clock on BNC for synchronising the DA-88. A dedicated unit with no other features, the simple plug-and-play box is expected to sell for an attractively low price.

◆ Apogee Electronics, US.
  Tel: +1 310 915 1000.

Denon DN-045R MD copier

Denon's range of MiniDisc products continues to grow with the introduction of a high-speed copier/replicator, the DN-054R. The self-contained rackmount unit incorporates master and slave drives and can transfer audio, TOC and titling at 3.6 times normal speed. Denon even expect it to improve the quality on transfer, as the system bypasses the ATRAC data compression, defragments the data on the master and lays it down as a linear track on the copy. This defragmentation should reduce access times and the possibility of errors.

◆ Denon, UK. Tel: +44 1753 888 447.

SADiE SASCiA

Alongside the first NAB appearance of Octavia, SADiE will be showing SASCiA, designed as a bridge to a real-time network capable of transferring multiple channels of digital audio between SADiE and Octavia DAWs. It uses ATM, and is initially available in a peer-to-peer networking version with a central server solution expected later in the year as a software upgrade.

◆ Studio Audio & Video, UK.
  Tel: +44 1353 648888.
◆ SADiE, US. Tel: +1 615 327 1140.

apt codec and digital audio cards

apt has two new products for NAB. The apt-Q audio-coding system, shown in software form at AES, will be demonstrated in its hardware implementation, the result of a technology cooperation with AT&T. The new system is designed to deliver exceptionally high quality stereo at low bit rates over a single direct-dial ISDN or Switched 56-channel. Also to be launched is the ADK200, a new range of integrated 16-bit or 20-bit digital-audio PC cards, designed to cater for a wide variety of applications from audio workstations to radio-station automation systems.

◆ apt, UK. Tel: +44 1232 371110.

DAVE stereo upgrade

ASC's DAVE2000 newsroom editing system has

Sony's DNE-50 portable editor

been upgraded to incorporate full stereo recording and playback as well as file exchange with the PC's universal .WAV file standard. This means that any notebook computer equipped for .WAV audio can be used as a portable news-gathering device for later transfer into the DAVE newsroom network.

◆ ASC, UK. Tel: +44 1734 811000.

ART mic pres and FX

ART has announced several new products, with three of particular studio interest. Two are valve microphone preamps, the simpler being the Dual MP, incorporating two channels of ART's Tube MP into one rackmount chassis. The circuit configuration and front-panel jack-input allow variable degrees of valve colour to be added either to the microphone input or to a DI'd instrument. The Pro MPA takes the preamp design a stage further, with a sound ART described as 'similar to the popular classic preamps of early European consoles'. Large front-panel vu meters are complemented by LED Tube Character meters to show the degree to which the valves are affecting the signal. At the other end of the technology spectrum is the Effects Network, a fully programmable multieffects processor, featuring a choice between fully dedicated reverb-delay algorithms and multiprocessing or Dual Processing, applying independent effects to the two channels. A simple front panel suggests ease of programming, and the reverb algorithms are derived from ART's ARM (Acoustic Room Modelling) research.

◆ ART, US. Tel: +1 716 436 2720.

WTN on Asiasat II

With the Asiasat II satellite now successfully launched, Worldwide Television News is now on line with news available to broadcasters in a footprint extending from Iceland to New Zealand. Test transmissions are now complete, and any broadcaster with the relevant decoder can see WTN's daily schedule of transmissions as well as booking ad hoc satellite feeds for particular regional stories.

◆ WTN, UK. Tel: +44 171 410 5200.

IN BRIEF

Precision Devices drivers

New for Musik Messe were neodymium versions of the high-power 21-inch and 18-inch cone transducers from Precision Devices. The neodymium magnets, only available in these large sizes in the last 12 months, give weight savings of 30% and 37% respectively, and their heat generation difficulties are dealt with by direct coupling to an aluminium chassis and a venting system.

◆ Precision Devices, UK.
  Tel: +44 1709 360200.

Heavyworks v2.0

Upgrades to Heavyworks software in v2.0 incorporate a major expansion of supported storage, with up to 350 hours instantly accessible, and a new networking capability. New performance features include a rendered effects package with blue-screen capability, cropping, resizing and graphics import-export, and a new real-time effects package with around 20 different real time wipe patterns as well as ramped dissolves and real-time superimpositions.

◆ Lightworks, UK. Tel: +44 171 636 4000.

DK Audio MSD100

DK's MSD550 metering display is now joined by a budget version, the MSD100. The display combines a phase meter, and audio vector oscilloscope and a level meter to provide the main parameters necessary for monitoring a stereo signal. The self-contained swivel-mounted LCD screen is back lit and will interface to reference levels from +4dBu to +18dBu.

◆ UK: Plasmec Systems.
  Tel: +44 1252 366300.

Digital Genius DACs

Following the release of the ada 20.16 bidirectional 20-bit convertor, Digital Genius has added two new 20-bit D–A convertors to its range. The dac 20/4 and dac 20/2 are 4-channel and 2-channel versions, respectively, of a convertor design which is claimed to have a complete absence of zero crossing errors and glitches, giving a near perfect low-level performance.

◆ Kinetic Systems, UK.
  Tel: +44 181 386 1001.

The latest arrival in the Digital Genius range—the new dac 20/4 and 20/2 D–A convertors

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PRODUCT PREVIEW

Soundelux U95

A new microphone unashamedly modelled on the valve classic has been introduced by Soundelux. The U95 has a 1-inch capsule and all-valve electronics, using both sides of a 6072A dual triode, one for the voltage amplifier and one for the cathode follower output transformer driver. High quality components and direct connections are used throughout for optimum signal quality, and nine polar patterns are available, switched from the remote power supply. Impressive specifications are quoted, including an equivalent noise of 18dB(A), low-frequency response down to 20Hz in cardioid, and a maximum SPL handling of 135dB for 0.5% THD.
◆ Group One, US. Tel: +1 516 249 1399.

Steinberg Cubase Audio XT 3.0

Version 3.0 of Cubase Audio is now available for the Mac, with all the sequencing and scoring facilities of Cubase Score 2.0. New facilities include audio-to-MIDI and MIDI-to-audio match quantising, and even audio-to-audio match quantising; extraction of groove templates from audio; new time stretch and pitch-shift algorithms and recalculation of the tempo master-track based on audio performance, and support for Audio Media II, Sound Tools II, Session 8, Pro Tools and OMS 2.0. A TDM patchbay environment is included, which allows audio routeings to be drawn from input to output and plug-ins to be opened, closed, inserted and deleted while audio is playing. The price has been reduced, with an introductory offer including Timebandit free, and existing users can download the program free from CompuServe at GO MIDICVEN.
◆ Steinberg, Germany. Tel: +49 30 211594.
◆ Steinberg, US. Tel: +1 818 993 4091.
◆ Harman Audio, UK. Tel: +44 181 207 5050.

New Frontier DSP 2010-EX

New Frontier has introduced a novel audio analyser, mounted in a single-unit rackspace and working with any standard black-and-white or colour monitor. From its single front-panel XLR-jack connector and further rear-panel inputs it can perform a variety of analytical functions including real-time spectrum analysis (in both linear and logarithmic modes), long-term SPL readings, dual trace audio bandwidth oscilloscope, envelope mode (typically used for RT60 measurements) and a 16-channel MIDI diagnostic mode. A full-function, audio-signal generator is also incorporated, and further connections include two line outputs, RS232 and parallel ports and MIDI In and Out.
◆ New Frontier Electronics, US. Tel: +1 215 862 9344.

Apogee AD-1000 enhancements

Apogee's AD-1000 ADC system has been given significant new features in a firmware upgrade. The most interesting allows full 20-bit stereo recording on four tracks of ADAT, using a bit-splitting technique called PAQRAT developed by the Rane Corporation. Further, 20-bit signals can be passed through the unit to have Apogee's UV-22 encoding applied as they are converted to 16 bits. Both enhancements are available in a comprehensive upgrade package which also adds ADAT input, and a special edition will also be produced which also incorpo-

IN BRIEF

Westwick AS2

Westwick Installation's AS1 active rack shelf, a rackmount shelf for consumer equipment incorporating an active level-matching interface, has been upgraded to the AS2, with a wider equipment capacity and improved electronic performance. Input and output levels are adjusted by front-panel screwdriver multiturns and can accommodate levels up to +26dBu, with other parameters selectable on internal jumpers.
◆ Westwick Installation, UK. Tel: +44 1526 352950.

Fostex DMT-8 upgrades

Alongside a price reduction for the whole system, Fostex has announced that the DMT-8 hard disk digital multitracker has now reached v2.0 of its software, providing several additional features. These include a program-change function, the ability to slave to MTC via optical ports, Move editing, multiple copy-move pasting, pasting across tracks, higher display resolution and digital I-O connections.
◆ Fostex, US. Tel: +1 310 921 1112.
◆ Fostex, Japan 196. Tel: +81 425 45 611.
◆ SCV Electronics, UK. Tel: +44 171 923 1892.

Waves L1-Ultramaximizer

Originally developed for the Mac, Waves' acclaimed L1-Ultramaximizer is now available in a Windows version for Sonic Foundrys' Sound Forge 3.0. As before, it combines an advanced peak limiter, a level maximiser and a re-quantiser with IDR (Increased Digital Resolution) dithering technology to give control over peaks while maximising levels and optimising quality.
◆ ks Waves, Israel 65165. Tel: +972 3 510 7667.
◆ World Marketing, UK. Tel: +44 1637 877170.

Spendor SA200 now shipping

Spendor launched a new close-field monitor system at AES, and deliveries of the SA200 are now available. As with all Spendor Reference Monitors, systems are matched to within 0.5dB, and the biamped enclosures are magnetically shielded to cope with a wide range of applications
◆ Michael Stevens & Partners, UK. Tel: +44 181 460 7299.

Chevin Black Box

Chevin Research plans to launch a range of digital signal processors this year to complement its amplifiers, and a prototype black box, Chevin's first nonamplifier design, was shown at Music Messe.
◆ Chevin Research, UK. Tel: +44 1943 466060.

QSC Powerlights

QSC's Powerlight range of amplifiers has seen the addition of the PL4.0, delivering 2000 Watts into 2Ω despite a weight of only 30lb.
◆ QSC Audio, US. Tel: +1 714 754 6175.
◆ HW International, UK. Tel: +1 181 808 2222.

A *Work of Art* can easily be spoiled by digitizing...

...so can *Music*

PRODUCT PREVIEW

...rates the new FC-8 format convertor launched at NAB.
◆ *Apogee Electronics, US. Tel: +1 310 915 1000.*

Audix Broadcast expands

Audix's monitoring system for on-air broadcast has been expanded to provide more input sources. The rackmount monitor-meter module allows checking of connected sources via a front- panel loudspeaker, headphones or an external output while the signal appears on a stereo vu or PPM. The original model had 12 stereo inputs, while the new version can handle 24.
◆ *Audix Broadcast, UK. Tel: +44 1799 520022.*

Sony's sons

NAB will see the release of a slew of new Sony equipment. Top of the list is the News Network, a 'complete end-to-end news solution for broadcasters'. News Network is an all-digital setup allowing acquisition, editing, server storage, archiving, broadcast operations and transmission management. Central to the Network is the DNW-series recorder-editor which, with the DNE-1000 newsroom server, allows programme material to be assembled and presented for broadcast as part of a streamlined and highly integrated system.

Sony has incorporated its high-speed Serial Digital Data Interface to allow faster than real time transmission of 4:2:2 Studio Profile video from the field and between stations, and 4 times uploading and downloading for both editors and servers.
◆ *Sony Broadcast & Professional, Europe. Tel: +44 1256 55011.*
◆ *Sony Corporation, US. Tel: 201 930 1000.*

Korg SoundLink

Las Vegas will witness Korg's 'integrated system' approach to its SoundLink nonlinear recorder-editor. Called SoundLink DRS—Digital Recording Systems—the system includes the 168RC recording console; the DRS 1212 PCI multichannel audio I-O; 880 A–D and 880 D–A convertors. The 168RC is a 16:8:2 console which features 18-bit conversion, two ADAT, eight analogue inputs and full console automation. The 1212 features 12 audio ins and 12 outs presented as two analogue I-Os, one SPDIF I-O and eight channels of of ADAT I-O on optical. The 1212 also incorporates OSC DECK II software. The 880 A–D and D–A convertors are 8-channel units available to expand the 168 and 1212 via ADAT optical cables.
◆ *Korg, US. Tel: +1 516 333 9100.*

Telex KP-12

Telex is to use NAB to debut its new KP-12 key-panel series for use with the RTS CS9000. Offering a fully-programmable comms panel, the KP-12 handles high quality audio comms in a single-rack package and comes in both push button and lever key variants, each with 12 assignable talk-listen keys.
◆ *Telex, US. Tel: +1 612 884 4051.*

Audio Ease BarbaBatch

Meeting the growing demand for CD-ROM sound-file manipulation is BarbaBatch from Dutch software group Audio Ease. The

The ARM monitoring system from Audix Broadcast

Macintosh package converts to, and from, a number of formats including Sound Designer I and II, AIFF, WAV, VOC, NeXT-SUN, Amiga IFF-8SXV and AVR, and provides sample-rate conversion, normalisation to a user-definable ceiling, bit-rate conversion, stereo-mono conversion and optional dithering. All parameters on a whole batch of differing files, from all ten formats if required, can be set up before running the conversion as a background process, and the speed of the process is increased when run on a PowerPC or using a DigiDesign audio card on a 680x0 Macintosh.
◆ *Audio Ease, Netherlands. Tel: +31 30 2433606.*

Furman headphone amplifiers

A new headphone amplification and distribution system has been introduced by Furman, comprising the HA-6A amplifier and HR-2 remote box. The amplifier connects to any line-level source and incorporates its own power amplifier to feed six front-panel, stereo-headphone jacks, each with its own volume control. Its 20W per channel amplifier can also drive two pairs of close-field monitors. Linking to the HA-6A is the HR-2 Headphone Remote Box, which attaches to a microphone stand and provides outputs for two sets of headphones with independent volume controls. Multiple HR-2s can be daisy-chained to the HA-6A.
◆ *Furman, US. Tel: +1 415 927 1225.*

C Audio ST 1000

C Audio's ST Series of power amps now includes one of the largest professional amplifiers commercially available, in the form of the ST 1000. Designed for the installation and small PA markets, the amplifier is rated at 1000W per channel into 4Ω and 680 into 8Ω (2000W bridged mono). Forced cooling allows multiple units to be racked together, and C Audio describe the circuit topologies as audiophile grade in order to ensure sonic quality and stability into long cable runs and difficult loads. Features include Speakon outputs, built-in protection for both amplifier and speakers, and a ground-lift switch.
◆ *C Audio, UK. Tel: +44 1223 211333.*

Tapeless Audio Directory

The 5th edition of the the *Tapeless Audio Directory* is set to see publication at the forthcoming NAB show. Building on the reputation established in its prior editions, the new Directory covers both software-based and turnkey systems and details more than 300 nonlinear systems complete with contact details.

All aspects of nonlinear usage are covered, including broadcast, (video and film) post, audio tracking and speech along with generic developments such as networking.
◆ *Sypha, UK. Tel: +44 181 761 1042.*

www.americanradiohistory.com

Compact disc: the next generation

After a prolonged battle between some of the largest names in consumer electronics, the future of the CD appears to have been settled. In fact, the warring parties may have simply agreed to not to disagree writes **MARTIN POLON**

AFTER A PROTRACTED WAR of words and thoughts between Hitachi, JVC, Matsushita, Mitsubishi Electric, Pioneer, Time-Warner and Toshiba on one side and Philips and Sony on the other, the war over the future of the CD format was deemed to have been fought and won. But after an announcement made last September, all the various warring parties have actually agreed to, is not to disagree. At least, not in public. As to the adoption of a hard-and-fast single standard agreed by all the players, to move forward from the established audio CD (CD-DA), video laser disc (which shares many CD-DA standards) and CD-ROM, it appears that a true and lasting peace in Bosnia is more likely.

What the original CD partnership of Sony and Philips initially attempted to achieve was to upgrade the storage capacity and technical limits of CD. Upgrading the Red Book CD-DA, the Green Book CD-i and the Yellow Book CD-ROM to provide full record and playback capabilities and an implementation of a video capacity was the brief. Focusing on a single-sided disc with limited low bit-rate reduction coding, the Sony-Philips MMCD (Multi-Media CD) picked up momentum, albeit it with a capacity that would not accommodate all the motion-picture features so far released. Toshiba and Time-Warner, meanwhile, sought a carrier that would allow playback of any motion picture available and championed a new CD format known as SD (Super Disc). SD used a double-sided disc that promised extended playing time but no recording option. Excess storage capacity provided by the new disc was regarded as a bonus for the provision of motion pictures in a digital format using MPEG2 compression.

As the consumer electronics industry giants, their audio and film software affiliates postured, the players in the motion-picture-studio business took positions on either side of the future-CD-controversy fence. The 'compromise' was agreed after months of infighting had left the existence of yet another two competing standards exposed to a wary electronic consumer public, potentially damaging the viability of any future product.

THE WARRING CAMPS have agreed on some issues but have not reconciled several of their major differences. The first of these is the name for the new disc format. Many industry analysts and journalists have settled on the appellation Super CD; others

have chosen to use the term HDCD (High Density Compact Disc) although that term is also used for a technical-quality upgrade of the original audio compact disc. Also being used is the DVD (Digital Versatile Disc) name for movies on a CD.

At the present time, the status quo of the enhanced CD formats seems to be as follows (educated guesses are as good as it gets in the current climate). Toshiba used the Winter Consumer Electronics show in Las Vegas during January to unveil two or three DVD players for CD movies. These players will be available supposedly in the Fall of 1996 and retail in the $600–$750 price range. The DVD format is a version of the double-sided-disc format originally proposed by Toshiba/Time-Warner, but supposedly using the Sony-Philips EFM+ modulation scheme. These double-sided discs are made by bonding together two 120mm diameter 'prepressed' discs 0.6mm thick. They are bonded with special adhesives pioneered by 3M for double-sided laser discs and protected by a special coating with similar historical antecedents. DVD disc will have a relative capacity of about 5Gb per side yielding total capacity in the 9Gb–10Gb range.

DVD players are backwards compatible in as far as being able to play Red Book audio CDs and potentially any other format installed by the maker of the DVD player in question such as CD-i or CD-ROM. The enhanced discs in any of their various Super Disc formats cannot be played on current CD machines. Timing of the availability of DVD software is one of the many questions that depends upon which manufacturer you believe. Philips has indicated that the current lack of final agreement on the format(s) could slow DVD from a planned mid-1996 introduction. There is speculation that DVD will be available in both hardware and software by the end of 1996, but some sources inside the Super Disc consortium are hedging their bets by indicating the possibility of an early 1997 DVD launch.

Apparently, there will be two single-sided formats utilising a single-layer disc capable of carrying about 5Gb and a dual-layer disc which can store between 9Gb and 10Gb. The single-sided, single-layer system bears more than a casual resemblance to the original Sony-Philips proposal while the single-sided, dual-layer system appears to use Matsushita's technology. The layers are

bonded at the so-called pit sides to seal the layers inside the disc. Players will use a laser that can change it's relative focal length from a single position to read both layers on the dual layer disc. A normal CD-style ink label can be applied to the 'other' side of the single-sided discs. It is an important assumption that the players for the single-sided system can read either single-layer or dual-layer Super CDs.

A third format exists in the expected introduction of a recording option in 1998 or 1999. After the furore that erupted when the original agreement was announced, the companies participating recognised the need to try and engineer a 'recording retrofit' to the entire project. Ostensibly, the DVD-E platters will store about 2.5Gb on each of the double-sided disc's recording surfaces. The recording mode is expected to use 'phase-change' technology but careful engineering will be required to insure compatibility between conventional prerecorded DVD discs and the recordable DVD-E discs.

The given rationale for the development of the new format(s) encompasses a number of issues. Competition for the burgeoning recording Red Book CD marketplace (CD-DA, CD-ROM) is one of them. Another is the need for replacement of analogue VCRs, especially for home-theatre applications due to the extremely variable quality of both audio and video reproduction from the home hardware and prerecorded tapes. Indeed, it is the impact on home theatre and the future potential of these several new CD technologies to provide an ideal format for home viewing of theatrical features—that is driving the consortium more than any other vehicle for delivery (audio and-or computer data).

CD audio is, as we have seen, backwardly compatible for existing CD audio software only. Looking forwards, the compromise technology could provide at least three and as many as six conventional compact discs worth of music, if that is the choice taken, on one extended disk. This could see the record companies' marketing *Best of...* ploy evolving into an *All of...* ploy.

Precisely how the additional storage capacity is finally used remains to be seen. Many in the consortium see it as an opportunity to adjust upwards of the current 16-bit, 44.1kHz standard to a 24-bit 96kHz implementation, for example technically speaking, some at the record-label level postulate that it could provide the much talked-up, but never implemented, potential to deliver the consumer a premixdown 6-track or 8-track disc with all of the elements of a conventional one-hour CD available for custom mixing by the listener. The fly in the ointment here is a big one: everything on 'Super' CDs will be companded, compressed and low-bit-rate coded. It is difficult to ☞

comprehend why audiophiles would prefer this to the existing 16-bit to 20-bit CD system now in use—which provides the entire musical performance without the loss of the slightest nuance.

Market forces strongly suggest that the present penetration of CD-ROM technology will prevent a new CD format from making significant inroads into computer systems. The best that can be hoped for is to offer an alternative drive system. But many developers of consumer and professional software as well as computer systems manufacturers are eyeing the new disc systems for their multiple-gigabyte capacity—which can be as much as 15 times that of the current CD-ROM—for software distribution.

THE SUPER CD faces a number of issues and competitive products.

The very different technology from the similarly named audio HDCD (High Definition Compatible Digital) higher bit-rate enhanced fidelity CD system.

The lack of recording capability on the new disc, at least until initial standards are actually set and a recording scheme can be worked out that will satisfy all of the partners and their record label affiliates. What has been lost is the ability to record, in an environment (the home) that values recording above all. Even the most likely candidates for the successful adoption of the new CD video system—home movie aficionados—will not easily accept the loss of the ability to record that came with every VHS VCR. Then there is the dream of the 'one machine'; the ultimate goal of the consumer electronics industry for ten years or more that has been dashed, if not forever, then at least for several years. No more 'one machine' to record audio, video and computer data in the home.

The decision not to provide a recording capability as part of the new system, strikes many in and out of the record label and motion picture studio communities as a continuation of the battle being waged against 'illicit' audio copying. Every time a new audio and-or video recording technology appears with the appellation 'digital', the film and record communities are concerned over the copying implications. Further, the initial Super CD press release led its readers to assume that the decision to not offer recording capability was made because of the technical incompatibilities of merging the two incompatible systems. The smart money is still betting on the politics of illicit copying.

The cost of a player, originally estimated at around $400 US, will either rise or else will not be subject to initial savings due to the mechanical complexity of tracking the double-layer or double-sided disk.

The process of dual-layer, single-sided along with single-layer, single-sided pressing or DVD double-sided pressing, plus the attendant adhesive coalition and coating of the two layers makes the disc manufacturing process more complicated and expensive than for conventional CDs.

Playing a double-sided disc will require either two lasers and associated tracking systems (one for each side) for continuous playback or else utilise some mechanical arrangement to either turn the disk or move the laser and-or a depth adjusting technology for a single laser to read a double-layer or single layer, single-sided disc.

The emergence of the recording CD as a viable tool for advanced consumer users as well as for professional audio facilities, could draw off any consumer or professional interest in Super CD audio. Recording CD does not use any form of low-bit-rate coding, data compression or software codecs found inherently in the Super CD.

Super CD provision for theatrical motion pictures will have to be directly competitive with Digital Satellite Services (DSS) such as the Direct TV service. Since home-theatre systems have revolutionised in-home movie distribution with much higher quality video and audio than the best of the domestic cable systems. They are the hottest consumer electronic products in America today.

Considering that almost everyone involved in the Super CD or enhanced CD camp finds in-home delivery of motion pictures to be the pivotal role for the new product, the forthcoming digital VHS VCR does not bode well for the new disc. The likelihood of the public staying with a known quantity (the VCR) albeit it in a digital guise, is high. Here again we find the various divisions of the large Japanese consumer-electronic companies prepared to slug it out, each with their own divisions committed to one system or the other, competing for the same market and market share.

The new DVD 'movie disc' will require the motion picture rental community to create a complete second inventory of the new digital video discs to match their existing video-tape stock. In addition, the new systemology will require all holders of motion-picture-video libraries to rerelease all titles currently extant on video tape to the new DVD format and practically speaking obviate the elimination of the existing laser-disc-player user-base.

The bottom line here suggests a scene for a modern-day Laurel & Hardy comedy. Oliver Hardy in the guise of a recording studio or pressing-plant operator turns to Stan Laurel as the record-label or consumer-electronic exec and says, 'this is another fine mess you've gotten us into'. And indeed it is. Everyone to be touched by the new CD formats will have to pay the piper—big time—to continue to be able to play in the audio big league if the new formats succeed.

Recording studios will be faced with almost impossible levels of signal quality to maintain when producing masters for the 24-bit, 96kHz standard—if adopted. If multitrack mixdowns are going to be put on the discs, then the whole record-production cycle that has evolved will have to be changed and that will cost money. So will the upgrading of CD pressing plants to accommodate to the new standards. For many of us, the CD is our bread and butter—our past, present and hopefully our future. Let us hope and pray that all concerned fashion a final compromise as well designed and implemented as the one that brought us CD in the first place. ⑤

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Off the bea

Following *Studio Sound*'s recent account of Tom Hidley's controversial 10Hz room at Masterfonics, **DAN DALEY** explores its operational agenda

LATE LAST YEAR, several hundred party-goers swirled through the cavernous innards of Nashville recording, mixing and mastering studio Masterfonics' recently opened The Tracking Room. Nashville is no stranger to large gatherings, but this one was different in that the studio whose grand opening was being celebrated was large enough to hold all of them.

Nashville's traditional and primary base of country music productions now has a larger-than-ever array of area studios to choose from—over 20 new rooms have opened up there in the last two years —many of them smaller facilities aimed at overdubbing and mixing. And a sizable chunk of those new studios is located in personal environments, many of which are owned by artists and engineers themselves (not an unusual phenomenon for Nashville). Coupled with an historic dearth of large, modern tracking spaces, the need for a big studio in Nashville was readily apparent. However, these developments come at time when studio rates have been stagnant for nearly a decade, and the proliferation of new studios limits much in the way of change in that situation. Thus, when Masterfonics owner Glenn Meadows decided to spend $3m on The Tracking Room, a

5,500ft² facility which incorporates a large SSL 9000j console and designer Tom Hidley's first US 10Hz room design (see *Studio Sound*, December 1995), he was walking the wire between the need to expand in the face of a growing pool of studios on one hand, and the need to build a studio that would attract business from outside the Nashville on the other.

At least, so it would seem. While he acknowledges the current dynamics of Nashville studio business, Meadows believes that both Masterfonics' main facility and The Tracking Room are on a slightly higher, more unaffected plane of operation. 'I could look just at Nashville, but right now I'm looking at the planet,' he laughs.

Levity aside, Meadows means that both he and the rest of Nashville—while they could never overlook the currency of country music as the primary revenue base for the audio community there—need to project themselves into the larger world market of recording. And The Tracking Room can justly be regarded as the city's first step in that direction.

'Nashville is incredibly diverse in terms of studio technology right now, from ADATs and DA-88s to producers who wouldn't touch that stuff with a ten-foot pole,' Meadows observes.

'There are people making records on Fairlight and RADAR systems and people using 1-inch 16-track machines. But our point of view has always been to cater to the upper echelon of artists and producers. The rate wars affect the middle level of studios more than the upper ones, and that's the case in Nashville, as well.'

Meadows claims to be within 5% of his card rates in recent years, an accomplishment in light of the fact that rates themselves are mired at mid-1980s levels. The main Masterfonics facility, on Music Square East, has several recording, mixing and mastering rooms, including the Hidley-designed Mix Room, a 20Hz room with an SSL 4064E desk with G-Series computer and Nashville's only AT&T DISQ digital mixing core system. Studio Six is the main facility's primary tracking room, another Hidley design going down to 24Hz and with an SSL 4048E desk with G-Series computer. Otari DTR-900, 32-track, Pro-Digi format, digital, multitrack decks are standard at Masterfonics. Thirty-two-track working is also pretty much a Nashville standards, although it's slowly giving way to 48-track digital, of which Studer D-827 and Sony 3348 decks are available as options on the rate card. Analogue decks are the Otari MTR-100A with Dolby A and SR NR.

ten track

to take, now that it's built, it to educate the local community and then the larger postproduction and music industry that Nashville has a place that all these things can be done in now.'

Meadows decided, after looking over the work of other designers, to stick with Hidley for The Tracking Room's design.

'People don't realise the calibre of work that Tom can do because most of his studios are in Europe,' he comments.

The Tracking Room looks good—well-lit and trimmed in light-coloured maple wood —but its design did not spring full-blown. Meadows started with his original concept-ualisation in May 1994 and did a relatively rare thing in the studio business: market research. He showed it to various producers and engineers around Nashville and modified it as their input came back. For instance, the original design had four large isolation rooms one of which was a large drum room—that were visually connected by video cameras. Nashville producers and engineers noted that the famous local musician pool works best and most efficiently when they can see and easily communicate with each other, a group of people for whom a nod or a wink during a session could send a track in a completely different direction and make the difference between a hit record or not. The engineers also said that they would rather put the drums on the main studio floor. The plan was adapted to break the large drum iso booth into two smaller—but still quite capacious—isos, enhancing acoustical separation while giving the musicians direct eye contact with each other. Furthermore, an amplifier closet was built into the rear of one iso booth to provide another level of separation. 'Tom's original design would have been the preferred layout in Europe, but this is what works better in Nashville,' says Meadows.

The studio is Hidley's first 10Hz room and takes his patented concept of Infrasonics —the propagation of sounds below the threshold of hearing—to a new plateau by incorporating the idea into both the control room and the studio. ('The Hidley-Infra-sound era', *Studio Sound*, December 1995) The room has drawn rave comments from several engineers and producers, including Tony Brown (Wynonna Judd, Vince Gill, Reba McEntire), who said, 'This is the room we wished we'd had in Nashville for years. We haven't seen anything like it since the old RCA Studios years ago. The isolation booths will really lend themselves to experimentation in making country records.'

Steve Tillisch did the first mix session at The Tracking Room, mixing a new indepen-dent label country artist Brad Hawkins with Producer Steve Diamond. Tillisch was complimentary of both the room and the console, noting that the control-room

environment was even and articulate throughout the full frequency spectrum and volume levels.

'You don't get the usual bass build-ups at the sides of the room,' he observes. 'The listening area is huge and goes back a good ten feet behind the console. And the Mark Levinson [Cello] amplifiers really contribute to that. In fact, it's pretty easy to get the volume up to a very high level; you don't realise how loud it is because they're so smooth.' Tillisch adds that he mixed on the main Hidley-Kinoshita monitors. 'The only time I usually mix on large speakers is in Hidley rooms.' His comments on the SSL 9000j are equally complimentary, noting that the new automation computer allows cuts and rides to occur as separate operations. 'You can edit the cuts any time without affecting the level of the fader rides,' he says.

One overarching design concern was the studio's proximity to Interstate 40—a 6-lane highway about 15m away from the studio's east wall—whose low-frequency vibrations caused the noise floor to rise significantly in the lower range of the spectrum in predesign site-evaluation. Positioning the entire facility on damping springs was estimated at $250,000, nearly 10% of the overall budget. The solution was found in using Kinetics Engineering fibreglass insulators under the studio and control room floors and studio walls as isolators, and by designing the facility structure so that its own resonant frequency was 10Hz. An additional step was to cut the studio base slabs into sections that resonate above 20Hz, which would cut off any additional infrasonic energy—which was present on the original uncut base slab —below that threshold down as low as four cycles. This combination of techniques and materials brought the noise floor down to between 14.8dBa and 15.5dBa measured from 4Hz to 40kHz.

The control room's 10Hz design incorporates pit trapping of the type that Hidley developed for South Africa's BOP Studios, which has 12Hz control rooms. The floor trapping entrance is under large screen-covered floor ports in front of the console underneath the 16-inch control room slab, which itself is decoupled from the earth using nine 3.5Hz German-made Gerb Engineering structural springs. Similar trapping is built into the ceiling. 'This is what gives the control room the depth and three-dimensionality it has,' says Meadows.

Tom Hidley sums up the control room's asymmetrical-loading design thus: 'For 50 or 60 years, up to Infrasonics in 1991, everyone has a solid floor in their control room covering the surface area that's exposed after the acoustic treatment is brought in. You have a monitor that's omnidirectional at the extreme low end. Not controlled in terms of directionality ☞

GOOD AS IT IS, what Nashville needed was a bold stroke, and The Tracking Room is Meadows' response. It addresses the city's historical lack of an acoustically state-of-the-art large recording environment that offers a high plane of technology and addresses the privacy issues that many artists and producers demand of a facility. Hidley's design and the choice of the SSL 9000j deal with the first two; placing the new room in a former warehouse/rehearsal-studio space three blocks away from the main Masterfonics facility takes care of the privacy issue, as does the minimalist staff: Cindy Sutton is the on-site Studio Coordinator and Judy Reeves, former tech at New York's Right Track Recording, is Chief Technician.

But more to the point, The Tracking Room could also function as a lightning rod for Nashville in general, designed as it is for post scoring as well as music tracking.

'There's a insignificant amount of scoring or sound-to-picture work going on in Nashville,' Meadows says. 'There was simply no place to do it. This is a combined-use facility that can do both music and scoring well. For music, it has the great acoustics; for post, it has the space that can accommodate an 85-piece orchestra simultaneously with a five-piece rhythm section. What it's going

between 20Hz to 50Hz. We have dealt with it in the ceiling and sidewalls for years. We do nothing about the floor. What goes to the ceiling and sidewalls is dealt with by several means, depending upon the designer, and depending upon the design philosophy. But none of them have done anything about the frequencies. And when it hits the floor, it hits a non-symmetrical condition to what it hits in the ceiling and the walls. So is the sound from the woofer symmetrical? No, and if it's not phase-symmetrical it's not pure. You must deal with floor as you do the other surfaces.

'We dealt with it at The Tracking Room and in BOP. It gives you a depth of field on a vertical as well as horizontal plane. The bottom end comes alive and an honest reverberation characteristic that is symmetrical to the horizontal and vertical design. The room comes alive with definition. As long as the woofer goes down and sees a rigid floor and goes up and sees an attenuative ceiling, it's not being treated equally. It's truly the third dimension. The front of the room is a cloth-covered hole. The slab is three feet above the the floor below us. This allows the infrasonic inform-ation to travel below us. It's all part of dealing with the parts of sound that digital has brought us.'

The choice of SSL 9000j console was also based largely on experience. An 80-frame model was selected, initially outfitted with 64 inputs. A Neve was considered, but Meadows decided that the physical geometry of the SSL would provide more acoustical neutrality in the control room and allow that room's 10Hz design to better express itself. Not that the 9000j was the only console under consideration.

'It's an amazing console, both for mixing and for tracking,' Meadows comments. 'So was the Harrison Series Twelve, a sonically brilliant console which we experimented with at Masterfonics for a while this year. But the automation and work surface on the SSL were better prepared for music. And sonically, it gives the engineer what he needs to work with to get the room to work to its fullest.'

WITH THE TECHNOLOGY resolved, perhaps the hardest work lay ahead. The possibility of lucrative postproduction sessions on the horizon saw the studio management rapidly learning the ropes of the post market.

'We're reading-up on it, getting input from post engineers and producers,' explains Studio Manager Steve Tolson, who came to Nashville after owning a studio in Los Angeles, a city he remembers less than fondly thanks to its relatively low quality of life compared with Nashville. He is spared some of the irony of having to look to LA for post business; the staff is working in conjunction with Tennessee's and Nashville's respective film and video commissions, hoping that the inclusion of a facility such as the Tracking Room will add allure to the commissions' inducements of lower produc-tion costs, including lower taxes than California or other film-intensive states, and Tennessee's right-to-work statutes, which have limited union activity in the state.

'The thing about doing post is, you have to have your guns really loaded for those sessions,' Tolson says. 'The opportunities are great but those sessions are very

> Tolson described the studio management's mood as anxious, 'but not scared' of attempting to market the studio to the broader bases of artists and producers outside Nashville and to postproduction. 'This is what Nashville needed,' he says. 'It already has everything else.'

demanding and they require everyone to know what to do and a larger amount of equipment being there and ready. We're not going to move on that until we're sure we're absolutely ready.'

Tolson says the only barrier is the perception of Nashville by the other coasts as being a place where not much other than country music gets done, and the city's distance from LA and New York. A reduction in the number of flights between Nashville and those cities doesn't help; the largest carrier in Nashville is American Airlines, which has reduced the number of its daily flights from over 140 to fewer than 30 in the past year, and has cut service to New York to three flights a day and to Los Angeles to one flight per day. On the other hand, the interest Nashville and country music have stirred up in recent years is bringing Los Angelenos and New Yorkers down in droves—750 California drivers licenses a month were being exchanged for Tennessee tags as of a year ago, according to a local radio report. Actress Talia Shire, best known as Sylvester Stallone's wife in the Rocky series of films, held a screening of her directoral debut film in Nashville in October last year and toured The Tracking Room with an eye towards production and postproduction in Nashville. 'And,' adds Tolson—and you can sense him winking on the other end of the phone. 'There are no 10Hz rooms in LA.'

Music will remain the largest revenue base for the studio, however, and Tolson notes that the fundamentals of The Tracking Room's design take that into account. Limited advertising—within and outside of Nashville, and an intensive word-of-mouth campaign fuelled by getting as many influential engineers, artists and producers through its gates as possible, form the basis of the studio's marketing strategy. The room's card rate is set at $2,500 per day, high for Nashville but considerably less expensive than similar rooms in Hollywood or Manhattan. And an introductory 10% discount was in effect until 1st January 1996.

The first to take advantage of that were Wynonna, country singer Suzy Bogguss, jazz vocalist Etta James (produced by Muscle Shoals expatriate Barry Beckett) and sophomore, Asylum Records' country artist Terri Radigan, produced by LA transplant Brian Ahern. Vince Gill was also slated to

track there before the year's end.

Tolson described the studio management's mood as anxious, 'but not scared' of attempting to market the studio to the broader bases of artists and producers outside Nashville and to postproduction. 'This is what Nashville needed,' he says. 'It already has everything else.' Correspondingly, Meadows doesn't regard The Tracking Room as a gamble. He says he's worked the numbers, the leases and the loans till he can do them in his sleep. 'We're not leveraged to any degree of concern, and there's multiple revenue streams coming from Masterfonics now. If someone were to build a place like this in Nashville as a stand-alone facility, it would be pretty hairy. But it's part of a paid-for business, and I've always been the kind of studio owner that reinvests profits back into the business.'

OTHER NASHVILLE STUDIO owners have watched the progress of The Tracking Room closely, respecting Meadows' willing-ness to walk the line financially in a stagnant rate environment and knowing that The Tracking Room's success could be the start of a rising tide that lifts all boats. But they're cautious about whether Masterfonics can get the $2,500 daily rate that Meadows says he needs for this level of investment. As much as they'd like to see their own rates rise, they know that Tracking Room's success could also set off a technology spiral that could see many of them going into debt to finance new acquisitions, all pursuing a relatively fixed country-music base and what has been up to this point only the promise of consistent larger-budget rock and pop productions.

However, they're accentuating the positive aspects. Michael Koreiba, Manager at nearby Soundstage Studios, comments, 'This has the potential to take Nashville to another level. What happens as far as rates and technology [in Nashville] after that, though, will depend upon whether the new business that the studio attracts comes from within Nashville or from other areas.'

Lisa Roy, former Los Angeles studio owner, former General Manager of Masterfonics and now founder of international studio referral agency Studio A, based in Nashville, notes that, while there are a limited number of artists in the world who have requirements that can be filled by a facility like The Tracking Room, few of whom are based in Nashville, 'What this does is raise the aware-ness level of the entire studio community here, something it started doing the minute it was announced.'

Contemporary Christian producer Dino Elefante, co-owner with his brother Jon of LA transplant studio Sound Kitchen in nearby Williamson County, says most optimistically of the venture, 'Glenn wouldn't have done it if he thought the risk was substantial. If the existing studio couldn't have supported the new room if it were only booked one day a week, he wouldn't have done it. He's been doing this for a while.' ◎

Channel crossing: 3

In the concluding part of *Studio Sound*'s exclusive series on multichannel audio, **TOM HOLMAN** asks how many audio channels are really needed to create a satisfactory soundfield?

WHEN THE QUESTION of how many audio channels are needed to reproduce a complete aural experience is asked, the answer is variable and depends on the aesthetic goals of the music to be reproduced, among other factors. Since new high-capacity carriers are coming onto the scene, such as the DVD, it is time to examine what various channel configurations are most suitable for the range of recorded music.

I ask my university students to think about the following progression: Mozart, Beethoven, Mahler, Terminator 2. What links these apparently strange bedfellows into a logical line is their ever increasing use of wider frequency and dynamic ranges and greater use of the space dimension. It is interesting that the development of media followed the same course: we see increased frequency and dynamic ranges as the most prominent feature of the technical history of recorded sound, and today we are approaching space, 'the final frontier'.

The 5.1-channel system for audio accompanying a picture was made with specific limitations in mind, in particular those having to do with having a picture. Sound imaging in this system is emphasised where there is supporting picture imaging, and sound envelopment is emphasised everywhere else—that is, off-screen sound is 'surround' sound, by definition. Pure music recording, however, has a much less strict on-screen or off-screen division of the world, and sound images from instruments at many directions are suitable for some kinds of programme material. On the other hand, last month we saw that traditional quad fails miserably to produce sound images from four matched loudspeakers set up in the conventional quad manner.

Before psychoacoustic considerations can be accommodated however, the first requirement of any new multichannel system is an acoustically aesthetic one. Mahler uses off-stage instruments, and Frank Zappa found in his composition 'The Yellow Shark' that having six surrounding channels was useful to counterpoint, as it was possible to hear more clearly into a mix when the elements making up the mix were spatially separated. 'It's hi-fi to the nth degree,' he said. In order to achieve all-around imaging, you might expect to have to use a symmetrical loudspeaker placement. But we shall see that several factors work against a symmetrical approach.

THERE ARE a variety of ways to determine the number of channels needed to reproduce a fully immersive soundfield, one in which the listeners can move with only small restriction yet retain solid imaging from many directions, and simultaneously produce a good sense of envelopment. It is axiomatic that having a greater number of channels improves the spatial reproduction of all three of the soundfields in a room: direct, reflected, and reverberant. Michael Gerzon has suggested that a million channels would satisfy the need to produce a soundfield indistinguishable from the original one point-to-point in a listening space, so anything less may be seen as a compromise.[1] Thus the question becomes how to make a practical system, rather than how to absolutely transport the listener from one space to another.

Reduction in the million-channel requirement to something practical can be achieved in several ways. One is to look at the room acoustics of real spaces and see what directions are most likely to be used in producing the direct and reflected soundfields (the reverberant field is directionless, by definition).

This may weight our results in favour of reproducing the sound of real rooms over some synthetic space that may never have existed except as the artist's intention, but there may be valuable lessons to be learned from room acoustics which can be applied to the design of stereophonic sound systems. In particular, the findings of concert-hall designers about the relative merits of particular directions and types of reflections, with their roots in listening psycho-acoustics, may well teach us the importance of various directions around the listener sonically.

Fig.1 gives the conventional impulse response from a source on the stage heard at one seat in the Concertgebouw, showing the direct sound, reflections, and reverberant field in the time domain.

Fig.2 shows a 2-D image model for the same condition, with the diameter of the circles giving the relative level of various reflections, and their distance from the listener displaying the time. Since it is an 'image model', the sound images appear outside the hall itself, because of the long path length of reflected sound.

Fig.3 shows a 3-D image model, with the circles having the same meaning as in the 2-D case.

This data was collected by JVC engineers for their study of reproducing concert-hall acoustics. Probably the most obvious finding is that there are a lot of energetic reflections arriving from above and below the plane of the listener and source. So reproducing the height dimension in a stereo system may well reproduce actual soundfields better than systems without a height component. This is just one example, and others can be found in the literature on the design of concert halls.

Another method to reduce the large number of channels needed to produce a physical sound copy of one space into another is to weight the various directions for the acuity of hearing differences in angle versus direction of arrival, called the Minimum Audible Angle. This work has been done by psychoacousticians too, and ☞
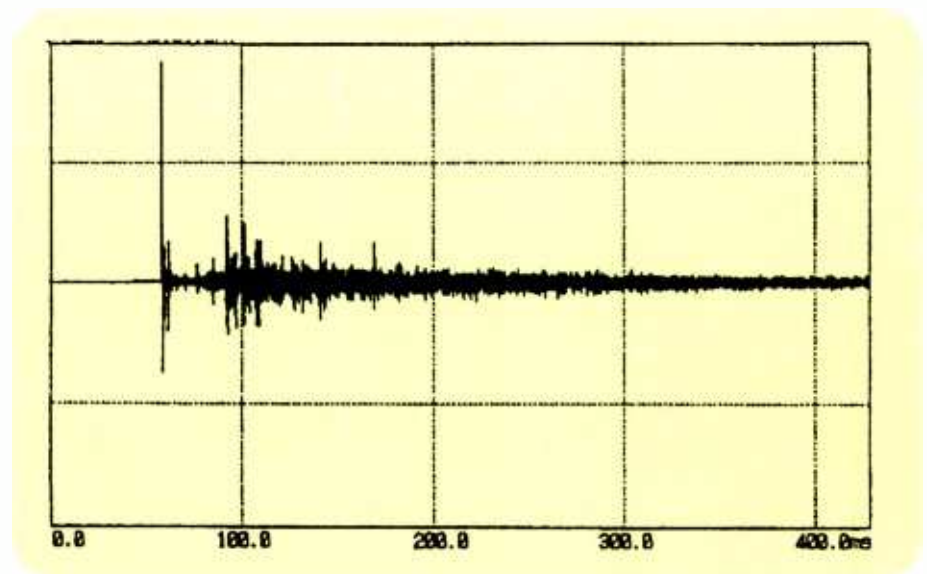
Fig.1: Time domain plot of the impulse response of the Concertgebouw in Amsterdam detailing direct sound, reflections and the reverberant field
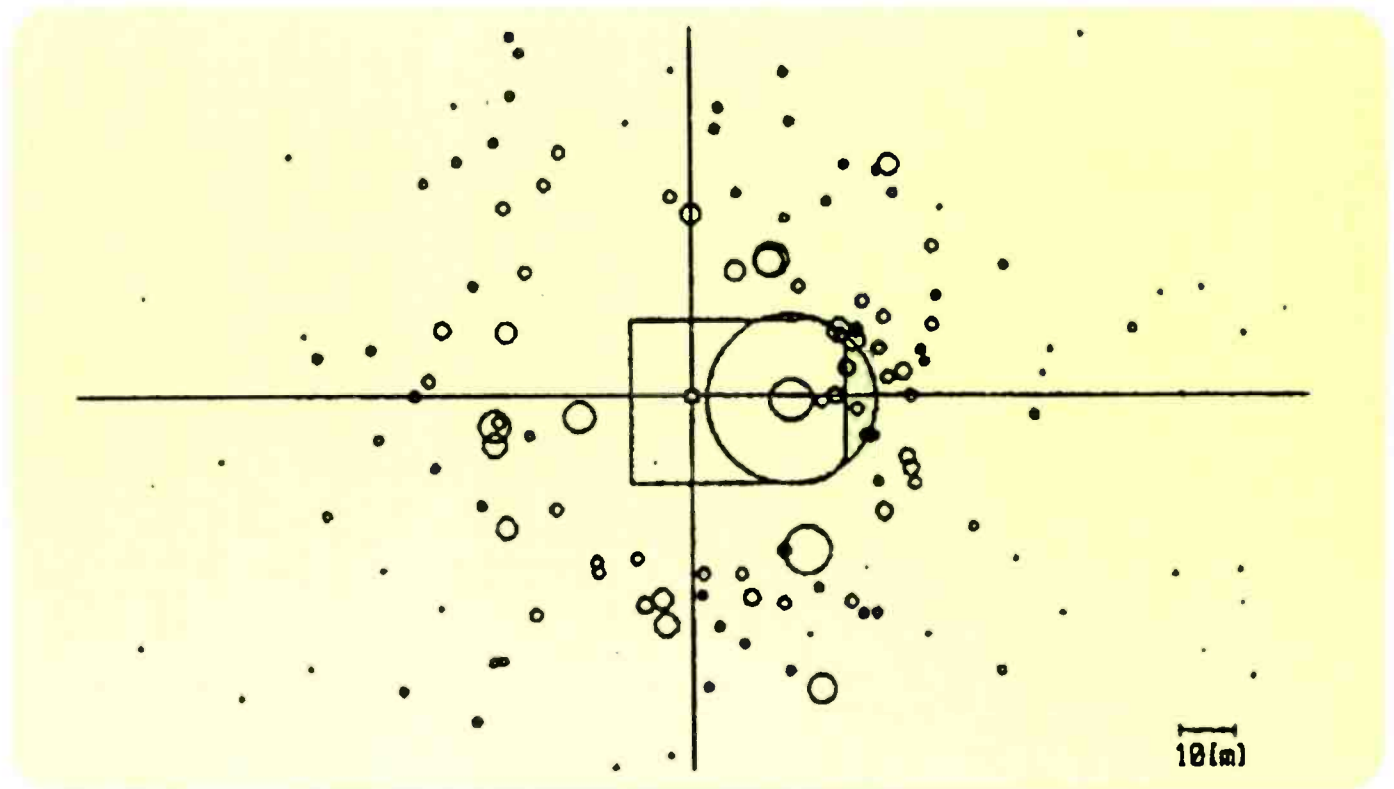
Fig.2: Two-dimensional model of the Concertgebouw impulse response—circle diameters indicate relative levels

the results of one of the most fascinating studies is shown in **Fig.4**.[2] We are looking in at a sphere around the listener from his front, side, and back respectively. The vectors show the average error from the actual position to the perceived position, and the circles or ellipses the standard deviation of the error in six trials. (The results are shown only for every other stimulus position, marked by the crosses.) The front view shows much greater acuity than the side or rear view. Front and centre shows the best acuity, with the MAA being 1 degree in the horizontal plane, and about 3 degrees in the vertical. Clearly, it is more important psychoacoustically to have greater accuracy in front than at the sides and back, so more channel resources should probably be spent in front.

Another method of alleviating the large number of channels is to use phantom images to produce sound from positions between loudspeakers. Although I showed last month that centre-front phantoms are not desirable, phantom images do work more or less between adjacent pairs of loudspeakers, so the problem is ameliorated somewhat. The problem is complicated by the fact that phantoms work better in front and back, and practically not at all at the sides. Combining what we know now about quadraphonic imaging and MAA, we could easily say that rear phantom images could be made to work reasonably well, compared to side or front ones.

ONE IMPORTANT soundfield has not been considered: the diffuse reverberant field. 'High subjective diffuseness is desired, giving the listener an impression to be completely 'embedded' in the sound'.[3] While it is nice to think that a multichannel

> Combining what we know now about quadraphonic imaging and MAA, we could easily say that rear phantom images could be made to work reasonably well, compared to side or front ones

system can easily produce a diffuse field, just by driving the loudspeakers available with multiple channels of uncorrelated reverberation, as it happens there are preferred directions which change for various numbers of channels needed to produce good subjective diffuseness. For instance, in a 5-channel system in the horizontal plane, the loudspeakers have to be at ±36°, ±108°, and 180° from straight ahead to produce high diffusion.[4] These do not wind up being in the same locations as the five channels of an imaging system for 5.1-channel sound, so either more channels are needed, or special loudspeakers emphasising diffuse-field sound may be used for this important soundfield. These loudspeakers are more important in systems having a fewer number of channels.

When all of these and other considerations are combined, a channel hierarchy of how important various directions are is found. Now curves can be drawn of the number of listeners for whom a given quantity of a given factor produces transparent sound. For instance, a 32kHz sample rate is adequate ☞
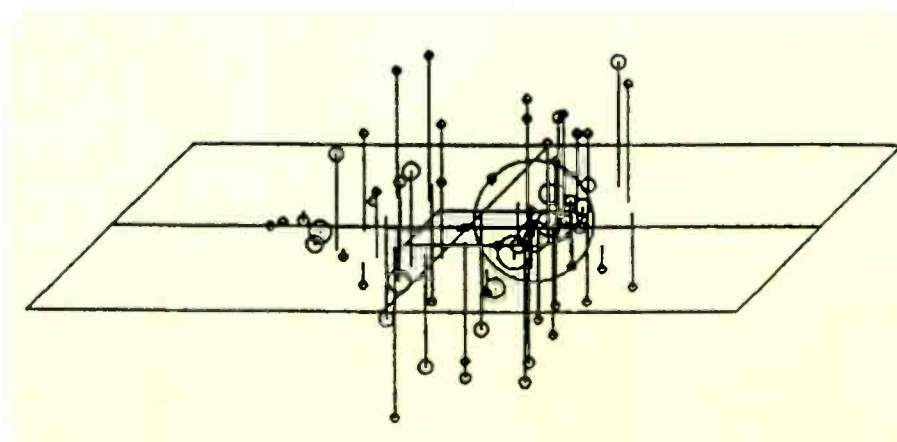
Fig.3: Three-dimensional model of the Concertgebouw impulse response—again, circle diameters indicate relative levels. Note that in both this figure and fig.2, the apparent sources lie outside the building itself

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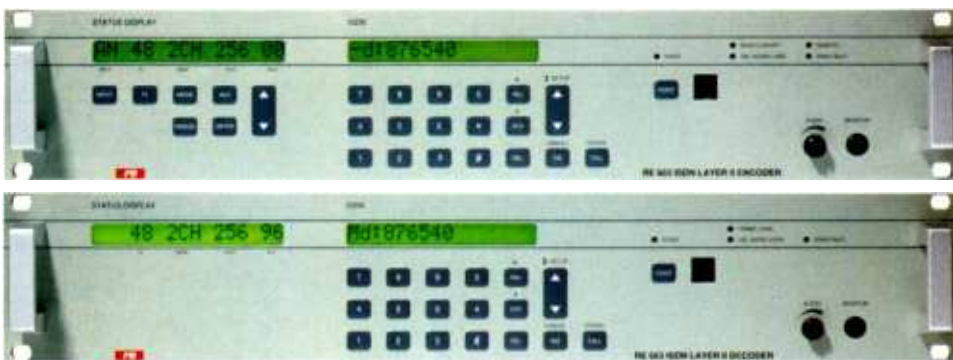
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...for some people, but many more are satisfied by a 48kHz rate. A curve of sample rate vs the number of listeners finding the reproduction audibly transparent is rather steep as more and more listeners get included in a pool for whom a given rate is adequate. Likewise, word length is a similar function, changing rather steeply as one goes from 12-bit through 16-bit to 20-bit words. The one factor which eats up bits but which changes more slowly in quality with quantity of bits is the number of audio channels. So there is no hard and fast rule for the absolute number of channels that are necessary, but some guidance can already be given regarding the uses of audio channels.

THE MOST IMPORTANT direction for sound presentation is centre front. It is here that directional acuity is best, so matching any system to human capabilities gives emphasis to this direction.

The next most important direction is that of the left and right-front channels. Various angles are possible but wide-stage
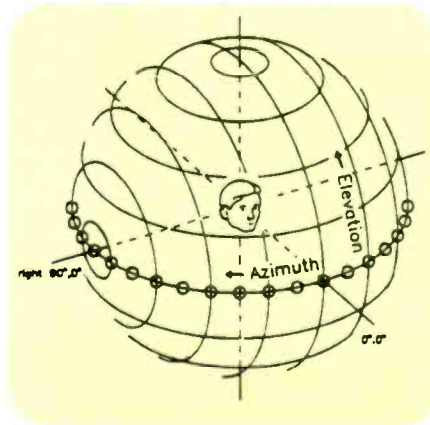
The spatial coordinate system. The subject is represented in the centre of the coordinate sphere: the viewpoint is from 30° to the subject's right and 30° above the horizontal plane. Isoazimuth and isoelevation lines are drawn in 20° increments

Fig.4: Views of spatial coordination system. (a) is from the front, (b) the side, and (c) the back of the listener

stereo is generally perceived as better than narrow stereo stages, until the image falls apart. With a hard centre, wider stages are possible, but staying with the ±30° traditional rule of thumb makes some sense. The reason for these channels can also be traced to psycho-acoustic localisation capability.

Following this, diffuse sound needs to be available to reproduce the spacious compo-nent of the stereo soundfield. While it can be separated spatially into 'leftish' and 'rightish' parts, it is useful to have a diffuse lateral soundfield.[3]

Beyond this, things get tricky. To produce good images at the sides, channels are needed. But also they are needed to produce images overhead, and lateral reflections in the range of ±55° have been shown to be useful too. The number and position of such loudspeaker channels to produce sound images from these unusual directions also depends to a large extent on artistic intention.

More channels will always be better than less in spatial reproduction, especially throughout a listening area. There seems to be no absolute limit within practical range, and many directions are as good as others beyond certain fundamental requirements. Ambisonics might seem to be one solution, but the imposition of a recording method is limiting to makers of more 'produced' recordings not involving the reproduction of a literal space. Further, Ambisonics may well make use of a large number of reproduction channels set up according to the hierarchy here. ⓢ

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Deaf, dumb and blind

Old habits die hard—and none die harder than those enjoyed by video engineers at the expense of audio engineers writes **KEVIN HILTON**

There are a number of inherent problems faced by people who earn their living from journalism. There are, for example, personality disorders (Hunter S Thompson, once wrote that journalism was a world inhabited by misfits, drunkards and failures) and there is the anxiety caused by waiting for cheques to turn up while the landlord asks politely pointed questions about the whereabouts of the rent.

Then–through the sheer volume of words one turns out during a career –there is the chance that certain phrases will find their way into almost everything a hack writes. We all have them; there is one sentence that I know I have used several times over–not because I can't think of anything else to say but because people keep telling me that it's true.

The sentence runs along the lines of 'audio was once the poor relation of visuals but that is now starting to change'. I've trained myself to watch out for it so that its repetition doesn't undermine its importance. But just as we start to believe that it's true because everyone is saying it, something comes along that instils doubt. This happened for me at a recent pre-NAB press conference given by a leading nonlinear video-editing-system manufacturer. Most of the leaders in this field have been offering extended audio functions in recent years, going beyond the token four channels that makers of VTRs ·thought was more than enough. The company concerned in this story is noted for its commitment to the sound . sector, something underlined by the managing director, who was once a sound engineer.

The reception was held in one of Soho's top media haunts (on my way in I bumped into Jon Pertwee, the third Dr Who) and started with an overview of upcoming product releases, which included an audio sweetening add-on for the company's top-of-the- range video editor. After this, and while assorted broadcast technology journalists sat round quaffing beer and eating sandwiches, there was an extended Q&A session. For the most part this concerned the visual effects and computer processing aspects of the new systems. Among the topics covered were disk capacity and file formats, with everybody contemplatively munching sarnies and listening intently to the discussion. Until I had the temerity to mention audio.

Well, things were slowing down and I hadn't made that much of a contribution to the proceedings up to that point so I felt that it was about time I made my presence felt. I asked my question and our host began to answer it. After about 30 seconds I sensed that we were losing the attention of those around us; they started to talk amongst themselves, leaving me and the managing director having something that amounted to a private conversation, only with him standing at the head of a table-full of other people. Even when I resorted to sarcasm, observing that everyone had lost interest because audio had been mentioned, they still carried on discussing the weather, plans for NAB and who would win the Five Nations Rugby Union Championship. The MD good-naturedly carried on and answered my inquiry, including the supplementary points, before conceding that it was time to bring a halt to the formal part of the day. He joked that he felt pretty much as I did, what with being an ex-sound person and all.

SO WHAT does this anecdote tell us about how audio is perceived in the obviously still vision-oriented world in which we live? Is it really so dull that it is only of interest to trained sound engineers (with their own beards and cardigans, as they are ribbed by their more glam vision counterparts)?

All it probably means is that my colleagues got a bit tetchy and maybe had wanted to move onto other subjects way before audio was mentioned. A more unfair judgement would be that there was a touch of arrogance in there, perhaps still regarding sound as the poor sibling. As one of the PR handlers remarked afterwards, some of these more high-tech journos maybe find audio a little too DC for their taste. The point of all this should be that it was quite amusing and shouldn't be seen as the beginning of a backward trend, where, through increasingly powerful processing, the visuals are the only things that matter. It also should not be seen as a slight against the hosts of this particular event, who were more than generous in their hospitality. Whatever the specifications, audio is now being used to add to the startlingly high-tech visuals that are possible. Through straight stereo or a more multichannel approach (with Dolby Surround or 5.1), illusions for the eye can be made to work all the more through illusions in sound. If anything, audio creation has more technological freedom than pictures.

Everyone had lost interest because audio had been mentioned, they still carried on discussing the weather, plans for NAB and who would win the Five Nations Rugby Union Championship

As Dr John Emmett of Thames TV observes: 'Video has to work on either a PC platform or its own specific platform. The audio attitude is freer than this, because it can be integrated as part of the buses.'

Areas of interest come into this matter but one element of the same thing (in this case TV or video) should not be derided or ignored in favour of its partner. After all, this may have happened because I have a boring voice.

But I doubt that.

Balancing

Reconciling new music with old, and old equipment with new,
has helped John Leckie identify the consistent values in recording.

PHIL WARD talked to him where it all started—Abbey Road

THE WORN STEPS that lead from Abbey Road's famous Studio 2 up to its control room provide the vantage point from which John Leckie considers British pop music history. In this room, as a young tape op and mix engineer in the early 1970s, he helped Pink Floyd, Marc Bolan and the various Beatles to fashion their sonic statements for an eagerly awaiting world.

Graduating to the producer's chair in time for a new wave of brash, intelligent UK pop, he left the studio in 1978, primarily as a result of EMI's policy of not allowing staff to work with nonproprietary artists. Leckie, naturally enough, had developed a taste for talent beyond the walls of the Abbey Road

stable. Since then, he has been single-handedly responsible for the generation of an indestructible canon of British guitar pop. The roll call of artists is a delight to the ear: XTC, Be-Bop Deluxe, The Fall, The Stone Roses, The Lilac Time, Verve, Ride and, of course, Magazine, whose 1978 release *Real Life* was arguably the finest record of its time. More recently Leckie's ascendancy has culminated in the 1996 Brit-nominated Radiohead album *The Bends* as well as production and mixing credits for debut album releases by Cast and Elastica. For these younger bands working with John Leckie is an honour.

For many of them the honour coincides with being introduced to a serious recording studio for the first time. Famous Leckie virgins include XTC, Magazine and The Stone Roses. Over the years he has noticed a surprising change in the way the new kids

approach his hi-tech block.

'I'm often surprised nowadays,' he says, 'but in the old days bands would actually be much more involved than they are now, particularly in the mix. It would be four people, hands on the desk—everyone would have their fader to move, or the pan pot, or the echo send. It may be just to do with automation but the band's involvement in the studio process would be a lot more than it is now. I often do records now where the band doesn't even touch the mixer.'

This is in complete contrast to what we are told about the rise of the project studio, and about artists taking control of their recordings. Leckie is full of these myth-puncturing observations.

'When they go into a proper studio with a producer and an engineer,' he continues, 'they are reluctant to get involved. Maybe it's just the bands I work with, but they can be very opinionated about, say, the sound of the guitar, or the reverb on the voice, and all they really mean is there's too much or not enough of it. And I'll say: "Well there's the knobs—turn it up."

'I am the kind of producer who won't discourage them. I like to involve people. I see it more as a team effort than the producer being a dictator. The studio is part of the band's trade that they have to learn. If they've taken up professional music and they get signed, it's not just about playing on stage and wearing the right clothes. You have to write the songs, sing the songs, play the music and be part of the recording process. You have to know when something needs to be double-tracked.'

Double-tracking, according to Leckie, is among many time-honoured studio techniques unfairly snubbed in recent years.

'There's this aim at an in-yer-face raw sound which means that a lot of techniques which have been built up over decades —double-tracking and other tricks—are kind of frowned upon. But very often it's because people don't work on it enough with the precision that is needed. A lot of the time singers say 'we want to sound like The Beatles but we want our vocals single-tracked. If they try double-tracking they get tuning problems, making it sound chorussy. And it's really that they have to work on.

'When The Beatles tracked things they would work on it—certain intonations and tuning things, fluctuations in pitch, the way notes are held, vibrato, not just getting the Ss and Ts right. When you do that you get really strong sounds. So, you shouldn't neglect the idea of double-tracking. It's frowned upon because people do it and don't get good results.'

Nevertheless, there's no disguising Leckie's basic belief in the value of a good performance. He may be a past master of the subtle methods of enhancing what ends up on the record; he may even mourn their current neglect. But his career shows, above

STONE ROSES: FROM MAGIC TO MAYHEM

THE FIRST ALBUM *The Stone Roses*: 'It was recorded at Battery Studio 2. We were put in at a cheap rate starting at seven in the evening. But of course, by midnight, the band were all knackered and couldn't do anything. So we were doing a five or six hour session each day. There was something about making the record under time restrictions: if there is a bit of a hassle sometimes you want to get in and out pretty quick. Those early sessions were the best tracks: we had 'She Bangs the Drum', 'Made of Stone' and so on, in a week or so. It was a case of everyone being of like mind, everyone doing their best. It had that feeling that we were doing something really special together.

'They were great ones for getting the vibe, the magic, the dynamic between the bass, drums and rhythm guitar. There is no click on the Stone Roses album. When Silvertone got all these remixes by Paul Okenfold and Adrian Sherwood they were phoning me up saying where's the click? There isn't one, it's a live band.'

THE SECOND ALBUM *The Second Coming*: 'They didn't intend to drift on and on! The idea was to do it all in a month. The first sessions were done on the Rolling Stones mobile. They wanted to record in the Manchester area, but they didn't really want to record in a studio, so I found an old brewery which was a residential rehearsal room. When I arrived I realised that the band hadn't rehearsed together for a year—Reni hadn't set his drum kit up for a year—so we did that for four weeks. Then they wanted to concentrate on one song for another six weeks, which was 'Ten Storey Love Song'. I told them they needed a deadline. There was complete free reign from Geffen, no pressure, no push. They had d to create an artificial deadline—a gig or a day in a demo studio. I said I won't go into the studio until you give me a demo. This is the way bands work with producers: send me the demo, and if it's any good we'll go into the studio.

'The demo was, erm, okay... but they booked six weeks in Rockfield anyway. Well, at least we were now in a proper studio. But when I went up there I realised I didn't want to do it. I wasn't feeling it any more. We've always been honest with each other. Well, change is another thing I always advocate, and I said, if I go, that's a change.'

'Mind you, they spent another 18 months in Rockfield after I left...'

act

all, just what can happen when a band at the height of its powers bumps into the right producer.

'No amount of knob-twiddling can equal someone putting their all into a performance, whether it's a guitar solo, a drum part, even a tambourine part. You can EQ the voice or the snare for four hours, but if they finally just go out there and hit it properly you have to turn all the EQ off. To me that live thing is to do with the vibe and the confidence from the band all the way through from the beginning. It's got nothing to do with the technology.'

WHAT HAS GOT something to do with the technology is how the sound of a record has changed since Leckie processed his first signal—the mysterious Leslie'd piano 'ping' on side two of Pink Floyd's *Meddle*. One thing he's noticed is how, as speakers have got smaller, records sound bigger.

'Since the birth of the NS10 every single studio in the world has got them. That's never been done before. All the Beatles records, *Dark Side Of The Moon*, Aretha Franklin—they were all monitored extremely

loud on big Altec, JBL or Tannoy speakers, not piddly little bookshelf speakers. Then the Auratones came along and everyone laughed—then the NS10. Everyone thought, no one uses the big speakers anyway. Before that, people would select a studio because of its monitors. Either that or the echo plate...

'Yet the definition in monitoring now means you can hear a lot more of what you're doing. With parametric EQ you can hear the difference between 6k and 8k. Everything is much more finely tuned and defined. I also think people are trying harder than they did back then because they've got a lot to live up to. In the old days, as long as it was loud it was good. Now they've got to be loud and defined and everything has to be given equal attention.

'I used to do sessions here at Abbey Road with Micky Most, and he used to love to get what he calls "value for money" on the vu meters, which was a sound that was electrically quiet but sounded loud. That's not to say it's middley, full of piercing 1k, but a bass guitar or a vocal that cuts through but retains its warmth and its

depth. When you listen to a track, what you hear coming out of the speakers is a balance between what's in the background and what's jumping out in the foreground. You can have screaming sounds that are really cutting, but if everything else in the mix is cutting they don't leap out. Here at Abbey Road they used to call the engineers "balance" engineers, because they balanced the sound. A balance engineer didn't necessarily have any technical ability; it was an ethereal thing. You balanced the controls—which goes right back to the 1930s when they only had two knobs'

'A lot of blues artists, like Robert Johnson, were recorded using one microphone. Where do you put that microphone? You've got this guy singing his heart out with an acoustic guitar. Do you stick one microphone on his guitar and one on his voice, or do you move it backwards and forwards? It's interesting listening to old 1930s recordings, whether it's orchestral or blues—where did they put that microphone? You listen to big bands, say Glen Miller, and when the saxophone players had a part they would stand up

> 'bands can be very opinionated about, say, the sound of the guitar, or the reverb on the voice, and all they really mean is there's too much or not enough of it. And I'll say: "well there's the knobs–turn it up"'

and lean forward so you wouldn't have to push a fader up. It would be totally natural. The same applies to recording a drum kit today. If the toms need more EQ it's usually because the drummer's not hitting them hard enough. The worst thing when recording a kit is twiddling each drum's EQ individually. When the guy then plays the whole kit it's meaningless. No amount of knob-twiddling can equal that little bit extra in a performance. That's probably why I choose the bands that I do: I get a feeling that they're going to perform for me, so I don't have to do too much!'

Leckie's favourite audio hardware reads like a directory of industry standards, as though venturing too far into the esoteric might upset the natural order of things.

'I'm more of an SSL man than a Neve man–with a bit more than 24 tracks, whether it's a slaved ADAT or ideally, another machine. As far as reverbs go, a Lexicon 480 or 240. I'm a great lover of SPX90s and SPX1000s. The SPX1000 is one of the greatest pieces of outboard to be invented. I could make a record just using Alesis Midiverb IIs and SPXs. You'd have to have lots of them racked up, though...

Very often it's a little bit of something that actually makes the difference. If someone wants something to sound dry, what you do is put something on it to enhance the dryness. It's not just a matter of having the mic up flat and dead. You need something that's got this very short delay, like you take the treble off a short room reverb, so that it sounds even dryer than it really is. I'd love to make a record one day on Shure SM58s and maybe on one valve 67 and maybe a V12 on the bass drum. In fact, I could probably do a whole record that would sound great just on 58s. The SM58 is a wonderful microphone. You don't need anything else: you can abuse

it; it's flat; it kicks; it cuts. It's got everything you need, which is funny because it's a £100 microphone...

'I'm also a great one for valve 47s, but they're all different. Usually when I hire a 47 I ask them to send three of them so I can choose the best. I booked RAK studios just to use their mic with Radiohead.'

Leckie's faith in the application of particular technologies readily extends to his choice of tape–and beyond.

'I always record on 499, 30ips, no Dolby. When I've had to use 456 with Dolby, I wasn't happy with it. If your meters don't move between 0 and -2, there's no point in using noise reduction. Digital tape is horrible, I don't see the point. Especially in a mix when you really want to dig into the EQ, when you start changing something to the point where everything else has to change. What you thought was a bright guitar is suddenly no longer bright because everything else is bright, so you have to really dig into some broadband EQ without making it too peaky, just really bright and sizzly. You can't do that on digital; you can't go as far. You can't turn the EQ on an SSL all the way, full on; the tape can't take it.'

The best progressive new wave record: Magazine's *Real Life*

RECORDING TECHNIQUES are the heart of Leckie's work. Like other producers who have served a useful engineering apprenticeship he is keen to identify the working methods behind his best recordings but his opinions of other peoples' successes aren't necessarily in accord with their own.

'I go for live recordings with no click but there's plenty of editing. Most of the Radiohead album tracks were edited together from the best sequences. Count the edits on the Beatles albums, there are hundreds. Mind you, there's something about recording live takes, picking the best and repeating them which is a bit weird, I don't think I do that a lot. You know, same chorus coming round every time. But that doesn't mean a good recording can't be pieced together. I try to take a good chorus, and put it in the right place–like maybe make it the first one. We can suffer something a bit less for the second one...

'And I edit with razor blade and scissors. Chop the 2-inch all the time. Chop the mix, as well. Even with an SSL. Probably the first thing you learn when you go 🔼

RADIOHEAD: A TASTE OF REAL LIFE

'RADIOHEAD approached me because I did Magazine's *Real Life* in 1978, one of their favourite records. Their music lends itself to peaks and troughs, light and shade. It's in the arrangements. Does it peak too soon? Is the first chorus sufficient? Do you leave the drums out until the second chorus? Very often the band has spent time getting their live show together, and when it comes to doing the record things don't come across in the studio—too much too soon, an instant powerful beginning designed for a gig. If you're listening to a record you need balance. With Radiohead everything was in the backing track. That's pretty much what you get on the finished record. There's nothing on the Radiohead record, apart from 'Planet Telex', which is done to a click.'

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into a studio like Abbey Road is that it's not all one take. It's like a film, with different shots all edited together. That's part of the art of recording. You don't need a sampler to fly in a vocal—just copy it off the half-inch and press the button. It's usually a shock to bands nowadays that you can do something as simple as that without getting bogged down in computers and technology.

'The funny thing is that if I got a computer to fly in a vocal or something, to sample a section and move it around, a guitar band is likely to say "what's going on, we're a guitar band not a computer band, this is not us..." But if I do it with tape, whip a guitar part straight off the half-inch, press the button and fly it back in somewhere else, that's

fine. It's the same thing; it's just the way I do it.'

But the job isn't over until the mix is complete...

'We'll finish a mix, everyone's happy, and I'll say let's do one with the vocal just 1dB louder. Everything changes. Suddenly the focus is on the voice. Now, what's happened to the rhythm guitar? What's happened to the piano? That's one of the things an experienced professional should be able to do—make the vocal sit on top of the mix, while everything else in the track is still there. If you just do a mix with the vocal up loud, all you get is a loud vocal thing with a track somewhere in the background. When you push the voice in a

track, everything else changes even if you don't touch it. The ends of vocal lines become really exposed, and you're tempted to push something else up to provide another focal point, maybe a little guitar figure.

'It's like moving a snare beat in the computer. If it's a sample triggered by the snare hit, the dynamic is the same no matter how hard you hit it. So you have to move it because the first hit is a little late and suddenly everything's shifted. Six hours later, you're still fiddling with it. Six hours! This is what goes on...'

Techniques—potentially they hold all the secrets of turning a performance into a record. But techniques are frequently at the mercy of equipment developments and the promise of progress can readily translate into poor performances and missed opportunities. And sometimes it becomes difficult to distinguish between innovation and indolence, direction and distraction, model and muddle.

'It's generally accepted that *Sgt Pepper* made a great contribution to recording techniques', states Leckie, 'but you could also argue that it was the worst thing that happened. It made the recording process abstract. It made the band's performance secondary. They always wanted to sound like something they weren't, especially in the later years, like 'Get Back', when they wanted to sound like they were recording in Memphis or Philadelphia. They used to demand of Geoff Emerick, why can't we sound like that? But they were actually in St. John's Wood...' ⓢ

PINK FLOYD: GOLD MEDDLIST

'YOU KNOW THAT one piano note at the beginning of Pink Floyd's "Echoes on *Meddle*? If it hadn't gone through the Leslie it would have sounded dumb. Or if it wasn't on fast vibrato... The whole song was written from that one note, the idea of what that one 'ping' conjures up in your mind.

'It was the first job I engineered on. I was a tape op here at Abbey Road. We recorded, on 8-track, 'Nothing Parts 1–25'. Twenty-five basic Floyd ideas, one of which was this one note which went ping through a Leslie... Then we copied the 8-track to 16 at Air, which had just opened. Three months at Air doing *Meddle*, just me and Floyd. I came back to Abbey Road and went back to being an assistant. Me and Alan Parsons were kind of rivals, both assistants trying to get balance engineer jobs. Alan did *Dark Side of the Moon*, then I did the backing tracks for *Wish You Were Here*. EMI had just installed a Neve desk—EMI made their own desks then, and this was the first none-EMI console they had—we were three weeks in and this guy called Brian Humphries came in and said "Hi, I'm Pink Floyd's engineer". He'd just started at Britannia Row, so I had to let him finish the record.'

www.americanradiohistory.com

The Safety Net

In the face of evolving comms systems **ALEX** **van** **SOMEREN** discusses digital audio recording, the Net and the growing need to understand and implement data security measures

DIGITAL AUDIO RECORDING: the saviour of the universe. It offers all those now-familiar benefits—excellent editability, fantastic fidelity, lossless layoffs, to say nothing of those seductive black boxes with mesmerising displays... But there is a problem with the dominance of the digital domain that we seem to be overlooking: if you can record and edit and reproduce the stuff so perfectly without even batting an eyelid then so can somebody else. I'm talking about piracy, not remixes.

Okay, it's not necessarily news. I mean, people have been ripping off copyright material for years. That's what the tape levy's for, isn't it?. And with digital media the pirates just get a cleaner copy in return for spending more cash on their kit. So what if they copy CDs onto DAT and make house records from the bits? So what if they capture a digital broadcast from the airwaves and cut their own CD master? So what if they hack into the approval mix you're playing to the executive producer in Barbados and sell it into Holland before the tape op has hit stop on the 48-track? Wait a minute—you mean they can do that?

Of course they can. Bootlegging's big business nowadays. People will pay good money for a high-quality copy of the new Oasis album on DAT in advance of its release. We're talking really good money, like enough to set up a whole studio just for this project and then move to the Cayman Islands afterwards. The days of nasty live recordings made at the back of a venue and packaged with grainy photocopied inserts are gone. Nowadays we're they've got squeaky-clean pressings on CD with glossy 4-colour printing and embossed diamond cases. From China. By the truckload.

I agree—it's about time for us to do something to reduce the risks. How about making our digital data truly safe from prying hackers, and I don't mean by dropping it down our trousers? It's time to think about using strong cryptography, that more recent saviour of the audio universe.

FOR AS LONG as there have been secrets, there have been people wanting to keep those secrets secret—spying is sometimes referred to as the second oldest profession. People are known to have practised cryptography, the science of codes and ciphers, since the time of Ancient Egypt. Crypto has long been the preserve of the military, the government and the intelligence services; the sort of thing people didn't talk about. Twenty or 30 years passed before the work on Enigma at Bletchley Park during WWII came out in the open, and it wasn't exactly a minor war effort: it kept the Third Reich out of Britain.

But that's all changing. Today you can download some of the strongest crypto-graphic software in the world from the Internet (for free). And by 'strong' I mean that by current reckoning the most powerful computers in the world couldn't decode it in (literally) a million years. Antigovernment rebels in Myanmar (formerly Burma) reportedly carry portable computers about in the jungle with their plans stored tightly scrambled on the hard disk. Not even a swift application of the rack or hot irons is going to get the data out again: modern cryptography allows neat tricks like requiring several people to be present at the same time before data can be decoded.

In the past, when you wanted to send a secret message to somebody you chose some obscure way of representing the message, like using groups of numbers to stand for words or translating the whole thing into Serbo-Croatian mirror-writing and putting it on a microdot. The problem with this approach is that at some stage (usually pretty early on) you need to tell the person you're trying to communicate with how to decode the message. And if the dog eats their code book, or it falls out of their underwear at the Customs desk, then they may not be there to meet you in a rowing boat when your submarine surfaces. This style of cryptography is called 'secret key' cryptography because it's critical to keep the 'key', which lets the message be decoded, safe from prying eyes.

Central to modern crypto techniques is the mathematics of so-called public key cryptography (PKC). PKC is a heavy mathematical technique for encoding messages using a key which can be made completely public and still not allow the message to be decoded. The encoding function is one-way—you can't get the encoded message back out even if you know the key which was used to encode it. Instead, a second key is required for decoding which is totally separate from the key used for encoding, so a secret channel for communicating the keys is no longer necessary. Key pairs can be generated, perhaps by the intended recipient, and the encoding key 'published' without fear of the integrity of the system being compromised; the decoding key is, of course, kept secret.

Most PKC techniques revolve around a simple mathematical truth: if you multiply two sufficiently large prime numbers together, it is impossible to determine which numbers were multiplied to arrive at that product from the product itself. The process of testing a large product in this way, known as factorisation, is a laborious one which has not been made significantly simpler even through thousands of years of effort of great mathematical minds. If the numbers involved are very large indeed, equivalent to hundreds or thousands of decimal digits, then even the fastest of modern computers cannot make significant inroads into factorising them within human lifetimes. The use of prime numbers is mandated by the underlying logic of the PKC systems; prime numbers are, of course, those numbers which have no factors other than themselves and one.

Besides the apparently fantastic ability of PKC to remove the need for a secret channel for exchanging keys, there are other fringe benefits. The same sorts of technique can be applied to establish definitively the identity of the originator of a message—important if instructions for the movement of money are involved for example. Equally, more contrived applications allow message exchanges to occur in ways which are at once completely anonymous but at the same time verifiably between two parties who can be shown to have the authority to participate. This sort of exchange is like using cash in that its forgery is hard but the banknotes themselves tell you nothing about who they came from.

One of the great things about digital data is that there is no limit to how it can be manipulated. Breaking things down into binary numbers permits a whole range of computer manipulation techniques to be applied to audio data, many of which are already in widespread use within the professional audio world. The same digital signal processors which allow infinite variations of EQ or the simulation of room acoustics can allow digital data to be made secure by using cryptography.

An important question to consider before ☞

> The days of nasty live recordings made at the back of a venue... are gone. Nowadays we're they've got squeaky-clean pressings on CD with glossy 4-colour printing and embossed diamond cases. From China. By the truckload.

☞ thinking about encrypting any kind of data is how likely it is to be intercepted and by whom. A side effect of recent scare stories about hackers seems to have been an increase in paranoia about the ease with which telephone interception can take place. Most of the media coverage is, quite simply, a load of nonsense. It's still just as illegal and just as difficult (or easy, if you know what you're doing) to place a wire tap and capture traffic over a telephone line as it ever was. Old tried-and-tested techniques such as bullshitting the receptionist into letting you 'service' the 'phone and the extensive use of crocodile clips (like Donald Sutherland in *Klute*) are still obligatory. But if you have limited interpersonal skills, severe halitosis or a recently dry-cleaned macintosh then you may be out of luck. Somebody is going to have to go to a lot of effort, do a lot of trespassing and break a lot of laws to tap a phone line, whether it's for voice or data. On all-digital circuits such as ISDN it's pretty hard to parallel tap the line without horrendous side effects anyway.

THE INTERNET, on the other hand, is a different story. The Internet was developed by the American defence industry during the 1970s as an experiment in networking computers. It is the latest over-hyped and over-here export from Nerdland into the real world, and it genuinely does suffer from an excess of egalitarian zeal which leaves it wide open to abuse. The computers forming switchpoints along the major trunks of the Internet are programmed to share the load between them and spontaneously avoid congestion if it develops anywhere within the network. Many of these computers are located in those well-known hotbeds of piracy and sedition: universities. If only the boss of Rip 'em Records knew, when suggesting playing out guide vocals over the Internet, that it was potentially handing the data to a bunch of scrofulous students

he might think twice.

Try to retain a rational perspective about this, there are some sharp-toothed laws against intercepting people's data over telephone lines and computer networks and people have been fined and locked through their use.

These laws ought to be enough to discourage casual hackers from becoming a nuisance so the real question becomes one of how valuable your data is in the wrong hands and whether you believe in organised crime. It's exciting, if difficult, to find out exactly where the computer hackers actually are when they're stealing your data, although it's not impossible—if they're in Uruguay and working by remote satellite uplink then the London Met is going to have quite a run for its money.

If you are interested in knowing about the process of tracking down and capturing hackers, it is worth reading Clifford Stoll's compelling account of one such episode called *The Cuckoo's Egg*.

If you stick to using the major telephone networks, then your data is travelling over well-secured routes which are in no real danger of being interfered with, except by the expenditure of truly horrendous sums of money.

Nevertheless, there's every danger that it's going to make more and more sense to

The internet is the latest export from Nerdland into the real world, and it genuinely does suffer from an excess of zeal which leaves it open to abuse

use the Internet for digital data traffic in the future. Already there is software which allows practically free telephone calls to be made PC-to-PC over the Internet between parties anywhere in the world. How long before it becomes both practical and cost effective to send professional audio data the same way? The Internet is cheap, it's flexible, it's fast (at least at times of day when American students are asleep) and everybody who's anybody is going to have a connection before long.

Ironically, the major national and international telephone companies are all queuing up to establish Internet services of their own and sell them to you. This means their otherwise secure telephone networks are going to be plumbed into the relatively insecure Internet. They'll be careful and wear a digital condom, sure, but you too might want to think twice about who you have unprotected interconnection with and seriously consider getting some data security systems in place. Ⓢ

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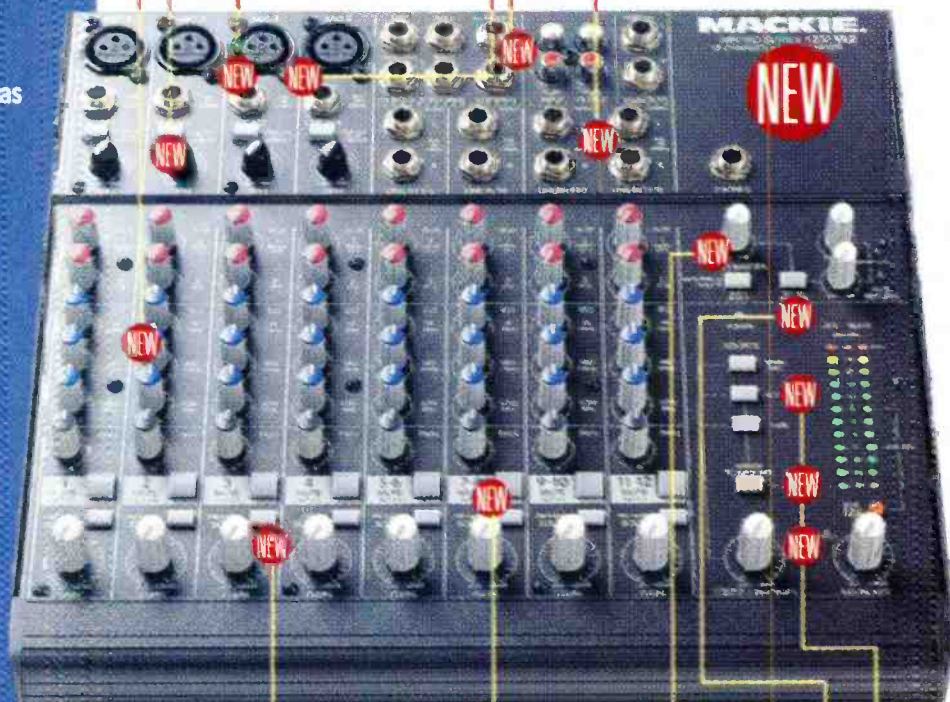
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Master Class

Programmes with a wide dynamic range presents a number of problems for broadcasters. **RICHARD HULSE** offers an alternative approach to broadcasting compression

IN AN IDEAL WORLD, the recording, transmission and reproduction chain would have a consistently wide dynamic range. In reality, the dynamic range of each stage is generally less than that of the preceding stage. The average analogue console has a usable dynamic range of about 110dB; 16-bit digital audio has typically less than 96dB; FM radio has around 55dB and the worst-case listening environment, the car, can have less than 20dB. The dynamic range of the human ear is about 120dB, although there may actually be a shifting range of around 60dB if other factors are considered.[1]

In broadcasting, there is a strict limit on the transmittable dynamic range—in FM radio it is around 55dB—but this is often exceeded by programme material. Before compression was available, recording engineers had to tailor dynamic range by

> Manual changes would be made as slowly as possible and resulted in transparent dynamic control. This method mimics approximately the apparent dynamic response of the ear

hand. This involved following a score (or working from memory) and making level changes in anticipation of the peaks and troughs in the music. These changes would be made as slowly as possible and resulted in transparent dynamic control. Interestingly, this method mimics approximately the apparent dynamic response of the ear mentioned in reference 1.

One of the main objections to the normal compressor configuration (**Fig.1**) is that it reduces the impact of loud musical passages. This is particularly noticeable when these passages contain transients such as percussion. It is the departure from a 1:1 gain law at high levels and the effect this has on the impact of the music that is most noticeable.

Some manufacturers have tried to address these problems with multistage or multiband techniques and programme-dependent attack and release times. On the whole these are an improvement, especially where there is time to fine tune the device to suit the material being processed. The BBC has developed a system called DRACULA that uses a 3-second look-ahead

to better place level changes in the best dynamic context.[2] However, there will always be listeners who find the action of these devices disturbing. The following compression system is subtle enough to go largely unnoticed yet still provide 'real' amounts of control.

THIS SYSTEM—first outlined some years back in *Studio Sound*—relies on a secondary compressed signal path running in parallel with the main signal path (**Fig.2**). The basic idea has been around since the early 1960s and dozens of companies have used it as the basis of a range of products. A colleague dubbed our implementation 'Side Chain Compression' or simply 'Side Chain'. These terms should not be confused with the internal control side chain of the compressor itself. This name has stuck and I will use it here, although in retrospect 'dynamic scaling' might have been better. ☞
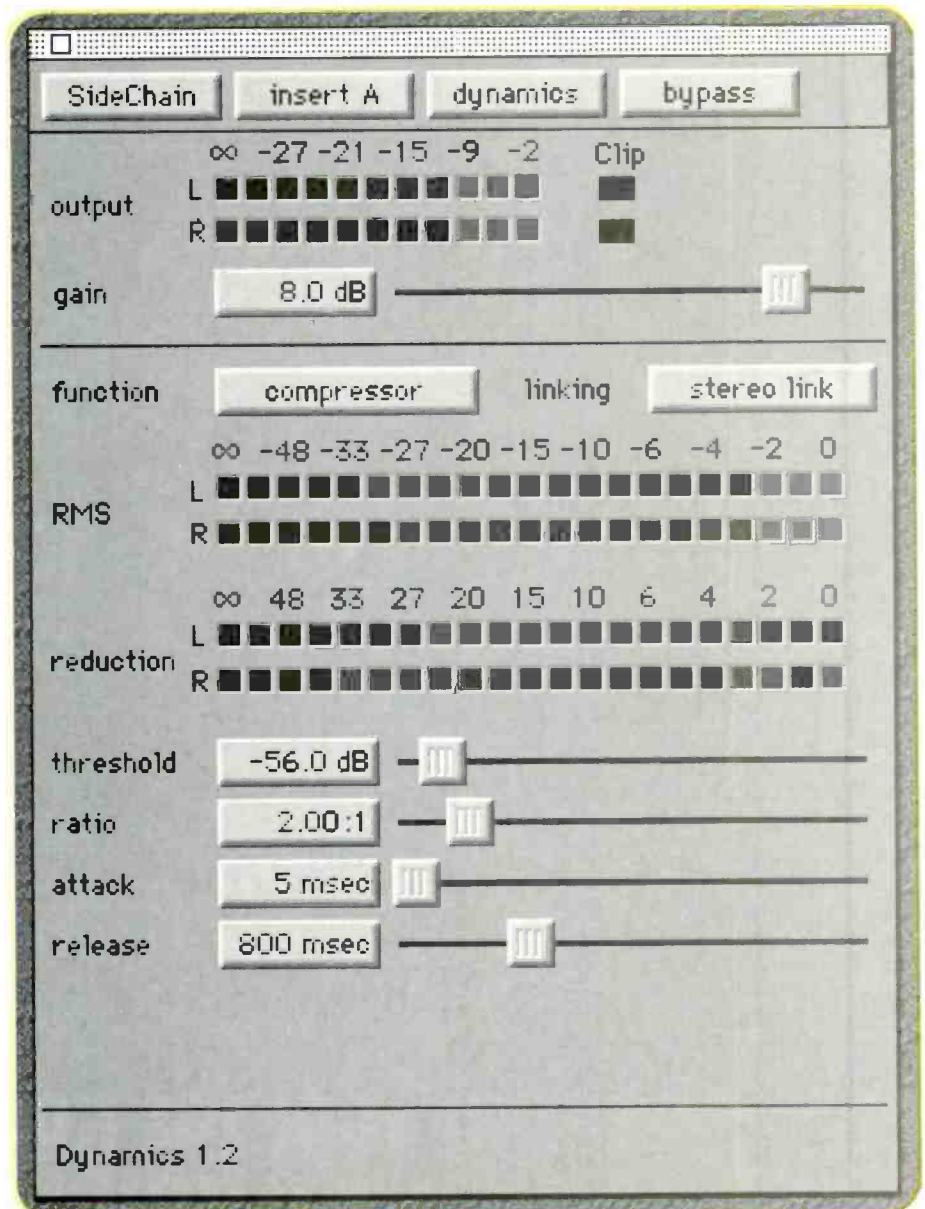
Compressor: A Screen Shot showing the compressor's settings
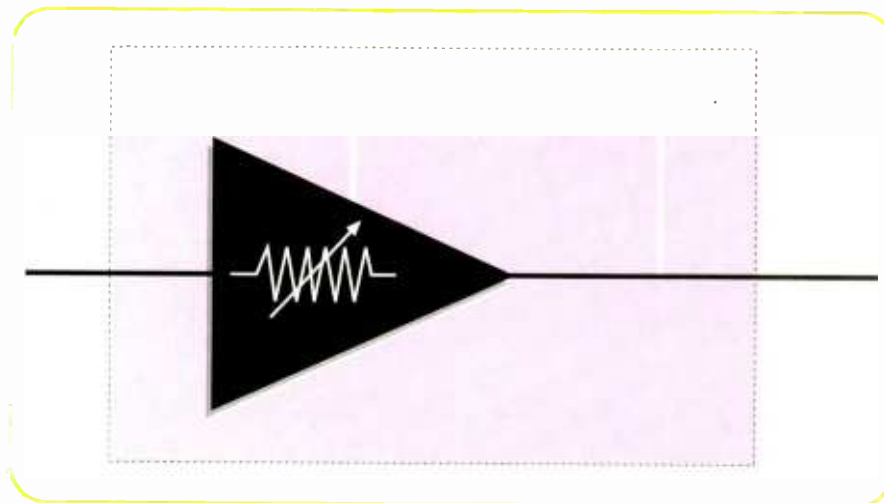
Fig.1: Standard compressor configuration
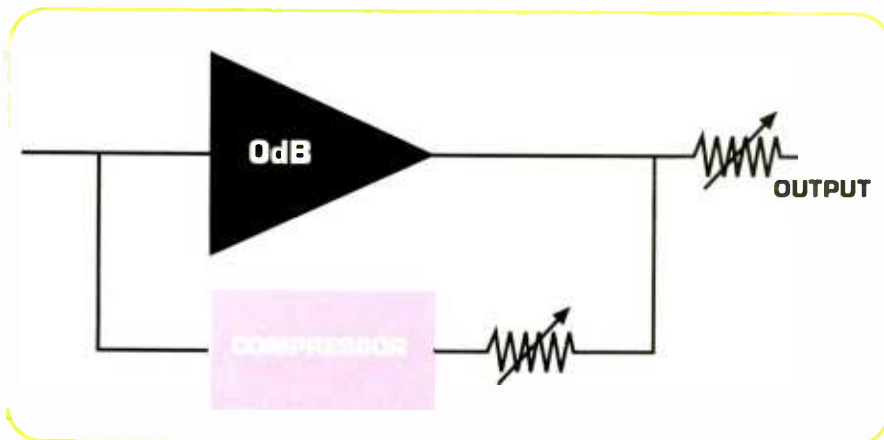
Fig.2: Side Chain compressor configuration

Setting up the system is fairly easy. I use a secondary stereo bus to feed the compressor (**Fig.3**). Any audio to be processed is sent to the main and secondary buses. A stereo channel is used to return the compressor's output to the desk which in turn is sent to the main bus only. The return must be in phase with the original signal.

Calibration is also straightforward and enables the user to achieve consistent results at different locations. A 1kHz tone at reference level (0dBr) is fed to the Side Chain bus only. This would normally be 0vu or PPM 4. The compressor is set for 16dB of gain reduction at a 2:1 ratio. Fast attack and slow release times should be used and in my case these are 2.5ms and 800ms respectively. Release times as short as 25ms can be used depending on the type of music and the effect required. Experiments with digital compressors within Pro Tools have shown that times shorter than these don't really work because they follow the waveform too closely.

The level at the compressor's output is made up to 0, in this case requiring 16dB of gain. The compressor's return channel is trimmed so that a setting of 0 on the fader gives a reading of 0 (dBr) on the main meters. In order to get a consistent indication of the amount of Side Chain, the compressor's return channel can be marked with a chinagraph pencil as in **Fig.4** To identify these points, put tone back on both

buses and add enough Side Chain level to increase the level on the main buss by 0.25dB, 0.5dB, 0.75dB, 1dB, 2dB, and 3dB.

Fig.5 shows the effects of '1dB' of Side Chain compression (dBsc). Through the middle is a 1:1 gain law representing the main signal path. Below is the output of the compressor at 1dBsc (-18dB set-point). Above is the resultant slope—a ratio of 1.14:1. As can be seen there is approximately 5dB of lift at the -32dBr threshold. A linear transfer curve (1:1 offset by 5dB) is maintained below this point. **Fig.6** shows the actual plots of the system I use. The main difference is the curve at the threshold due to the compressor's soft knee. Because the system has 1dB of 'gain' this can be corrected at the master faders. This shifts the slope down by 1dB giving +4dB at -32dBr. In practice it is only necessary to reduce the masters when the amount of Side Chain exceeds 1dBsc. The calibrated scale outlined above will show exactly how much this reduction should be.

The original 1977 description of this method called for 20dB gain reduction. In our case we were using Neve 2254s which have only 16dB of metered GR available and this became our 'standard'. **Fig.7** shows Side Chain at 1dBsc compared with 2:1 compression. As can be seen, the low-level lift at the Side Chain's threshold of -32dBr is the same in both cases. Both methods achieve 4dB of gain reduction. In the case of standard compression, this ☞

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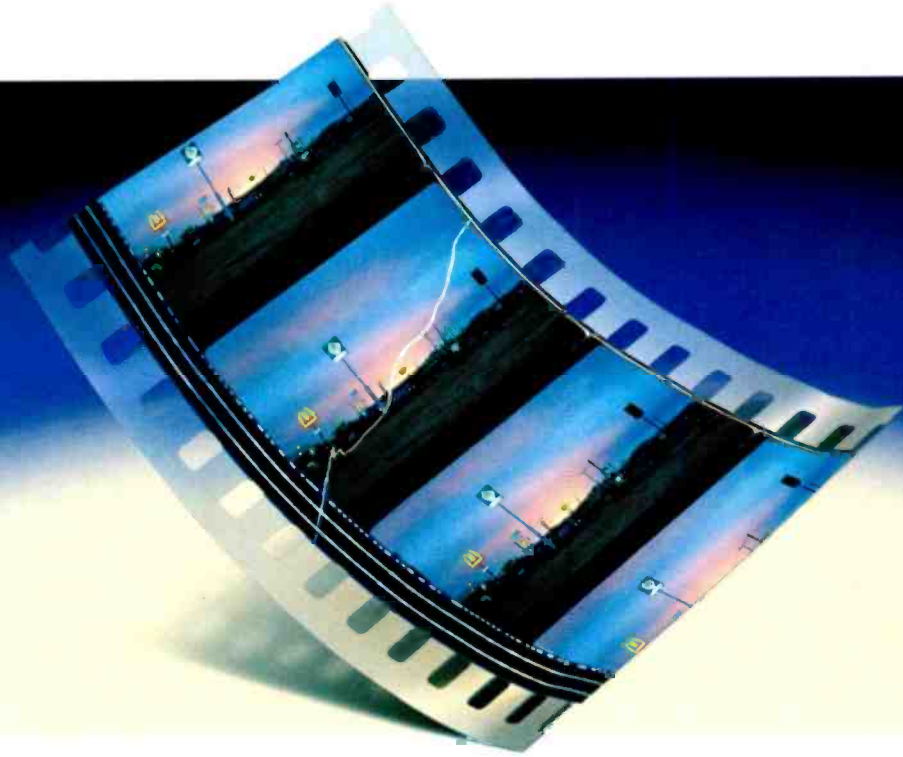
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Fig.3: Mixer set-up

Fig.4: Fader markings

that 2dB–3dB of Side Chain compression applied to an AM radio mix creates a denser-sounding track without affecting the transients too much. The result is a track which is less likely to be mangled by the station's own processing. Using 3dBsc or more can be used to make really dense-sounding tracks with a ratio of only 1.36:1. ☞
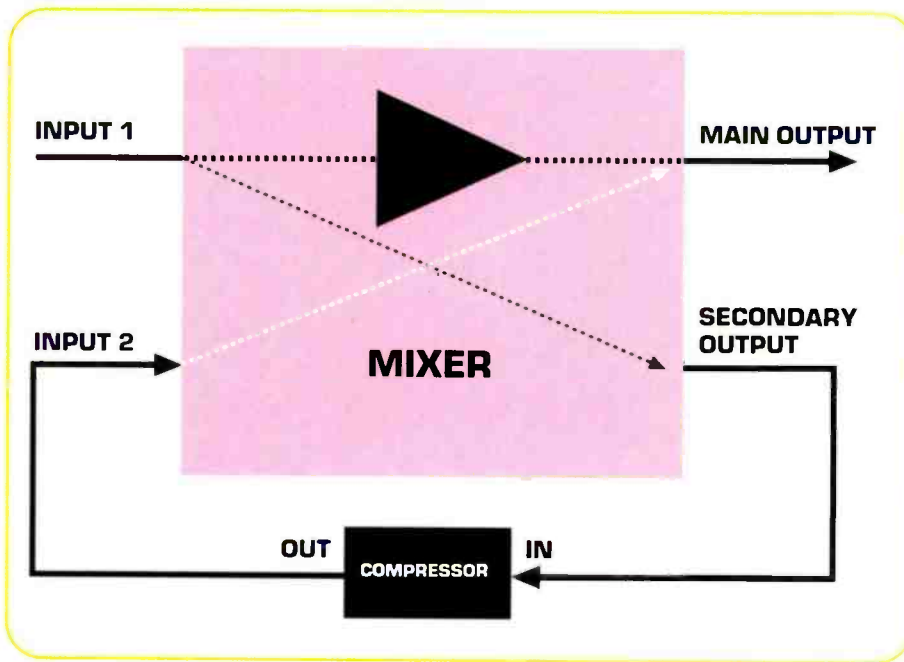
is achieved by scaling the top 8dB at 2:1. The Side Chain method spreads this gain change over 32dB, giving a ratio of 1.14:1. This is achieved by adding a highly compressed version of the signal at low level with itself. At high input levels the effect of the compressed path is small—nil if you reduce the master faders as detailed earlier. As the level drops, the additive effect of the Side Chain path increases and gives a gentle slope.

There are two main reasons why Side Chain compression sounds better. Firstly the gain law is gentler when it departs from 1:1. In fact, the threshold is so low that the system is in 'gain reduction' most of the time. Secondly, the method of implementation takes the compressor out of the main signal path. Standard compression derives its control directly from the signal and this control is applied via a feedback loop to the signal itself. With Side Chain, the main signal still controls the gain change but the gain law is derived indirectly by signal addition. Purists should still expect some smearing of transients due to nonlinear group-delay in the compressor and return channel but this is insignificant when compared with a compressor in the main signal path.

The great thing about Side Chain compression is that you can subtly alter the amount you add in real time. This is useful when recording or broadcasting classical concerts, as you can remove the Side Chain between movements so that audience noise is not increased. On the other hand, it can be used to create a cushion of atmosphere which sits nicely under announcements. Additionally, low-level passages can be lifted without the risk that unexpectedly loud passages will get out of hand.

Because I use a secondary bus to feed the Side Chain, it is possible to pick exactly what is processed. At the suggested setting, the audible side effects are virtually nil. Like any form of compression, though, it increases the density of the sound and this is particularly noticeable in the reverb. It can also appear to change the balance between a soloist and accompanying ensemble. This subtle lifting of low-level information is difficult to detect in isolation so I would suggest that you include its use in your session notes. Ultimately, because quiet material is lifted, the average level increases and the ear perceives that the sound is louder and somehow 'better.'

While I use Side Chain compression primarily on classical music there are other areas where it could be useful. The system could be used as the main on-air processor for FM radio with a couple of caveats. Because it reduces the dynamic where the peak level has already been correctly set it is not suitable as a gain-riding processor. It is best used where the levels are already properly set. The effect is sufficiently subtle that the result is very listenable, even when set as high as 3dBsc. At this level care is needed setting the release time. As with any overall station processing, the level of spoken material may have to be reduced slightly to maintain the correct balance between it and music.

This mode of compression works well on non-classical music too—both on whole mixes and individual tracks. I have found

TABLE 1: COMPARISON BETWEEN 16DB AND 20DB GAIN REDUCTION

SIDE CHAIN	SET-POINT	THRESHOLD		ACTUAL GAIN REDUCTION		LIFT @ THRESHOLD (dB)*		RATIO IN: 1	
(dBSC)	(dB)	16dB	20dB	16dB	20dB	16dB	20dB	16dB	20dB
0.25	-30.69	-32	-40	1.20	2.00	1.46	2.22	1.03	1.05
0.50	-24.54			2.20	3.5	2.77	4.04	1.07	1.09
0.75	-20.89			3.50	4.8	4.25	5.58	1.12	1.13
1.00	-18.27			4.00	5.9	5.08	6.92	1.14	1.17
1.50	-14.49			5.50	7.7	7.08	9.20	1.21	1.23
2.00	-11.73			7.00	9.9	8.88	11.10	1.27	1.29
2.50	-9.53			7.75	10.2	10.21	12.74	1.31	1.34
3.00	-7.69			8.50	11.1	11.64	14.19	1.36	1.38

* NO MASTER GAIN CORRECTION
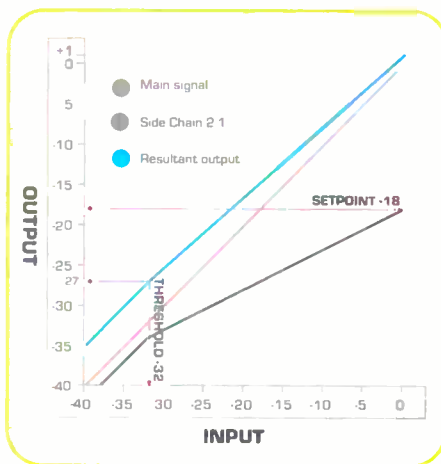
Fig.5: The theoretical outputs at 1 dBsc
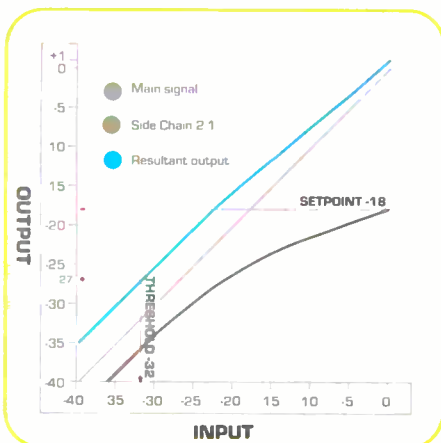
Fig.6: The outputs generated by the author's set-up

☞ It may be necessary to slow the release time a little depending on the desired effect.

Standard compression can be used creatively when tracking.[5] It is also relatively transparent at this stage as each source is processed individually. With Side Chain, transients are retained while the dynamic can be precisely controlled. While it works well with most instruments it doesn't work so well with singers. If you want to try it for vocal work, it is advisable to use a peak limiter prior to tape to catch big peaks. The vocal application in which it really excels is the recording of speech. I have used it for documentaries and the like to reduce the progressive reduction in level that can happen during long reads. This gives a more natural and open sound during lengthy passages of speech.

I have also found it a useful mastering tool—particularly for classical releases destined for cassette. The whole dynamic can be scaled to fit comfortably on cassette without audio being buried in tape noise a lot of the time. I apply peak limiting and then let the cassette duplicator know where the biggest peaks are, thus guaranteeing a hot print. The CD version is printed without it, of course. I occasionally use it when mastering pop and rock albums where the effect seems appropriate. Ultimately, it depends on what you want to achieve and what sounds best.[6]

AS WITH ANY compression system there are side effects which can be useful in themselves. In one case I used the system to repair the balance of a classical concert with solo vocalist. It was the kind of recording that inspires engineers' nightmares. Individually, the orchestral balance was fine as was the sound of the vocalist but the vocalist was far too loud and the recording was very dry. I added 3dB of Side Chain compression with a fast release time. This had the effect of lifting the orchestra between vocal passages. Two reverbs were used: one send was EQ'd to have mostly voice and the other mostly orchestra. The orchestral reverb was set to carry the sound through the start of vocal lines —hiding the fact that the overall level had ducked. This relatively severe treatment was acceptable for a one-off broadcast but would have been too much for a CD release. The untouched audio was broadcast with pictures elsewhere and listeners to radio were surprised at how much better 'our' sound was.

There are a couple of other things you might like to try. Putting a bass cut in the Side Chain return reduces the lifting of low frequency noise. In some cases I have EQ'd other areas for various effects. Boosting at 100Hz and 10kHz (both peaking) can give you a subtle contouring effect at low levels. You could also use a multiband compressor to create subtle dynamic equalisation effects. More severe EQ can be used to lift instruments in a mix.

One commercial production operator used to EQ the feed to the compressor. He would boost the speech intelligibility frequencies only and this gave immense HF density without resorting to lots of EQ in the main signal path.

Alternatively, you can send from the Side Chain channel to your reverb. The send level can even be set slightly higher than those of your audio source(s). This creates a cushion of reverb that seems to 'stick' better—especially with cheaper reverb units. It also stops recordings 'drying up' as the level decreases. I also insert a compressor in the send to the reverb to flatten quick peaks. This reduces excessive reverb hangover during repetitive staccato passages

It is also possible to process the sum and difference signals rather than the discrete left and right signals.[7,8] The idea is to enhance or alter the image width—it's probably more useful as a special effect than for serious work.

This system of compression provides an ☞

Fig.7: Comparison between 2:1 and 1 dBsc. Both are at 4 dB gain reduction

alternative to regular compression where subtlety is required and is a useful tool to have in one's 'box of tricks'. I would welcome feedback from anyone who tries this system themselves or has any suggestions. 🌀

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4. Fletcher, Ted, 'A Personal View Of Psychoacoustics—The Transparent Compressor', *Studio Sound*, November 1983, pages 70—71.
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Further Reading
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2. Morgan, Michael, 'Analogue Signal Processing for Digital Recording', *Recording Engineer-Producer*, April 1989 pages 32—38.
3. Caesar, Marvin, 'Digital Audio, Dynamic Range and the Real World', as 2. pages 26—30.

MATHEMATICAL TREATMENT

The mathematics are actually not too difficult. I worked it out by plotting the standard set up outlined (16dB GR @ 2:1 +1dBsc) and then worked out the relationships on a calculator.

Firstly we work out the system threshold. For this example I will assume the standard setup above.

$$\text{System Threshold} = [\text{Gain Reduction} \times \text{Ratio}] / \text{Ratio} - 1$$
$$= -16 \times 2\ (:1) / 1$$
$$= -32\text{dB}$$

Next we calculate the set point for +1dB of system gain (dBsc). The set-point is how much the secondary path is offset relative to the main path.

$$\text{Set-point} = [[\text{antilog (system gain increase} / 20) - 1]]\ \log 20$$
$$= [[\text{antilog} (1 / 20) - 1]]\ \log 20$$
$$= -18.27\text{dB}$$

From the set-point and threshold we derive the level added to the main signal path at the threshold.

$$\text{Compressor's effective output at threshold} = (\text{threshold} / \text{ratio}) + \text{set-point}$$
$$= (-32/2) + (-18.27)$$
$$= -34.27\text{dB}$$

Adding the input threshold and the compressor's effective output at that threshold gives us the total system output at the threshold.

$$\text{System output at threshold} = [[\text{antilog (output at threshold} / 20)) + (\text{antilog (threshold} / 20))]]\ \log 20$$
$$= [[\text{antilog} -34.27 / 20) + (\text{antilog} -32 / 20)]]\ \log 20$$
$$= -27\text{dB}$$

The difference between the system threshold and the output at that threshold is the amount of lift.

$$\text{Lift at threshold} = \text{system output} - \text{threshold}$$
$$= (-27) - (-32)$$
$$= 5\text{dB}$$

We work out the resultant gain law as follows.

$$\text{Ratio} = \text{threshold} / (\text{system output at threshold} - \text{system gain})$$
$$= -32 / ((-27) - (+1))$$
$$= 1.14:1$$

1 Changes in ratio with the same amount of gain reduction
2 Changes in gain reduction with the same ratio
3 Increasing ratio while increasing gain reduction to maintain a constant threshold
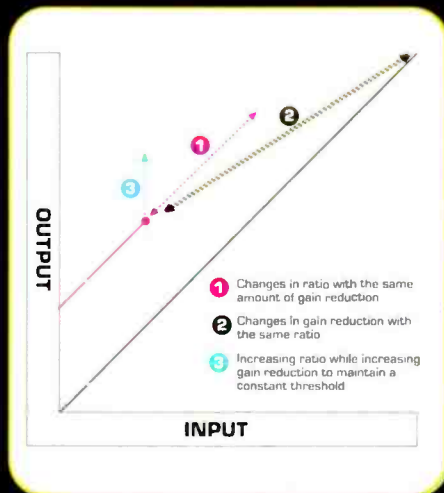
Fig.8: The effects of altering the compressor's settings

Some general patterns emerge if multiple settings are plotted against each other. **Fig.8** shows how altering different parameters affects the resultant transfer characteristic. The general 'rules' are as follows:

Increasing gain reduction lowers the threshold and increases the lift at that point.

Decreasing gain reduction raises the threshold and decreases the lift at that point.

Tighter ratios on the compressor with the same gain reduction and set-point give the same threshold with a greater amount of lift.

Given the same GR and set-point the steeper compression ratios give progressively less boost at the threshold.

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◆ **June 10th–15th**
Americas TELECOM 96
Rio de Janiero, Brazil.
Tel: +41 22 730 6161.

◆ **June13th–15th**
Infocomm International
Philadelphia, US.
Tel: +1 703 273 7200.

◆ **June 19th–21st**
Audio 96: Technology
& New Media (APRS)
Olympia, London, UK.
Tel: +44 1734 756218.

◆ **June 20th–22nd**
World Lighting Fair 96
Pacifico Yokohama Exhibition
Hall, Yokohama, Japan.
Tel: +81 3 3706 5687.

◆ **June 25th–30th**
ITS 9th Annual Forum
ans Exposition, Chicago,
US. Tel: +1 212 629 3266.

July 1996

◆ **July 10th–12th**
Pro Audio & Light Asia
96, World Trade Centre,
Singapore. Tel: +65 227 0688.

◆ **July 12th–14th,**
Summer NAMM, Nashville.
US. Tel: +1 619 438 8001.

August 1996

◆ **August 7th–10th,**
MacWorld Expo

Boston,
US.

◆ **August**
15th–18th
Popkomm, KölnMesse.
Cologne, Germany.
Tel: +49 221 8210.

◆ **August 26th–29th**
Windows Solutions
Expo & Conference
San Francisco, US.

September 1996

◆ **September 3rd–6th**
Broadcasting China 96
Exhibition & Symposium
Beijing, PR China.
Tel: +86 10 609 3775.

◆ **September 4th–8th**
CEDIA, Dallas, Texas, US.

◆ **September 6th**
International Monitor
Awards, Beverley Hilton
Hotel, Beverley Hills, California.
US. Tel: +1 212 629 3266.

◆ **September 7th–10th**
British Music Fair
Earls Court, London, UK.
Tel: +44 1442 215435.

◆ **September 8th–11th**
Plasa, Earls Court, London, UK.
Tel: +44 171 370 8179.

◆ **September 10th–12th**
6th Australian Regional
AES Convention, World
Congress Centre, Melbourne,
Australia. Tel: +61 3 9682 0244.

◆ **September 12th–16th**
IBC 96, RAI, Amsterdam,
The Netherlands.
Tel: +44 171 240 3839.

◆ **September 16th–20th**
Seventh International
Symposium on Electronic
Art (ISEA 96), Rotterdam,
The Netherlands.
Tel: +31 10 477 8605.

◆ **September 17th–19th**
Interactive Multimedia
Association Expo
New York, US.

◆ **September 18th–23rd**
photokina, KölnMesse,
Cologne, Germany.
Tel: +49 221 821 0.

◆ **September 21st–23rd**
cinec, MOC Events Centre,
Munich, Germany.
Tel: +49 89 51070.

◆ **September 24th–29th**
Live 96, Earls Court, London,
UK. Tel: +44 181 233 9306.

October 1996

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

November 1996

◆ **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**
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 15th–18th**
Tonmeistertagung
Stadhalle, Karlsruhe, Germany.
Tel: +49 2204 23595.

◆ **November 21st–24th**
LDI 1996, Orlando, US.
Tel: +1 212 229 2965.

December 1996

◆ **December 3rd–5th**
Online Information 96
Olympia 2, London, UK.
Tel: +44 1865 730275.

◆ **December 8th–11th**
Broadcast Cable &
Satellite 96 including
Pro-Audio India, Lighting
India and TV India, Pragati
Maidan, New Delhi, India.
Tel: +91 11 462 2710.

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ROCKET SCIENCE

Realistically speaking

Long the subject of intensive research, the increasing power of DSP is only now proving its worth in imitating and manipulating the human voice. **CHRIS EDWARDS** explains how the unconvincing becomes convincing

The problem with pitch shifting is that you can only take it so far before munchkinisation (the chipmunk effect) sets in. Even neat tricks such as diatonically correct shifting to give you perfect harmonies will not help if you want the result to be a long way from the original pitch.

The problem is that pitch shifting does exactly what it says. Any pitch in the signal gets shifted and by the same amount. The reason why this can sound so bad when applied to vocals is that this is not how the human vocals handles changes in pitch. There are certain fixed elements of any voice that will affect the overall sound, no matter what notes they were trying to hit. Fortunately, it is not that difficult to recognise which parts of the voice are fixed and which move with the sound of a consonant or vowel and the singing pitch.

THE HUMAN VOICE can be split into two major elements. The first is the signal produced by the larynx–this sounds important but it is the interaction of the throat, lungs and mouth that really shape the voice. These interactions act like a complicated filter bank that shapes the harmonics produced by the larynx. This signal is generally quite rich in harmonics because it is a train of short pulses. The closer the pulses are together, the higher the pitch of the resulting sound. The harmonics emphasised by the filter bank are known as 'formants'. (Get used to the word, you are going to hear a lot more about formants as the new generations of pitch shifters roll out.) Next, the action of the tongue and the mouth in producing vowel and consonant sounds acts like another another filter bank. This is the situation that vocoders have taken advantage of: they are basically multiband parametric EQs that can move fast enough to replicate vowels. They sound metallic or synthetic because the signals that they are fed with have unnatural formants. Change these formants to those of a

human voice and you should get a convincing 'voice'.

THE KEY ASPECT of realistic pitch shifting is that it links up a number of separate techniques, all the way from vocoders to physical-modelling synthesisers which use a generally similar model to replicate the behaviour of acoustic musical instruments. What realistic pitch-shifting techniques attempt to do is break the voice signal down into its component parts. They can then pull out the formant-related parts, adjust the pitch of the core signal with the correct formants in place and then put it all back together without introducing any artefacts, such as the 'underwater' effect associated with time stretching.

The reason why it has taken so long to arrive at formant-corrected pitch shifters is that, in terms of DSP power, it is an expensive job. This is largely the reason why speech synthesisers can be so unconvincing. Another problem lies with the voices of women and those of children: due to their high pitch, the fundamental frequency produced by the larynx is often so close to the first formant that the processing software has difficulty in telling them apart. However, the core techniques have been around for some years and are now becoming affordable. And recent research has indicated the way in which it may be possible to turn voices that can clear karaoke bars into virtual virtuosi. Once you know what the formants are, you can apply more DSP to the job of what the results should be for correct singing. It turns out that it is not just the ability to hit a note that is important, but how quickly the voice reacts to the change, that determines the difference between a professional and a willing amateur. One study indicated a 70% improvement in the shift speed for a professional singer, something that a correcting pitch shifter could inject given information about just when the note should be hit. Another study which looked at how far the jaw opened during singing–it cannot have been comfortable–also

showed an interesting relationship between that and the note. The results indicated that when the singers sang the vowels 'a' and 'o', they systematically increased the jaw opening as the pitch went up and the fundamental approached the first formant. The movement was not as marked for other vowels, but shows that even the shape of the mouth, and its corresponding filter bank in any simulation, can have an impact on the realism of pitch shifting

Recent research has indicated the way in which it may be possible to turn voices that can clear karaoke bars into virtual virtuosi. Once you know what the formants are, you can apply more DSP to the job of what the results should be for correct singing

The research mirrors that of other projects looking into the behaviour of musical instruments with the possibility that the techniques may move from MIDI-controlled synthesisers and virtual guitars to processors for real instruments that make them sound like something else. Not surprisingly, a lot of research focuses on capturing that Stradivarius sound. However, getting this sort of complexity into a rackmount effects unit might take just a bit more work by the DSP companies.

The present fortunes of Apple Computer are making many audio engineers and systems manufacturers nervous. **MARTIN POLON** poses the question: 'Is there an Apple in your future?'

Forbidden fruit

IF YOU TAKE a moment to stop and think about the use of the personal computer in modern pro-audio, it quickly reveals itself as the most dramatic change to occur in the studio business since magnetic tape recorders appeared during the 1940s. Think a little harder and you'll probably realise that you first conclusion is based largely upon the application of Apple's Macintosh. Beginning with the first 'Lisa' from Apple in the early-1980s, the Macintosh has intrigued the audio community with its potential. Today, it is estimated that approximately 80% of all audio recorded, manipulated, edited and postproduced is done on mass market personal computers with Apple getting the lion's share of the workload.

The year 1996 is being hailed by the world's computer industry as the 50th anniversary of the development of the computer. The accolade may be off a year or three in ignoring the gestation of Eniac, Brainiac and other US WW II machines that were developed for ballistics calculations, and Alan Turing's 'Bombes' that helped break the German Enigma codes at the Government Code and Cypher School at Bletcheley Park. But the same 50-year point has also seen a virtual firestorm explode around Apple Computer's viability, some 12 years after the introduction of the first Macintosh machine. This enormous challenge to Apple's survival has been marked by a news media frenzy at virtually every level—newspapers, magazines, radio, television and on the Net.

A select few news sources—such as the domestic US financial newspaper barons and the international financial news magazine *The Economist*, or the BBC World Service on shortwave—even attempted to offer an even-handed report on Apple's 'descent from grace to computer hell'. What ever territory you happened to find yourself in recently, the issue of Apple Computer's imminent failure and its vilification at the hands of the consumer press was raised to a level of unbelievability at the beginning of this year. There is no user, reader or viewer of the world's mass-news media who has not witnessed the vision it has presented—of a metaphorical Apple dangling in the wind while hundreds of thousands of frightened Macintosh users rally below, bearing pitchforks and burning torches. In addition to the media, many Wall Street experts and analysts continue to prophesy gloom and doom for Apple's future, as they have for the last several years. The company continues to be regarded as a marginal investment and Apple's downgraded status at several major brokerage houses has not altered materially. Worse still, the actions of

rating organisations Standard & Poors, and Moodys in downgrading Apple's long term future to virtual junk bond status is still common in financial circles. Some analysts feel that Apple stock is over-valued at it's current level—yet its price is already down by nearly half from the recent Apple stock share highs of 1995.

SEPARATING THE REALITY of Apple from the perception of Wall Street and the portrayal of the news media, it's difficult to justify intolerance of or prejudice against the Californian computer innovator is because in the computer marketplace, the media are indeed the message. But the decision makers who currently employ Apple Macintosh computers to their business ends have certainly been forced to reassess their situations.

The various 'niche' applications that have been the staple of Macintosh computers—in our case the sound recording studio and the television, video and motion-picture postproduction facility—are now destined to face agonising re-examinations of their choice of computer platform. The extremely loyal nature of the Macintosh 20m-plus user-base makes the prospect of using the competing Wintel (Microsoft Windows running on Intel Pentium chip-based machines) platform approaching the unthinkable. The 'plausible' options for the Mac's future proffered recently by the news media (including having a virtually unsupported computer platform is what so terrifies the Macintosh user base.

In reality, the immediate future of Apple Computer will be more closely determined by the following points than it suits the media and Wall Street currently to admit:

Firstly, there is no question that Apple's end-of-1995 sales numbers were not perceived as desirable by either Wall Street or the press but the reality is that Apple Computer was the first major equipment maker to be hit by the end-of-year sales slow down but the not the last. Many, if not all, of Apple's current problems have as much to do with the bad press and poor financial analysis directed at Apple as with actual 'per quarter' financial performance. Witness US film-and-camera manufacturer Polaroid, who took a $119m 'hit' at exactly the same time Apple announced its $69m 'problem.' Apple posted an 11% increase in gross sales while Polaroid's dropped. Yet there was nary a column inch in the press about Polaroid while Apple roasted in the media spotlight.

Further, other computer companies are having the same problems of slowing demand, declining profit and sales figures, and expectations for the future that do ☞

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not meet Wall Street's hopes and perspectives. The truth is that the personal computer industry has in all likelihood 'hit the wall'. Such well known names as Compaq, Digital, IBM, Intel and Microsoft have revised their expectations of their respective financial futures. Certain computer manufacturers, such as New England's Digital Equipment (DEC), have chosen to leave the non-office marketplace completely. The desktop computer industry may have sold computers at such a prolific rate that the industry now has become as conservative as, say, the television set industry. This prospect implies flat growth and profit expectations coupled with elongated 'turn' (replacement) predictions for new computer sales. As we speak, DEC is undergoing the same battering previously enjoyed by Apple, with the company's stock

value precipitously deteriorating. The bottom line is that Apple is not alone in having changing future financial expectations in a changing computer universe.

An alternative method of evaluating the future is to consider the alternatives. Microsoft Windows 95 and-or Windows 3.1 operating systems (OS) are not superior to the Macintosh OS. One reason that the Macintosh user-base is so committed to staying with the platform is the unnecessary and arcane complication and differences found in the Windows products. Windows still has at its base, what is, essentially, a machine language translator dating back to the late 1970s–the Digital OS or DOS. In the case of extreme difficulty, Windows 95 users will have to default to the basic machine language or DOS. Apple's forthcoming Copeland and Gershwin operating systems

(System 8.0 and 8.5), will further widen the distance between the MAC OS and the basic Windows line.

And Apple, despite the current difficulties, has a user-base that is, to a large extent, reluctant to change platforms even if it means staying with an alternative system to that used by the majority of personal computer users. In a recent study, a large percentage of Macintosh niche publishing, audio and video users cited the history of Commodore's Amiga as a justifying analogy of a user-base clinging to a superior system rather than transferring its allegiance to a mediocre 'standard'.

The devotion of Apple Macintosh owners transcends any other brand loyalty relationship in the world's electronic-business industry –it almost assumes the cult status enjoyed by prestige automobile and hi-fi manufacturers. Certainly, the enthusiasm of the 60,000 attendees at MacWorld conventions more exactly matches that seen at *Star Trek* gatherings or at Beatles' reunions than at most computer shows.

The well-publicised shortage of top-end professional products such as the PCI-slotted Power Mac 7500 and 8500 and key niche Power PC Performa products such as the 6300, all of which have significant design features for audio and video applications, take no account of the fact that availability has since improved markedly. Additionally, Apple is expected to upgrade these and other machines to the 604 Power PC microprocessor chip this year, enabling speeds of 160MHz to close to 200MHz. Dual 604 processor options will also be available in the 8500 and 9500 models. The likelihood of some, if not all, of these machines reaching 250MHz speeds in 1996 is also high.

Similarly, the well-publicised setbacks to initial software and hardware release time frames, belie the fact that Apple's technological progress has continued unabated. Forthcoming technologies such as imaging tools like 3D Quicktime and Quicktime VR, Internet utilities such as CyberDog, communications enablers such as Open Transport, the Open Doc software suite and so on, are both on-schedule and on-course. Let's not forget that Microsoft's much promoted and much heralded Windows 95 was over two years late–with three different public delays of the final release date.

The current presence of Power Computing Corp, Radius, UMAX Technologies (selling as SuperMac) and DayStar Digital, signifies the continued presence of Macintosh via these licensed clone manufacturers–no matter what changes are made at Apple.

Even more important to Apple's future, major personal computer manufacturers previously committed to the IBM-PC platform are expected by many industry analysts to join the new Macintosh platform 'clone family' later this year with the introduction of PPCP machines. PPCP is the Power PC Platform –formerly the Common Hardware Reference Platform (CHRP) and before that the PowerPC Reference Platform (PREP).

These new Apple ROM-less Macintosh units will be built using standardised PC and Mac parts and are based on the Apple-IBM-Motorola PowerPC chip with a new robust software operating architecture. They will accommodate multiple operating ☞

NEWS FROM TUBE-TECH EQ 1A

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systems and incorporate options for both Mac and PC peripherals and plug-ins. The new PPCP Power Macintosh and Mac clones will be capable of running Power PC compliant versions of the Macintosh OS, Windows NT, Solaris (Sun Unix), AIX (IBM Unix), OS/2 and Novell NetWare. You may even see IBM enter the Apple Macintosh fold with the PPCP introduction, along with such major PC computer manufacturers as Canon, the Taiwan New PC Consortium and Zenith Data Systems, Yet other large 'name' computer companies such as Compaq Computer, Dell, Gateway 2000 and Toshiba may follow, according to analysts. PPCP chip sets will be available to the computer industry from chip makers Motorola, National Semiconductor, Texas Instruments and VLSI Technology.

Apple has enabled Power PC chip supplier Motorola to sell complete chip sets to computer makers in China and to sub-license the Macintosh operating system. The potential for sales in China can be measured in the billions of dollars, from this arrangement. And it's worth noting that Apple Macintosh dealers are making money selling the line and will not desert the product either immediately or in the short term. As long as the dealer base can obtain products to sell, they sell well.

But despite these strengths, there are other issues still threatening Apple:

A 'brain drain' is one of the greatest dangers to the survival of Apple. The bright dedicated people who have made Apple what it is, are not located as much at the top as at the middle or core of the company. Of the top executives who have left the company, some will be missed and some won't but the core engineers, software writers, product managers and other dedicated and long serving individuals within the company are irreplaceable. These people are rethinking their positions and resumes are still flying about the Silicon Valley and elsewhere; albeit at an less alarming pace than at the beginning of the year.

New buyers, who are basically not in tune with the Apple cult or culture, will think twice, or perhaps three times, before buying from a company that everyone around them predicts will not be around in six months or a year. Apple's disappearance in total is not likely, but to the uninitiated, the news media barrage directed against Apple and the Macintosh is fearsome.

The company's transition to PCI add-on slots and connection bus has not always created a satisfactory user-solution for the problems facing the niche user base involved in the important niches of desk-top publishing, audio, video and graphic creation. The lack of supporting 'third party' developers and product suppliers to utilise the PCI bus has left Apple's existing NuBus machines—such as the first generation power PC 6100, 7100 and 8100 plus powerful 68040 units with DSP such as the 660AV and the 840AV—to carry the volume of these niche users. In addition, the Wintel platform has made real advances with the third party developers porting their programs from Mac to IBM compatibility. But, Apple is solving these problems and new machines to be released throughout 1996 will further enhance studio functionality while third-party vendors finally place PCI software and peripherals in users grasp.

ULTIMATELY, MANY FEAR that Apple is no longer an Apple and is beginning to look like a cranberry or a side of British beef instead. The problems that befell the lowly US cranberry industry when greatly exaggerated claims and inaccurate media reporting of chemical contamination of the US cranberry crop caused the public to virtually abandon cranberries for several years. To some, Apple could be in that state and having to deal with a media barrage that is more sophisticated and damaging than that aimed at the cranberry does not help.

Apple has bought itself a fair amount of time to form a cohesive strategy for the future with the replacement of it's Chief Executive Officer and President, Michael Spindler. The new Apple Chief Executive and Chairman of the Board Gilbert brings a strong physics, chip production and computer industry background, has a substantial reputation as a turnaround expert and company saviour. What Apple must learn to do is to satisfy both its shareholders with acceptable sales numbers which include licensing from an ever expanding Mac clone market and sales of it's operating system software and satisfy it's dedicated niche computer-buying customers, especially like the audio studio community. So, if the question is asked, is there an Apple in your future? The answer is probably yes.

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Disagreement by design

Studio acoustics has long been a contentious subject, continues to be so, and will doubtless carry on being so for a long time to come. **ANDY MUNRO** picks up Tom Hidley's gauntlet

I RECENTLY READ Eric Stark's article on the latest generation of the Hidley 'concept' studio (*Studio Sound*, December 1995). It appears to me that his message is completely at odds with common wisdom and the basic physical laws. The idea that reverberation and diffusion are bad, and anechoic environments are 'ideal' is an anathema to virtually every sound engineer and musician I have ever met.

The definitions of clarity, stereo imaging and precision are meaningless concepts in an anechoic environment; a one-dimensional sound stage which has a left centre and right can be achieved with a $10 pair of headphones and a knowledge of acoustics which extends to locating ones ears. The art of recording and mixing requires control of perceptual cues robust enough to survive in any reasonable soundfield, and to this end the monitoring environment becomes part of the monitoring equation.

Equally false is the notation that sonic reality is somehow 'extended' by increasing the monitoring bandwidth to 9Hz. I have carried out detailed Fourier analysis of many recordings and environments and have found nothing of any harmonic value, just the inevitable aural flotsam of our polluted environment. The fact that subsonic noise breaks into studios is largely negated by the fact that our hearing is 80dB less sensitive at 20Hz than in the mid band. In simple practical terms, if an Apple Mac is whirring away in the corner of your control room at around 25dBA then the air-conditioning would have to roar away at 105dB to cause even the smallest masking effect.

As a matter of interest, one of my colleagues in the Institute of Acoustics has found a direct link between HVAC resources of over 100dB at very low frequencies—with Sick Building Syndrome. I know well SCD (Sick CD Syndrome) but it is easily avoided by good mastering practice which simply cuts unwanted subsonic signals.

Extending bandwidth down to 9Hz may be fine—even laudable—for the recording chain because in the electron domain it does not pose any particular difficulty. When it comes to moving air things are radically different. To begin with, the wavelength of sound at 9Hz is 35m, give or take, so the fundamental room mode will be 70m. To the best of my knowledge even Hidley doesn't build control rooms that big, so we can assume that there would be direct pressure coupling between room and speaker. In this case the room becomes largely irrelevant, except to say that it needs to be extremely solid to contain the movement of such energy. Also, building materials behave badly at such frequencies, therefore justifying the $1m per room budget to build a shell solid enough to contain the sound of amplified 10Hz rock and roll.

I have been reliably told that the people who worked on the control rooms at Bop had great difficulty working with the sub-bass system, trying it both in and out of phase with the main monitors and then ultimately disconnecting it. I can understand why.

The idea of extended response is not new. B&W and others have made 20Hz monitors for years—half the world's classical producers use them.

When I designed PUK studios in 1984, we installed 30-inch drivers working down to 15Hz, complete with ground reflection traps. This wasn't a new idea and was first shown to me by Don Davies (at Synergetic Audio Concepts) in an entirely different context in 1975. It had been found that floor reflections cause bass cancellations at around 100Hz as a result of the path length difference of around 1.6m. The phase shift at 10Hz was 45°, not enough to cause significant interference and actually helpful to break down modal wave patterns. The

> *The claim that reverberation is bad because it cannot be repeated consistently and produces mixes which 'do not travel well' suggests a misinterpretation of a complex problem—that of good acoustic design*

results at PUK were fine but the compromises required to achieve such results were difficult to justify. Moving air at 9Hz is easily done by fans, trucks and aircraft but not by loudspeakers which are required to move eight times more air than at 40Hz, to maintain power bandwidth. It is true that it is possible to create very impressive physical effects when moving so much air and I can see plenty of justification for such systems in the cinema. We already work with film dubbers to produce 20Hz systems.

The statement that only the Hidley-Kinoshita system addresses 20Hz is rubbish; JBL, Dolby, THX et al have been doing it for years. It was also stated that 'very little testing is done down to 20Hz'. Actually we measure all systems from DC to 50kHz, including every currently available commercial monitor system so I have plenty of evidence, based on over 500 room tests.

The claim that reverberation is bad because it cannot be repeated consistently and produces mixes which 'do not travel well'

suggests a misinterpretation of a complex problem—that of good acoustic design.

A well diffused soundfield (where the intensity, the net vector flow of energy, is zero) can only be sustained in a room which has carefully designed geometry and dispersion. Of course, this is more difficult than building a box full of rock wool but I know which I would rather work in.

THE PROBLEM WITH many control rooms is that they have the wrong combination of monitors and acoustics—a horn-loaded speaker in the average, deadish room will create a Critical Distance Point (the distance from the speaker where the inverse square law reduction in the direct sound pressure is equal to the reverberation sound energy level) somewhere in the next town; the concept of reverberation is not relevant. In the same room the low frequency Critical Distance may be as little as 1m. So even close-field monitors (there's a can of worms) can be strongly influenced by their immediate environment.

This could be argued as a case for reducing LF reverberation, but to eliminate it completely means throwing away most of the energy that the monitor can produce. Most 15-inch bass speakers produce a sound power-output (Lw) of around 100dB. In a control room, at 3.5m from the speaker, this will produce a level of 100dB for a completely live room, and 81dB in a completely dead room. A typical 'classical' room, with an RT of 0.3s will produce 86dB. To throw away 5dB of level requires doubling the size of the monitor system or halving the headroom.

Such a wide discrepancy in Direct-to-Reverberant Ratio is the culprit in the matter and one which can be addressed by good design. Dead rooms suck energy, so you need horns which beam at high frequency, so you need to make the room ever more dead at low frequencies, to maintain some kind of balance, but call this 'the Real World'—give me a break guys.

One last point; Mr Stark states that we don't mind vertical reflections because there were no bombs or land mines in our early revolutionary, sorry evolutionary, period. Actually its because most people, with the possible exception of those in Los Angeles, have their ears mounted horizontally enabling lateral discrimination at the expense of vertical resolution, which relies on pinnael comb filtering. In any case, what about all those primeval swamp creatures and snakes in the trees?

Anyone interested in the early application of some of these concepts may wish to refer to *Studio Sound*, October 1980, and also November 1985 where the PUK Project is described in detail, complete with floor traps. ◎

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