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On the road and on the air

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Space at home

THE EVOLUTION OF MANKIND has also influenced the evolution of the environment in which it lives. In the same way that our ears have settled roughly in the same repeatable position on our heads in order to best identify the direction of that charging mammal or taxi, and the placement of our eyes has happily coincided with the direction in which we walk, so too we have happened upon a character of acceptable living space.

A house move has caused me to experience a variety of prospective different abodes and I am convinced that the first acoustic impression of rooms is the biggest contributing factor to making a place feel good, followed, it has to be said, by sympathetic light and the lack of offensive smells.

For the most part these rooms have been empty and I have been surprised at just how nicely perfectly plain houses can sound. Our last house slipped to a bare shell of all furniture and fittings reminded me of why we had liked it so much all those years ago.

The trouble is that you can’t live in an empty room and a life’s baggage and paraphernalia soon clamps a rooms response down to undisguised and inflexible. Which is all the more pity given that the acoustic demands that are about to placed upon the average living space through the reproduction of multichannel domestic formats are orders of magnitude greater than the obstacles placed in the way of stereo reproduction.

The TV and the stereo invaded living rooms, but the progress of entertainment media has taken it to the very limit of what that space can comfortably absorb. Perhaps we are approaching a time where an entertainment zone in a house will become as much a consideration to architects as multiple phone points, storage space and an extra downstairs cloakroom. I can’t see any better evolutionary route to the problem.

Zenon Schoepe, executive editor

The price of progress

IT USED TO GIVE US PLEASURE to hear the sophistication of professional audio equipment compared to that of the space programme. After all, we’d all rather be sporting a Purdey than a Rossi, and driving a Ferrari than a Trabant. But after a disturbing conversation with a tame mathematician—who reluctantly concedes that he would rather work in applied than pure circles—I wonder how long we can hope to hold on to the accolade.

His specific concern was over the commercial sector’s reluctance to recognise that commercially valuable research comes at the cost of plenty that isn’t. The incentive to those with research grants to allocate is that someone, somewhere makes good at some time, but that it’s impossible to be sure who, where or when. To play the research game is a bit like doing the lottery—or riding in aeroplanes. As a New Zealander currently on his way to an academic post in Toronto via Cambridge, England, my mathematician has some insight into the differing research climates around the world. His experience is reflected in the research efforts of pro-audio companies who are increasingly dependent on other areas of business for their technological progress.

While it is fashionable to point out that the early recording studio business was driven by personal passions rather than business plans, there’s no doubt that those passions were directly responsible for much of the progress that was made and the experience we now possess. Had the early private studio owners taken their case to any reasonable financial backer, we might still be recording on metal wire in stairwells since the acceptance of the general rules of business inevitably brings with it a tempering of what we might achieve—technically if not artistically.

Perhaps the recent award of Fields Medals to two Cambridge mathematicians, and the return of the inspirational John Forbes Nash from the brink of insanity will help mathematicians recover the glamorous mystique that attracts investment of all kinds. It’s the same quality that once fuelled music recording, and we could do with a fresh delivery.

Tim Goodyer, editor
Born to Broadcast ...
and On Air Around the World.

Digital broadcasting brings many new challenges, not least in ensuring operational staff are trained to fully maximise the advantages on offer. For broadcasters seeking to ease the transition to digital, Aysis Air offers the unique SSL combination of advanced signal processing and a familiar and intuitive control surface.

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Stig Nordahl, Sound Technician. TV2, Norway

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www.americanradiohistory.com
board of directors spearheading the Coalition are Manley Laboratories, Drawmer, Braun and SoundField USA (Transamerica Audio Group) and API. Others included are Esoteric Audio Research, Purple Audio, Empirical Labs, D.W. Fearn, Earthworks, Crane Song Ltd, Soundelux Microphones and ProAudioSearch.com (the pro audio internet information source). ‘Every one of these companies is unique,’ said Brad Lunde of Transamerica. ‘The Coalition brings personally run, customer committed companies with unusually high performance products together to help each company reach more people than any one company could alone.’ The Audio Underground Coalition can be found at AES Booths 575 and 676.

Analog Town Hall
The forthcoming San Francisco AES Convention will provide the setting for a Town Hall meeting to discuss the future of analogue recording. The meeting is open to producers, engineers, studios and suppliers, and is sponsored by Studer, Quantegy and Emtec. Discussion topics are likely to include demand for analogue recording, equipment maintenance costs and recording hardware availability—coordinator Bruce Borgerson being anxious to draw attention to the fact that only one professional recorder manufacturer is still supporting analogue technology. The meeting is scheduled for Monday, 28th September at the ANA Hotel.

Bruce Borgerson, Tel: +1 514 488 5542; Email: bbwavecompuserve.com

Training places
The world of training schools and courses saw the launch of Oxford Audio Training recently. Set up by John Gallen on the strength of some 20 years in production and engineering, OAT also draws on the expertise of Brian Gayler, with his 15 years in pro-audio, and LA-based producer-engineer Mark Deaner to offer training in all aspects of sound recording.

BBC’s Centre for Broadcast Skills, meanwhile, will unveil new courses at the forthcoming BIC. These will include the flagship 5-day Digital Terrestrial Television for technical staff and broadcast engineers involving lectures and practical demonstrations addressing all elements of the digital broadcast chain. In the interests of validating the growing number of training courses, the British-based APRT has announced its initiative for industrial accreditation of professional audio training courses and the first courses to carry it—Surrey University, the Gateway School of Audio Engineering, Music Technology and Music Business Studies (Kingston University), the University of Westminster and the Liverpool Institute for the Performing Arts.

The scheme is the result of 18 months work which involved ‘pilot schemes’ and a team of assessors that included Dave Harris (Rumour Associates), Pete Fielder (CTN-Lansdowne Studios), Simon van Swaenenberg (SSL) and Joe Hosken (Studio Sound/Pro Sound News Europe).

Oxford Audio Training, Tel: +44 1491 614695, Email: Oxfordaudio@btinternet.com BBC CBSTel: +44 1386 420216, Email: cbst.admin@bbc.co.uk APRS.Tel: +44 118 975 6218, Email: info@aprs.co.uk

September 1998 Studio Sound
Tomorrow's TV schedule

Formalising their long-standing relationship, Japanese electronics giant Matsushita and US software giant Microsoft are to jointly focus on developing digital television technologies for next-generation PCs, including support for the reception and display of digital TV broadcast signals and associated interactive programs.

In addition, Matsushita plans to deploy analogue Web TV terminals in Japan targeted at the 1998 winter-selling season, and the two companies will collaborate on the development of digital cable advanced set-top boxes for public networks and retail customers.

The agreement demonstrates the companies' intention to encourage the convergence of digital AV technology and the PC and falls into four key areas—the PC's ability to handle digital television broadcasts, Web television, porting Windows CE to Matsushita's AM33 chip for forthcoming AV equipment, and licensing Windows CE for Matsushita's use in AV computer products.

- Hollywood's The Post Office has installed Fairlight's (UK) digital audio workstation to pursue its niche approach to postproduction services. Its brief includes cable TV and home video post-production as well as mainstream work for the likes of The Disney Channel. Culver City's 20th Century Fox is to open the Darryl Zanuck dubbing theatre complete with 2 BGW M200 active subwoofer systems, bringing it in line with various of the studio's screening rooms.
- 20th Century Fox, US. Tel: +1 310 369 2873.
- Fairlight, UK. Tel: +44 131 387 1400 BGW, US. Tel: +1 800 466 2677.
- The Latvian Radio SWH Group has purchased an Orban Audicy workstation as part of the relaunch of its contemporary music station. One of three stations in the group, Radio SWH is the territory's leading commercial station reaching around 2m people with its 24-hour cocktail of music, news, current affairs and magazine programming.
- Radio SWH, Latvia. Tel: +371 731 3286.
- Orban, US. Tel: +1 501 351 3500.
- London specialist recording studio Bunk Junk and Genus has upgraded its DynaudioAcoustics M44MM monitors with a pair of XTA DP100 digital system controllers.
- BJJ, UK, Tel: +44 171 381 6298. Muro Assoc8ates, UK. Tel: +44 171 403 3808.
- Channel's TV station, TVB, has installed 6 Jünger d.o.s digital dynamic processors with adaptive pre-emphasis for processing of its transmissions.Along with New York's ABC Networks,TVB is the first broadcaster to employ the updated processors. Across the border in Hong Kong, DK Audio's Master Stereo Display digital meters have found favour with a number of broadcasters including What's Cable, Star TV, Media Watcher and TVB with a number of 8-channel surround units. Jünger, Germany. Tel: +49 30 877721 00.
- DK Audio, Denmark. Tel: +45 44 53 0355.
- Montreal-based post facility AstralTech is to install a Soundtracs DPC-II digital console in a new digital film dubbing stage currently under construction. The 96-fader desk will be used in conjunction with a 48-track Augen Master Recorder and a 64x8 Adat programmable surround monitor routing system.
- AstralTech Canada. Tel: +1 514 939 5060.
- Soundtracs, UK. Tel: +44 181 388 5000.
- Augan, Netherlands. Tel: +31 73 78.
- New Delhi Television has purchased 9 Micros UHF radio microphone systems for its ENG teams. Each system is composed of a TX50 belt transmitter and SDR70 miniature diversity receiver. NDTV in India's largest news and current affairs production service, broadcasting to over 50 countries within Asia and producing news programming for Star Plus and NDTV India. Tel: +91 11 621 6820.
- Audio Engineering, UK. Tel: +44 181 341 3500.
- The regional ferry service is to open its new TV facility by been billed as the UK's largest outside London. Complete Television is part of Andrew Sumner Associates, recent £2m expansion and includes 5 on-line suites and a broadcast studio. ASA's portfolio of clients includes the BBC, Channel 4, Yorkshire TV and Granada.
- ASA, UK. Tel: +44 161 228 0330.
- Brazil's Teatro São Pedro has installed a Stage Accompany SR system as part of a complete restoration. The installation includes 8 Per former P77, B15 and 4 Enformer E2400 cabinets, PPA 900R and SA1600 amplifiers and 4 SA2310 graphic EQs. Founded in 1917, the theatre is São Paulo's second oldest and reopened with the participation of Rossini's La Cenerentola earlier this year.
- Stage Accompany, The Netherlands. Tel: +31 229 822 000.
- Here companies' recent activities include Britbit-based Cinevideo's acquisition of Soundserver's EMI 1994 radio mic systems which have seen use with OBI provider Teleogene nonlinear hire specialist Salon's 2 PCI Audio/Visions and Pro Tools systems.British PA company South West Audio's inaugural use of the Crest 8002 amplifier at the Glastonbury festival and Yorkshire's Desirable Sound purchasing an A&H GL3300 desk, 4 Neo PSI speakers and 2 Neo L50 LS1, Gearhouse Broadcast's pre-rations for the Malaysian Commonwealth Games in Kuala Lumpur include 22 Audionics SC1 soundcheck monitor units for audio monitoring in the Tech Ops cabins at each of the event's 22 venues.
- Cinevideo, UK. Tel: +44 181 743 6762.
- South West Audio, UK. Tel: +44 1454 633635.
- Desirable Sound, UK. Tel: +44 1226 380817.
- Audionics, UK. Tel: +44 114 252 2533.
- London-based Black & White TV Mobile is to install a second Carvin 5-serise console in its new £13.6m OB Unit 5. The 48-channel desk will be used for music productions and keeps the company of the earlier 5-serise which is in use on sports and music assignments.
- Black & White, UK. Tel: +44 1222 350959.
- Carvin Audio, UK. Tel: +44 1442 842159.
- Sydney branch of the School of Audio Engineering has equipped a new 24-track studio with a 32-channel Maclce desk, Studer 827 multitrack machine.A new 16-track room houses a Yamaha QZ2 desk and ADAT XT700 DDM machines while the multimedia department has added 13 Apple Mac 7600-200 compters and 2 Silicon Graphics Indigo systems running Softimage software. In Auckland, recent purchases include Foxtel DS DAT, Focuenteen Ecomix and a selec of AKG microphones.
- SAE, The Netherlands. Tel: +31 20 421 7575.
- London audio broadcast specialist 2nd Sense has chosen HHB's CDR8000 CD recorder for use on its sound design for a variety of TV and radio projects including Top of the Pops and Top of the TV, and a special commission for digital radio.
- Fly the Moon. Future digital programming prompted the selection.
- London post activities include Offline Editing's purchase of Pro Tools and Audio Vision's systems and TVL's 2 Supplemental scheduling and a selection of AKG microphones.
- 2nd Sense, UK. Tel: +44 1923 492761. HHB, UK. Tel: +44 181 962 5021.
- Europe's cinemas have benefited from heavy weight sound systems recently months. the Dutch Pathé group has chosen Stage Accompany 526, 57 and 545 Screen Seres loudspeakers to be driven with the new model LS20 amplifier.Pathe Eindhoven joins the chain's multiplexes in Maasbracht, Groningen and Rotterdam in this selection. Belgium's Kincope group, meanwhile, has opted to install BGW Performance Series 3 amplifiers in its cinema expansion in Madrid and on the island's main continent, as well as others across Europe.
- Stage Accompany, The Netherlands. Tel: +31 229 282930.
- BGW, US. Tel: +1 800 468 2677.

UK: The acoustic hood in Birmingham's prestigious Symphony Hall has been improved in order to accommodate the increased demand made on it by additional lighting rigs. The 35-tonne hood that provides the principle means of controlling the auditorium acoustic is now suspended from 28 new cable assemblies designed by Midlands-based PCM, that incorporate strain gauge load link transformer capable of monitoring the load on the hood and an alarm facility.

PCM, UK. Tel: +44 1726 864444

Studio Sound September 1998
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<td>Berlin, Germany. Contact: Martin Woehr, Germany. Tel: +49 30 357 5839. Fax: +49 30 357 5839. Email: <a href="mailto:aes15@des.de">aes15@des.de</a>. Net: <a href="http://www.aes.org">www.aes.org</a>.</td>
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<td>Berlin, Germany. Contact: Martin Woehr, Germany. Tel: +49 30 357 5839. Fax: +49 30 357 5839. Email: <a href="mailto:aes15@des.de">aes15@des.de</a>. Net: <a href="http://www.aes.org">www.aes.org</a>.</td>
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<td>Hall 19, National Exhibition Centre, Birmingham, UK. Contact: 1449 1491 838575. Fax: +49 149 132575. Email: <a href="mailto:dmsv@pointscom.co.uk">dmsv@pointscom.co.uk</a>. Net: <a href="http://www.sbs.exhibitionsindia.com">www.sbs.exhibitionsindia.com</a>.</td>
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<td>Intex Shanghai, 88 Loushangan Road, Shanghai, China. Contact: Marcus Berner, P&amp;O Events. Tel: +49 14 171 370 8211. Fax: +49 14 171 370 8143. Email: <a href="mailto:vdt5@bom2.vsnl.net.in">vdt5@bom2.vsnl.net.in</a>. Net: <a href="http://www.aaes.org">www.aaes.org</a>.</td>
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The Freedom principle

It was good to see the excellent article on the Freedom Series wireless microphones written by Neil Hillman (Studio Sound, May 1998). It’s great to see someone who is at the heart of the sound recording profession carrying out such rigorous tests and concluding with such accolade.

Could I clarify a couple of points in order to enthuse your readers even further about Freedom Series? The ECM-77BM is the version of lapel mic designed for use with the Freedom Series and the ECM-301 referred to in your article should have read ECM-310BM, a head-set mic also designed specifically for these transmitters. There is no Sony option to fit a Hirose connector to the Freedom Series transmitter WRT-805A. (Hirose manufacture the connector used on our professional WL-800 Series.)

Neil Hillman mentions ‘the fiddly plastic slider switches’ which operate phase reversal and input level. These are intentionally small to avoid accidental misoperation by the user; imagine the effect of an accidental extra 20dB during the latest conference on The History of the Ancient Greenly in Medieval Mesopotamia—it might wake someone up!

While I can understand Neil’s test suggested a reduced operating range from Freedom Series, wireless mic operating range can be adversely affected by anything from body absorption of RF to the number of railway lines running nearby. If you want greater range we will happily provide any existing user with a new wave whip antenna, if they get in touch. Freedom products were never meant to replace the existing, and very successful, WL-800 Series wireless mic systems from Sony. Freedom wireless mics are an extension to the range—a more affordable alternative for users in environments that are not as demanding as professional broadcast.

In breaking new ground with such an attractively priced wireless mic system, there is always going to be debate about quality versus price. At around £3000 (UK) per channel, the Pantechnicons are on their way.

Ciaran Doran, Professional Audio Marketing, Sony Broadcast & Professional Europe.

Standard issue

With great pleasure and satisfaction I have read the comments and statements made by David Bell and Scott Dorsey on surround sound principles and common standards for practice (Studio Sound, July 1998).

Allow me to give two answers:

- Just this need was the reason to establish the open working group, Surround Sound Forum (SSF), during the 1996 Tonmiestertagung in Germany. It was recognised that the high standards of EBU, ITU and other international broadcasting standardisation organisations need more background and-or comments for the audio community to transform the techno-aesthetical rules into the daily practice for the reference basis. During the last two years the (SSF) has prepared two recommendations which should be finalised before the next Tonmeister (Karlsruhe, 20th–23rd November, 1998). These are SSF-01, Listening conditions (Requirements of soundfield, monitor loudspeaker, listening room, and so on) and SSF-02 Multichannel Sound Recording in the 3/2 format. The present interim status—unfortunately only in German language—can be found on the internet and we are thankful for any comment. (www.tonmeister.deforen.surround-sound@tonmeister.de)

- The SSF cannot give final answers to any problem concerning ‘surround sound’ but is on the way to support the work and to help within the variety of confusing formats (formats for recording, reproduction, delivery-coding, and so on) and recording methods (here you also can find basic requirements for control rooms and for monitor loudspeakers).

At this time the SSF consists of more than 60 colleagues in Germany, Switzerland, Finland, Denmark, Austria—and the aim is to publish the documents in English. We shall inform you on the progress of work of the SSF at the coming Tonmeister.

I have been working in the audio field since 1947—sound engineer, R&D in audio broadcasting labs (Telekom) and now as audio consultant, working in the international standardisation bodies since 1955 OIRT, ITU-CCIR, AES, ISO-MPEG etc. I am also engaged in multichannel stereophony. Starting in this domain in about 1960, we developed (in Berlin) one of the first 4-channel systems ‘stereoambiophony’ (see 100th AES Convention, Preprint No. 4286, 1996 Copenhagen: Surround Sound—The New Phase: An Overview). But there was no recording medium for the home application. I may say we have collected a lot of experiences (good and bad) which we now can use in this discussion when we work with the common 3/2 standard reproduction format of ITU-SMITE (with or without picture, with or without any perceptual coding for recording/delivery). The 5.1 format feels useful only for cinema productions. When we (my colleagues and myself of the broadcasting labs) worked with 2-channel and 4-channel stereophony (note: a large difference for us to the misunderstanding of quadraphony in the seventies) we gained also experiences with multichannel control rooms. But we will not forget that such experiences were also made by BHC Research (David Meares and team) too, and published.

You see, your ‘call for discussion in surround sound’ is falling on fertile ground everywhere and with the help of your readers maybe we can compile the best ideas of all to such practice standards to be updated if necessary.

Gerhard Steinke, Surround Sound Forum, Berlin

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September 1998 Studio Sound
**THE MIX** Alan Branch:
"Comping some backing vocals in the computer and printing them on to multi-track, they were a little bit sibilant, so were processed with the MX50. With a few seconds fiddling they sounded much more natural. It's an excellent tool for recording or mixing. This box can reduce anything that is up in the sibilant range, from the fret string noise or twang of an acoustic guitar to a hi-hat within a loop. When looking for the final polish to problem vocals this should be in your rack."

**SOUND ON SOUND** Paul White:
"Not only is the MX50 a very good dedicated de-esser, it is also affordable when you look at what other manufacturers are charging for a comparable product...there can be few de-essers that work as smoothly and unobtrusively as the MX50, and which are so straightforward to operate. The floating threshold system is also extremely clever, as the input signal can vary over a wide range and still be treated effectively. If you suffer from sibilance problems, this is probably the best budget solution around."

**AUDIO MEDIA** David Mellor:
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**Digigram Xtrack; PCX440np**

Like many French companies, Digigram has achieved consistent success but not the high profile of its US or UK rivals. **Rob James** looks at its processing card and recorder-editor software.

The French have always had a reputation for going their own way. Their gastronomy and wine culture are legendary, as is their art and their music, but French engineering has always seemed, well, slightly wacky to the rest of the world. What, you may well ask, is the relevance of all this to professional sound recording and editing on laptop PCs?

The PCXnp range is the latest introduction together with two cost-effective LCM cards. The complete range is now huge with around 13 audio boards and a couple of sync boards. Software offerings are, thankfully, less prolific. Digigram offers Xtrack as a multipurpose multitrack editing package with some interesting additions. The cards have found a considerable following in the OEM market and Digigram supplies drivers and comprehensive software toolkits to aid system integrators.

The list of OEM partners is approaching 100 worldwide with names such as Studer, Orban and Siemens joined by a host of less well known, vertical market companies. Applications are mostly of interest to the radio fraternity, but there are some interesting tangential products. Gervasi Elettronica makes the Cinetrack editor and recorder that, as the name implies, is aimed at sound for picture. IDMS offers a Digital Voice Announcement system for airports, bus stations and the like, and Multim has a complete control system for sound and light shows. Sonotopia even produces a musak system using the Digigram boards. The radio applications are too numerous to detail here, but everything is covered from on-the-road acquisition through editing, asset management, playout and full networked news systems. One area worthy of note is ISDN, increasingly used in radio. Maycom has developed a PC-based ISDN codec based on a Digigram board.

Despite all this Digigram is not terribly well known in the UK. I can remember seeing a stand at trade shows and dismissing it because of the emphasis on compressed audio. I know I am not the only one with this antipathy to any form of compression for professional purposes. The reason I believe it is imprudent to originate material in this way is due to the probability of the recording being passed through subsequent compression codecs of unknown type which may have unpredictable
effects on a previously compressed recording. For example good-old Nicam, familiar as the delivery format for stereo audio in UK broadcast TV, can react badly in conjunction with various types of compression. I have heard it and it is not at all nice. However, this is a rather unfair prejudice since Digi-gram's current line works with linear PCM as well as MPEG and will export Real Audio for Internet delivery.

The PCX440 (new performance) series is a family of boards with Motorola 56301 DSP chips providing the processing horsepower. The various models cover 1 to 8 outputs and 2 to 16 inputs in analogue or digital flavours. If required, and if you have the budget, not to mention sufficient free slots, several boards can be installed and synchronised. The standard PCX440 model has 20-bit converters and is analogue only. AES/EBU 1-0 is optional as is a small daughterboard for SMPTE LTC time-code input and MIDI. This uses real estate on the computer's rear panel, but mercifully does not require a slot.

The card is a full-length PCI type with a deep cutaway to miss the ram banks on some motherboards. It went into my ATX-based machine with no problems. In fact, I really liked the 'board on both sides' construction which keeps the card looking clean and avoids cables snagging on components. The top board is neatly silk screened with the pin outs of the breakout connector. The breakout cable attaches to the card via a massive 62-pin sub-D connector. This connector is beautifully designed with three separate cable exits for analogue, digital and wordlock. It may well be an in-house item since it is embossed 'Digigram'. The cables are short and terminate in XLRs for analogue and the optional AES/EBU 1-0 with a BNC for wordlock input.

An often used line of argument said driver installation was as painless as the hardware. Drives are supplied for Windows 95 and NT4. I selected to use Windows 95. Unfortunately the install wizard is convinced you will have the drivers on diskette. Mine arrived on CD-R and, despite repeatedly informing the wizard of this fact I had to manually locate the files for it and even manually decompress the WAVE driver with Winzip. Once all this had been accomplished everything was just fine. Just for once plug-and-play actually managed to configure the card without any conflicts and the Xtrack software installed without fuss. However, much to my disgust Xtrack requires a dongle. These devices really do not have any place in a professional installation. There are other, highly effective, methods of protecting intellectual property without resorting to an (easily stolen) hardware excessence.

The manual for Xtrack is now only supplied as on-line help. This is all very well, but I really feel a professional product should come with a full printed manual in addition to the 'help'. Some of the nomenclature is unfamiliar, but no more so than with some of the other long established DAW manufacturers. A project in Xtrack speaks is a Title with a soloiliated file extension. A variety of other file extensions identify the type of sound file or waveform file. These take the form of first letter equals type, subsequent numbers equal sampling rate.

The screen is remarkably clean and uncluttered, and the 3D graphics on buttons and faders are sharp and elegant in an unusually satisfying way. The main screen follows the now familiar model of horizontal track display with transport controls at the bottom and a floating window containing meters and 1-0 faders. This has tabs to switch between the various analogue and digital and digital and monitoring functions. The monitor windows when recording. Alternatively this window displays 'professional vu meters' for ins or outs.

The Track display allows for fixed head with scrolling tracks or vice-versa. On a Pentium 233 the scrolling is commendably smooth. The usual transport keys are supplemented by a performance key. Three buttons in the toolbar determine the action of this. The options are cut, copy and paste. The cue between the yellow and red markers, cut and paste which skips the marked area or copy and paste. This plays a snatch of audio up to the second mark then after the first. Rudimentary dynamic automation is achieved by choosing Draw mode. Once this is selected, a continuous envelope, which appears as a black line on the track display, defines the level at various points in time. Nodes are added by pressing the key or clicking with either the Head or Insert markers attached to the mouse pointer. Once a point is defined it can be dragged up or down to produce a change in level. Fades in and out are made by marking the area of the intended fade and clicking the appropriate icon. Crossfades are similarly achieved by defining the area of the fade, selecting the track(s) to fade in, and hitting the button then selecting the track(s) to fade out and repeating the exercise without changing the markers. A bit cumbersome and cross-fading overlapping material on the >
same track is not possible. A right mouse click over any of the virtual folders in Xtrack restores the setting to unity, a very nice touch.

With a Motorola 68501 on board the card you would expect some time-domain effects and the Digigram does not disappoint. Time stretch, pitch shift and sample-rate conversion are all offered as off-line processes. The resultant audio files either replace the existing or are put on the clipboard where they may be inserted wherever you wish. Other processes included are a broadband, noise print modelling, dynamic noise reducer and a noise gate that will break up speech into discrete blocks without user-intervention. For each of these processes there are enough controls to enable worthwhile results to be achieved with a variety of material.

T
HE SOFTWARE also supports ActiveMovie (Direct X) plug-ins. Much to my amazement and delight all my Sonic Foundry and Waves plug-ins appeared on the menu without any intervention from me. This delighted turn to frustration when I could see audio on the meters of the plug-ins, but not hear it when previewing effects. After rummaging through the online documentation and much head scratching and playing about I tried enabling the card as the default in the multimedia section of Control Panel. Lo and behold—preview audio. This is the fulfillment of the promise for anyone writing software with the right hooks can take advantage of any correctly written plug-ins. It is just a shame the on-line manual does not tell you how to get the preview working. Xtrack includes a rather nice 4-band parametric EQ, just to get you going. As with the other effects the results of any of these processes either replace the existing files or go to the clipboard.

Xtrack also includes some nifty applets in support of the main functions. Catalogue or CATS is a sound database that allows dragging and dropping of audio into Xtrack. It enables you to keep track of a vast library or libraries of sounds across many local or network devices. Sounds are playable from within Catalogue for auditioning purposes. This includes audio CDs, but via the audio output of the drive. The CATS manages to read the TOC (Table of Contents) and use the information in the database. In theory it would appear to be possible to drag and drop CD tracks into the editor; although I did not manage to divine how to actually get the audio from the on-line help. I have a suspicion this is only possible via an audio connection from the CD drive since this is the way it seems to work. I would be happy to be proved wrong since audio extraction from CD-ROM drives usually works well and is faster than real time. If the software can read the TOC why not the audio? CATS keeps track of CD changes and updates the display accordingly.

Search facilities are more than adequate to keep track of a large library. The second applet, Purge, removes sound files unused in the current Title. It does this in an intelligent manner and allows 'dry runs'. It will also generate reports on what has been done.

Test EDL management is also dealt with via an applet and another one, Syncrho, manages synchronisation of an Xtrack workstation when equipped with one of the optional sync boards. The sync board manages the synchronisation of all installed audio boards and provides a variety of options including RS232 (Novell P2 protocol) 9-pin.

The PCX4400NP card is a good performer with all the features required to integrate easily in a professional environment. The rest of the Digigram range has enough building blocks to produce a very serious DAW—if you can afford it. The real catch is that the cards look expensive when compared with the competition and do not offer sampling rates beyond 48kHz. Perhaps this is not unrelated to the emphasis on MPEF. At the time Digigram started producing hard disks were extremely expensive. Now they are virtually given away with packets of cornflakes, and I do not believe the complexities and risks of compression are worth the candle on a serious DAW.

In view of all this I cannot see the Digigram boards appealing to a mass audience. Where the company will continue to succeed is in the vertical market integration field. Radio, in particular, requires a range of dedicated solutions to problems not easily addressed by general purpose products. The catalogue of building blocks and the various software development packages all aid in this. Digigram are clearly geared to supporting this kind of endeavour and do it well. The relatively high cost of the boards disappears into the overall cost of the complete solution. In fact, the use of off-the-shelf boards and high level development tools will tend to reduce costs in these applications. This is really just an extension of the idea of using a PC for audio—why re-invent the wheel?

Xtrack software is robust and obviously the product of considerable development and refinement after extensive experience in the field. The price is thoroughly reasonable for what is on offer. Once I became accustomed to the Xtrack way of doing things it became possible to work quickly and accurately and it still manages to exude Gallic charm. I shall miss it.
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Yamaha DSP Factory

Yamaha's conception of the future of mixing consoles is further defined with the launch of its DSP Factory. Rob James takes a walk along the product line.

Once in a while, a product captures the imagination of almost everyone who hears about it. Yamaha's announcement, a few months ago, of the DSP factory concept and the initial cards was one such. Word spread quickly—24 channels of 02R on a PCI card plus hard-disk recording, for how much?

This specification, and the price alone, should ensure success but there is a lot more to it than this. The significance of the concept goes way beyond the obvious.

Perhaps even more remarkable in a world cursed with vapourware and late deliveries, a shipping date was announced and Yamaha has pretty much kept to it, and initial units are available as you read this. Yamaha's contribution to making digital mixing widely available is undeniable. The first affordable alternatives to analogue mixers in 1985, the DMP series, was followed by the well-regarded professional DMC 1000. The lineage continues through to the current 01V, 02R and 05D consoles and the company has continued to innovate. The current range of 'hardware' mixers are available at price points which other manufacturers are still striving to meet.

Yamaha has not achieved this by compromising its build quality or unnecessarily limiting the feature set. The remarkable achievement is that the sound is generally considered excellent, and the few operational compromises required are a small price to pay for what is on offer. Some people complain there are insufficient steps on faders, and other controls, but every piece of kit has its critics.

The DSP Factory takes hardware building blocks from other Yamaha products, specifically the LSI (Large Scale Integrated circuit) chips and offers them in a more affordable and potentially flexible form. Yamaha is currently only supplying the PCI plug-in cards and driver software for the appropriate hardware platform. Initially Windows 9X and NT on the PC, to be followed by Macintosh drivers. All the initial application software will be supplied by third-party vendors. This brings me to the one warning about the PC-Mac way of doing things. The reliability is unlikely ever to be of the same order as dedicated hardware products. Leaving aside any Windows, NT or Mac OS problems this is the price paid for the more flexible PC and Mac-based approaches.

Having said that, I had no problems whatsoever during the review period.

The first results of the DSP Factory concept are the DS2416 digital mixing card and AX-11 Audio Expansion Unit and the SW1000GX PCI Audio-MIDI card that combines an XG-series synthesiser with a full-duplex soundcard and MIDI interface. The SW1000G will have the option of three associated daughter boards which will provide a DX7, a VL synth or a high-spec time-domain, effects unit.

I had one of the first production DS2416 cards...
and an AX-1.1 interface. Installation is completely straightforward, simply a case of find a vacant PCI slot and plug it in. The driver installation poses no difficulties either. The only other software supplied with the card by Yamaha is a diagnostic program. After running the diagnostics as suggested, which takes just a few seconds, I tried using it as a simple I-O card with Sound Forge and it worked first time.

Fitting the AX44 is equally straightforward. This neat little unit attaches to the DSP factory card with one ribbon cable. A spare disk-drive connector is used for power and the unit conveniently mounts in a half-height 5 1/4-inch disk-drive bay. This leaves the connectors where you need them, on the front of the PC.

The DS2416 is a half-length PCI bus card populated with five large, surface-mount DSP chips and an even larger gate array, all of which are custom-designed Yamaha devices. Three of the DSPs are used for mixing, one for audio streaming and one for effects while the big custom LSI handles interfacing to the host computer's PCI bus. Apart from the rest of the board is remarkably sparse compared with the majority I have encountered. This level of integration makes for reliability and can only really be attempted by companies with huge resources. Onboard audio I-O consists of stereo 20-bit 128x oversampling A-D converters and stereo 20-bit 8x oversampling D-A's. These terminate on unbalanced, gold-plated phono sockets which are accessible from the rear of the PC or Mac. Nominal level is -10dBV.

Stereo digital IFC 60958 SPDIF I-O is provided on two further phono sockets. There is no external socket for wordclock input so sync is either taken from one of the inputs or derived from one of the internal connections. The 4 further connectors on the card >

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There are two common approaches to adding audio workstation capabilities to a general-purpose PC; both involve adding a card or cards. The difference is in the function of the card(s). In one approach, the card handles getting audio signals in and out of the computer. In this case, summing signals (mixing) or processing—such as EQ and effects—are undertaken by the main host computer's CPU. The alternative approach adds DSP to take some of the load off the computer when summing signals or processing. The majority of currently available cards following this approach use general-purpose DSP engines to provide a pool of processing able to do mixing, EQ and effects. Whichever route is followed the actual functions and ultimately the way the machine sounds is defined by mathematical algorithms. These algorithms are translated into sets of instructions to the hardware to get the required audio, and crunch the numbers to perform the intended changes. Obviously, this works, but there is a snag. A lot of power is required and this will vary depending on what function(s) are required concurrently with recording or playback of audio. It is difficult to guarantee a consistent level of performance using these models. Yamaha's approach is different—by using the same, purpose-designed, DSP engines as the OIV, O3D and O2R, it has provided a kit of parts to construct a digital console with all the signal-processing algorithms and the horsepower to carry it out built in, leaving software development to provide the user interface. There is a risk with this approach, if the concept is as successful as I think it will be, that the 'Yamaha Sound' will become too common. However, as with the overuse of synthesiser presets, this does not have to be the case. It largely depends on the intelligence of users and the laziness of programmers.

The result is 24-input channels, the first 20 of which are full channels with much the same facilities found on an O2R. This means, DC cut as required, an attenuator, 4-band parametric EQ with 12 EQ types per band, dynamics in any of 6 flavours with gain reduction metering, delay up to 2,660 samples, mute, fader, pan, channel meter and bus and aux sends with pre/post options. Channels 21-24 have everything except the programmable delays, and Channels 19 & 20 will automatically apply de-emphasis if required. The 16 buses are organised as 8 bus outs/sub groups plus a stereo master and 6 aux sends. The bus and aux masters can have a fader, mute and meter. The stereo output also has the same EQ and dynamics options as the channels. Input and output patchbays are also built in, again only requiring a user.
interface. On the input side Channel 1-8 sources may be PCI bus playbacks 1-8 or I-O connector B2 1-8 respectively. 9-16 source options are PCI bus, I-O B1, I-O A2, sub, and so on. This is, perhaps, a little restrictive as physical inputs are constrained to specific channel strips. On the other hand, I can already see a couple of ways around this that could be implemented in application software. On the output side any bus can be patched to any destination. When the card is used with software that is not specifically written to support it the patching assignments are fixed.

Two onboard effects units complete the picture. These use identical processing to the renowned ProR3 and Rev500 stand-alone effects processors. As you might anticipate, there is a vast palette of adjustable parameters, that again only require appropriate application software to manage them.

All processing is 32-bit or better. EQ uses a 44-bit data path. Sampling rates available internally are 44.1kHz or 48kHz (6%) or externally from 32 to 48kHz plus or minus 6%.

The first multichannel interface to be released is the AX44 expansion unit. This connects to the I-O A socket on the DS2416 card via ribbon cable. Two AX44 units may be fitted per card to add a total of 8 channels of analogue I-O. Sockets are unbalanced ½-inch jacks. Nominal levels are -10dBV, but inputs 1 and 2 are each equipped with a small sliding switch to change the gain to a nominal -50dBV that will facilitate the use of microphones and low level instruments. A further stereo jack allows headphone monitoring of Outputs 3 and 4. The available level is more than adequate and a volume control pot completes the panel.

In a conventional review we would...
One problem with using general-purpose mixers or DAWs in broadcast or installation work is they actually offer far too many options to the operator. Using this card it should be possible to quickly develop software or even hardware front ends with a reduced feature-set for applications where this is appropriate. Equally, by using two cards, more complex solutions can be conceived, designed and constructed. One of the ironies of digital implementations of audio devices is you are almost invariably paid more for machines which are only capable of a limited set of functions. All of this would be of no consequence if the hardware did not deliver—but it does. The audio quality is, so far as I can determine, the same as a current -set hardware console. The card works reliably, both as a PC soundcard, and with specifically written software. I generally distrust putting analogue audio anywhere near the noisy environment of a PC, but Yamaha has managed the difficult trick of keeping things quiet. Using the AX-4 I was achieving input-to-0-threshold ratios of around 92dB and the output was managed 90dB easily. I admit my measurements were not particularly scientific but the card is subjectively quiet which is the real point. Other interface options will undoubtedly appear, Yamaha has already announced an API-like interface and I am sure if demand exists interfaces will be provided to accommodate special formats, balanced analogue and AES-EBU.

The initial point of the DSP factory is the applications which previously relied on a heavyweight PC to provide a reasonable level of performance can be adapted to take advantage of the power of the DSP116. The implications are that this will lead to a raising of performance expectations at the price point. New applications will be written specifically tailored to the capabilities of the card. These will be both generally applicable and highly specific, I also expect to see further additions to the DSP factory range. The concept is clearly scalable and I wouldn’t be surprised to see an up-market console or workstation from Yamaha using the same core technology. Meanwhile, I think Yamaha has done it again with a concept that offers potential for growth in directions which at this stage we can only guess at.
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Lab Gruppen DSP24; dba audioView

Increasingly, audio amplifiers are becoming an integral part of a powerful new breed of DSP-based systems incorporating EQ and control into the amplification chain. **Terry Nelson** investigates the Swedish option.

The Lab Gruppen DSP24 processor with View software from dba audio made its appearance at PLASA in 1997 and can now be considered to be shipping. In its most basic form, the DSP24 can be considered a stereo 2-way or mono 4-way crossover-processor with 16 preprogrammed setups for different loudspeaker configurations. The unit is aimed at both the sound reinforcement and studio monitor sectors, so it is a shame that the unit lacks a dedicated mono subwoofer output as this would enable its use in 5-way surround sound systems.

The DSP24 is a 1U-high rack unit with the front panel divided into 3 main sections. The first features gain controls for Input A and Input B together with switches for the associated Clip Limiters; the second contains two 9-segment LED meters for the A and B inputs and 4 further 10-segment LED meters for the A and B low and high frequency sections (or Sub, Low, Mid, and High bands) together with an alphanumeric display and 7 status LEDs followed by two push-button for Setup selection (STEP and ENTER). The third section consists of 4 sets of level and volume controls for the Low A and B, High A & B outputs or Sub, Low, Mid and High outputs. Each output features a MUTE switch and the front panel is completed by a mains rocker switch at the far right. All rotary controls have a centre detent. The review unit was also fitted with the PCMCIA memory card option and this is situated over the centre of the output section.

The rear panel is simplicity itself and features PLRs for the inputs and outputs, a 9-pin XLRs socket for external computer control, an RS232 port and IEC mains connector.

The processor comes loaded with 16 preset configurations, ranging from mono 4-way to various versions of stereo 2-way, and can be used as a stand-alone unit. The different presets are selected by pressing SETUP and View (Visual Interactive Equalisation Workbench) is comprehensive digital filter design software for Windows 95/NT4 and enables the user to maximise existing equipment as well as aiding system design—it can be regarded as the other half of the DSP24. The program is supplied on a CD-ROM and includes both View and View Pro, the latter being targeted at contractors, installers and loudspeaker designers.

The disc includes both programs, a Help file, a tutorial overview and user manual. Loading the programs into the computer presents no problems, however, you do need to install the supplied dongle for the computer and DSP24 to communicate. As mentioned earlier, there are two programs—View and View Pro. The latter is aimed specifically at design and installation, and has the ability to import measurement files plus other features suited to design work.

Upon booting, it is a pleasant surprise to discover that the software is really quite intuitive and user-friendly. I did go to the trouble of printing out the user manual and the help files as recommended, but found only minimal reference necessary to get into the system. This said, I am still of the opinion that a good manual is worth its weight in gold, and while it might look ecologically correct to, supply everything on a CD-ROM, I am sure that more paper will get used up in unsatisfactory printouts and lost sheets. Having consulted on and written tech...
New delivery channels, including DVD, satellite, cable, digital TV and the internet, are providing an explosive increase in the number of routes available to deliver material to an ever-more enlightened audience. Demanding complex levels of audio format.

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There are two basic modes of operation for the system: Control and Design, and these are accessed from the opening page. The connection status with the DSP24 is shown by one of three icons in all of the control windows and these indicate Connected, Unconnected and New Ox (Operating System) required. We can note that the presence of an icon means that communication is open as no icon will be present should the DTE connection be faulty. Connection to the computer is initiated by pressing enter on the processor.

First-time users will want to see what is actually installed in the processor and the computer database, and this is two clicks away with Setup Configuration-Setup Explorer. The left-hand column displays the setups in the database and the right-hand column (of the DSP24 is on-line) the setups in the unit. A setup can then be selected as required and this can either be from the database or the memories loaded into the DSP24. Loading a setup from the database into the processor consists of a simple click-and-drag operation.

Once the setup is defined and loaded, the Setup Configuration shows a block diagram of the signal path and the settings of the various hand-pass filters, delays and equalisation. All settings are combined in the digital domain and implemented with high-resolution pseudo FIR filters for precise response control. The filter operation may be minimum, maximum or linear phase. At this point you can either use the configuration called up as it is or edit the settings. You can also design a new setup if this is required.

The Design Sequence takes you through the various stages and is, again, intuitive to use. The steps include: Loudspeaker, Amplifier, Target Band-pass Response for each output (all are totally independent), Limit (sets maximum SPL for chosen components), Target EQ (with an additional window for individual driver equalisation). Graphical display of group delay and physical offsets of the drivers. Analogue output gains for correct summation at the crossover points, and Exit (including setup name).

There are also a number of optional serial and parallel interfaces. All digital consoles can be programmed with serial, parallel or combined interfaces. These include two kinds of graphic equalisers, parametric equalisers, high-pass and low-pass filters and shelving filters.

When designing a setup, the loudspeakers and amplifiers may either be selected from the database or entered in as required. As might be surmised, there is a lot of automatic calculation by the software depending on the parameters entered and required (such as driver alignment delay) though manual adjustment is still possible. The program features numerous possibilities but points such as the ability to combine different crossover characteristics or set different hand-pass responses to have spaced crossover points are particularly appealing.

I was able to try the DSP24 under fire rather than just in the workshop — I programmed four different setups for a stage monitor wedge in a festival environment and was able to toggle the settings from 'standard response' through the variations user settings. The difference was quite amazing and shows what the system is capable of. Not everyone carries a computer to shows, and the fact that you can program off-line, as it were, and then select the presets on the job is very handy.

Space precludes a more thorough overview of the DSP24—View combination, but this report may have whetted your appetite. The system finds applications in all areas of sound reinforcement and for active studio monitors (which I would like to investigate further) and merits the attention of all who are interested in a hardware-software solution for design and control of loudspeaker systems.

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Power. 8Mb of RAM expandable to a colossal 256Mb for over 25 minutes of CD quality stereo sampling, 128 voice polyphony (standard on the S6000 and upgradable from 64 on the S5000), 26 different filter types, new 20 bit multi channel effects (optional on the S6000), stereo digital I/Os and the capacity to handle up to 16 individual analogue outputs which are also configurable as stereo pairs. Two MIDI In/Out/Thru ports give 32-channel multi-timbral operation and the two SCSI ports mean flexibility in a SCSI chain. A Wordclock connector allows the new samplers to be integrated into an all-digital environment and an optional adat™ interface provides stereo digital inputs and sixteen digital outs for direct connection to digital mixers.

Function. Record to RAM or directly to hard disk for seamless transparent replay. Recognising that .wav is fast becoming the world-wide standard for audio interchange the S5000 and S6000 use .wav files as their native sample format allowing files to be loaded directly for instant playback from any PC formatted hard or removable disks attached to the samplers. The ‘Virtual Sampling’ function lets you assign disk recordings to keygroups so that long recordings may be triggered directly from disk within the context of a program. Not only does this enable disk recordings to be processed via the sampler’s filters, LFOs, envelopes, etc., but a program containing ‘virtual’ samples appears in a Multi just like any normal program where it may be mixed, tuned and sent to the effects, elegantly integrating traditional sampling and disk recording. But we haven’t forgotten our existing customers as the new samplers will read S1000 and S3000 sound libraries as well, making the decision to up-grade your sampler even simpler.

Ease of use. Both models have a large 6” graphic display with all common user-needed data such as number of items loaded and available memory space shown in the centre of the screen but unique to the S6000 is the seductive removable front panel. 16 function keys read against on-screen parameter boxes and with a large data wheel for adjustment give a ‘Touch and Tweak’ system that virtually eliminates the dreaded cursor trawl. Extensive use of graphics, icons, pop-up windows, progress displays, drop down menus and the inclusion of a PS2 port for ASCII keyboard attachment for naming give a computer like familiarity. And for more intricate operation we added a ‘Window’ function to allow power users deep level access to the sampler’s heart.

Incredible pricing. Evolution has gone backwards here. The S5000 is £1799 (inc VAT) and the S6000 £2799 (inc VAT). Call for a brochure or visit your Akai dealer (look for the store with the queue outside).

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WHERE WERE YOU in 1972, when I was 12 and my birthday party became a true-blue-jean rock 'n' roll event? My mates and I played Sweet's Blackbuster over and over after Ginger Davies discovered that the tunable on my parents' teak radiogram would repeat indefinitely if you left the magnetic auto-change arm open; we ate salmon boots and sausages on sticks, drank Dandelion & Burdock and were blissfully unaware that there were only boys at our house that night. It was also the year that my Rover PSB car was built in Solihull, David Bowie became Ziggy Stardust, and Audio Engineering Ltd launched onto an unsuspecting world the Micron pocket radio transmitter, packaging it in such a beautifully rugged case design that it became the radio equivalent to the classic Coca-Cola bottle.

How many of these have stood the test of time? Well, my PSB has undergone extensive and expensive surgery, while certain cherished childhood friends were destined not to travel with me as far as Bulging Midriff, PLA, and I sure miss them not being around. But the Coke bottle is still with us, and so is the Micron radio transmitter, little changed in its appearance, but brought up to date by the introduction of new features including what many might feel to be essential for a UHF radio channel—a diversity receiver.

A UHF receiver, operating on shorter wavelengths than its VHF counterpart, is much more likely to be susceptible to the signal drop-out 'dead spots' that occur when direct and reflected signals cancel each other out at the antenna, an event that is possible at regular H-wave intervals, as the person wearing the transmitter moves. The advantage of a diversity receiver is twofold; with the opportunity to process the strongest of two—rather than—one signals through a comparator being the most obvious, but also where the two signals are of a similar strength this Micron 570 mixes the two inputs to the comparator to improve the signal-to-noise ratio and increase the output by a useful 3dB.

The 570 is designed to offer the location recordist maximum flexibility; although for the designer we should recognise that this can present considerable difficulties as different sound recordists will prefer different personal microphones, each with their own characteristics, so while Sony or Sennheiser, for instance, can tailor their systems to their own types of mic, as an independent manufacturer a compromise must be reached between popular makes like the Sony ECM 77, Trum T750, Sennheiser MK E2 or Sanken COS-11. All credit then that the Micron test units fitted with ECM 77's sat comfortably in a mix with Sony 810s also using their dedicated ECM 77 microphones.

If you have ever used a Micron in the past, then the 570 transmitter will feel like you have stepped back through time. It differs visibly only in the improved battery compartment lid, now hinged on the single 9V transmitter model, rather than being secured via the reluctant 'tearaway' screwdriver clasp of the double-battery version, and the rotary frequency change switch situated on the front face, while the features available are unchanged with the proven limiter-Automatic Gain Control option available from a recessed set level switch located on the top face of the transmitter beneath the line-up tone button. When set level is depressed, the AGC system is disabled and the modulation level LEDs alongside the tone button act as a simple volume level indicator. My starting point with this setting is to light the -10 LED with normal speech and just cause the 0 LED to blink on higher volumes;
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adjustments to the input level being made by a recessed pot labelled 'set level', underneath the unit and calibrated 1-7. Depressing tone causes the transmitter to radiate a line-up tone at 2dB below the limiting modulation level and like the set level button, it may be latched into the down position.

The transmitter is fed with audio via a 8-pin connector offering the options of a high-sensitivity input for dynamic microphone, condenser microphone, a direct powering connection for all popular electret microphones, powering for 12V T-powered condenser microphones, powering for 12V phantom powered condenser microphones, a dedicated low-sensitivity (-40dB) input, for use with high output condenser microphones, a high-pass filter—to avoid overload from wind rumble or from vocalists' plosive poppings, a line input and the opportunity for external powering with a suitable input lead, but a word of warning to existing Micron owners—the earthing of the transmitter connector has been changed with this model from positive earth to a more conventional negative earth to take care with your older configured leads.

The overall frequency response of the transmitter is superb to the ear and is specified as being commendably flat at 2dB between 100Hz and 15kHz, a signal-to-noise ratio in excess of 100dB is possible given a full signal of 500µV and a current drain to the 9V battery of around 50mA, gives a useful operating life of about seven hours between changes.

The BIGGEST differences come with the receiver and its associated double-antenna housing, the DDH2 is designed in conjunction with radio mic specialists Mattissen in the Netherlands, and offering the opportunity to hold two Micron SDR Small Diversity Receivers with one pair of antennae feeding both receivers via the built in Band Pass Filtering and signal splitters, with additional RF-DC outputs that allow multichannel operation from a single pair of antennae. The DDH2 can also power the receivers via the antenna sockets, obviating the need for internal batteries. With these useful facilities of separate power switching for accessories, such as a mixer or a DAT recorder, and internal balanced transformers on its line-level audio outputs, the DDH2 housing lets the side down by its agricultural appearance of a black steel case, unmarked connectors and seemingly vulnerable connector cables.

The whole receiver itself, apart from the frequency selector slider-switch on its front face like the transmitter, carries its controls on the top face of the device. In the centre is a 6-pin Lemo multipin connector that provides the audio outputs and DC input connections to the receiver, flanked on the outer edges either side by the A and B 50Ω Lemo coaxial antenna input sockets. The receiver can also be phantom powered from a 7-15V DC supply via these sockets. A recessed high level output volume control is screwdriver operated and situated below a 1dB push-button that when operated changes the LED display to register internal battery cell voltage.

What Microns have always offered is comprehensive feedback monitoring for both the transmitter and receiver, albeit in a unique version of 4-light versaphone (see there’s four), which is brought up to date by the use of 3-colour LEDs albeit in the familiar star configuration. The display offers four indications: received signal strength (in normal use the height of the display indicates the received signal strength with the bottom pair of LEDs acting together); tuning (the bottom pair of LEDs act as a tuning indicator and show if there is any mismatch in the transmitter and receiver frequencies as normally both lamps should be lit). With the transmitter off, the display shows the presence of an interfering signal; transmitter battery warning (when the battery voltage in the transmitter falls low, a sub-audio tone is emitted to the receiver causing the bottom two LEDs to flash alternately 2-3 times per second); receiver battery condition (when BATT TEST is pushed the height of the top LED column indicates the receiver battery voltage).

Back in 1972, the Micron cornered a huge slice of a small market by being the first and best for location filming particularly when tucked away in the front pocket of a leather Nagara case. With the widespread introduction of tape in the 1980’s, and the huge growth of out-and-about, portable single-camera shooting, other players subsequently entered this lucrative arena and have successfully challenged the Micron product on quality, features, ease of use, flexibility and price.

So did video kill this radio star? Nearly, very nearly, but not quite, and this latest 2-frequency crystal version embodies a purity of processing that none of the more flexible synthesised systems can offer due to their fundamentally higher inherent noise. But that clarity comes at a cost, and the price to be paid is the sensitivity of multiple channels available in hand, should the need arise, afforded by a synthesised system. However, for a purist that may not be considered too high a price to pay.
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Custom and modular are terms frequently associated with desks,

**Dave Foister** discovers a Lego approach to the business of mastering

It ought to be very simple. Managing half a dozen sound sources, routing them to various destinations with or without processing, and listening to the results—effortless. Yet outside the confines of a mixing console, most of us will have encountered all too many fash-ups and tangles, where even a patchbay does not seem to manage the job of interfacing a set of disparate boxes properly. This is becoming more common with the widespread use of DAWs, with various sources, recorders and processors vying for access, but the basic problem has always been best demonstrated in the cutting and mastering room. Here there is no mixer as such, just a bank of playback machines, a selection of EQs, compressors and limiters, and destinations ranging from a workstation to a cutting lathe. Fast, flexible and reliable routing and monitoring around this lot has always been vital, yet there is almost no off-the-shelf equipment available on the market to deal with it. Or there wasn't.

Crookwood's modular approach to, well, anything is well known, as is its Home Improvements nomenclature. Its Paint Pot became better known than the one on Phil Collins' piano, and the idea of modular consoles assembled from bricks was forecast some time ago. Now the concept has become reality in the form of Crookwood's Mastering Bricks. I visited London Facility Tape to Tape to see the system as installed in George Lambert's cutting room.

It is important to make the point straight away that the Tape to Tape installation is just as much a one-off as any other Mastering Bricks console would be. The system has been designed to be as flexible as possible, with a wealth of configuration and operational options available. There is no such thing as a 'standard' console, which means by extension that there is no premium to pay for having it just the way you want it. Every one will be a custom build with differences coming down largely to software programming and simple printed panel inserts.

The work is done in an anonymous rack that can reside anywhere convenient to minimise cable runs. This can accommodate a number of cards to carry out the switching, routing and other processes, and is connected to one or more control panels via serial links. Since routing is the primary function the cards' main job is switching, and this can be done in the analogue or the digital domain. Digital crosspoint switches operate with AES or SPDIF signals and can be ganged together up to a 64x64 matrix, and for analogue signals the switching is done with relays rather than any active electronics. Analogue outputs are, of course, buffered to allow the system to feed multiple destinations—a glorified distribution amplifier—and Crookwood's analogue expertise entitles them to feel confident of the signal integrity. A wide range of cards is now available, covering options such as microphone preamps, A-D-Cs, meter drivers, and monitoring for up to 8 channels.

The user need never see the rack; operation is carried out by small stylish gold control panels, that come in a few distinct types and can all be customised to suit the application. Tape to Tape has four cards in two basic types, with three selection panels, and one to handle the monitors. The button layouts are similar, and, although the three selection panels perform quite different functions, they are mechanically identical. Each has 16 selector buttons with a display panel showing sources and destinations, an ENABLE button for safety interlock on certain functions and a button for invoking additional functions. Where these are fitted they are shown on the LCD screen at the top of the module, which otherwise shows messages chosen by the user. The obvious custom element is the labelling panel, but as this is printed to order in houses—no silk-screening or fancy metalwork—it's simple and cheap to have what you want and even to change it afterwards. Even so it looks very slick, slipped beneath a transparent panel and backlit to show current status.

The real customising is in the software inside, and, again, the whole setup is designed to deliver maximum flexibility. The Tape to Tape system shows some of the possibilities simply in the three different uses of the selection panels.

One selects a dubbing source for transfer, and routes it to all the available destinations with the exception of the chosen source's own input. Obviously this avoids feedback in any situation, but when a cutter head is on the end the results of a mistake could be costly. The second panel deals with inserts in the chain, and handles both analogue and digital devices transparently. Tape to Tape has an interesting mixture, and the units can be inserted in any order, the only limitation being that the digital stuff must come first. This is partly because the final system output is analogue to drive the lathe, and it makes efficient use of the available converters. More converters would increase the flexibility still further. Note that button pushes can do more than simply switch a signal; effectively each button runs a macro that can include things like automatic wordclock routing and momentary mutes to cover resync glitches.

The third selector panel determines what will be heard on the monitors, and can take its feed from the chosen transfer chain or any of the other machines. If the chosen source is digital a D-A converter is automatically inserted, and since Tape to Tape has two Prisms the monitor selector will use whichever.

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is not being used by the insert chain. This signal then comes under the control of the monitor panel, that has buttons for selecting which set of loudspeakers or cans is to be used, and a large black volume knob with a ring of tan to show its position. Level matching between monitor outputs is preset, and buttons are provided for QUT, DIM and LFT functions. DIM can be set to be absolute (always at the same level regardless of the volume control setting).

Provided everything is properly labelled it puts a whole room full of equipment under the fingers for fast setup and reconfiguration.

or relative, and LFT gives a similarly preset high level for checking tails of fades and other low-level details.

All the panels have an ENABLE button and its function is determined by the software programming. For safety’s sake, the transfer source selection panel can only have a new source selected after the ENABLE button has been pressed, and this is also necessary to change the insert configuration. The two panels relating to monitoring have no such risks attached, so their buttons are always live. Even here, the buttons can operate in different ways, perhaps requiring both buttons to be pressed together or possibly a sticky shift type of arrangement. This kind of setting up is performed quickly and easily by Crookwood to suit the individual user.

Put all this together and you have a system that is intuitive, neat and powerful. Provided everything is properly labelled it puts a whole room full of equipment under the fingers for fast setup and reconfiguration, in a way that seems so simple you wonder why you have not seen it before. But the possibilities do not end there: there are modules available with faders, P&G belts, meters, and controllers for mic amps, EQ and compressors. The whole lot is surround ready, as a control panel can be linked to as many as eight signal paths in the rack to switch them and monitor them simultaneously. Eight-channel linked compression is on the way, and cards are already available for surround down-mixing.

The control panels are so small and neat that the word console almost seems inappropriate even for a 4-panel system like Tape to Tape’s. They will fit four across in a 4U rack chassis, and also have individual pods available for table-top use, but the most popular arrangement is flush mounting in studio furniture. This is how Tape to Tape have them, in a 2x2 layout, and in among the other equipment it almost disappears. This, of course, is quite an advantage, as everything required is right under the hand without taking up large amounts of real estate in the monitoring sweet spot.

The possibilities of the Mastering Bricks are seemingly endless. A simple one-panel controller for a workstation set-up would be very reasonably priced, and just as customisable as a larger system, while the power and convenience of a big surround system would be hard to match with anything else. New applications and special requests are emerging all the time, and once Crookwood has done something new for one customer that option is available to anybody else. Tape to Tape is currently installing a second Mastering Bricks system in a new room, which speaks for itself, and I came away wanting one of my own for sorting out the equipment around my SADIE system, as the savings in time and frustration were immediately obvious. Its aims are simple, but in terms of flexibility and delivering what is really needed it breaks new ground.

Thanks to Tape to Tape for the opportunity to spend time on the system preparing this review.

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tc electronic Gold Channel

Amplify the mic traditionally and then convert the signal for processing.

Dave Foister investigates an elaborate digital voice-channel of sorts

WHAT GOES around comes around. For years analogue manufacturers have had to explore digital technology as it marches relentlessly on, and it sometimes feels as though before long the only analogue gear we'll have left will be the old stuff. But, of course, it becomes ever more apparent that there is a need for both, and occasionally we are treated to the sight of a digital specialist walking the scenic route—diversification works both ways.

tc electronic is known almost exclusively for its digital expertise. From benchmark delay lines to effects and digital dynamics control, and even software plug-in versions of its hardware, it has done everything the numbers way. While its products can provide all you need from digital conversion to recorder and back again, they can not quite complete the whole chain; before you start you need to preamplify a microphone, which has become a specialist area all to itself. tc has decided to fill that gap and take on the specialists, and the result is the Gold Channel, combining tc's attempt at high-spec mic pre with its established digital processing capabilities. This take on the established voice-channel concept contradicts the usual format by having no valves, no reliance on fancy analogue EQ, and no opto-couplers; it goes digital at the earliest possible opportunity and does all the hard work there. In the process it provides all the digital advantages of program memories and compact size; although those who expect a voice channel to be 3U-high and punctuated with black knobs may feel slightly short-changed.

A brief run-down shows just how much is squeezed into the 1U-high box, which not surprisingly resembles the Finalizer and the M2000. Its front panel is silver rather than black, and its nature demands more controls, but otherwise it clearly carries the tc electronic hallmark. Two fully fitted mic pres are followed by 24-bit conversion, and this leads to a broad palette of digital processing including various flavours of EQ, compression, limiting, expansion, and gating. The analogue front end has the expected collection of discrete knobs and switches, while the rest is controlled with tc's familiar setup of on-screen adjustment.

The microphone preamps themselves are straightforward yet comprehensively equipped. Microphone and line-level inputs are provided, and the full complement of switchable phantom, pads and polarity inversion follows. The character of the preamps is quite anonymous; although this is meant as a compliment and I believe tc would take it as such. This is not a preamp with a character to impose, rather a clean accurate circuit that simply sets out to deliver the microphone signals to the converters, and this it manages very well with good noise, distortion and frequency response figures. Gain adjustment is on a single small knob, and three different high-pass-filter frequencies are offered. A switchable soft limiter precedes the converters for safety. Each channel has a switch, and this is also operative when a digital input is selected.

Digital signal handling is equipped to allow either a single process on both channels or three processes on one channel only.

The available processing algorithms encompass a comprehensive spectrum of treatments, from the obvious to the obscure. Two types of EQ are available, described as Easy and Advanced, and the difference is just that—Easy gives high and low-shelving equalisers with one swept mid and filters, while Advanced is a 5-band EQ with shelf-selector selection on the end two, and variable slope or width on all five. Boost and cut is a powerful 18dB.

Two dynamics blocks are available simultaneously, one for expansion and the other for compression-limiting; although it should be noted that there is no separate limiter. Like the EQ, two flavours are offered for each: the expander has a choice of Easy Gate and Advanced Expander, the latter having variable time constants and gain reduction ratio, while the compressor can be configured as Soft or Vintage. The first designation is slightly misleading as it has a hard knee available, but its point is fully variable parameters across the

This is not a preamp with a character to impose, rather a clean accurate circuit that simply sets out to deliver the microphone signals to the converters, and this it manages very well for high-resolution recording. The onboard converters will only run up to 48kHz, but they are 24 bit and the digital path can handle 96kHz, albeit with restrictions on the amount of processing that can be used. At 44.1 or 48, four processes can be chained together on each channel, either in stereo or as two independent paths. High sample rates board, with everything you might expect including the possibility of using channel 2 as a side-chain input for channel 1. This allows the channel 1 input to be split across the two paths and the channel 2 EQ used for controlling the compressor operation. The Vintage mode removes the knee selection and the side chain, and, most importantly, the Threshold control; the threshold is fixed at -24dB and the degree of compression is set by the Input Drive level. This is quick, dirty and very effective, and the two alternatives cover just about everything you're likely to need. In this context in particular it's worth noting that if both channels have the same specific algorithm placed at the same point in the chain, the two can be stereo linked for setup and proper dynamic control. The last block of processing, described as Tools, offers a more varied choice of special treatments. One is a clever de-esser that relates its threshold setting to the average level of the material, making it, according to the

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equaliser

The Gold Channel is not a "souped-up" Finalizer. It does not have the multi-band dynamics that make the Finalizer so powerful, but the comprehensive mix amps and bigger choice of processes mean it is not only a superb complete microphone-to-recorder signal path, but an intensely desirable general-purpose processor as well.

I have sometimes found it frustrating that the Finalizer, for all its range of analogue and digital inputs, cannot use some of them as splits or inserts. This is put right in the Gold Channel.

shows what is happening in inimitable style with clever combinations of text and graphics. Here the overall configuration can be built up as a set of blocks, and editing a particular block can be invoked either from the screen's associated buttons, or from the function buttons on the channels themselves: these use a clever idea I've not seen before where the buttons that select or bypass the steps in the chain can be double-clicked to call up the relevant setup display. For such a small screen it is remarkable how much it shows and how effortlessly it is to navigate it once you know what's on offer.

One hundred factory presets are provided, and the intention that the unit should be used on individual tracks as much as complete mixes is evident from the names, which include EQ and compression settings for a range of classic mics and micphones. Besides the 200 user presets are available, and up to 999 more can be stored on a PC card. For any stored setting, any individual block can be recalled to have other bits added, as well as full recall of everything.

In the Gold Channel, inserts can be added, but they cannot be used as splits or inserts.
Class A
The heart of the Red 3 is Focusrite’s proprietary Class A VCA technology which delivers outstanding low distortion performance to both Compression and Limiting functions, whilst increasing the compression ratio provides the punch and warmth from second order artifacts.

Dual Mono or Stereo
Red 3 is a two channel device switchable to true stereo operation under the lower set of controls, so you can use it for recording (great on vocals and instruments alike) and across the mix to make it jump out of the speakers.

Extraordinary Build Quality
Red Range build quality is the envy of the industry. The extruded aluminium front, side and rear panels are milled, finished and anodised a rich burgundy red whilst the legends are anoprinted into the finish so they cannot wear off. The brushed stainless steel covers conceal glass-fibre circuit boards, the shielded power supply and the wealth of electronic components employed to ensure reliable, outstanding performance, for life. High grade potentiometers and switches complete the bill of materials. Even the control knobs are individually machined and hand polished before anodising to feel smooth as silk to the touch.

It’s no accident that the Red 3 was awarded the 1995 TEC Award for Outstanding Technical Achievement - many of the nominators actually own one!

Make an informed judgement. Before buying your next compressor ask your pro-audio dealer for a Red 3 to evaluate or ask him to set up a comparative demonstration. There’s no substitute for your own ears!
Spendor SA300

Studio Sound's 'bench test' loudspeaker reviews continue with the SA300. Keith Holland reports.

The Spendor SA300 is a 2-way active loudspeaker with a built-in amplifier and crossover package. The low frequencies are handled by a nominally 200mm diameter cone driver, and the high frequencies by a 25mm soft-dome tweeter that fires through a shallow horn waveguide. The cabinet has dimensions of 470mm x 340mm deep, a slot-shaped port on the front panel beneath the woofer, and front panel edges that chamfer towards the top of the cabinet to a depth of 34mm. The rear panel has a large heatsink with vertical fins, an IEC-type mains socket, an XLR-type input socket, and controls for gain, LF cut, LF tilt and HF tilt. The LF cut switches the cut-off frequency of a 'subsonic' protection filter to either 50Hz, 70Hz or 90Hz. The LF tilt gives up to 3dB boost or cut to frequencies below 500Hz, and the HF tilt to frequencies above 4kHz, both in 1dB steps. The measurements presented in this review were carried out with the LF cut control at 50Hz and the tilt controls flat.

The on-axis frequency response (Fig.1) lies within ±3dB between 40Hz and 19kHz; a commendable result. The low frequency response rises by about 3dB at 50Hz before a very steep roll-off through -20dB at 30Hz—a consequence of the use of a ported enclosure and a high-pass protection filter. The acoustic centre result (Fig.2) confirms this rapid roll-off with the low frequencies suffering a group delay that corresponds to a shift of about 3m relative to the high frequencies. The rise at 50Hz may be tuned somewhat in practice by adopting a higher frequency setting for the protection filter. The harmonic distortion results shows levels below -10dB (0%) at low frequencies but the third harmonic rises above this level between 200Hz and 1kHz with a further, narrow peak at 2kHz. There is also a narrow peak in second harmonic at about 6kHz to -10dB.

The horizontal directivity (Fig.5) is well controlled with only a slight narrowing in the upper frequency range of the woofer and there is no evidence of side-lobes in the directivity pattern at any frequency. The vertical directivity (Fig.6) is similar except for the inevitable dip at the crossover frequency due to the physical spacing of the drive-units.

The step response of the loudspeaker (Fig.3) shows good time-alignment with a high frequency rise occurring only about 100μs earlier than the main response. The power cepstrum (Fig.4) shows only low level and time-smearred echoes at about 500μs and 750μs. Cabinet edge diffraction problems appear to be kept to a low level by the adoption of chamfered edges and the short tweeter horn—a result that is confirmed by the smooth on-axis frequency response. Fig.7 shows the waterfall plot for the SA300. The most dominant feature of this plot is the ringing at low frequencies due to the rapid roll-off. At higher frequencies though, the plot shows rapid decay except for some slight ringing. The third harmonic at 300Hz and 400Hz.

Overall the Spendor SA300 is a very good performer. The on-axis and off-axis frequency responses and time response are commendably good, but the mid-frequency harmonic distortion is higher than that found in some similar loudspeakers. The rapid roll-off at the low frequencies seems to be very common in compact monitor loudspeakers of this type; it is, perhaps, a necessary compromise where good low frequency extension along with reliably high output level is required.
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**NEW TECHNOLOGIES**

**Dolby Digital price slash**
Dolby has slashed the price of Dolby Digital encoding with the release of the $5,000 (US) DP569 at the San Francisco AES, and paved the way for entry-level multichannel working in light of the promised explosion in DVD and DTV. The new encoder costs some 74% less than its PC-based predecessor, but has even more features.

Dolby, US. Tel: +1 415 558 0293.

**Soundfield goes 5.1**
Soundfield is to unveil a surround-sound processor that enables the conversion of B-Format information generated by Soundfield microphones into 5.1, Pro Logic, and other surround formats. The 1U-high processor makes it possible to mix live surround broadcast or recordings from a single SoundField MkV System or ST250 Portable System. Four high-resolution bargraph meters display the incoming B-Format information with individual bargraphs showing the variable front left, front right, centre, rear left, rear right, and sub-bass output levels. Further stereo 'front width', 'rear width' and 'rear focus' controls are provided to compensate for speaker positioning and room acoustics.

Soundfield, UK. Tel: +44 1192 201 1089.

**Sonifex reds**
Sonifex has released a range of budget Redbox interconnection units. The range includes the RB2DA6 6-way stereo, 2 x 6-way mono or 1 x 12-way distribution amp, the RBMA2 dual mic preamp, RBMSM2 dual stereo to mono converter, RBBL2 balanced to unbalanced bidirectional converter, and the RBLS2 twin mono or stereo limiter. All units are housed in aluminium boxes which can be screw-mounted to any surface and can be powered by 115V, 60Hz or 230V, 50Hz.

Sonifex, UK. Tel: +44 1933 650700.

**Drawmer digital DC Line**
Drawmer is to introduce digital products for 24/96 processing and analogue-digital conversion at the AES. Three products will offer 'refined analogue' sound quality to TDIF, ADAT, SPDIF and AES-EBU interfaces plus a host of features. The DC2476 Digital Mastering Processor has 24/96 inputs and outputs, and the inclusion of 5-RAM card slot allows settings and parameter to be memorised and recalled on any Drawmer DC2476 or Drawmer DC2486 units. The Drawmer DC2486 TwinScreen Processor is a digital mastering 2U-high processor.

Studio Sound September 1998

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**dbx DDP**

Renowned for its analogue units, dbx recently introduced its first digital dynamics processor. George Shilling puts it through its paces.

DIGITAL REVERBS AND DELAY are far more prevalent than their analogue forebears, yet in the area of dynamics the analogue design reigns supreme. Only very recently have any significant inroads been made into this market by digital processors and this has mainly been restricted to all-digital setups. The DDP (Digital Dynamics Processor) is the first stand-alone digital processor from dbx.

The latest dbx Type IV 24-bit conversion process is included, which does away with the 0dB brick-wall concept, instead letting you operate after the limiter. All processes take their key from the initial input, less than the output of the previous element in the chain, so that tweaking one part does not affect another section's key level. Unfortunately, there is no separate side-chain input, so the only way to use an external side-chain is to sacrifice an input. The two channels can be used as stereo if you select a 'linked chain' effect blocks. Alternatively, two completely separate mono chains can be used.

The EQ is a 3-band parameteric with bell curves. A bargraph displays the changes. The Gate includes a ratio control rather than a range in dBs providing treatments from gentle expansion to brick-wall gating. The Compressor is the most fully featured section, including 10 levels of Over-Easy (soft-knee) and Hold as well as Attack and Release. the last unusually calibrated in dBs. TheDe-esser and Limiter sections are fairly simple to operate, the De-esser working particularly well.

The front panel features up meters switchable for Input or Output level, accompanied by 4 stepped rotary controls for input and output trimming of both channels. These controls have a wide range of adjustment, so no switching for -0dB to +4dB operation is needed. A large illuminated cross-fader dominates the front panel. This contains a huge amount of information, some of it in tiny lettering, and not as contrasted as it might be. There are additional meters on screen, a large program number indicator, and a whole host of other information, depending on the mode. When adjusting any dynamic processes a graph indicates all thresholds and limits. Alternatively, you can display real-time threshold indicators for all dynamics processes. A Data Wheel is used for parameter adjustment and takes you through the 50 preset programs: an additional 50 are available for storage of user settings.

The multidoit presets are somewhat of a waste of time as they always need tweaking for a given situation. The bank of push-buttons include a program button, that really should be larger or more obviously located. Edit buttons for each of the processing functions, and a select button to choose between up to three visible parameters. Rear page and priority page buttons take you to further parameters. The whole system is simple to navigate, but it is extremely slow to use compared with any dedicated analogue unit.

The DDP is a powerful package, priced very competitively. However, some ergonomic features leave much to be desired. Subjectively, the sound quality has something of a 'closed-in' digital sound to it, compared to the warm openness of the best analogue units. However, dbx has done its best to make it sound as good as, if not better than any similar digital processor.
Alesis LX20

Lowest of the new Type II breed but better than the old Dave Foister goes 20-bit again

Like BUNES, you wait ages then three come along at once. Alesis jump to 20-bit recording on ADAT is an assured commitment, with a complete product range from the outset. We have already seen the XT20, adding 20-bit capability to the established XT, and we've looked at the M20, which takes a leap into the major league with more facilities than ever seen before on ADAT. But it would be a betrayal if Alesis were to neglect the entry-level end of the market that helped so much to establish its position, and with that in mind there is now the LX20, bringing us affordable 20-bit ADAT streamlining for the application.

The LX20 could look cheap, but it does not. In fact it looks sleek and desirable than the original ADAT, and, although it is lighter than the XT20 it still seems to have a solid construction, and good-quality buttons and controls. The back panel shows the most obvious sacrifice for the first time an ADAT has no balanced pro-level analogue connections. The EDAC is gone, leaving only the phones that the original machine. The main display window shows the familiar set of information, with the eight big meters surmounted by the time read-out, and flanked by status indication such as sample rate, input selection and monitoring mode. To the right is the 5-VIB transport slot, with a couple of rows of soft-touch buttons beneath. These deal with the setting of the above functions as well as the basic set of four locator positions, it comes as a surprise to find that the whole system of looping and auto-punching is carried across to the LX20 complete with rehearsal mode. The most prominent controls are those for the transport, which are large and responsive with indicators above. The mechanism they control is clearly faster and smoother than the one in the original ADAT, like that in the XT20, although it seems to make a little more noise. This may be because of the lighter overall construction of the machine, which is fairly flimsy, but still apparently robust. Going from wind to play is fast and smooth, but the meters always overshoot several times before setting in the required place, much like many analogue multitracks.

The quality of the recorded audio is what this whole range depends on, and like the XT20 and M20 the LX20 uses the Type II ADAT format which records 20 bits on tape. It's at the bottom of the tree as far as on-board converters are concerned, the M20 is 24-bit both ends, the XT20 has 24-bit ADCs but 20-bit DACs, and the LX20 has 20-bit conversion both in and out. The sound through these converters shares the very slight brightness of the other machines, but overall it is quite astonishing that 8 tracks' worth of this quality can be had at this price. This is what will score heavily in the smaller installation, where the feature complement of the bigger ADATs will not be required, but this level of audio fidelity will be a huge bonus. One or two of these hooked up to a computer-based system, via one of the simple synchronisers that use ADAT's on-board sync code, would be a very impressive set-up indeed.

Alesis is clearly out to make a clean sweep of the market with a machine for everybody, placing 20-bit MDM on the agenda in all areas of the business. The new range certainly seems to offer the tools for the job and the LX20 sets the seal on a suite of machines that should make the technology available to, and appropriate for, everybody.

Contact
Alesis, US. Tel: +1 310 558 4530. UK: Sound Technology. Tel: +44 1462 480000.

Euphonix multitrack
Euphonix will unveil the 81 multitrack digital recorder at the AES that is expected to start shipping in the first quarter of next year for around $25,000 US for a 24-track system with MADI interface and remote control. The system claims 24-bit conversion and storage combined with 40-point floating.

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CD dupe
Otari's CDPS50 CD duplicator is aimed at short runs, requires no external PC, and can continuously copy 50 CDs using pre-installed screen-operated application software. In auto mode, master CDs and blanks can be mixed in the copy stack, the machine automatically detects prerecorded material and will continue to copy blanks until it detects a new master allowing multiple small runs to be copied in one pass. Features include write verification, error disc detection, and a separate reject disc magazine. The machine supports CD-DA, CD-I, CD-Video, CD-ROM, CD-ROM-XA, CD-Plus, enhanced CD and Photo CD. Expanding on the stand-alone Lightwiner LW10 system for PA use, the LW50 fibre-optic wiring system is intended for uses where setups have to be changed rapidly and stored. It also supports video camera signals. Lightwiner uses fibre-optic cable to transmit 128 channels of up to 24-bit, 48kHz audio 5km between units and 16 units can be chained together for a maximum transmission length of 48km. LW50 includes control and off-line editing software for PC remote control of routing, mic-line gain and setup. Interface options include video, A-D and D-A, AES-EBU, 4-channel intercom and control data 1-0.

Otari, Europe. Tel: +49 21 5905861.

September 1998 Studio Sound
Menus belong in restaurants not on mixers

Spirit's unique E-Strip avoids the need for tedious menus, so 328 is as easy to use as an analogue desk.

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CLM Dynamics DB200S

Less than four, but better than one, Dave Foister discovers a dual channel version of an established product

It's difficult to come up with something that sets your microphone preamplifier apart from the competition, but CLM Dynamics is determined to give it a go. A while back (Studio Sound, June 1997) we looked at the DB400S, which scored extra points for putting four preamps in a 2U box, and now here is the DB200S, essentially two of those preamps with a few modifications. While the attraction of the four-pack has gone, other details remain to mark the unit down as a little unusual, notably the built-in M-S decoding.

Its appearance, for a start, is mildly functional rather than sleek. Green panels on a black background, simple plastic knobs and a slightly muddled screen print do not suggest the employment of a style consultant, for which these days we should perhaps be grateful. Instead we have a practical design that incorporates all the standard functions expected of a microphone preamp, but with a couple of extras. The functions and facilities are, by and large, self-explanatory and straightforward.

The main inputs and outputs are on rear-panel XLRs. The inputs are for microphone signals only, which is slightly surprising in view of the M-S facilities, and the nominal output level can be switched independently between +4 and -10. A helpful extra here is an insert point for the addition of external EQ, compression, or other processing not provided on board. This is a standard unbalanced 3-pole jack. Otherwise the back panel is pretty sparsely populated, with only the mains input and a thoughtful signal ground lift switch to break up the monotony.

For its operation each preamp depends on three rotary controls and a bank of seven push-button switches. Two of the knobs are for input gain and output level; an output fader often seems superfluous on a simple preamp like this, placing an unwanted source of misalignment and potential trouble in the signal path, but in this case it can much more easily justify its presence in the context of the M-S decoder. CLM meters show final output levels, and as before only go as far as +9. This is apparently referenced to 0dBm, so full scale on here is +11.5dBu, but this is still some way below the levels one would be wanting to drive into the destination machine (unless you are calibrated to 0dBFN+12dBu). With most machines on the end the preamps' meters are pegstoping most of the time, rendering them less useful than they might be.

All the standard switches are present; 48V phantom power is independently switchable, as are 20dB pads, polarity reverse and a high-pass filter. Rolling off 12dB per octave below 8kHz this is not too aggressive, but just enough to deal with the odd tumble, hum and close singer. Independent muting for the two channels is useful in the kind of direct-to-tape situation the unit is intended for.

Purism demands the minimum possible in the signal path, so there is no EQ, and no compression, when the DB200S does have is, perhaps, the only process allowable on a box like this as it can save an awful lot of red faces: a limiter. This is only turned off by taking its threshold control to maximum, so you're taking it on trust that it is effectively out of circuit, but when it's introduced by reducing the threshold it does a neat and tidy job of keeping the lid on the levels. There is a switch for giving it a fast attack, and this provides all the flexibility it needs.

An ion at the top of the signal meter shows when the limiter is doing its job and unless it's horribly overdiven this is the only way you would know.

Channel 1 has a switch for gainng the limiter side-chains for stereo or M-S operation, and the corresponding switch on channel 2 selects the M-S mode. This allows an M-S array to be converted to L-R stereo here within the preamps, with full control of the resulting stereo image. The M signal is connected to channel 1 and the S to 2, and decoded stereo when the output is the right way round this time, unlike its predecessor. This means that the level controls on the S channel adjust the width of the stereo output. The stereo linking of the limiters works in this mode as well, which is thoughtful: if it didn't, limiting of either input signal would result in a widening or narrowing of the image.

I would be interested to know how many people need to convert M-S to L-R at this stage. I know many classical recordists use the technique, but for location work the whole point is to record the raw M-S signals (perhaps with a decode matrix for monitoring) and sort out the stereo back at base. Here, perhaps, CLM has missed a trick by not allowing the unit to accept full line inputs for straightforward decoding. At the same time, for those who don't want stereo straight off the mix, the DB200S can be used as a single-gang solution that provides just the right combination of quality and features.

NEW TECHNOLOGIES

Audix large capsule

The Audix CX101 is a true large capsule cardioid condenser that uses a 1-inch gold vapour diaphragm and boasts a noise floor of 17dBA. The housing is brass, finished in black satin, with powering via phantom. The more expensive CX111 adds a -10dB pad and a bass roll-off switch, but both mics come with a shockmount standard adapter and an aluminum carrying case. A 2-channel power supply is optional. Also new from Audix is the TB40, described as a test and recording mic, that employs a 1/2-inch diameter prepolarised condenser capsule with an omni pattern. Frequency response is claimed as 20Hz—19kHz+1dB. Price in the US is $249 complete with carrying case and clip while stereo matched pairs are available for $359.

Audix, US Tel: +1 503 682 6933.

Westlake monitors

The 5-way bookshelf LC35W10 monitor from Westlake has extensive dampening, an integrated passive crossover and a claimed high output. The speaker contains a 10-inch polypropylene woofer, 5-inch midrange and 1 1/2-inch tweeter with an 80W power handling capacity and sensitivity of 88dB in 1m for 2.83V input. Frequency response is claimed as 42Hz—20kHz. Westlake has also announced the LC255.1 centre-channel speaker driven by dual 6½-inch woofers, 5-inch coaxial mid and 1-inch tweeter.

Westlake, US Tel: +1 805 499 3666.

Digital mixers

Italian company LEM has previewed two affordable digital mixers which are expected to start shipping later this year. The entry level model is the Falcon which has 4 mono mic/line channels, 2-bit A-DGs plus two stereo line channels, and one AES-EBU SPDFD input. Eight outputs are covered by two auxes, a digital output, and main mix analogue buses. Each channel has 3-band fully parametric EQ with 4-band graphic.

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The THAT 2002 is available to owners and operators of consoles from SSL, Neve, Sony, MCI, and Harrison.
The concept of the plug-in extends to the Yamaha 02R mixer.

**Zenon Schoepe** reports on adding even more value

**YOU O.N.C.F* WOULD* NOT have believed that a time would come when another manufacturer would be allowed to develop and insert an effect board into a Yamaha digital mixer, but this is precisely what tc electronic has done. It says as much about the increasing openness of the Japanese giant as it does about the resourcefulness of the Danish processor company.**

The Unity card contains two engines. In tc speak, which amounts to two separate processors or a whole lot of power if you are acquainted with the potential of the M2000 16-in/16-out digital mixer, this card is said to be handy. Unity cards can be supplied with or without AES-EBU I/O, and when supplied comes with the same 25-pin Sub-D connector that is provided on Yamaha's own AES-EBU card. An I/O extension card for the Unity can be bought as a retrofit.

Installation is, perhaps, not as straightforward as might be assumed as it requires the changing of two V08D boards in O2R processors, regardless of what software version you are already running. Access to the chips is gained after removing the desk's standard wordlock, digital I/O panel and the process is simple; although fraught with the painstaking care that multilegged chip removal and insertion instils. Thereafter you are free to insert the Unity card into one of the O2R's four slots by simply sliding in the board and locking it in. Power up reveals a small tc electronic credit on the boot-up screen and you are then into the desk as normal without having to bother with the Unity if you don't want to. After all the tc chips are V08D variants, and if you haven't got around to upgrading you 02R to this standard yet then you are likely to spend some time enjoying some of the finer points of this substantial software upgrade.

Indeed, the Unity does seem to be tied in to v2.0 quite heavily, and the biggest relief is that the plug-in behaves remarkably like any other part of the O2R's excellent operating system. The operation of the Unity resides in one of the v2.0 remote screens, which incidentally also takes in a variety of control for such things as Pro Tools. Other Yamaha digital mixers and effects. There are three screens—one for each engine and an output page—that deal with the management of Unity parameters and the simple way to access these is to simply hold down the desk's true switch with a repeat of the same manoeuvre taking the desk back to its previous status.

Screen parameters as in tc speak are adjusted by desk cursor key and dial, and faders that are adjusted by using the desk's faders, and predictably these parameters are dynamically automated. One of the best inclinations, and one that demonstrates the strong link to v2.0, is that if the desk's Touch Sensing mode is activated you can move between the two engine screens and the output mix screens by pressing a fader from the relevant group of assigned controllers. A word about the routing possibilities with and without the aforementioned optional extension I/Os fitted: in both instances the sends to the two engines are selected from the four signals sent to the desk's card slot Buses 5-8 and returns from the effects units arrives at faders that are allocated according to which slot you choose to install the card in. The real difference resides in the fact that external inputs can be mixed in with the engine returns when the I-O is included and the Unity's output can be broken out from the desk. From here on in it is really all down to the sound of the effects that tc has brought to the party. There are 50 ROM presets and 150 user locations, and within this you encounter reverb, delay, pitch change, chorus-flanger, phaser and de-esser.

There are, of course, programming possibilities being offered here with some of the busier algorithms housing some 18 different parameters. As it happens access is clear, although the simplicity of moving a fader to change parameters is revealed as being markedly superior to the comparatively laborious routine of cursor and dial adjustment of the remainder.

The quality is quite excellent. But then you would expect that from a card that closely resembles the M2000 in performance terms. Reverbs in particular are decided thicker and more complex than those found in the O2R. What impresses most is that it works at all inasmuch as I am still struggling with the concept of plugging in third-party processing hardware into another manufacturer's product and the result functioning seamlessly. The true triumph is in the co-operation that made Unity a reality and for this both manufacturers should be applauded. It is a fabulous idea, and it is a fabulous product. Users of O2Rs may be quite content with the already fairly slick onboard processing of the desk, but for those who want something more sophisticated and different the Unity is the only choice. And you won't have to compromise the fundamental appeal and operation of the O2R to do it. A masterstroke.

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**NEW TECHNOLOGIES**

< EQ for the stereo channels, and the desk can be equipped with an optional ADAT extension interface. Two internal effects processors handle reverb and multi-effects on internal aux sends while eight other types of effect are available including dynamics, additional EQ, and enhancers. The larger Eagle digital mixer provides 12 mono mic-line channels, four aux outputs, two reverb processors connected to two aux sends and an additional 30 different processors. This version also has motorised faders for inputs and outputs.

**General Music, Italy. Tel: +39 541 959511**

New from Brauner

Brauner is introducing two new mics at the AES—the VM1u and the Valvet. The former offers the same performance as the VM1, but is restricted to omni and cardioid patterns and has no -10dB switch, although the mic can be upgraded to full VM1 spec as a retrofit. The Valvet uses a Lundahl transformer and is a cardioid and omni valve mic and is cheaper than the upgradeable VM1u.

**Brauner, Germany. Tel: +49 2856 9270.**

**HHB speakers**

HHB has launched its own Circle 5 studio monitors. The active 2-ways employ an 8-inch bass driver, the cone thickness of which is varied across its diameter, which is mated to an aluminium voice-coil. The tweeter is a fluid-cooled soft dome. The monitor is powered by an amp pack that delivers 125W to the woofer and 60W to the tweeter. HHB's ADAT45 tape is now available as a boxed library cassette in 45 and 60-minute versions. HHB Bulk CD-Rs are shrink wrapped in 100s and packed 600 to a box.

**HHB, UK. Tel: +44 181 962 5000.**

Location preamp

FEL has released a low noise mic preamp that is smaller than an XLR plug, boosts by 20dB, and is intended for location recorders. The mono version is built into an XLR and terminates in a 3.5mm plug to replace the mic lead while the stereo version is cable mounted and terminated with a 3.5mm >

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Roland VS-840

Applying the principles of its up-market VS-880 to a lower level of the DAW market has given Roland the VS-840. Rob James reports.

LAST YEAR I had a Roland VS-880 for review and found it rather hard to part with. Since then Roland have expanded their range of digital multi-trackers—the VS-880 has been joined by the powerful mobile VS-1680 and a more economical version, the VS-840.

Although the VS-880 is designed primarily as a successor to the ubiquitous cassette-based personal multitrackers, the quality of the package and the sound it can produce positively encourages more professional use. I am a little more equivocal about endorsing the VS-840 for professional use due to the absence of any uncompressed record modes.

Like its siblings the unit is small and neat, and contives to feel solid and well built. Rear panel connections are few: 1/4-inch mono jacks facilitate the four analogue inputs while an additional high impedance 1/4-inch jack on Input 1 provides direct connection of a guitar. Inputs 3 & 4 have alternative phono connections and 4 further phono inputs take care of analogue master and aux-mix outs. A-Ds are 20-bit, D-A is 18-bit, and files can be compressed.

There is no digital input, but there are both optical and co-axial SPDIF outputs. An optional board allows connection of external SCSI storage devices. With a footswitch for punch-ins, a pair of DINs handle MIDI In and Out, and 1/4-inch jacks deal with headphones.

The headphone output level is generous—an essential feature on this type of unit. Competitors please take note. The mixer has 6 channel strips and a master strip. Input sensitivities range from 50dBm to +4 dBu. Peak limits light at 6dBu before clipping. Faders 1-4 are mono, but 5-6 and 7-8 are only available as stereo pairs. This pairing extends to the record and virtual or 'V' tracks. Some may find this a little limiting, but I feel it helps keep things simple.

Transport controls are prominent and positive. Monitor levels are set by six track cue pots. The back-lit display is small yet effective. Grouping of other function keys is logical and reasonably easy to follow without frequent recourse to the massive 190-page manual. The chosen recording medium is the ion mega Zip disk. It is a good compromise, offering limited storage at reasonable cost although a second drive would be a good idea if you want to copy disks. It is possible to copy using the internal drive, but the number of disk swaps required may quickly send you insane.

The VS-840 has 4 recording modes. Each of which can be set at 2kHz or 4kHz sampling rate, although all use data compression. The Multitrack 1 and Multitrack 2, Live 1 and Live 2, which translate to record times of 37, 50, 60, and 75 track-minutes per 100Mb disk at 4kHz sampling. There are limitations on the number of simultaneous record tracks depending on the chosen mode and whether variable pitch is used. MTI is restricted to 2 tracks or 1 , w,y variable pitch. The other modes to 4 and 2 respectively.

Each of the 8 tracks incorporates 8 virtual tracks to give a total of 64 within which punch-in can be manual or auto and you can set the VS-840 to repeatedly loop over a section, recording on each pass, until you get it right. The Scrub function loops a section of sound (25ms-100ms).

Internal processing is 24 bits facilitating 3-band EQ on all channels with provision of 12dB of boost and cut. The built in effects are well up to Roland’s usual high standard, making much of the company’s algorithms, and I was delighted to discover the inclusion of a Wavetone and some RNS (Roland Sound Space) presets. I also enjoyed the guitar patches.

Snapshot automation gives 8 scene-memories. Scenes can only be recalled when audio is not playing. There is no dynamic automation. MIDI allows the transport and record functions to be controlled in a suitable sequencer using MMC commands. Alternatively the sequencer can simply be synchronised to MIDI clock or MT.

Since Roland introduced the VS-880, other manufacturers have entered the same field and Roland have the same field and competing models. When considering the options it pays to remember the practical price paid for a big feature set is the complexity of the user interface. Roland has made a good effort to produce an entry-level digital multitracker that is not simply a cut down VS-880 and a number of the VS-880 features need help to make this unit an ideal introduction—particularly to the idea of having a proper D1 input for guitar. The bus structure and separate track level controls make the whole easier to understand. The clever EZ Routing functions accessed from a dedicated button helps set the unit up for various tasks and should go a long way to demystifying the process for newcomers.

Distribution system
XTA is to launch the DS800 8-input, 32-output audio distribution system. In standard format each input is split into two transformer-isolated outputs and two actively balanced outputs. The unit can be internally reconfigured to give more than the standard four outputs from each input up to a total of 32. Each input has a remote activated pad, adjustable input gain, mic, line switch, phantom power, 5-segment LED metering and a listen facility. A headphones socket and level pot are included and the listen function can be ganged together when using multiple units.

THX crossover
Lucasfilm THX has introduced the D138 monitor-Digital crossover which has an 8-channel input stage in analogue and digital and a software configurable output stage for compatibility with all formats. Output stages also offer adjustable delays and digital EQ for surround and subwoofer channels. For postproduction it contains numerous functions in one product including customisable computer control for channel solo and muting, digital delays for all channels and THX standardised subwoofer monitoring of screen and surround channels for discrete and matrix sources.

September 1998 Studio Sound
There are a few things you’ll miss with the Opal Series
Like noise, distortion, lack of headroom, limited facilities, and no pedigree.....

It's obvious that the better you protect your sound, the better your mix will be. When you buy an Opal Series processor by BSS Audio you're getting not only the best in audio quality, but a heritage and proven track record that has benefitted artists, sound companies, studios, broadcasters, theatres and clubs the world over.

From this pedigree, BSS engineers have brought you a range of products designed to solve your sound problems without adding either impurities or complexity in operation. And by exploiting the latest advances in component technology and manufacturing techniques, we can now bring you this thoroughbred processing power for less than you'd think. Don’t miss out on an Opal - call your BSS Dealer or Distributor for more information.
Thermionic Culture Vulture

Bringing useful valve distortion into the studio environment has proved an entertaining chase. George Shilling is close to a kill.

When the Ti Audio indulge valve distortion unit appeared, I predicted that we would see further expansion in this area of outboard equipment. And, perhaps belatedly, the Culture Vulture has fulfilled my prophecy. While not an essential in every studio, a distortion box is something for the enthusiast recording engineer to really get his teeth into as evidenced by the hours of frustration logged by engineers struggling to connect guitarists' stomp-boxes with inappropriate impedances to a pro console. Realistically, few studio multi-effects boxes include distortion effects, and those that do often possess all the character of a microwave oven.

Directly following Thermionic's remarkable Phoenix compressor (Studio Sound, August 1990), comes the Culture Vulture, which looks similarly marvellous. The black, 2U-high front panel includes a host of vintage-style knobs and toggle switches. The unusually ugly meters are not vus, but ammeters measuring current flow through the valves. The unit's construction is similar to the Phoenix, with a sealed metal base-section containing smaller components, on top of which are mounted the larger parts. These include a large old-fashioned bobbin mains transformer, and an array of obscure valves, almost conceivably from the designer's attic.

This top section is covered with a metal mesh casing. The unit is surprisingly lighter than it looks, especially as this is an all-valve unit. There are no ICs in sight, and not even any printed circuit boards.

The front panel layout is irritatingly messy. The two channels' controls are arranged side-by-side, but as a mirror image of each other, which can be confusing. The two meters are valves that are themselves mounted just below the main chassis, as their glass casing protrudes from the front panel. Each channel has controls for bias and drive, scaled 1-11. A 3-position switch selects Triode or one of two Pentode modes. However, all these controls labels are below the large knobs, and are therefore invisible if your eye level is above them or you are sitting on the floor. The knobs are owl-like toggle switches for each channel. These appear to be something of an afterthought, mounted extremely close to the new knobs and without proper legend—just a stuck-on label. Between the meters is a large ox-eye toggle with accompanying light, which is a torch bulb behind a green cover that takes a few seconds to brighten at switch-on. Below the meters are chunky twin metal knobs, similarly legended 1-11. Also on the front panel are unbalanced jack sockets for inputs and outputs of each channel, their layout following the mirror imaging of the controls. On the back of the base section are duplicate connectors: both sets of outputs can be used simultaneously, but the front input jacks override the rear ones.

The Drive function behaves as you would expect, working similarly to a premix gain control on a guitar amplifier. Bias is a little more esoteric—as you turn it up it swerves the valve, the level drops, and a brightening tonal change takes place, depending on the mode selected, and a gating effect starts to occur with only the peaks getting through and sounding fairly nasty. The meter displays the bias current, and as you turn the bias knob, the current decreases. The meter wiggles slightly upwards if you are changing the value downwards if the valve is overdriven. The latter setting (with bias set low) gives a warmer, more even distortion. In use, the bias knobs have to be constantly adjusted in conjunction with the Drive and bias knobs. The Triode distortion mode (even harmonics) and Pentode 1 mode (odd harmonics) can vary from gentle warming to raucous, with the latter mode—seemingly slightly richer and creamier. P2 mode provides an even more drastic distortion, with a really horrible thin sound at high bias settings. In any mode, flipping the cathode toggle gives a boost to the Drive and Bias for more radical distortion effects.

At extreme settings, there is a remarkable lack of self-induced noise. Any buzzing or humming I came across was generated by the source plugged into the inputs. This is thanks to the use of a very high-quality military-grade valve (CV4010) that gives plenty of gain with very low noise.

I found the Culture Vulture most useful for distortion guitar parts that were lacking a bit of sparkle, and warming up plain-sounding electronic keyboard D-I-1s. I tried the manual's suggestion of putting the whole mix through it and after some experimentation it was possible to add some subtle warmth in Triode mode with a low Bias setting. However, without outboard bypass switching it is difficult to set up and use. Also, the unusual mirror-image layout takes some getting used to. It is easy to find yourself adjusting the wrong knob, particularly when you cannot see its label.

The Culture Vulture is a peculiar piece of equipment, and a luxury rather than a necessity. It does what it sets out to do, but is a little awkward in operation, and comes across as slightly unrefined in terms of usability. However, its quirks will probably endear it to potential owners, and Thermionic Culture should be applauded for their unique approach.

Reference cassette
The BASF Reference II Master audio cassette is described as an innovative product in the chrome class with outstanding output in the high frequency range. Output levels at low and high frequencies are around 3.5dB above those of the IEC reference tape while overall dynamics are 7.5dB higher. Other features include a high-precision shell with a separate sound head and an Anti Stick Slip Pad.

EMTEC, Germany. Tel: +49 621 5920406.

Zone expansion
Citrone now offers a 5-zone distribution mixer, the Z-SDM, and the Z-2AS 50W power amp. The former can be used independently or as a zone extender for an existing system and is a stereo line mixer with five stereo outputs to which either of the inputs can be mixed. Mono or mixed distribution can be selected and while 1-0s are balanced, inputs have enough gain to handle unbalanced connection. The Z-2AS is available in low voltage and 100/70 Volt line versions and has a protect mode indicator that shows when the amp is shut down on switch on, thermal overload, DC on the output or a short circuit.

Citrone, UK. Tel: +44 1225 705600.

Monitors and sub
M-Tech has introduced four new products. The 2-way PM30 is designed for medium to large rooms while the 2-way PM15 for small to medium rooms. The PM70 subwoofer is designed to integrate with the aforementioned Extends response to below 20Hz and employs an 8-inch x 12-inch ellipsical long-throw driver. Input to the internal 100W amp and adjustable crossover is by phono or XLR. Additionally a Z-2A-15 rackmount monitor is available in active or passive versions with an equalised frequency response and magnetic shielding.

Musical Technology, UK. Tel: +44 1656 842000.

MOTU
Mark of the Unicorn has launched a 24-track expendable hard disk, recording system for PC, or Mac, with a UK suggested price of 850 plus VAT. Model 2408 offers 24 simultaneous inputs and outputs, with the ability to expand to 72 channels and include real-time effects. The system's PCI card connects via Firewire to an external 1U I/O module containing six Alesis ADAT fibre-optic.

September 1998 Studio Sound
## GS3000

**OK... or Outstanding?**

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<td>Wide ranging mic preamp</td>
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<td>Optional 21 segment input bargraph meterbridge</td>
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<td>Interface for three 2-track recorders</td>
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**SVT VALVE PREAMPS**

GS3000’s 2 valve preamps can be patched to individual channel inserts, groups or LR. The valve preamps incorporate Symmetrical Valve Technology, allowing them to be used in balanced mode for regular inputs or in single-ended ‘guitar’ mode, which drives the tube valve to give the type of pleasing harmonic distortion much sought after by guitarists. Other valve preamp features include valve drive control, pre-valve swept frequency EQ, hi-cut filter and output level trim.

Most modern recordings sound OK - but few sound really outstanding. The new GS3000 gives you truly great sounding mixes, with all the analogue warmth, sparkle and power that set the best apart from the rest. The new GS3000 - sounds glorious.
Audio-Technica AT4060

Everyone is after the valve effect and the valve premium.
Dave Foister reports on a less blatant implementation

MoST PEOPLE currently making valve microphones are recalling former glories. The valve tradition, so enthusiastically rediscovered in recent years, has few specialisations with as much continuity as microphone design, and the people who built the classics are mostly still around to bring them back. The rest are divided between those who genuinely think valves are a good idea, and those who just think selling them is a good idea, and usually the difference shows in the finished product.

Audio-Technica has made its name as a studio contender very recently, and has done it impressively quickly and effectively. It has produced microphones that compare favourably with established standards right from the outset, and at competitive prices. It has won the approval of major names and major organisations with its first catalogue of serious microphones, and continues to build on its strengths apparently driven from within. And yet, unless I've missed something, its heritage does not go back far enough to include valves, and indeed its perceived selling points have been the kind of parameters done best in solid state. Nevertheless, the latest model in the all-encompassing 4000 series is a valve microphone whose chief claim to novelty is its valve as it has an unusually restricted set of features. In fact it seems to set out to be a valve 403, with the intention of combining transparency and valve character in the simplest possible package.

Putting it up consists simply of mounting it on a stand and hooking the bits together. It is a big microphone, albeit small in appearance—there is no attempt to make it look as though it is 35 years old. The styling is very similar to the 4033 and 4050, the main difference being a longer body to accommodate the valve. Its mount is elaborate and secure, holding the body with two rings, one with screw clamps and the other fitted with elastic bands. The resulting mechanical isolation is good, yet the suspension is tight enough to hold it where you put it.

The link between the microphone and its power supply is terminated in 5-pin XLRs, and the supplied cable looks to be of a good quality even if the folding up to fit it in the box has been done in such a way as to impart the maximum possible twisting effect. If the microphone were mine I'd hang the cable from a high ceiling and just use it to get the tangles and knots out. The power supply has an input for this lead, an XLR output, and IEC mains inlet, an on/off switch and a ground lift switch, and that is it—no adjustments for the microphone at all. Considering it this is surprisingly big, and comes as standard with rackmount fittings to slot into 1U of space.

Like the 4033 the 4060 is cardiod only, and the simplicity of its configuration goes even further, in that it has no pad and no filter. Yet the internal design is surprisingly complex, as the capsule has two diaphragms, a state of affairs normally only found in variable-pattern microphones—indeed, the need for only one diaphragm in a cardiod microphone is generally the big cost saving that makes it worthwhile. The rationale behind this is not made clear, except that it is supposed to improve the SPL handling and the evenness of the frequency response. Like the other Big A-T condensers, the 4060's diaphragms are pre-aged to ensure stability with time, and the same applies to the handpicked miniature valves.

The resulting specification does not read like a typical valve microphone. The printed frequency response plot is unusually flat, more like a good solid-state design, and the distortion, and SPL handling figures could leave you wondering why they put a valve in at all. Clearly the aim has been to deliver the kind of extended flat response that has made the existing model so successful, with just a hint of valve character.

And that is what the 4060 produces. The initial impression is of a smooth quiet microphone with more than its fair share of HF extension, coupled with the kind of clean transparency that extension needs. It is not a clinical microphone but neither is it flattering in the way many valve models. I tried it primarily with a jazz singer whose album we had just finished; all her vocals on that had been recorded with another modern valve microphone, and we both felt that the 4060 had a less pronounced signature. Its avoidance of overstatement left it sounding slightly thin by comparison, but in reality this was simply showing its flat response. In all there was very little sound difference, with the 4060 just having the edge at the top end while retaining the slight fairness of the sound. It is also quiet and clean in the best Audio-Technica tradition, so could be said to be offering the best of both worlds. Valve microphones do not have to be retro and obviously coloured, this is a microphone of serious quality, and so further A-T's already enviable reputation.

NEW TECHNOLOGIES

< connectors and three Tascam TDF connectors, either of which will provide 24 tracks of digital interfacing. There is also a bank of eight unbalanced phone outputs, as well as one stereo SPDIF input, two stereo SPDIF outputs, and two balanced outputs on TRS jacks. As many as three 2408 I-O units can be connected to a single PCI card, giving the full 72 tracks. Macintosh versions will be shipping with audio workstations software supporting 16-bit and 24-bit recording. The Windows version will have a driver compatible with audio applications that support multichannel Wave drivers. The Mac software will apparently include multichannel waveform editing, automated virtual mixing, graphical editing, real-time effects plug-ins and support for third-party plug-ins in MOTU Audio System and Adobe Premiere formats. Although the PCI card handles I-O processing—and also acts as a digital router—audio processing is handled by the host computer. MOTU says that a 200MHz Power Mac with provide 16-24 tracks of audio, with the very fastest machines allowing 32 tracks or more.

MOTU, UK. Tel: +44 1462 812010.

Directional mixer
Shure's Directional IntelliMix direction sensitive mic activation is used in the AMSR100 automatic mixer which activates mics only for sounds originating within a 120° acceptance angle. Additionally the system can gate only one mic per sound source or multiple mics when multiple sound sources are in use. Designed for the broadcast market, the SM63LB is a black version of the company's SM63L omni dynamic mic while the VP64AL is identical to the existing VP64A except for a longer handle.

Shure, Europe. Tel: +44 7131 72142 23.

Small hooters
Designed by Ted Fletcher of JoeMeek processor fame, the Hooter Sound B1 is a single-channel mic preamp with phantom power, a volume compressor, peak limiter and variable noise gate for -5dB + VAT UK. Compression can be bypassed and ratio is variable up to 6:1, and includes automatic gain compensation, while the gate has variable threshold. LEDs indicate compression, limiting and gate status and line-level output is tracked on a bar peak meter.

Hooter Sound, UK. Tel: +44 1626 333234.

Digital radio desk
The Audiotronics NuStar 3001 affordable digital radio console has been redesigned following the acquisition of the company by >

September 1998 Studio Sound
At last the promise of digital mixing is fulfilled with TASCAM's extremely powerful, compact TM-D1000 - the lowest cost digital mixer ever, at just £899 inc vat.

- 8 ch integral TASCAM (TDIF) digital I/O
- 8 ch Alesis digital I/O (with optional interface unit)
- 8 mic/line analogue ins, inc. 4 balanced XLR
- Optional rack mount kit
- AES / EBU and SPDIF stereo digital I/O
- Stereo analogue and separate monitor outs
- 4 configurable aux / group outs
- Assignable on-board digital effects
- Optional additional FX board doubles effects DSP
- 128 snapshot memories
- MIDI sequencer automation of console
- MIDI I/O, MMC and MTC slave and Word sync. I/O
Bellari LA120; MP110

Low cost no longer means no valves. George Shilling investigates Bellari's tube compressor-limiter and direct drive tube mic preamp.

SELECTIVE OF smaller devices supplements the Bellari range of full-size and mounting tube studio equipment. The two units reviewed here are each smaller than a 1U-high half-rack box, being 19mm wide, 40mm high and 135mm deep. Potential purchasers surely include electric and electronic musicians on the move: this must be one of the cheapest and most portable ways to incorporate valuable studio features into a panel. And any studio setup can always find space for devices such as these, although the lack of any feet is an oversight. As PC-Pac users won’t be happy with a scratched computer table. As with their larger cousins, these Bellariis described as tube kit might be considered slightly misleading. They each contain just one valve, and much of the serious business is carried out by solid-state circuitry. Both units require on-board power. The review models came directly from Utah where they are made and were supplied without adaptors. If purchased in this country one would expect the necessary 12V AC 300mA rated power transformers to be included. This is a fairly uncommon rating and there is a danger that someone might try plugging in a more common 9V DC transformer; as these units feature power sockets more commonly used for such supplies. Also on the back panels of both boxes are Input and Output sockets, monopaks and XLRs. Oddly, for budget units, both operate at 4+4B, but the MP110 will happily output a lower level. The LA120 will operate at 10+4B, but without an input gain control you will not be making use of all the available headroom.

The LA120 features a brushed-aluminium silver front-panel with clear black legends. It sports two damped knobs of the same plastically varied as those found on larger Bellaris, and two push-buttons of retro style. The active button lights an LED, when switched out the unit is in bypass mode. Strangely, when the unit is bypassed (active out) the signal is boosted by a few dB; while the (output) gain control is inoperative. Another quible is with the legends of the active knob, marked from +100 to 10dB full up. In fact, there is a boost of about 18dB with the knob fully clockwise with no compression applied. This is useful, as otherwise you would have no makeup gain after compression. The only compression controls are a threshold knob (marked -20dB to 12dB) and a gain push-button. I initially found the compression very harsh, but then discovered that the compound button operates the opposite way round to my expectations. Nowhere in the manual or on the unit is this explained, but for the record, gentler Compression is apparently achieved by pushing the button in, with harder Limiting achieved when the button is out. Compressor mode enables a ratio of between 2:1 and 20:1 whereas Limiting mode varies from 20:1 to 10:1. A small round vu meter indicates Gain Reduction, giving a good indication of compression, and its illumination indicates power on (there is no power switch). Thankfully, this unit has a (very) soft character unlike other Bellari compressors, which I have criticised for their unexpectedly extreme 'hard' knee characteristics. In Compressor mode the unit worked very well on vocals and instruments, although the time characteristics were a bit last for my taste, especially compared to what one might expect for a photo-electric compressor billed as 'tube'. Limiting mode is useful for heavy compression on drums and where extreme squashing is required.

For no obvious reason, the MP110 has a slightly shinier front panel than the LA120. This is no problem, but the smaller, more spidery legs are a disadvantage. Considering that this model gains just one push-button and LED over the LA120 and lacks the latter's vu meter there is no excuse for the inferior lettering. On the rear the XLR input is marked 'MIC', while the jack input is marked 'INSTRUMENT, claiming a difference in source. On the front panel are push-buttons for a handy 30dB PHANTOM POWER and useful PHASE INVERT button. Tiny LEDs indicate Clip and Power. The two knobs are GAIN and VOLUME, the latter usefully acting as a fader, something lacking on certain outboard mic channels. Plenty of gain is available, the leg- ending indicating a range of +17 to +5dB. In combination with the no switch this covers everything you might want to plug into the unit. In use the MP110 performed extremely well, providing a full-bodied, clear tone with low distortion and a very low noise floor. In a brief comparison I preferred the sound to a more expensive competitor. Driven hard, volume overload is audible when the Clip LED glows, which is quite fun when hooked up with a guitar. However, noisy transistor overload can quickly take over. You must be wary in mind that the Clip LED indicates tube circuit clipping rather than output overload.

The LA120's manual erroneously states that the unit includes side-chain connections, but apart from this glaring error the manuals adequately explain the functions. For budget units, audio performance is remarkably good, and these Bellaris are very useful little boxes.
We proudly present to you our new series of purely analogue audio processors. Taking account of many of your suggestions in terms of expand-on and functionality, the BEHRINGER PRO SERIES features advanced versions of our Standard Audio Devices which have proven their reliability in applications throughout the world.

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Mixing with the right company

It is summer, it is early morning and after an absurdly punctuated journey of four hours you have made it to the Port of Dover, or more precisely the ferry departure lanes, in a vehicle heaving with a spouse, three children and a multitude of items each member of the family has considered to be indispensable for a 2-week camping holiday in mainland Europe. A quarter-way though Day 1 and the atmosphere has become more tense than tents.

Take a look at the car next to you, and to the one next to that: take in the sight of the cars surrounding you in a sea of shiny metal and notice that these people are just like yourself. They too have driven here in a car—maybe its not the seven-year-old model that yours is, or maybe its a people-carrier rather than a saloon—but they will also be packed to the gunwales with their essentials. In essence their vehicle will have been purchased with the same set of objectives in mind as you as when you bought your car, and by means of this strange analogy we may draw a parallel between real-life and that time, space and dimension occupied by us, the aurally anxious.

So come with me, if you will, and let us escape the squabble of siblings and stretch our weary legs in a figurative car-park of audio mixers; where workhorses flick their tails next to thoroughbreds, and witness how the adage of being 'built for speed not comfort' might have been coined by the audio industry. In no order other than alphabetical, and where size matters less than what you do with it, here are the light and heavyweight contenders. The truth is in here.

Audio Developments

**AD-261**

4 channels in, 2 out. MSI-AB

Arguably the most elegant of all the popular mixers, the AD-261 offers the ability to stereo-gang the XLR balanced inputs and 1/2 and 3/4 decibel MS inputs to AB or vice-versa and comprehensively monitor all stages of the mix. Slugging it out for the top-spot with the SQN-45 in the over-the-shoulder stakes, the essential features of a lightweight mixer are all here: mono-stereo limit, 48V phantom and 12V T-power, 90Hz and 150Hz high-pass filters, phase reverse on Channels 2 and 4, plastic conductive rotary faders, unbalanced output feeds with main outputs available on a Hirose 10-pin connector or two balanced XLRs, switchable between mic or line level, all neatly packaged into an intuitive layout.

**Making informed purchasing choices relies most essentially on being properly informed — whatever the market. Neil Hillman surveys the available location mixers and their facilities.**

Audio Developments

**AD-140 series**

6-24 inputs, MSI-AB.

A range of chassis sizes and modules allow the purchaser to precisely define the configuration of their mixer with the legendary AD flexibility and audio integrity. The film flagship, AD-149 offers an adjustable limiter on all inputs, insert point on inputs, direct output from each mic-line module, 3-band EQ with a choice of four centre frequencies in the MF section. Two man and two aux outputs with a limiter on Left and Right which can be linked. Comprehensive MS facilities in both man and monitor paths. The full range is AD-144, a 4 output plus 4 aux sends. 12-24 input mixer, AD-149, a 4-output. 6/12 input mixer, AD-147, a 2-output plus 2 aux. 6/12 input mixer, AD-148 4-output. 6-12 input, Edit. Suite mixer, AD-149 2-output plus 2 aux sends or 4 output. 6-12 input film drama mixer and the AD-245 a 2+1 output. 6-12 input mixer.

Cooper Sound CS 104

4 channels in, 2 out, MSI-AB

Smaller brother to the highly successful CS 106 trolley-top location drama mixer, all controls are sited on either the top or sides of this 4-channel mixer with channel pan-pots, high-pass filters, peak indicators and PFL. The output section includes slate microphone, limiter and tape-direct switches, master level control and twin vu or peak meters. The 4 XLR inputs have line and 4-position mic gain selectors, 48V phantom or 12V T-power and phase reverse for channels 2 and 4. The outputs are via the obligatory 10-pin Hirose video connector on balanced Left and Right. XLR outputs are unbalanced Left and Right outputs. A mix-bus connector is available to daisy-chain additional mixers.

Cooper Sound CS 106

6 (+ 1) input channels, MSI-AB.

The supremely flexible 106+1 offers 6 input channels plus the option of a stereo input module with MS decoder. Each input channel offers a limiter, PFL, an aux feed pre or post fader selectable and a mix bus that allows a channel to be used discreetly from the mixer output. EQ is provided by 2 high-pass filters of 70Hz and 100Hz with high and low-frequency shelving of 10KHz and 100Hz and mid-frequency selectable from 50Hz to 5kHz. Both 12V and 48V phantom power and 12V T-power is available. There are 3 discrete monitor outputs with tape returns, talkback, communications and a remote-roll facility.

Fimmtech LSP-4

4 channels in, 2 (+4 direct) out, MSI-AB.

The LSP-4 is a 4-channel portable mixer with several features in addition to the usual facilities on the type of mixer. Slightly larger than some of the other over-the-shoulder mixers on the market because of the design philosophy to get as many of the controls as possible onto the front panel for ease of use and access, the unit is built into a strong aluminium enclosure. The LSP-4 has 4 transformer balanced input channels with 48V phantom or 12V T-power available, rotary attenuator switches and phase reverse on Channels 2 and 4. Each channel has a 4-position switch that routes the signal off, left, right, centre via the MS matrix or via the adjacent pan pot. Each channel also has an AFL switch and LPF cut filters. The transformer balanced output appears on a 10-pin Hirose connector and two male XLRs with

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< attenuation switches. There is also a stereo unbalanced line output, a mono unbalanced mic. level output and a cascade socket for ganging two mixers together. Each channel on the LSP 4 also has an electronically balanced direct channel output that appears on a 12-pole Hirose connector.

**Shure FP-33**
3 channels in, 2 channels out. AB stereo.
The baby of the Shure Brothers range of mixers, the stereo FP-33 can be powered by two 9V batteries or externally via a 12V-30W DC source. Each channel offers 12V or 48V phantom or 12V T power and pop-up pan-pots. The 3 mic or line inputs are transformer-balanced and accept a wide range of signal levels without the need for stepped attenuation. A link switch couples inputs 2 and 3 as a stereo pair and the left and right outputs are on transformer balanced XLRs, with a max-bus available for connecting further FP-33 mixers.

**Shure FP-42**
4 channels in, 2 channels out. AB stereo.
As the adolescent version of the baby FP-33, just the additions of an extra XLR input and pull-pots cueing for previewing each input are welcome substitutes for the growth of facial hair and spots.

**Soundcraft LM-1**
6, 8 or 12 input channels. MS-AB.
The LM-1 may be configured by the purchaser into a chassis designed to accommodate 6, 8 or 12 input channels of which these may be either mono or stereo mic level or stereo line level. Optional isolating transformers may be specified and the switchable 3-band EQ section is supplemented by an independent 80Hz high-pass filter.

Two auxiliary sends are available. Aux 1 being switchable pre or post fade while aux 2 is a switched post fade send. Inputs may be paired for MS working and stereo placement is via rotary pan controls. Four long-throw faders control the Left and Right and axues 1+2 output levels which may be shown on the dual PPM or vu meters. External talkback audio and control signals are interfaced via 6-pin XLR whilst DC power is provided by 12 D cells for the 6 or 8 channel version and 16 D cells for the 12 channel mixer.

**SQN-4S series IV**
4 channels in, 2 out. MS-AB.
The 4 transformerless balanced inputs of the highly popular SQN-IV use XLR-3 type female connectors while uniquely channels one and two also share an XLR-5 type connector for stereo microphone operations. While offering all of what may be considered standard features, the mixer is intended to work in several modes: Twin Mono in which the channels are un-jungled and each input channel can be routed or pan-potted to either or both of the output channels. Stereo Ganging in which the first channel fader controls the gain of both channels and the first channel pan-pot acts as a stereo balance control. Mid-Side Ganging in which the first channel fader controls the gain of both channels and the second channel fader acts as a sound level control. Mid-Side Matrixing in which each channel pair has an MS matrix which will convert an MS input into AB stereo or vice versa or finally Panidding Mode where the Channel 3, Channel 4 pair has its output treated as MS encoded stereo and the channel signal is panned to move across the equivalent AB stereo image.

**TL Audio PM-1 4/2**
4 channels in, 2 out. MS-AB.
TL Audio’s objective with the PM-1 4/2 was to produce a mixer offering features similar to those found on models offered by the more established manufacturers such as Audio Developments or SQN, but at a significantly lower cost. The PM-1 employs an all-metal chassis sealed plastic conducive ports and gold-plated switches. Inputs for the four channels are via combined 3-pin XLR jack sockets with 20dB pad on jack inputs. Individually switchable 48V phantom and 12V-T power is available as switchable high-pass 12dB per octave filters, selectable for roll-off frequencies of 75Hz or 150Hz. The balanced XLR inputs are transformer-isolated and the mixer also offers stereo linking of inputs, stereo limiting, MS decoding and a choice of PPM or vu metering. The outputs are electronically balanced on 2 3-pin XLR connectors, duplicated on the Beta-Cam standard 10-pole Hirose multipin connector that also provides a tape-return monitor path. Some mixers are destined to lead a hard life as a means to an end; for others they will become like one of the family. A piece of advice, however, packing them for the annual vacation is definitely taking things too far.

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There’s a tendency in music to think that we’re educating people or coming up with a cure for cancer, but we’re not,’ says Bruce Swedien. The only thing you can do to music is listen to it, and I have always felt that the real value of what I do as an engineer or as a producer lies not in the technical or acoustic sense, but in what the music signifies to the people who listen.

Swedien does not regard himself simply as an engineer, for as someone who also produces, arranges and composes he is not quite so easy to define. Indeed, among his 13 Grammy Awards (starting with Frankie Valli and the Four Seasons’ single, ‘Big Girls Don’t Cry,’ in 1962), there was one for his co-writing of ‘Jam’, the opening track on Michael Jackson’s Dangerous album, and one of three songs which he has thus far composed with the King of Pop. Swedien also recorded and mixed 11 of the tracks on Dangerous while co-producing five of them, so there within a single project several of his skills were put to good use.

Born in Minneapolis in 1934, he is fast approaching his sixth decade in the recording business, and the credits keep rolling in—from pop to classical, jazz to blues and almost everything else in between. Swedien has lent his talent and expertise to projects with Lionel Hampton, Count Basie, Stan Kenton, Duke Ellington, Woody Herman, Oscar Peterson, Sarah Vaughan, Dinah Washington, John Lee Hooker, Buddy Miles, Muddy Waters, Mick Jagger, Paul McCartney, Nat ‘King’ Cole, George Benson, Barbra Streisand, Herb Alpert, Donna Summer, Patti Austin, Lena Horne, Natalie Cole, The Chi-Lites, Edgar Winter and Jackie Wilson. He has won Grammys for his engineering of Michael Jackson’s Thriller, Bad and Dangerous albums, as well as for Back on the Block and Q’s Jool Joint courtesy of his long-standing professional relationship with Quincy Jones, and he has also worked on the scores of movies such as Running Scared and The Color Purple.

It has been quite a career, yet Swedien is not ready to wind it down just yet. ‘What else would I do?’ he asks, before musing, ‘I guess I could drive a tug boat. You see, I’m such a junkie, I love this business just as much today as I did the first day I walked into Studio A at Universal when I was 20 years old.’

Adding to the list of legendary blues artists with whom he has worked, Swedien is currently scheduled to record B.B. King together with a large orchestra. This is the kind of music that he has a passion for, having been instilled with the spiritual sounds that reverberated around a black church which he and a friend used to sneak into as children.

When he was just 10 years old, Swedien was given a disc recording machine by his father, and, in his own...
words, '10 minutes later I decided on music recording as a career.' To that end he acquired an evening-weekend-vacation job in a small basement studio at the age of 14, and after graduating from high school he then bought a Magnavox PT-6 pro tape recorder. Entering the University of Minnesota, he studied electrical engineering with a minor in music, and at the same time worked in and around Minneapolis recording jazz groups, choirs, polka bands and radio commercials.

A job running the Schmitt Music Company's recording facility led to work assignments with major artists such as Tito Puente and Tommy Dorsey, and eventually Sweden bought out the business and relocated it to an old movie theatre, where it would evolve into a world-class studio. In 1957 he and his family moved themselves to Chicago, and there for the first 11 months Sweden worked for RCA Victor's facility and recorded the Chicago Symphony Orchestra. Then he transferred to Bill Putnam's new Universal recording complex and he remained there for the next nine years, working with the aforementioned luminaries in the world of jazz.

Bill could be called the 'father of recording,' he says. 'Many of the techniques that we use today originated in his mind. He believed in me as a kid, and for the first few weeks at Universal I would follow him around, bring him his coffee, set up mics, and generally try to learn how things were done. I would sit down and get the song started and Bill would finish things up. Well, we were doing a Stan Kenton session for Capitol and he must have arranged something with Kenton, because at one point Bill said, 'Bruce, come on, you sit down and do this song. I've gotta go take a leak.' I didn't see him again for five years. So that was my baptism of fire. Evidently he thought I was ready, and so did Stan, because I did the rest of the record.

That was in 1957, yet it was several years earlier that Sweden first became truly inspired by the thought of recording and mixing.

I didn't really get excited about pop music until I discovered that it was possible to use my imagination, he recalls. That came with a record which I myself didn't work on. Les Paul and Mary Ford's 'How High The Moon' in 1951. Up to that point, the goal of music recording was to capture an unaltered acoustic event, reproducing the music of big bands as if you were in the best seat in the house. It left no room for imagination, but when I heard 'How High The Moon', which did not have one natural sound in it, I thought, 'Damn, there's hope.'

You see, being a Scandinavian from Minnesota, it was okay to like your life's work, but not to get too excited about it. Well, I was absolutely in love with music and with what I was doing, and when I got to Universal in Chicago and did the first couple of albums with Duke, and I had the chance to talk to him and to spend a little time with him, all of a sudden he made me realize that it's perfectly all right to love what you do. So things evolved from there and I did some exper-
IOW

Nothing beats advance-Swedien's sonic sculptures is desirable. was not an important part of music recording and probably wasn't even desirable. I kind of see my mixes as sonic sculptures—I would like them to be in this world, but not of this world.

Nevertheless, having been present for many of the revolutions that have taken place in the studio—starting with the switch from disc recording to tape—Swedien feels that technological advance hasn't necessarily played a large part in helping him to realise this goal.

Does digital multitrack recording make it possible to produce material that is devoid of musical errors? he asks. Or, more importantly, does digital technology make it possible to produce a recording that is free of technical errors? I think the answer lies somewhere in between, yet what is more important than all of that stuff put together is being able to use my imagination; to create a sonic canvas that could not exist in reality, only in my mind.

In that respect Swedien cites Quincy's Back on the Block album as, perhaps his most well-realised project, and the title track in particular as an example of a 'sonic canvas.' What's more, he also regards music mixing as an extension of arranging, and as to back up this theory he points to the arranging credit that he once received on a Sergio Mendes album. Indeed, it's not so much the equipment that he uses, but the way in which he uses it, as well as his willingness to extend the accepted boundaries, that has placed Swedien in the esteemed position that he finds himself today.

'I learned early on with some of the older ribbon mics, like the RCA 441X, that moving them from a vertical plane to a more horizontal plane added more high-frequency response,' he recalls. 'Gravity put a little bow in the ribbon and brightened the sound up. This was when we had little—and, in many cases, no—EQ in the recording path. Well, that was a great time to be working. We were learning so rapidly and things were moving so quickly.'

'I remember in the late fifties, early-sixties, when the moguls of the record companies said there was no future in stereo, and they took it to the point where they'd refuse to pay for the tape to record some of these incredible bands in stereo. So, a bunch of us guys—myself, Phil Ramone, Al Schmitt and so on—paid for the tape ourselves. In fact, I have a lot of stereo recordings of Oscar Peterson, Duke Ellington and all of those major artists that I worked with in Chicago, and now the record companies come to me, of course, to see if I have a certain recording and I'm always happy to sell it to them.'

It was in 1969 that Bruce Swedien went independent in order to concentrate on album recordings and film scores, and shortly after moving to Los Angeles in the mid-seventies he encountered Michael Jackson.>
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< while working on the movie, The Wiz, with Quincy Jones. The rest, I suppose you could say, is History.

'I can't be in the control room when Michael listens to a mix,' says Sweden. 'He plays it so loud. We'll be in the middle of one of our huge mixes and he'll turn to me and say, 'Bruce, hurt me!' So I'll turn up the speakers and leave the room. He'll then leave me a little laundry list, signed 'Love, Michael' at the bottom.'

Leaving the room when Jackson asks to be hurt brings me to another interesting point about Bruce Sweden; namely, the fact that his hearing—even at the top end—is still intact after so many years in the studio. Now, considering that some of those years happened to be the 1970s, when people often monitored at outrageous levels, his present state of well-being is all the more remarkable.

'I have been very, very careful,' he says, 'and I have very critical monitoring parameters that are set up in the studio before I start. For near-field speakers I use Westlake BSMBs and I never run them at an SPL above 87 or 90 at the absolute most. Then there are my Auratones which will never be above an SPL of 83. I've worked that way for 25 years.

'Of course, when we're doing an R&B track or a big kick-ass thing I like to play it loud once in a while, but if it's at an SPL of over 100 I'm very careful not to listen to it for more than five minutes out of the hour.

'About 15 years ago I did make a major change in terms of the way that I listen, in that I previously used three parameters: large studio monitors, near-fields and small-scale. Now I've eliminated the large studio monitors and I go just with the near-fields and small-scale. I love my Westlakes, not only for their accuracy, but also their imaging which is to die for. You can tell probably every 15° of position in the panorama, and that's pretty amazing. At the same time they have low impedance in the mid-range, so they're powered by four amps out of Norway which are fantastic. I also love Auratones; I have a lot of them and the older ones are the ones that sound really good.'

As for microphones, Sweden does not experiment too much during a session these days, being able to pretty much discern what will work best with a specific voice as soon as he hears the artist speak. 'As the saying goes, The first million hours in the studio are the hardest, and after that it gets easier,' he quips. Nevertheless, he does still like to move the mic around himself, and in his home studio in Connecticut, where he has been based since 1994, Sweden continually experiments with new equipment.

'I'm lucky in that—because of my visibility within the industry—people constantly send me new equipment to try, often before anyone else, so I'm able to remain on the cutting edge,' he says. 'You should see the gear that I now have—Quincy says moving me around is like moving the 5th Army. I have two 75k racks of effects and some very exotic equipment, and wherever I go I bring my own monitors and my own wire, monster cable which looks a bit like garden hose. On top of that I have 17 cases with microphones.'

'Some of which go all the way back. Listen to old Nat King Cole performances and, while the orchestrations may adhere to the kind of standards that you would expect from 1950s recordings, marvel at how the velvet-smooth vocals often jump out. Several of these were captured by Sweden with a Neumann U47.'

'Nat was great,' he says. 'I'd place >

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PreSonus [audio electronics]
< the microphone in the studio, hit the button on the tape machine and he would do the rest. It was so easy, I still have the "1" that I used on him and in fact I also used it on Michael."

Still, while the "1"s enduring qualities are beyond dispute, Swedien recently took a strong liking to another brand that he was introduced to while lecturing in Germany.

"I was in Munich, and I met a guy over there named Dirk Brät neuropur," he says. "He has come up with a hand-made series of condenser tube mics, and I have never heard anything better. He gave me one and I brought it back and used it, and it's just phenomenal. It sounds like the best U47 that you've ever heard in your life."

In his home studio Swedien has a Harrison 42 Series console that is similar to the one that he used on Thriller. "It's absolutely beautiful," he says. "I bought it from Roy Clark in Nashville, and it's like the day it came out of the box. Still, being that it only has 32 inputs. I'm looking for a second one in the same kind of condition so that I can put the two of them together.

"On the other hand, I also love the SSL 9000 which, although I don't like that the SSL 9000, which is also a good choice. Working with a great artist and great material brings more responsibility to a project, and I think the most important thing that I've learned from Quincy is never to take that lightly. We've worked together for over 40 years, and I promise you there is no one like Quincy in terms of the quality, and the musicality, and the good taste, that he brings to every project. In the beginning I was 21 and he was 25, and one of the first recordings that we did was Dinah Washington's 'What a Difference a Day Makes', which was not a bad start.

"There again, both Quincy and I bring all of our experience to every session, and so another thing I've learned from him is that the kaleidoscopic approach is really the most efficient. When you play one of my mixes, for instance, you can hear it in a certain way, but then you can play it again and listen for something else. I mean, they're still trying to figure out some of the techniques that I used on Thriller, and I've got stuff buried in there that people will be studying for years."

Indeed, having conducted a master class in music engineering at the University of California, Los Angeles (UCLA), and conducted lectures and recording seminars for numerous universities, colleges and industry organisations both in America and overseas, Swedien is more than happy to disclose his work methods.

"I don't believe in secrets," he asserts, "and I'm criticised by both Quincy and Michael for that. Michael especially, he'll say, 'Bruce, you can't go telling people all this stuff', but I don't believe in that, the reason being that I have yet to find anyone who really understands what I'm talking about. As a result, I'm more concerned about finding people who will understand my approach to what I do than I am about anyone stealing ideas. That's never been an issue. So, microphone technique, multitrack recording technique, I go through all of them."

In that case, what techniques did Swedien use on Thriller and what kind of sounds are buried in there? After all, he did just say that he doesn't believe in secrets, and as for Michael, well, he'll get over it...

"There's the title track, for instance," comes the immediate reply. "On the intro there's a little rhythm track that commences the music, and I purposely limited the bandwidth on it so that as you listen to it your ear adjusts to that spectral response. Then, all of a sudden, the real bass and kick drum come in..."
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and the effect is really startling. So far I've told people about this but nobody has verbalised to me how it actually happens.

Maybe, but hasn't he heard anyone else initiate the effect?

'Oh yeah, I love that.'

So there's a little nibble about the title track, but what about one of the other hit songs on *Thriller*?

'Well, when we recorded 'Billy Jean', for instance, Quincy told me, "Okay, this piece of music has to have the most unique sonic personality of anything that we have ever recorded". Now, that's a hell of a way to go into a recording, so I thought and thought about it, and when it came time to do the rhythm track—consisting of drums, bass and so on—at Westlake Audio on Beverly Boulevard (in LA) I did everything I could imagine to really make it sound unique, even to the point of calling my old pal George Massenburg and borrowing a super-high-quality small recording console from him.

That was the first time that I began using a specially-built, 8ft square, plywood drum platform, courtesy of the carpenters at Westlake, which is a great studio by the way. I also had a special huss drum cover made, and I took the front head off the drum kit, put cinder blocks in there to hold it still, put the cover on and slipped the microphone through. Then I made a special little isolation flap that went between the snare mike and the hi-hat mike in order to give much better imaging. Consequently, I think that track really is unique, see if you can think of any other piece of music where you can hear the first three drum beats and know what the song is. That's what I call sonic personality.

'Separately these elements are all small things, but as Quincy told me early on when we began working together, everything is important. Every little detail. Anything you can do to enhance the image is important, and I admired that approach so much early on that it's kind of become a part of my everyday life, and it drives people crazy.'

This amounts to spending a lot of time in the preparation, but not when the musicians are around. For, once a session is underway, Sweden is intent on retaining the spontaneity.

'Something else that Quincy brought home to me is to learn to listen to my instincts and believe in them,' he says. 'Also, from Duke Ellington I learned a lot about listening for the primitive elements in music, and being sure that none of them are overlooked. Primitive is important, and I would hate to be the first person in the world to make a perfect record. I don't think that would be very interesting.

'It's all about feeling responsibility to the music that we're working on, because anything that we do in a studio is, for all practical purposes, carved in stone. I mean, it will be there forever if it's important enough for people to examine, and like a good photographer I always try to look at my best pictures.'

CONTRARY TO the popular proverb, Sweden is of the older he gets, the more he knows, and he also states that, in his experience, the bigger, more successful artists are also nice to work with.

'Michael is a doll in the studio,' he says. 'You could not possibly ask for anyone better to work with. At the same time, I'm an only child and I would love to have Quincy for a brother. The same goes for Burt Bacharach. Oh man, the list just goes on and on.

'Although there are a few choice names that didn't make it on there...

'Sometimes I think that I would have loved to work with Sinatra, but Quincy's worked with him and so did my pal Phil Ramone, and as a result I think that it's better if I just remember his music. I recall a time when Quincy and Phil were working with him while I was in LA producing a band named Missing Persons, and they called me and said, "Bruce, just be glad you're not here."

'Other than that I usually hear from Phil Ramone on my birthday. You see, we have an old joke. We used to call each other at Christmas and say, "Up is louder..." and hang up.'

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What more do you want from a mate?
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I HAVE BEEN spending quite a bit of time recently travelling between what feel like parallel worlds. And it seems I have a confession to make. I will try to explain.

If somebody offered me an album's worth of 'far too cheap' multitrack tape—it fell off the back of a lorry guy—to record an album, I would not want to know about it. The album project would be too close to my heart and far too expensive to even think about the risks let alone the karma. Yet when it comes to software, until recently, my computer seemed to have amassed truckloads of applications, some inherited from employers, others borrowed from friends, but most of which I never quite got around to buying: and that made me a pirate and guilty of some pretty serious crimes in most countries. The funny thing is, that many people I know in the industry could probably say the same thing at one time or another.

Some time ago I was in my office in Nashville, thinking about Bill Gates' new offering of Microsoft Office 98 for Macintosh. I'd read the reviews and it looked pretty good—corrected many of the tiresome bugs of earlier versions and for the first time was not artificially hobbled to run slower than the Windows versions (maybe something to do with him being an Apple shareholder?). There was only one small problem—the respecatled one wanted over $100 for his new software and I was a little short of cash that week. So I looked on the internet, shopping around to see if there was any special deal available. Soon enough I came across a pirate site offering a complete image of that CD for free download at a scary fast T-1 speed. I confess, the temptation was significant: well, can you think of anybody in this world besides the Sultan of Brunei who needs $100 less than Bill Gates? But of the simplest of analyses will reveal that morally, ethically and legally, the bottom line is that piracy is piracy, period.

A little bit more poking and prodding my way around the net revealed that it was not just the mega business titles which were being pirated; almost every other kind of specialist software was there for the illegal taking. Without exception, all of the software that many of us use as our primary tools in audio production were also being cracked, hacked and illegally distributed. I talked about this subject with many software-developer friends at the AES in Amsterdam earlier this year. It was immediately clear that any attack by pirates on intellectual property and copyrights strikes at the heart of free enterprise. Many software developers are 2-man or 4-man companies with highly innovative products, some of which have taken years to bring to market. There is not a Bill Gates among them, although if the criteria for riches was based upon quality of product they should all be richer than they are. The reality is, that like most of us, they are running their businesses on tight budgets with development and marketing costs factored into a final selling price. Piracy throws all of their business equations and models into a cocked hat.

There's a paradox here, in that we make our living in the music business either directly or indirectly from royalties earned on sales of recordings. We fight hard to defend those royalties and yet to date, as an industry, we've done almost nothing to defend the copyrights of the tools that help us to earn those royalties in the first place. So in the best tradition of pockethanded gamekeeper I've been getting involved in putting together an organisation which will be up and running by the time you read this article. It's specifically targeting these issues in our industry.

The most reliable figures for software piracy (from a study conducted by the International Planning and Research Corporation, funded by the BSA and SIA) show global software piracy figures of 43% and in excess of $1bn in 1996 (for all business-related software excluding operating systems and client-server applications). That is almost one in two of every software application in use. Okay, so you say the figures are distorted by China and Hong Kong or South America. Actually, the truth makes surprising reading with 27% of all software in the US being illegal to the incredible value of $2.5bn, 54% in the UK ($55m), 34% in France, 5% in Netherlands or $221m, 55% in Italy for $340m, and 11% in Japan for a whopping $1.2bn. The sheer size of these numbers is as staggering as is the breadth of the problem. It would be simply too naive to think that we in the pro-audio industry are alone in being immune to piracy.

The perceived value of software is no longer the price of a diskette, but its intellectual worth is still in doubt.

Dave Powell dives deep into the pool of bootleg 'warez' and its consequences.

The pirate's fear of the penalty

IN THE US, if caught with pirated software, the infringing company may be liable under both civil and criminal law. Authorities can seek civil damages in the amount of their actual or statutory damages of $100,000 per work infringed. Criminal penalties for copyright infringement include fines of up to $250,000 and jail terms up to five years, or both. Last December, President Clinton signed a law called the 'No Electronic Theft' (NET) Act that allows for criminal prosecution of copyright infringement, even where there is no profit motive, closing a loophole in US copyright law. FAST points out that, 'breaking the law could have serious consequences for your job and the organisation you work for; threatening both your own and your employer's reputation and future prosperity'. In the UK, legal penalties include unlimited fines and up to two years in prison. Personally, you could lose your reputation, promotion prospects or even your job. There are no mitigating circumstances and no organisation would condone or defend either illegal copying or the use of unauthorised software. Also, most countries have now subscribed to a UN agreement designed to harmonise international copyright law. This agreement should be ratified later this year and come into local law shortly thereafter.
pirated copies of software. If you purchase software installed by someone else, ensure the vendor provides all the original materials and proof that the product is licensed and itemised in the receipt. Equally, we should purchase genuine software from reputable dealers or manufacturers and install it in accordance with the license agreement. Additional tips for the purchase of software can be found on the BSA-SPA-FAST and other websites. The license agreement that comes with each piece of software explains what you are entitled to do with it—it will not include making copies of unlicensed software for colleagues. It will help to know what software is in your company as the company is liable for the actions of its employees and can be sued for copyright infringement even when the management is unaware that unauthorised software is in use or being downloaded.

And if the logic of supporting our industry is not sufficient, the potential penalties for using pirated software (even unknowingly) make for sobering reading. FAST explains simply. The IUKI Copyright, Designs, and Patents Act 1988 states: The owner of the copyright has the exclusive right to copy the work. That means it is illegal to copy software without the copyright owner's permission.

Using pirate copies of software is a civil offence. This means you could be sued by the copyright holder. Buying and swapping pirated copies is aiding and abetting the criminal offences of distribution and sale, for which you can be prosecuted and thereby obtain a criminal record. According to BSA, it's illegal to distribute 'warez', which is a slang term for pirated software. Downloading or purchasing pirated software is illegal.

Legal software comes with required serial numbers or keys to install the software. Providing them for others to use with pirated software contributes to copyright infringement and is illegal. The BSA also states that you should not use 'cracks', which are programs that will modify the copy protection in an application, enabling you to use the program after a trial period has ended, or access features of the program restricted to fully licensed users. It is illegal to make these modifications.

Lastly, SPA has indicated that the additional concept of Contributory Infringement can be quite wide ranging. Anyone who knows or should have known that he or she is assisting, inducing or materially contributing to infringement of any of the exclusive rights by another person becomes liable for contributory infringement.

For those who need further convincing (convincing!), there are also technical risks. For your own sake, don't use pirated copies. They are usually the work of clever, but often mischievous programmers, who, once they solve the protection problem, might try to stir up more trouble by inserting viruses. Also poorly worked hacks can potentially damage your system.

To cut to the chase, Jeff Bloom of Synchro Arts fame, Julian Searle, an enormously talented network programmer and reformed hacker, and yours truly have put together Copyright Control Services to educate professional software users, and target blatant pirates for action. If you'd like more information we'd love to hear from you.

It's our industry and computers and software will play an ever-increasing role in it. It is up to us to decide how what sort of future tools we want to be able to enjoy.
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The Whole Truth And Nothing But The Truth
Craig Leon: best of both worlds

Tempering style with flexibility is an enduring challenge facing both artist and producer. Phil Ward corners producer Craig Leon and grills him on news of artist Deborah Harry

Working on both sides of the Atlantic, Craig Leon revels in producing records from almost any genre of music. With credits ranging from punk to country and blues, from industrial to folk and jazz, and from mainstream pop to esoteric classical, the only common denominator throughout his producing and arranging career is individuality. His roots lie in the R&B days of Criteria Studios in Florida, working with Alex Sadkin. In the early 1970s they built a small demo facility, while Sadkin was a mastering engineer at Criteria. One such demo was for Richard Gotteher and Seymour Stein, who ran Sire, the seminal New York label. These sessions—with the Climax Blues Band—led to sessions in New York for the label, and by the osmosis of the industry Leon closed up in Florida and became Gotteher's full-time assistant.

In those days, especially in America, "A&R" and "production" were the same thing, Leon recalls. You came to it from a musical background, whereas in the UK you worked your way up through the studios.

As a case in point, Leon discovered Blondie while working as an A&R scout at Sire, having snapped up local luminaries Talking Heads and The Ramones. As an A&R man, he was in the thick of New York's punk explosion. As a producer, he supervised the recording of every significant band of the era during the making of Live At CBGB's for Atlantic in 1976.

Blondie lived across the street from CBGB's, and became the 'soundcheck band.' Leon recounts, "routinely their set every week while we prepared for that evening's recording. Richie Gotteher and I had formed our own independent production company, and the soundcheck tapes prompted us to make a single with Blondie—a project which turned into a year during which I arranged and rehearsed every song they had.

By the time the band was signed—to Chrysalis—around 40 tracks were ready. Although Gotteher is credited as producer of the first two albums, it was this material—honed by Leon—that constituted Blondie and Plastic Letters, while much of what was left over—including 'Heart of Glass', known to Leon as 'Disco Song'—was inherited by Parallel Lives producer Mike Chapman. At a time of anti-school idealism, with multitrack analogue recording established as the sensible way of making records, Leon's production was characterised by an innovatory approach, in which he took unstructured ideas and pieced them together into songs. The method of tape splicing, which enabled him to dovetail grooves into pop structures, presaged today's sampling techniques directly.

Most bands in the US were just trying to get their live performances represented on record," he says. "Blondie were different. They brought a layering, sampling approach; we'd do a riff, and work that into a verse; then we'd do another riff, and work that into a chorus. These would be chopped together into a whole track, and then Debbie would take that away and come up with the vocal.

Most of these sessions took place during downtime at Plaza Sound, where Leon was making the first Ramones album. There was a big rehearsal hall for the NBC Symphony Orchestra on top of the Radio City Music Hall, and we'd do stuff there at night. Another >
blag was Bell Sound—now Scar Sound—which was the old home of The Shangri-Las, The Four Seasons and all those early sixties recordings.

It wasn’t all punk, though. Country artist Rodney Crowell. Tex Mex pioneers Sir Douglas Quintet; and folk feminists The Roches were among credible clients, and there was even an album co-produced with Bob Marley and Lee Perry featuring The Wailers.

By the early eighties, however, the American music scene was losing its flavour for Leon and he was drawn to the UK, where he is still based today. Following his muse for idiosyncratic, original material (‘Those that want more of an arrangement approach’) he wound up producing several bands on an indie offshoot label at Virgin called Statik, including Gun Club, Flesh for Lulu, The Sound and his own wife Cassell Webb. From this basis he has become one of the UK’s most eclectic producers, with acts as varied as The Fall, Jesus Jones, The Primitives and Mark Owen benefiting from his touch. For good measure, Canadian noise grinders Front 242 and nutty Japanese girl trio Shonen Knife also figure on his CV.

Next month, though, a new Blondie album is scheduled to appear, picking up where Leon left off, piecing together Chris Stein and Debbie Harry’s arty ideas in that Warhol way. This time his multitrack renderer of choice is the Otari Radar, which allows him to continue his art of assemblage with the luxury of digital cut and paste, where the Radar scores over other DAWS, according to Leon, is sonically.

The Radar doesn’t have a sound, that’s why I like it. When we were doing Clem Burke’s drums on the new Blondie album, we tried the Studer A827, the Sony 5504 and the Radar. I needed to get in to a digital medium because of the amount of editing I do during the work in progress, and the best sound out of those three—and the closest to the real analogue sound, without that change to the top end of the cymbals, or that loss of depth that you normally get with digitally recorded drums—was Radar. You would be hard pressed to play the analogue tape next to the Radar and hear a difference.

Recall last-minute editing equally suit Leon’s work-in-progress approach. But there are no hard and fast technical rules in Leon’s domain, in which an old Neve console or a Capricorn can provide the solution.

Right now I like to take everything from Radar into a Capricorn digitally, just to check balances, but the mix may not end up that way; it takes 25 seconds to recall something on Capricorn. And needless to say we may be in Electric Lady working on an SSL, so there’ll be a different mix from that.

No gear is universally right: at the same time, the range of gear available for different sounds is fantastic. I love an old MCI sound, for certain analogue sounds, and I love old Neves, but if they’re not available I can make do with something else. To me, the feel of the

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Deborah Harry with Craig Leon

actual music is a lot more important. I came from playing music; I learned engineering by the seat of my pants along the way—mainly because people wouldn't make records as loud as I wanted. I watched Tom Dowd at Criteria, and figured I'd have a go at that and just make it a little louder.

Leon chooses a studio by the quality of the live area and the atmosphere for the band. Studios are right to emphasise their unique creative vibes over what gear they have. If you've got the right music and the right feel, it doesn't matter if it's recorded on one console or another, or one tape-disk format or another. I've done records entirely on MCIs. The Ramones record was done on a now-extinct API console, with 550A EQs, mixed manually, 16-track to a 2-inch M 79—one of the worst machines ever made—and mixed down to another M, with a bunch of odd compressors and stuff. And you get other records mixed on SSLs which sound great, too.

"Monitors are the most personal thing; you have to get those exactly right for you. Same with amps: if you go with a Fender amp or a Dynaudio amp, they'll sound radically different. But in terms of EQ and desks, well... Nowadays you can chop and change so much. If I don't like the desk EQ in a place, I'll bring in my own stuff, so I would never say I'll only go to a studio which has, say, a Neve 8800 desk. As long as your favourite records sound right to you in that room, that's basically it. As long as the band is comfortable to be there, too, of course."

Reacting against the decadence of albums by Steely Dan, The Eagles or Emerson Lake & Palmer, Leon drew upon the rootsy heritage of Florida R&B and his early radio influences when tackling mid-1970s New York punk.

"All the pioneering producers of country, blues and folk were the guys that I thought were really cool. There was Alan Lomax and the recordings of Muddy Waters in The Library of Congress; the Chess brothers and Howlin' Wolf's 'Smokestack Lightning'—one of my favourite records. I was listening to those on the radio when they first came out. Hank Williams, Chuck Berry... The appeal of those records was that they were very direct. Phil Spector blew me away; here was a guy who could make a little Wagnerian opera condensed into three minutes on a bit of plastic, using studio technology."

Other influences were the record company executives who were the precursors to today's independent producers. "Richie Gottehrer was one of those old-timey songwriters from the fifties; he wrote 'My Boyfriend's Back', and so on—again a primitive, rootsy style of recording."

"Once I moved to New York, the fifties jazz producers really hit me. I wanted to make records as direct as Sketches of Spain by Miles Davis, whatever the genre. As little interference with the music as possible. Even people like Tom Dowd had made very slick records, not raw at all. Because I did punk first, my first records were raw—because the music was raw. But I'm equally proud of some of the country records I've made, that had the same philosophy. Things like Rodney Crowell, Guy Clark, Sir Douglas Quintet. Take the Roches record I did with Robert Fripp—acoustic guitar and vocal, that shit. Oh, Tony Levin plays triangle on one track..."

"We would switch the studio..."
When we changed to SM 911, 7 or 8 years ago, we got that sound back. It has a really good musical edge.

When BASF SM 900 maxima came out we started to use that on the 24 track – it gives me that sound I want.

Producer of the Brit Awards “Album of the Year” 1997 “Everything Must Go” by the Manic Street Preachers and Winner of the Music Week “Producer of the Year” 1997, Mike Hedges has produced hits of artists such as Texas, Everything But The Girl, Siouxsie and the Banshees, The Cure, The Beautiful South, Geneva and McAlmont and Butler.
Leon admits, ‘It was seen as very slick,’ and we were distancing ourselves from all those smooth productions of the seventies. And all the old country, jazz and blues records that really liked were very natural-sounding, where you would only use reverb for a specific effect.

It’s a big mistake, for example, to think that Phil Spector’s wall-of-sound records were washed in echo. Sometimes there are overall effects on voices, but in general it’s pretty dry. There’s just a whole lot of things going on at once, and what you hear is the ambience of the room. I’ve always been fond of placing room mics in odd places to enhance that; you walk around the room with one ear covered so you’re listening like a microphone—sometimes I’ll move a couple of grand pianos in front of the drum kit, and put a mic into each one. I did that with The Ramones—you get a very dry sound, in fact, but a big sound as well.

Experimentation with what you had was the thing. Technology was very expensive in those days, and most of my records were on shoestring budgets. We had the Cooper Time Cube, which is all over the Ramones album, but that was about it. Level and tape compression were the keys; what it did to bass and drum sounds was amazing. You have to remember that there were only a few primitive compressors and limiters in those days. Tape compression was a specific effect, it wasn’t just the punk ethic dictating loudness. You got a big, round bass drum sound from it. Even jazz producers would use it for the same effect.’
I thought, yeah—this is my kind of band.'

All recording is sampling, according to Leon—it's just that the mechanical means to do it have become more efficient.

Recording is a manipulative process in the first place. When they put a mic in front of Robert Johnson sitting in a hotel room playing guitar, the result was not just Robert Johnson. It was Robert Johnson caught inside a medium. Something was changed. You can try to duplicate as much of the input as possible, but you're making a transfer. 

You're just trying to package the spirit of what's being expressed in the most suitable way for the final product, whether you start with a bottleneck guitar or a Fairlight.

People in recording get hung up on analogue or digital, one kind of reverb or another. Anything which improves the picture is okay. But you can't do it to the point where you overwhelm the sound. It has to come through on its own.

One thing that makes the modern era so much better than when those guys were making those old records is that the sound of the nineties can be the sound of anything. You have this whole storehouse of information to draw on. The way that dance music samples antique sounds and incorporates them into modern recordings has been really positive. The world is not so structured any more.

It is arguable that the role of the producer is to find a structure in this confusion—but if the producer, like Leon, is a good A&R guy too, things should fall into place naturally.

'Every time you go into the studio you should hope to bring out the most individual characteristics of what you're doing. Producers don't make the music; you're working to bring out the vision of the artist on a mass communications medium. The best producers are attracted to you because of some old record you made. And while I'm very glad to have worked with The Ramones or The Fall, I have to say to them, well, I really like your music and I want to make your record, not theirs again.'

INITIAL meetings with bands Leon won't talk about studios or records, least of all his own records. 'You're going to spend a lot of time with someone, and you've got to establish some common ground. But I'm only interested in them, and what's happening now. With this new Blondie project I was very keen to document what they're doing now, and have none of that, 'gee, wasn't it great in the old days.' We did that already, and it documented those ideas perfectly well. The new record should not refer back to that period at all.'

'Now comes the guy in the loudest band in the world will tell you that his favourite album is a Nick Drake, or something like that. It's very important to focus in on personal details like that. You can take it as given that you like the band's music once you've heard the tape or seen them live and you've agreed to work with them—and later on you can get to the specifics of each individual sound, but in between, you've got to establish a general understanding of what they're trying to say.'

A new project with Ray Cooper, Elton John's percussionist, is already underway. 'It's going to be live drum loops with various ethnic drums, with Ray directing, then using those loops as a basis for tracks with invited musicians. Maybe a string quartet on one, a vocalist on another. Eric Clapton doing straight country blues on another... Then we'll make a 'Ray Cooper Project' band and do some live shows. 

'It goes back to an album I made in 1980 using a prototype Linn drum, on which I made up a load of North African rhythm loops and fused these with simple, ancient Greek music on synthesisers. 

For trivia buffs, this was called Nomos, released by Chrysalis in 1981. But to find out where Leon's tireless search for originality may lead him next, you'd better just keep an open mind. I'm not averse to having different engineers at different stages, because different people are good at different things—even for different songs on the same record. It's an ongoing process, and there are always people that you'd like to work with if you get the chance.'

And just to prove his eclecticism, Leon answers the obvious last question with a less than obvious choice... 'I'd really like The Prodigy to do some remixes of this new Blondie stuff.'

Wouldn't we all?
Colosseum

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Introducing the first digital multitrack production studio in 24 bit
Lavender Castle

The new Gerry Anderson television series is a space adventure with a difference, as Simon Croft discovers when he talks to Daren Cox, sound editor at UK facility Hullabaloo.

The 21st century already has a few pioneers working full-time in outer space. These bold adventurers devote their lives to pushing back the boundaries of what is possible. One such man is Daren Cox, sound editor.

Cox is currently working diligently on *Lavender Castle*, a 26-part television series from Gerry Anderson. Mr. Anderson is no stranger to space himself; after all, it was Anderson who inspired generations of children to run round the school playground yelling, 'You take Thunderbird 3', 'FAB, Virgil', or just plain 'Wooloomooloo!'!

*Lavender Castle* may not be a Thunderbird, Stingray, Fireball XL5, UFO, Space 1999 or Captain Scarlet remake, and its plot occupies more fanciful territory, as Cox explains: 'It's about these space travellers who are looking for Lavender Castle, which is the centre of the universe, and the place where goodness radiates,' he enthuses. 'They are the goodies, but they come across baddies along the way. The main one is called Dr. Agon, who flies around in this big, menacing looking ship. He tries to stop them by blowing them up and there are various other baddies along the way.'

So, even by Anderson's standards, *Lavender Castle* is no 'ordinary' space adventure. Visually, the spacecraft make even the implausibly dimensioned Thunderbird 2—the retro-rockets on which used to help the lumbering craft descend—look like an orthodox link in aerospace history.

Cox and his Hullabaloo partner, Simon Hall, wanted to give each spacecraft a unique sound and to move away from the 'tub boat at half speed sound, which has become the de facto spaceship effect. It soon becomes clear why a rethink was in order.

'We've all seen space ships living around in films and they all look and sound similar,' Cox explains. 'One of the ships in *Lavender Castle* actually looks like a thatched cottage, so creating a sound for that was quite a challenge.

'Obviously, you needed the sound of a rocket underneath it all but then also the creaks of all the wood,' says Cox. 'There's also a kitchen which you can see sometimes, so we've got pots bubbling away in there and a fire crackling away. There is an old Rolls-Royce engine in there as well, so that it sounds quite old, but there is power there.'

Armed with samplers, keyboard instruments and other sources, Hullabaloo went about making just the right effects for each spacecraft and each planet visited. It turned out to be quite a job in itself.

For that one spaceship, there's probably about 16 tracks, just to make up the one noise. It's quite a complex job,' Cox reveals. Many of the other craft were no less involved.

There's one pirate ship in space, so you have the noise of the sails but also the noise of a spaceship, says Cox.

At the time of writing, Cox is editing Episode 24 of 26 but, as the episodes have not been worked on in exact sequence, he actually has another five to work on.

The chance to pitch for the sound posting on *Lavender Castle* came through Cosgrove Hall Films, which won the contract for the model animation (see side-bar). Previous Cosgrove Hall animation contracts have >
< included Danger Mouse, Count Duckula and Wind in the Willows.

Hullabaloo came recommended but Cox notes, 'If Gerry Anderson didn't like us or thought we weren't up to the job, he would have the clout to go somewhere else. But we seemed to hit it off straight away when he came here to have a look and he liked what we were saying about the project, some of the ideas we had.'

Ask Cox about the difference between working with animation and 'real' footage, he initially jokes: 'I suppose the actors are less temperamental — although some of the animators might disagree when puppets fall over in mid shot!' I've worked on dramas and documentaries but I actually think that animations can be a lot more challenging. You are starting from absolutely nothing, so you are creating every single sound.

In a drama, you would have the atmosphere of where it was recorded on location. With animation, if the character wears a cloak, you have to create sounds for that. You have to make that cloak move and come alive. That is true of every effect: if they move something, put a cup down, you have to do it. All you've got is two tracks of dialogue.

'We use over 70 odd tracks for a 10-minute programme. With computers, it is quicker than it used to be. A programme of this length used to take about five weeks for four dialogue editors to work on. I can now do it on my own in five days.'

One thing animation does have in common with feature films is the difficulty in getting to a locked off cut and the flexibility of new technology has been used to keep some fluidity in the process for longer than would have been the case.

'Sometimes, we work to an off-line picture, but then there are still scenes missing where they want reshoots,' says Cox. 'Whereas the models are shot in the studio, the space ships are computer generated, so there are time limits and sometimes we may be waiting for the computer shots, or reshoots of the model.

'So we can be working on shows where all the scenes are not there or you know there are scenes which are going to change, so you don't do anything for that part. It can get a bit higgledy-piggledy but normally, a good 70% of the scenes are there for us to work on. Then we go over it again when the new scenes are there, then go through it with the director just to make sure it's all okay for them.'

Although postproduction started around 18 months ago, the majority >
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< of the dialogue recording was done two and a half years ago, before the second Hullabaloo studio was built. As with most animation, the dialogue was cut to the storyboard before the animation was created. Cox explains:

'The dialogue is cut to the length of the body of the programme. The body is nine and a half minutes and the titles make it up to the full 10 minutes.

'You come into body with all the gaps for action, scene changes and everything else. Then it goes to a process called barcharting, which means that someone counts each line of every syllable so that if "La-ven-der Cas-tle" lasts three seconds, at 25 frames a second that's 75 frames. Someone has to go through every frame so that the animator knows how long to move the mouth and to move the body for action.'

With every other sound added in post, it's small wonder than Cox does not attempt every aspect of it single handed. Mixing is handled by Simon Hall and again, maximum advantage is taken of computerisation.

'On the Avid AudioVision, you may make up sequences of the same scene over and over again,' says Cox, 'so we lay one sequence just of dialogue, which Simon will premix. Then we do a spaceship premix, with them all whizzing around. Then there is an atmos premix, followed by an effects premix. We put all that together with the music that's specially written and go for the final mix.' Just the premixes take about a day to complete, although there is occasionally some scope for reusing sequences.

'There are things I've done and kept which we can use again, such as an atmos they keep going back to,' he says. 'If it's premixed on the computer, it can bring it in again to save Simon's time.

'This is proper film mixing,' Cox concludes, 'It is a complicated little process. It's not quite and it's not easy. It takes another whole day to do the final mix for a 10-minute programme.'

Hullabaloo facility tour

Part of the Cosgrove Hall media complex in Manchester, England, Hullabaloo was set up two and a half years ago. Founders Daren Cox and Simon Hall formerly worked at John Wood Sound in Manchester.

Hullabaloo has grown since 1996 to employ three more people. In addition to the Dolby Surround dubbing studio, it now has a stereo studio, set up for multi-mic and Foley work. It also has a number of edit suites and a machine area, which includes the computers and video systems, so that working areas have the minimum of heat and noise generating equipment.

John Wood Sound was an early adopter of the Avid AudioVision system, so it is no surprise to discover that they are central to Hullabaloo's editing and dubbing set-ups.

'The main studio has the Avid AudioVision 16 track and an Akai DD1500, so we have 32 tracks which we can play off and record onto at any one time,' explains Cox. 'We can also put into that Hi-8 and DAT and desk we also have the Amek DMS.

In fact Hullabaloo became something of a test bed for Amek, which is based in nearby Salford. Although it was Amek's first digital console, the technology developed was soon adapted to some of Hullabaloo's more arcane postproduction techniques.

The second studio has a recording booth which will house about five people and the paraphernalia necessary for Foley recording. Although Hullabaloo obviously benefits from its connection with animation experts Cosgrove Hall Films—and other media companies in the complex, such as nonlinear film editors Flix Facilities—it also works for UK broadcasters the BBC and Granada, as well as independent advertising agencies and clients in the corporate sector. Nonetheless, Cox estimates that about 50%-60% of Hullabaloo's work is sound for animation.
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"It's the overall flexibility and expansion capabilities of Avant that convinced us."

Alan Swelling, Avant Post Production, UK.

"Mrs Brown" image courtesy Ecosse Films.

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That networking is a hot issue is, in part, driven by the interest of computer systems companies such as Hewlett Packard, Silicon Graphics, Tektronix, Oracle, Apple Computer and Microsoft. These companies are extending their existing data processing and transaction models to support video, audio and associated metadata. And they are not alone—the push from the suppliers is endorsed by radio and TV chief engineers worldwide and the traditionally conservative film people have been equally quick to see the possibilities.

If a satisfactory architecture for the transport of massive amounts of digital video data can be established, it follows that the same technology will be adopted in many applications for audio networking. To do otherwise would be perverse. Accordingly, it is worth taking a look at some of the manufacturing solutions. Although it is becoming an increasingly difficult category to define, the companies here may all be considered manufacturers of 'hardware' DAWs.

Akai was one of the first digital audio manufacturers to use existing Ethernet network technology to allow the various items in their product range to be interconnected. No audio is transferred over the network. It is designed to be a sophisticated means of remote controlling many of the functions of multiple machines. As an example, this allows multiple DD8 dubbers and DD1500 DAWs to be controlled from a single DL1500 remote. A DL1500 in a dubbing theatre can 'dial up' any DD8s-DD1500s in a facility, put them on line and control all functions. Audio is delivered to the theatre by conventional tie lines. Project interchange is efficiently dealt with via 'sneakernet' and this includes direct access to audio disks from Avid systems among others. By keeping it simple Akai have delivered a solution that is stable and useable. It employs today's technology with very little management overhead. Akai feels the use of high-bandwidth networks is still in its early stages.

Many of the larger studios have already investigated the installation of networks to distribute video to various locations within the facility. Given the high costs involved in installing and maintaining such a system, and the high degree of management it requires, a logical step from this may be for the studio's audio departments to use any spare bandwidth on the video network for audio distribution. This gives audio manufacturers a problem in that the amount of available bandwidth on the network for audio may vary depending on the level of video traffic.

Network technology itself is rapidly changing both in terms of physical specifications, cabling and the maximum bandwidths achievable. Given these changes, it is understandable that none of the major studios have yet committed to the installation of large-scale networks. Those that have tried have been beset with problems (for example, BBC News 24).

Akai has been following the developments in network technology but feels it is too early to commit to a particular medium as many of the current solutions will not be able to achieve the full expectations of users at a sensible cost. Akai's digital audio products have been developed with an open-architecture. As such, it is possible to quickly add support for any network technology that becomes established as a standard, whether this is Fibre Channel, Gigabit Ethernet or some other variation.

Akai has a robust, practical and economical system that is field proven and they intend to ensure future developments to extend the functionality, meet the same criteria.

AMS Neve's Audifile first appeared >
The company has remained at the forefront of technical innovation, and is stillimitated more frequently than one might suppose. Today’s third-generation network capable version of AudioFile is based on the latest 24-bit hardware platform.

Due to success in major broadcast facilities with multiple rooms working on various aspects of projects, AMS never understands that elegant technology is not enough. Users require complete process solutions. In company speak this is referred to using the umbrella term of ‘WorkFlow’.

The WorkFlow audio resource management system consists of three layers, any or all of these may be specified. The Audio layer handles distribution of digital audio signals from multiple sources to multiple consoles/editors. The Control layer is a control bus that provides shared access to machine resources (ATRs, VTRs), plus the control of remote I-O units. The Project Management layer deals with networking of AudioFile editors to enable sharing of projects and sources, plus off-line centralised archiving. Much use is made of industry-standard technology such as MADI, ESI-S and OMFI.

StarNet is a network designed to enable multiple AudioFiles to share access to any project or sound source on the system. Any AudioFile control surface can be connected to any hard disk array. When connected to StarNet, the direct connection between the AudioFile processor and its local disks is replaced by connection to any storage device on the network. Users can attach to any disk on the system just as if it were a local disk independent of traffic elsewhere on the network. The actual disks may be physically installed with each AudioFile or centrally in a 'disk farm'. Disks are often housed in a removable chassis to enable 'sneaker-net' project exchange with other facilities. Different projects can be stored on different drives. This simplifies tracking a project through postproduction as all the elements are in one location rather than distributed across several disks in a central server. Since each AudioFile processor and each disk bank has its own 400MHz optical connection, the performance is consistent, even with all users playing audio at the same time. The directory of all disks may be viewed by any user with the correct authority, whether attached to the disks or not. Shared disks, such as effects libraries can be browsed by any user and recordings auditioned in real time. The selected items may then be copied to the project drive at high speed. If multiple rooms are working on the same project they may all wish to record and playback using a single drive containing all the project elements. Each user is given access to a number of tracks of the disk which ensures that all users can playback their allocation of tracks at all times. Independent of other users.

Projects from OMF and Lightworks edit systems can be played back and edited directly without file translation. The Media Toolbox provides network-wide off-line backup and restore of either disk images or individual projects. This enables the 'housekeeping' aspects of the system to be carried out independently off-line. The Media Toolbox also enables off-line conversion of OMF and WAV files to AudioFile format.

AMS Neve believes existing IT network technologies are unable to provide the required functionality, so StarNet uses a hybrid of proprietary and generic technology. The drives are grouped together logically to form a large server for the network, without requiring a stand-alone server. The central network router provides a high bandwidth path for directly connecting disks to AudioFiles. Other network functions are administered by network controllers which are interconnected via the central router. The StarNet system is scalable and may be expanded by the addition of multiple routers and network controllers.

As well as performing network management functions, the network controllers provide an interface to the outside world. The controller is based on:
The mixtreme PCI card has a completely different concept, as not only does it provide 16 channels of 24 bit digital I/O via two industry standard 8 channel TDIF (Tascam Digital Interface) ports, but it also has 24 bit custom digital mixing and real-time DSP effects plug-ins from some of the world's leading audio companies. mixtreme performs like you expect, without the annoying in/out delays that make native mixing and effects impossible to use in real time.

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For ADAT users, there is the SS8IO-2 TDIF to ADAT converter, and you can connect two of these, or combine one ADAT and one Analogue interface. Optional SPDIF completes the range of Digital Connections available.

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The current hardware underlying D-net is a network hub using 100-Mbit Fast Ethernet. The technology employed is really irrelevant to the user and can, in any case, be upgraded as better things become affordable. The point is what this networking offers the user in functionality. In this case audio transfers across the net are typically 10 times faster than ‘real’ time.

The AXIS server uses a hierarchical filing system to manage projects and sound libraries. Furthermore, the server may be used as a central archiving station, backing up material from remote DAWs to Exabyte or whatever. A standard Windows PC can be used to organise all this. There is no need to tie up a workstation. It is also possible to integrate ISDN links into the network.

The DAR approach is a perfect real-world compromise between providing a reliable solution and maintaining reasonable cost. By intelligent use of local storage the required bandwidth is minimised. CD Advance, DARs term for the use of a CD drive as online storage, which also allows CD tracks to be scrubbed as if they were conventional segments can also be accessed across the network. The desirability of this will increase still further when and if DAR develop a CD jukebox as network device.

A lot of work has gone into getting the management aspects right. The system is easily able to take advantage of the rapid developments and falling costs of IT networking technology to improve throughput.

Fairlight, one of the early players in the DAW market, has recently introduced MediaLink which, together with Audiobase, a database to manage libraries, is its first networking effort. Their goal is real-time full-bandwidth networking with no local drives together with multiple simultaneous access to the same data. Currently, the hardware supports 86 tracks of concurrent real-time audio which equates to two or three MFSs plus other machines working side by side with local storage.

There are five states: Project can possess: Media Read, Project Read, Marked for Backup, Append Write and Modify Write. A second operator can save his version on your project using an Extension in Fairlight parlance. To assemble tracks for, say a final mix, dialogue, effects and music blocks edited as separate Extensions can be merged into a single new project.

Fairlight’s server uses a dual Pentium based PC with Windows NT as the operating system running proprietary networking software. Topology is cur-

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Currently 100 baseT Ethernet using Category 5 cable, but with greater bandwidth. A RAID disk array is required to provide the real-time bandwidth. Fairlight expect users to operate with a mix of real-time and store-and-forward. Real time is used for applications such as auditioning library material, at the one end of the scale and final dubbing at the other. While bandwidth remains at a premium, store-and-forward is used for other purposes where real time is not an essential requirement. The use of an NT server enables Fairlight to provide direct connectivity with Avid Media Composers or AudioVisions and with SGI servers used by some facilities for OMF file storage. Together with Fairlight’s ability to use OMF audio only files this considerably extends the appeal of networking.

The company is predicting a big increase in bandwidth capacity, that is the number of simultaneous real-time tracks in the near future. However, European top man Nick Cooke, believes video and audio networks will remain separate for the foreseeable future. Meanwhile Fairlight is offering a practical solution.

**Solid State Logic**, another long-term player in the DAW market, has championed the cause of networking since it pioneered networking for audio production and post-production with SoundNet in 1989. This system provides multi-user digital audio networking by means of a central mass storage unit comprising up to 16 SCSI devices—either hard disk or magneto-optical drives—with tape backup. In 1989, a 300Mb hard disk (approximately 1 track-hour of digital audio) was the zenith, but 1Gb disks quickly replaced 300Mb, and growth has been exponential ever since. It was not long before a SoundNet system could offer up to 96 hours of hard disk storage, with 8 tracks per disk.

Any user with an SSL ScreenSound (and later Scenaria and Omnirax) workstation can select any disk via the SCSI switcher and have sole unrestricted access to that disk for as long as required, and by slaving together several ScreenSound workstations up to 56 simultaneous channels can be provided. Once the user finishes work on that disk, control can be relinquished, thus allowing another user to carry out further work on the material without physically moving the recording medium. This direct connection of the user to the disk is the key to the system, providing full time access to eight simultaneous tracks.

SSL believes the expectation of professional audio users is that a network must operate in real time, and that a source should be available to any number of users simultaneously. As discussed in the first article this has little direct relevance in a classic packet-switched IT network. The real-time requirement is why SSL developed its proprietary HiWay system which carries up to 95 digital audio signals for up to 100m via a single coax connection, or up to 2km via their FreeWay optical fibre option. Combined with SSL’s Hub Router, the HiWay-FreeWay system can provide real-time networking with access to a potential 2280 sources and destinations, and multiple Hub Routers may be interconnected to share sources between groups of consoles and input-output sources and destinations.

Until the vision of seamless interconnection between different proprietary systems becomes a reality, Solid State Logic regards transferable media as the most satisfactory technology for the interchange of large amounts of multitrack digital audio between compatible systems. John Andrews reckons, ‘A reel of tape is still the safest bet at the moment’.

What all of this amply demonstrates is there are wide variations in approach. The common themes are the use of generic IT technology where appropriate and to reduce R&D costs. The current lack of any obvious single answer is a question which arises. ‘Do I wait for a video networking standard to emerge or get involved with audio networking now?’

I do not believe it matters except to major broadcasters and their symbions. For everyone else, the essential point remains: ask very searching questions about the management and security of any system under consideration. The functionality will rapidly improve with developments in IT technology, but an efficient technical solution is no guarantee of a good operational outcome.

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Plagued by a drugs scam, Le Tour de France went from sports to news coverage

Kevin Hilton reports

T is the world's greatest cycle race, a French institution that is recognised in other countries as one of the highest physical challenges a person can face. It is also the most watched of all sporting events, attracting over 10 million spectators along the way. The Tour de France, Le Tour, la Grande Boucle—everybody knows it. But this year it became the Tour de Farce in a 3½-week period that threatened to undermine 95 years of history. Prima donna participants and then a protracted drugs scandal were all captured by the attendant TV cameras, as the high point in the cycling calendar stopped being purely a sports broadcast and became an international news event.

First held in 1903, the race—for professional, individual riders; although sponsored teams of 12 take part, with a ‘star’ supported by others—covers approximately 4,800km, 3,000 miles of road, including the gruelling mountain stages. The majority of the competition takes place in France, but occasionally stages are held in other European countries. This year everything started in the Irish Republic, which, despite the supremacy of French, American and Spanish contestants recently, has made its mark on le Tour through Stephen Roche (the 1987 winner) and Sean Kelly, who took the green jersey—awarded to the rider with the most consistent points record during the race—in 1982, 1983, 1985 and 1989.

When the Tour has ventured outside of France, it has generally been on the same land mass. This tradition was broken a few years ago when, soon after the opening of the Channel Tunnel, some stages were held in the south-east of England. The trip to Ireland was an involved logistical exercise. France Television Sport, the host broadcaster of the event and a collaboration between public service TV channels France 2 and France 3, chartered a ferry for all the vehicles involved and four aeroplanes to get the various personnel back to the race’s home after the Irish stages were completed.

Le Tour is an institution in France and hundreds upon hundreds of people line the course of the various stages, just to catch a glimpse of a brightly coloured, Lycra clad man on a bicycle zip past them. Much the same happened in Ireland. The race started in Dublin; the Grand Depart, as the opening stage is known, began in the Irish capital with the Prologue, the opening time trial that started at College Green and then ran for 5.6km around the inner city streets. Residents were trapped in their
outside broadcast countries. working Television was estimated the 13th leaving Ireland for France. which he organisers and although Sean county. Ireland's was expected Wexford: Kilkenny was next, with Placido. Soccer hanging around strangely predictive Tenors played <This Roscoff for what many Back 'COMPRESSION' was the birthplace of Stephen Dublin Wed ad non-linear non-averted Dublin...and e' /was the 1626 activated a second Wescam helicopter and an aeroplane with additional video equipment. Signals from the mobile units were received by two trucks, one positioned along the route, the other at the finish line of the stage being covered. Also at the finish was a digital production truck, equipped with five cameras, a super slow-motion camera, and three slow-motion video units, among other gear. France Television assembled its main feed in SFP's digital scanner, offering four lightweight cameras and a further portable unit, with a crew of four camera operators, plus a remote control camera. Other equipment included a 1-channel Profile hard-disk player/recorder, three used for replays and slowmo, and six Betacam video tape machines. Full editing facilities were provided on site, with a multifunction truck for postproduction and archiving of the hundreds of tapes used during the course of the three week production. Two vehicles were also used for electronic newsgathering purposes, one to carry video tapes around, the other to ferry the journalists covering the story from place to place.

As with all host broadcaster productions, the signal produced centrally is then made available to the rights holders, who are provided with facilities to customise the coverage for their own purposes. These were supplied under the auspices of the EBU Eurovision service and included a truck with 16 commentator positions, ENG equipment for use by the overseas broadcasters (more romantically known as TV étrangères in French) and an edit unit.

The French passion for le Tour may not be matched in all European countries, but there is always intense interest from TV étrangères if any of their country's riders are in with a chance. After his win last year, Jan Ullrich of Germany, and his Deutsche Telekom team, was the focus of attention for his country's broadcasters. Sunrise broadcaster ARD, which was additionally sponsoring Ullrich's team, contracted an independent facilities company—which, for reasons that have not been made clear, cannot be named—to provide specific coverage, opting for the digital route to maintain quality through the number of different tape passes that occur on a production of this nature.

Pool pictures and sound were received from France Television and loaded onto a Sony Betacam SX DNY-A100 hybrid recorder, equipped with an additional six hours storage time. The anonymous scanner company provided two mobiles, between them offering BVE-910 and BVE-2000 editing systems, five DigiBeta DVW-A500 machines, the DNY-A 100, three DigiBeta machines.

Ullrich did not fully repay this attention, coming in second to Marco Pantani, the first Italian in 33 years to win the Tour. Generally considered the best rider never to have won a major tournament, the distinctively shaven-headed Pantani should have been the only true story of this year's Tour de France. Regrettably it became about a tawdry drugs scandal as French police heavy handedly raided the bases of leading teams, causing the other riders to stop racing and sit in the road in protest. It is unfortunate that the abiding TV image of this year's Grande Boucle will not be Pantani and his team mates with their triumphantly blond dyed hair (or goatee beard in the winner's case) but a mass of sullen cyclists riding very slowly through France.
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Aerial warfare

SIX MONTHS is a long time in broadcasting, particularly in digital radio. In March, Studio Sound examined the background and current status of digital audio broadcasting (DAB) worldwide, coming to the conclusion that, although the technology was in place, the format was still a long way from general acceptance, or even understanding, by the listeners until suitable affordable receivers came onto the market. In-car digital radios started to appear at the end of August, with the promise of hi-fi units. Walkmans and PC-plug-ins for the end of the year.

Just to prove that broadcasters have driven things all along, adoption of DAB is increasing around the world, while countries with services are seeing them develop and expand.

The big breakthrough of the last six months has been the eventual emergence of digital radio receivers. Previews were given in Germany and the UK during July, with the BBC hosting a reception to introduce five in-car units. The Corporation's director of radio, Matthew Bannister, commented: The BBC was founded in 1922 by six radio manufacturers who wanted to develop some programmes to be heard on their cutting-edge wireless technology. We are here today with five radio manufacturers to celebrate another leap forward for the medium.

Bosch-Blaupunkt, Clarion, Grundig, Kenwood and Pioneer were the manufacturers present; the new units range in price from £99 to £189. (UK) with further developments to come. Bosch has produced a prototype miniDAB portable radio, a PCMCIA receiver and a PCI plug-in card: a low-cost PC digital radio has also been announced by UK company RadioScape, with demodulation and decoding software running on Windows 95 and NT.

With the BBC's pilot service now established, the Corporation announced plans for other programming to join Radios 1, 2, 3 and 5 Live, BBC Parliament and BBC 5 Live Sports Plus on its multiplex. This will be pilot schemes based around the BBC's extensive rock and pop archive, a travel data service and a nationwide version of the BBC Asian network, which is currently confined to the Midlands region of England on analogue.

The UK commercial sector is similarly moving ahead. The Radio Authority, which licences and regulates independent radio, advertised the licence for a single national commercial multiplex earlier this year, receiving an application from Digital One, a consortium owned by GWR Digital Radio, NTL Digital Radio and Talk Radio UK. Local multiplex licences will be advertised once the mix of programming on the national multiplex has been determined: Digital One is proposing a 24-hour classic rock station, a rolling news service, a dedicated sports channel, plus teenage-style programming, book-club and cookery themes.

A focus for progress so far in DAB was the Radio Academy Festival, held in Birmingham towards the end of July. Here the BBC gave the first ever live transmission of a multimedia radio music programme, plus airing two other shows made specifically for the new format. Eurotika 147 is being used by the majority of countries establishing digital radio networks. The UK's fellow pioneer, Sweden, estimated that 75% of the population would be able to receive its national services, of which 85% having access to regional services, which also offer private broadcasters. Germany has more than 200 DAB transmitters and France is catching up, with public broadcaster TDF running an 8-programme multiplex, which additionally carries five commercial services. NRJ is also running a service, and TDF Networks, run by Radio France and other companies, is offering multimedia output. During the World Cup, transmitters and receivers were available in three major towns, covering 20 people.

Other European countries are at different stages of development, but all have some digital radio initiative planned. The Australian government has said that services will start in January 2001. Radio Television Hong Kong was due to launch a test service in August, and All India has a test system in Delhi and is looking into satellite distribution.

WorldSpace Corporation, founded by Noah Samara, formerly a senior adviser to the ITU on global business and regulatory matters for analogue and portable communications, is pushing ahead with plans for global digital sound broadcasting; based around three satellites. The first of these, AfriStar, is set for launch in October, and will cover 75% of Africa using three beams. AsiaStar and AmeriStar are due to launch in the first half of 1999 and will transmit to India, China, South-East Asia, Mexico and Central and Latin America respectively. By the end of next year WorldSpace intends to be transmitting news, education and entertainment to 4 billion people within this network of markets.

In the US, the Federal Communications Commission held an auction in April of last year for what it calls satellite digital audio radio services (DARS). The sale raised $175.2m for two licences, which were awarded to Satellite CD Radio and American Mobile Radio Corporation. However, the terrestrial situation in North America is confused. While Canadian broadcasters have enthusiastically adopted Eureka 147, with BBC intending to cover 75% of the population over the next five years, the US is moving towards the in-band on channel (IBOC) system. Although Eureka 147 performed better in laboratory and field tests, the NAB opposes its adoption, citing lack of new spectrum, a dislike of sharing transmitters in the multiplex and concerns over new competition.

At the BBC launch in July, there was solidarity from broadcaster and manufacturers, but there is still the acknowledgement that more, varied receivers need to come onto the market and at prices that will encourage listeners to move to digital.

Although digital radio is seen as a way to free up existing analogue spectrum, there is still the question of where to park the multiplexes. At present, most countries have chosen the L-band for digital radio, but there is an on-going discussion as to whether the FM spectrum should be used for the purpose once simulcasting is no longer necessary. In a recent speech, Sir Peter Gibbings, chairman of the Radio Authority, encouraged European manufacturers to seize the opportunity that this new technology offers, adding, "we also hope that every effort can be made to ensure that more frequency spectrum is made available, both in the L-band and Band III to allow the RA to offer the prospect of local digital radio in all localities across the UK."

As receivers appear in shops, it is up to market forces, led by how broadcasters and manufacturers promote the technology and its benefits, to decide whether digital radio will be an instant hit or not. Ultimately there is no other choice, but everything needs a good start in life.
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Risky a new digital console for an exacting live broadcast and surround recording of a prestigious opera production, finds Tim Goodyer back stage at Glyndebourne.

The days of theatre 'flats' and languorous scene changes have given way to ingenious sets given to a level of historical accuracy unknown to early opera-goers. Lighting effects have grown in sophistication to challenge today's rock circus, and opera's slopping stages offer an audience unstructured views that would be envied by the first 30 rows at the average stadium gig. Opera broadcasts, meanwhile, are regarded around the world as one of the most exacting tests of audio quality and expertise, while its recordings are preparing to show other musical genres—including modern film soundtracks—the way forward with surround capable delivery formats.

A recent season at Britain's prestigious Glyndebourne Festival Opera managed to involve not only a new production of its second ever opera—Mozart's Cosi fan tutte, first performed in May 1934—but the first broadcast use of a Soundtracs DPC-II digital console.

Taking a diversion between Glyndebourne's car park and auditorium for the subsequent production of Verdi's Simon Boccanegra reveals a collection of trucks housing the broadcasters—including the Visions scanner. NHK Satellite ops centre and the Soundtracs audio mobile. Inside the latter, audio director John Middleton, sound supervisor Andy Rose, and Soundtracs owner and engineer Steve Williams have foreseen the evening's sunshine in the interest of meeting the audio brief. The barely finished mobile contains only the third DPC-II, and has cut its teeth on an earlier production of Mozart's Don Giovanni. The project is so demanding that Soundtracs Todd Wells had earlier considered it to be beyond the capabilities of the desk, but Williams' confidence won the day, and Soundtracs is again ready to provide 24-bit audio for live television broadcast to Japan, the recording for Channel 4 TV to be broadcast two days later, and a digital 5.1-channel surround mix for later DVD release. To further complicate matters, while NHK is working to a 20kIIPS 0dB reference, Channel 4 is working to -18kIIPS.

Inside the 800-seat auditorium (increased in two stages from its original capacity of 500), John Gunter's set depicting 14th century Genoa occupies the vertical space between the London Philharmonic Orchestra and the superstitious for those whose Italian is wanting. Neumann KM140 microphones are arrayed across the front of the stage with more down each side and off-stage. In the orchestra pit there are Schoeps on the strings, B&Ks on the woodwind and large-diaphragm Neumanns on the cellos. Rose—whose last encounter with this particular opera was recording Kim Kanawa's performance at London's Royal Festival Hall some 10 years back—describes it as 'all pretty standard, really', with the exception of three radio mics hidden in a table in the wings. Where Boccanegra sits at the end of Act 1, you can't see the mics. 'He elaborates, and if you can't see the mics, there's no pick up at all. So we got Glyndebourne to cut a hole in the table and in their very expensive table cloth and mounted three radio mics in the table.' The total of 59 mic feeds arrive at the console as 24-bit digital signals and are treated, subgrouped, and routed to a pair of Sony 3348HR DASH machines before reappearing over 100 monitor faders for the NHK mix. In addition, 16 further tracks of DA-88 running at 24 bits take care of the overflow from the 3348HRs and a virtual 48-track made up of Tascam DA-88s and Sony PCM800s offers a 16-bit safety recording. John Middleton takes care of the orchestral mix on the left-hand side of the desk (so he can tickle an oboe solo if he wants...). Andy Rose takes care of the on-stage and off-stage mics on the right. Their moves are exhaustively rehearsed as the Hi-8 machines have been put to good use prior to going on-air.

'The key to this is lots of rehearsing otherwise all you can do is get a general sound,' Rose explains. 'John and I don't go for a general - put some stereo mics up, add a bit of orchestra and a bit of stage,' we go for a tighter, cleaner sound. The only thing is that when it goes wrong, it goes spectacularly...
The operation is so polished that it is easy to overlook its pioneering nature. For the earlier Rodiniana, a 16-fader demonstration DPC-II with the same processing power and number of channels sat in the trailer next to the live desk. 'It would have meant that the orchestral mix would have remained static and we would have a limited amount of mixing on the stage mix,' Williams comments, but it might have saved the day. Had it not been an unused Calrec console in one of the TV scanners it was paralleled of the analogue outputs from one of the 53484 D18s.

Today the Calrec policy prevails, but the second digital console has gone—the stability of the DPC-II having pleasantly surprised all concerned.

'We considered all the big players,' says Williams of the choice of the desk. 'We had a long hard look at the Sony Oxford and it wasn't suitable for our line of work; we also took a look at the SSL AMS Neve and Studer D950 desks, all of which were either grossly over-engineered and grossly over our budget, and in some cases the technology wasn't the latest as it is with the DPC-II. I have to say that Soundtracks have got it right; you can have the best sound and the best technology under the bonnet, but it's also about how you use it. From an engineering point of view, we can make it do what we want very quickly and easily, and from an operator's point of view, within about two hours somebody who's never seen the console can do basic input selection, routing, panning, EQ and mixing with no problems at all.'

Posting Simon Bocanegra involves taking the mobile back to Soundmovies' Surrey base and moving the console and monitoring into a more spacious room.

'The postproduction requirements are that the mix produced on the day is used as a guide for the NVC Arts video people and John Middleton to make a decision as to which is the best performance or which parts are the best from the two nights we've recorded,' Williams explains. 'They make an editorial decision about cutting the pictures together to produce the international video release, and that comes back to us as an EDL, and we produce the audio mix. That was done using the desk straight into a SADiE 2i-96 system to give both the stereo format and the 5.1 format—we have both sets of monitors set up to handle those.'

The stereo mix is then relayed back onto Digibeta as a wide dynamic and compressed using a te electronic M5000.

'Although NVC have no contract to do the production in 5.1 for DVD, they are prepared to spend money on things like the HR machines,' Rose adds in obvious appreciation. 'The extra cost to do 5.1 as we're doing it this time is not a lot where going back and doing it all again would obviously be a lot of money.'

On the detail of the mix, he opens: 'It seems to me that the rules are that there are no rules. I have four mics at the back of the hall to drive the rear speakers and we add a bit of orchestral reverb to taste. Most of the reverb goes to the front, but a bit of that and the same with the voice went to the back.'

In this opera there is quite a bit of off-stage stuff—trumpeter, choir, percussion. Various things—and depending on what is appropriate, we make them start at, say, rear right and what you'll hear is them effectively walking down to the front right and then they appear on stage.'

'With a lot of off-stage activity, you can maintain stereo compatibility and simultaneously make something move from the rear right to the front left with the DPC-II,' Williams adds. 'And those moves are very easy to accomplish and it's all recorded and edited against time code.'

'The centre speaker took a little bit of getting used to,' Rose continues. 'And we won't use the sixth channel at all. At one point we fancied using it for some of the off-stage stuff or telling people on the sleeve that the sixth speaker should be hung off the ceiling or something, but it's a bit crazy to do that really. If we had a cannon then I think I'd have taken the bottom end of that to the sixth channel—it's film stuff really. But there are operas that have gunshots and things, so I'm sure it'll come.'

With the broadcasts out of the way and a completed 5-channel mix ready for marketing, Middleton and Rose can rest assured of having maintained Glyn-debourne's reputation for artistic excellence and Williams can be confident of his new mobile's viability.
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Like a modern Odyssey, the story of Athen's Concert Hall is a voyage of adversity and achievement. Caroline Moss hitches a ride

Megaron

THE STORY behind the Megaron Concert Hall in Athens has all the elements of an historical saga. The impressive world-class venue, which incorporates an innovative recording centre, stems from a concept originated in the 1950s by the cultural group Friends of Music, which began raising money for Athens' own concert hall. Construction began in the 1970s but suffered severe setbacks when funding ran out. In March 1991, after 15 years of sitting empty, the Megaron opened with its first performance following government and private fund-raising efforts, and the huge edifice is now established as a symbol of the country's musical heritage.

Headed by president Christos D. Lambrakis, the Megaron is a joint venture between the Greek government and the private sector Friends of Music. It is a vast complex connected by labyrinthine corridors: a beehive of enterprise employing around 250 full-time members of staff. To give some idea of the venue's turnover, between October 1997 and June 1998 over 250 events were hosted, including musical performances, conventions and seminars. The longest-running musical event will be staged for an absolute maximum of seven nights, with particularly successful productions brought back for another run at the end of the season. An example of this was last October's production of Gluck's opera Orpheus and Eurydice which was restaged at the beginning of June for an extra four shows, when it was shot for video. The hall is also used for TV productions, such as a recent LWT programme about John Taverner which was filmed in the large hall using Megaron's in-house orchestra Le Camerata, with extra footage shot all around the building.

The Megaron boasts two concert halls—the 1,961-seater Hall of the Friends of Music and the 456 capacity Dimitris Mitropoulos Hall—as well as a dedicated recording area, a control room installed with the third AMS Neve Capricorn digital console off the production line, machine room and editing suite.

The Hall of the Friends of Music has an overall volume of 19,400m³ with a ceiling height above the stage of 13m, an orchestra pit that can accommodate up to 100 people and room behind on three separate tiers for a 100-piece choir.
Mounted at the back of the stage is the Klais church organ with a total of 6,080 pipes. The hall's reverberation time is 2.2s when empty, 1.8s when full, and 1.3s when modified for opera. Acoustics can be changed by means of moving roof, stage and balcony elements, making the hall suitable for a wide range of musical performances which encompass symphonies, chamber orchestra, lyrical performances, jazz and opera. The smaller Dimitris Mitropoulos hall has a volume of 3,510m³ with an overroof height of 7.24m and reverberation times of 1.3s when the hall is empty and 1.1s when full.

It had long been intended that Megaron should have its own recording facilities, and when the time came for its design and construction somebody with a comprehensive knowledge of acoustics, electronics and the studio world was required to oversee the project. The ideal candidate was Nikos Espialidis, who had been working as a freelance recording, PA and broadcast engineer, having returned from overseas where he had studied electronics in London and recording at Full Sail College and Centre for Media and Arts in the States. Apart from being instrumental in planning the studio's design and construction and selecting the equipment, he now runs the studio's daily operations and participates as an engineer on many of the recording projects. He joined Megaron in 1990, less than six months before the venue's projected opening the following spring.

One of his most pioneering moves was to consider digital console technology, choosing the brand new Neve Capricorn as the studio's desk.

Explaining his decision to install the Capricorn, a 198-input, 80-output version MADI delay unit, Espialidis says: 'I had faith in Neve and decided to go with this new console despite worldwide scepticism regarding digital technology. The desk was selected after a lengthy assessment period to determine whether digital or analogue would be the way forward for Megaron. This involved several visits to studios in Germany, the UK and the Netherlands to see analogue and digital technology in use. An important visit was to Bayerischer Rundfunk, a typically classical setup which had been using a Neve DN82 digital desk for a number of years. It was this visit that finally persuaded Espialidis to accept digital technology.'

A LONGSIDE the Capricorn, Espialidis chose to install an Otari 3824-24 digital multitrack, a Studer A820 24-track analogue multitrack with Dolby SR, a Sony PCM 9000 24-bit MDR recorder, Sony PCM 7050 DAT units, a Sony PCM 3402 DASH recorder and Studer A820 1-inch, 24-track analogue machine with Dolby SR. Monitoring is via PMC BBM5 XRD speakers with ATC electronic crossovers and drivers, and there is a wide selection of outboard and peripheral equipment. The control room, which floats on a sprung floor, is a totally symmetrical design in the form of a pentagon to provide good stereo imaging, with wall surfaces lined by polyurethane diffusers. All the equipment installation and wiring was handled by Elliott Brothers, and was carried out under main designer Andrew Ridley. The company was also responsible for the adjacent machine room equipped with systems for controlling air conditioning, ventilation and humidity, maintaining the array of racks for the Capricorn together with all recording devices in optimum conditions.

Apart from being used to record music played in the Megaron's two halls, the studio also has its own recording area which at 30m² can accommodate small ensembles and solo musicians to record under controlled conditions. This was built in a space originally earmarked for the air conditioning system. It took almost two years to convince the powers that be that we needed something next to the control room for overdubs and other recording projects, says Espialidis.

The recording area was designed by Munich acoustic design consultancy Müller to provide controlled acoustics in the small area by means of RPG diffusers covering four of the room's corners and foam polyurethane diffusers on the ceiling. They were used to design large rooms, so it was a learning curve for them, and they managed to get it right,' says Espialidis. The acoustic behaviour of the room can be adjusted by means of rotating reflectors, solid vertical panels which can convert the recording room from a dead area with a reverber time of 0.22s to a room with up to 0.4s reverber time and which virtually behaves as a larger space. This is vital for the wide range of recordings undertaken by Megaron, allowing anything from the sound of a classical concert being played in a hall to suitable acoustics for Greek folk music which is traditionally played outdoors.

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The rival merits of competing computer platforms have proved contentious to the point of straining friendships and stalling business plans. Martin Polon draws the battle lines between the PC and the Mac.

Although many in pro-audio are savvy about computers in general, and computers in audio in particular, they are an almost insignificant part of the story. There are many for whom a 'hard drive' is an all-paved stretch of road leading from avenue to garage (a 'soft drive' being made of gravel). Similarly, a 'floppy disk' is something treated by a male doctor with the new wonder drug Viagra and 'RAM' is a kind of sheep with large curved horns. And to the same crowd, a 'processor' is something you use to make goose liver and brandy pate.

More seriously, for many in audio, the complex and constantly changing world of computers is so off-putting that they simply ignore the new technology. People who change the subject, run to the bathroom, answer non-existent pager calls or ignore pertinent conversations are today's equivalent of those who could not read 15 or 20 years ago and had to hide it.

For our collective benefit, the personal computer (PC) was born as an IBM project to popularise and enable computing from the home, begun during the latter half of the 1970s. The software for the project was eventually contracted out to a small firm who produced the PC's digital operating system (DOS). That firm was Microsoft; the processor chips came from a company called Intel and the rest is, as they say, history. Apple Computer on the other hand, was started by Steve Jobs and Steve Wozniak in a garage and the Macintosh (or Mac) later appeared on the scene in 1984 as the 'people's computer.'

The two systems continued to grow in different directions, with IBM ultimately creating an open system for which anyone could make hardware, while software for Apple's Macintosh has remained a more closed system for which hardware was and is limited to Apple and those companies licensed by Apple.

Perhaps the first step in making the two platforms have common elements was the effort of John Sculley, who was brought in to head Apple in the mid-1980s, who shared Apple's appearance, GUI and mouse technology with Microsoft as part of a deal between the two companies. The two platforms have

The similarities

THese ARE elements that the two major computing platforms (Apple standard and IBM PC standard) share more or less in common.

GUI—Graphical User Interface—has become the very reason that computing has become successful for the masses. You no longer have to enter lines of code to communicate with a computer. Instead, you use a mouse on a screen filled with icons and other similar symbols that represent software applications, files and other elements of computing at graphical symbols. There is much evidence that this technology was the outcome of research conducted at Xerox Park in Silicon Valley, California during the early 1980s. Both systems use similar colour GUl and mouse technology with slightly different modes of mouse use and file access.

Hard Drives—Although initially hard drive technology (recording semi-permanently on several levels of magnetically coated moving metal platters hermically sealed with caliform dye heads) was used differently by the two platforms, drive design itself has always remained the same and the drives themselves function analogously for each platform. Today, the drive connection technology (to the computer's internal processor) has flip-flopped so that IBM PC standard IDE connections are used by Apple for Macintosh computers (including the new iMac) and Macintosh standard SCSI connections are used by larger PCs to access hard drives of unusually significant capacity. Today's drives range in size from about 2GB (2000MB) to 20GB or so (without resorting to RAID drives) on available systems for both platforms—a far cry from early drives that held 40MB to 100MB in the early 1990s.

RAID—Redundant-Rapid Array of Independent-Inexpensive Drives. A much used scheme of building smaller drives to provide relatively expensive high-capacity/highly-protected storage. RAID uses SCSI connections and is available for both platforms.

The differences

THE TWO systems have significant differences in how they are designed and how they operate.

Operating Systems—Windows uses an overlay on an underlying non-graphic system called DOS, with the exception of Windows NT. Windows 93-98 presents a graphical interface for users and is now the platform of choice for several successful audio recording software. Windows NT is the professional version of Microsoft Windows and in many ways emulates UNIX and other sophisticated languages. Some audio recording software already written for Windows 95-98 are being optimised for porting to NT and some are already NT ready. The Mac OS is a constantly revised operating system that has hosted audio recording software since System 7 days nearly 10 years ago. Today, with the release of System 8.5, which is optimised to run on G3 machines, the Mac operating system provides more processing power for audio but is hardware platform limited (slots).

Hardware—Windows machines at the top end of the cost curve come with a large number of slots for upgrades, support of audio recording, etc. Macintosh computers have, to some extent, surrendered the computer audio marketplace to Windows machines with the G3 line of computers that have only 3 PCI slots to start with and two of those are already occupied with SCSI and video cards.

DOS—Microsoft's Digital Operating System is the underlying layer of Windows 95-98 and the default location users reach after an especially nasty crash. One reason that Windows NT is so popular for power users is that it does not use the near 20-year-old DOS.

Pentium—Optimised to run Windows operating systems but will also run alternative operating systems such as the Be and UNIX. The current Intel chip of choice to power PC computers with its 400MHz clock speed. Expected from Intel are 450MHz and 500MHz versions of this chip.
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The similarities
Floppy Drives—Mechanically, like hard drives, virtually identical in operation and writing with a magnetic head on a portable disk housed in a plastic case, for either computing platform. The disks themselves are spinners (unrecorded) and read (played back) using the different formats for each platform, to the approximately one and a half megabytes of storage available. There are software available to exchange the contents of disks from the competing platform formats.

Monitors—At one time, the monitors used by each platform were not usable by the other platform's computer. Today, however, the use of adaptors allows Macs to use PC monitors, if desired. There is usually a price advantage to such a combination.

Internet Browsers—Both the Netscape 4 Communicator and the Microsoft Internet Explorer 4 are completely compatible on both platforms, although offered in the different and appropriate code for each platform. They look, feel and run more or less the same on both the IBM PC and Macintosh platform.

RAM—At the onset of personal computing, the Random Access Memory which stores application software and data in use, was different for each platform but Mac machines have been engineered to use more PC compatible (and less expensive) RAM today.

SCSI—Small Computer System Interface, a connection scheme for external (and sometimes internal) hard drives, scanners, CD-ROM recorders, other kinds of external storage media and other peripherals limited to seven devices—all pioneered by the Macintosh system. SCSI comes in several flavours including fast and wide. It can transfer data at a rate as high as 40MB per second but in its basic form is limited to 5MB per second. It is also used via plug-in boards for PC systems needing larger and faster storage options.

IDE—The basic connection system for drives used on PCs. IDE stands for Integrated Drive Extension. Macintosh computers are now using IDE configured hard drives instead of SCSI.

PCI—Personal Computer Interconnect. The new standard for plug-in boards for both PCs and Macs. Although boards using the PCI technology are supposed to be interchangeable for both systems, in practice the board has to be manufactured for either PC or Mac at this time. Macintosh computers have switched over to PCI from the earlier Mac only NuBus card slots for plug-ins.

USB—Universal Serial Bus. A new scheme for peripheral connectivity (keyboard, mouse, hard drives, etc.). USB operates in a low speed mode of 5MB per second and a high speed mode of 120MB per second. It does not have the cable length, termination and peripheral 'hot-switching' limitations present in SCSI. All Macintosh computers from this point on will be equipped with several USB ports, beginning with the iMac. New PCs are being equipped with USB ports as well.

Level Two Caches—Memory chips, usually valued at 256K to 512K or 1MB that serves to store recurring instructions to the processor. Often identified as 'pipeline burst' cache for PCs and as 'backside' cache for Macs. By storing these much used instructions and supplying them to the processor chip as needed, overall system speed is enhanced—especially if this L2 cache is mounted near or next to the processor as the two modes mentioned above do. Both computing platforms use Level Two Cache in a similar way.

Modems—Electronic multi-component interfaces between the computer and the telephone system. Moderns are probably the most 'sexual' component in the entire computer system, determining their allegiance to Windows or Macs by connectors and lines of software instructing the modem how to connect.

< continued to evolve: Microsoft moving from DOS through v3.0 to v3.1 and then into Windows, Windows 95 and now to Windows 98. On the other track, the Macintosh became System 7, then 7.5 and 7.6, now System 8 and 8.1 with System 8.5 due out any day now.

Those using the two platforms have argued the superiority of one over another for at least 15 years. From the standpoint of the recording studio user, Macintosh computing had the market pretty much too itself until the middle of the 1990s. Now, with a competitive marketplace for sophisticated recording software on the PC platform, the two systems have become evenly matched with the PC presently pulling slightly ahead, according to some observers.

The final decision on which platform is best suited to your needs is an intensely personal one. So, no matter how useful and informative this article has been, it must come down to a careful comparison of your own preferences as compared with the opportunities offered by the machines in the Windows world versus those in the Macintosh universe. The PC today offers far more value in purchasing a system. There is a competitive choice among high quality recording/editing software for the PC. PC hardware and software is leading in the evolution of several categories such as voice recognition.

On the other hand, Macintosh computers are easier to use, easier to reconfigure after a crash, wrote the book on audio recording and editing and are preferred by studio audio users two to one at this point in time. New G3 Macs can supply the six PCI slots needed for most...
The differences

Intel—The company that has supplied what some analysts estimate as 85% of all chips used in PCs in the last 10 years. Intel will provide new state-of-the-art microchips in the year 2000 that will yield performance reaching 1.008 GHz (1GHz).

AMD-National-Cyrix—An alternative PC processor chip suppliers whose products are being received favourably by such manufacturers as Compaq, IBM and Packard Bell-NEC (among others). AMD's 65 and the National Semiconductor Cyrix Media QX are considered by many to be at least equal of Intel's current Pentium chips.

Power PC—Apple's choice to replace the Motorola 68000 family of processors that it has pioneered with the Macintosh. Consider the processor chip as the engine that powers the computer. The bigger the processor (engine), the faster the computer. The very successful switch to the Power PC was made in mid-90s. At least as measured with benchmarks or measuring points for computer performance, the Power PC processor chip family has outperformed comparable Pentium chips optimised for Windows since its inception.

G3, G4—Apple's current and anticipated replacement chip for the latest in the family of Power PC processors. The G4s could reach the one GigaHertz mark — similar to the forthcoming "Merced" processor chip family from Intel.

IBM—One of Apple's partners in producing Power PC chips. IBM is a major captive and merchant chip foundry. Translation: IBM actually produces some of its own chips for its computers. That's something no other US domestic computer maker can boast. IBM also sells chips to other computer makers such as Apple. Although Apple was influential in laying down the original specifications for the Power PC chip family and supplied staff for the AIM (Apple, IBM, Motorola) Power PC design facility in Texas, today the two merchant semiconductor partners (IBI and Motorola) are going it alone. IBM has built a sustainable business with Power PC by using it for embedded processor control and for IBM's mid-line minicomputer family. IBM does not subscribe to the Motorola-developed set of multimedia instructions called AltiVec but is "responsible for basic improvements in the Power PC design including the use of copper substrates.

Sound cards—PC computers have until this point used plug-in sound cards for audio. Macs come with built-in sound functionality including analogue-to-digital and digital-to-analogue chips. There is a move afoot to put sound functionality on the PC motherboard which also houses the processor and RAM chips. However, the flexibility to choose your own sound board — with whatever level of MIDI and whichever quality of analogue-to-digital conversion one wishes — is very attractive to users.

Copper Chips—A technology developed by IBM and others to replace the long used surface or substrate that processor and other chips are "built on" from Silicon to Copper. This would improve speed and all around processor functionality since Copper is a far superior conductor. It is expected that Power PC chips will go Copper much sooner than Pentiums for the PC.

Cost—There is no real comparison for the PC versus the Mac platform. Macintosh computers are at least twice as expensive as comparable PCs, especially when you add the factors for monitor and computer speaker systems, which are usually included with new PC platforms as a 'bundled' software suite, usually absent on new Macs.

Modularity—PCs are totally modular. Every thing plugs into the PC's chassis and failure-replacement generally means upgrading to the latest technology available for that machine. Macs are not as modular, and failure/replacement is dependent on Apple and/or Apple approved third-party vendors who refurbish major boards (such as motherboards).

In 1948 in Karlsruhe, two men joined forces to produce a microphone designed to set a new standard for sound recording.

50 Years of Innovation

50 years later the men may have changed but the design goal remains the same.

Studio Sound September 1998

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Leading the move into digital radio stations is a move girded about with promise yet fraught with danger. **Tim Goodyer** visits some of those brave enough to accept the challenge.

In **RADIO**, many commentators would have us believe that everything is about to change. As digital broadcasting takes effect, the whole structure of the industry will be redefined and the pressures on radio stations to reinvent their technical systems and practices will be irresistible. But those of us who know just a little bit more can tell you that it's not true...

That broadcasting is about to undergo a revolution is indisputable, but what the less attentive observers have missed is that in a handful of stations—the smart ones—many of the changes necessary to welcome the revolution have already been made. Take Europe 1 in Paris: the recent commissioning of its flagship Coluche studio (named after a famous French comedian and broadcaster who began his career at the station and died tragically in a car crash) marks another chapter in its ongoing modernisation. But make no mistake, Europe 1 is no state-funded indulgence or nationalistic tour de force, it is a private operation thoroughly dependent upon its commercial success for survival.

In fact, Europe 1—along with sister stations Europe 2 and RFS—is one of three radio stations under the ownership of the Matra group whose other interests include communications, space, and military technology. Within the radio group it operates entirely independently, however, running as a fully manned station all day, every day, and complementing the predominantly music output of Europe 2 and RFS with its mix of news and current affairs programming. Established in 1955 on LW and moving to FM in 1985, Europe 1 now reaches all the major French cities and beyond to Algiers in the south and across the water to London in the north—where it enjoys the kudos of being recommended listening for students of the French language.

The three stations share a common address and 12 studios in central Paris, although Europe 1's production head, Olivier Bauchard, is at pains to stress the independent nature of their day-to-day running and operation. Europe 1's share of the technical resources extends to four main studios and two production studios; its technical staff numbers around 20 with a similar number involved in operations and a further 100 journalists in the newsroom. There is also an engineering staff that helped keep the costs of the present work down to the extent that the only external contractor involved was a wiring company.

The basement of the radio complex also houses the television studios of Disney Channel France where you can witness the kind of organised chaos that plagues the average family's Saturday morning. What is readily apparent from a tour of the facility is the absence of the familiar continuity studio—it's traditional duties of switching and summing having been assumed by a powerful custom analogue switching matrix. Bauchard identifies the functions of the matrix as being difficult to achieve with a digital matrix and that it will be necessary to put some intelligence between the console and the matrix before it is possible to take this area of the station's operation into the digital domain.

The re-opening of the Coluche studio marks its complete redesign to be a showcase—the glass walls along one
side of both control and live rooms allowing unhindered viewing of its operation without interfering with its working. The refurbishment also sees the installation of an Amek DMS console running alongside a Numisys system for news material, D-cart for advertisements, a 960 Systems Instant Replay for jingles, three Studer D730 CD machines, a Studer D750 DAT machine, and two Studer 520 analogue open-reel machines for the remainder of the audio playout. The surface of the console carries no less than six screen displays accommodating two each for Numisys and D-cart systems as well as one each for the desk’s audio metering, and a phone cueing system. All the screens are flat and output little heat and EM radiation—all of which are welcome attributes in a broadcast control room setting.

The news programming has been prepared in-house by the station’s extensive newsgathering and editorial facilities using an Arcs system feeding the Numisys for some time. Similarly D-cart enabled Europe 1 to be the first commercial television to digital playout when it was installed some seven years ago. The key decision to the refurbishment, therefore, was the choice of console to replace the 16-year-old custom Digitec desk. The specific decision to go with a digital console had been taken by a new management team last year, bringing the choice down to models from both Studer and SSL, with the time frame between choice and installation being critical—the selection process having begun as recently as September last year.

Subsequently, Bauchard describes the Studer 950 as having been judged ‘very interesting’, but too old, and the 971 as being prohibitively expensive. The high level of flexibility that helped win the day for the DMS causes Amek’s French distributor, Edge’s Jean-Luc Gérard, to describe it as ‘needing specifying before it exists’, a description Bauchard translates as ‘future proofing’, either way, the DMS is able to meet the specific demands of a broadcast installation. The physical size of the console was, as is often the case, another critical factor, the studio having to accommodate an engineer flanked by a producer and assistant engineer for certain broadcasts, yet be operable by a lone engineer for others. Here the small size of the console monitors and the power afforded by the DMS’s macros have helped make it possible to bring all the operational aspects of the control room into a physically small area.

The installed desk boasts 52 inputs, 40 outputs and in excess of 256 macros written by Amek to address the station’s specific requirements. The studio’s only commercial station to use digital format of the DMS concerned the limitations of the metering, as it ordinarily requires the channel PPMs to be displayed on a monitor as one of the desk’s operations pages. Since the operations staff felt it important to constantly check the signal levels, an additional, dedicated, metering screen was provided. The flexibility offered by the DMS software ‘toolkit’ is also to be the key to solving the problems of replacing the old analogue switching matrix with its less capable digital counterpart. It also offers the security of being able to adapt the desk for future applications without the need to know too accurately what they might be. In this way a digital console can begin to address its shorter lifespan as compared to an analogue design. Certainly Bauchard is aware of these considerations and believes that after 10 to 15 years in service the Digitec desks gave a good return on the investment. And that after the same asking price in 1998 as the Digitecs were when new, the Amek desks potentially represent both a cheaper and more powerful option.

Control room monitoring in the control room is by Genelec by way of Rogers, JBL and Urei, and is regarded by Bauchard as still being ‘not ideal’. The acoustic of the programme room could also be described as not ideal due to the considerable amount of glass along the viewing wall—that it only becomes noticeably reverberant in very close proximity to the wall itself, however, a tribute to the expertise at work. ‘It’s the first time in 10 years I’ve seen an acoustician,’ comments Bauchard.

Europe 1’s DMS is not the first digital console to find its way into a French radio station—the first went to Radio Classic and was followed by Radio France’s SSL Axion. Intriguingly, it’s only the first to become operational at Europe 1 by a few months as it was delivered as one of an almost identical pair of DMSs.

The selection of the console for Coluche, therefore, was made doubly important by the intention to refurbish Studio 7 in similar fashion later in the same year, completing this 40.5m phase of the modernisation project—a figure Bauchard estimates to be half that of using a ‘traditional’, non-digital solution. Europe 1’s third main studio—called Merlin—presents something of a different challenge as it will accommodate a small orchestra and a 30-strong audience. Although Studio 7 is not yet ready to accommodate it, the second DMS has been delivered to Europe 1 and is currently being used for testing and also to train staff of both Coluche and Studio 7 in its operation (following up on the 12-day course presented by Amek). So while no decision has yet been made regarding Merlin’s next desk, it may well be that Studio 7’s DMS will play a part in securing an order for a third similar console.

Of course, it is true that the world of broadcasting is set to change dramatically in the coming years. It’s simply that those smart enough and brave enough to address the future now stand to benefit from those changes and be best placed to move with the developments that will surely follow. That Europe 1 has adopted such a decision so early in the days of digital radio is testament to its confidence not only in the available technology to take it into such an uncharted area but also to its own ability to make it work. Best of all is the fact that this confidence is being displayed against the background of commercial imperatives that will punish it severely if it is mistaken. The confidence of the staff is infectious, however, and a day spent watching them work and listening to them talk gives some insight into how they have secured the support of their management. Other radio stations may be well advised to take note.

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In its 30 years, LA's Village Recorder paralleled the careers of many of the classic artists who used it, riding the rollercoaster of rock. Dan Daley pays a visit to the once and future palace of sound and finds a comeback tale worthy of Hollywood.

A instantly recognisable mural adorns the entire side of the Village Recorder's four-storey studio building in a converted Masonic Temple on Butler Avenue in west LA. Painted in 1971 by artist Terry Shanehoffen, it is the apocalyptic vision of Los Angeles after the Big One, with the jagged edge of a freeway leading off into earthquake-shattered empty space.

The mural, with paint now faded and barely visible, graced several album covers and bookplates over the last three decades. It also serves as a metaphor for what has happened to any number of classic recording studios that failed to make the transition from the astronomical rock 'n' roll budgets in the sixties, seventies and eighties to the leaner, meaner streets of the content-driven nineties.

Not so for The Village Recorder, which this year celebrates its 30th year in continuous operation. The studio, opened in 1968, has a client list of classic artists and records that could cover yet another side of the building, a who's who array that spans the modern rock era itself, including the Eagles, Eurythmics, Phil Collins, Bob Dylan, The Allman Brothers Band, Joni Mitchell, Sly & The Family Stone, Supertramp, New Edition, the Rolling Stones, Tom Petty, Heart, Fleetwood Mac, and in finitum (well almost), through contemporary stars including Smashing Pumpkins, Cracker, Sneaker Pimps, Tori Amos, Fugues and Cowboy Junkies.

However, The Village Recorder struggled during the early 1990s, labouring not only under the tightened studio market of the times but also from its own legacy as part of that immoderate and unbridled era. In addition to everything else Los Angeles had to throw at it, from earthquakes to riots to immature rockers. The Village passed through its own near-fatal transition to adulthood, surviving several treacherous generations of the music business, some of which had to be literally shown the door.

You can say The Village had fallen from its high point, says Jeff Greenberg, the studio's CEO for the last four years and the prime mover of the facility's tremendous turn-around during that time. Greenberg was recruited in 1994 by his friend and founder-owner Geordie Hormel's daughter, Julie, who was for all intents and purposes raised in the facility, at one point being regarded as a 'step-daughter' by Mick Fleetwood. Hormel and Greenberg waged a determined battle to reclaim the the moribund studio from the damage of several previous management regimes. Hormel was also up against the studio's acquired collection of pharmacologically inclined music business hangers-on, the accumulated detritus of the excess-laden 1970s and 1980s, when Village was at its peak, who often simply walked into the building's labyrinthine concatenation of offices, studios, hallways and secret Masonic passages and did unsanctioned business.

Which is precisely what Hormel did, and quite successfully. Within a few years of his arrival, he had embarked on what would become a hugely successful career scoring music for television series such as The Adventures of Ozzie & Harriet, Lassie, Rin Tin Tin, The Untouchables, Huckleberry Hound and The Fugitive. In 1968, he decided to create what we would now call a project studio in his home—a recording facility for his own use that would give him greater creative capabilities while allowing him more control over costs.

A few generations of successful meat packing had infused him with a higher degree of business acumen than most musicians. However, the bankers at the time had less foresight and instead advised him to open a commercial studio—but keep in mind that consoles and tape machines at the time were not bought off-the-shelf at the local music store. He agreed and bought the former Masonic Temple. But he applied the Hormel business touch to that transaction, as well—the building was bought for $125,000 with no money down, though Hormel subsequently pumped in over $6m in improvements, starting with the design of Studio A by Rudi Brewer.

But Geordie Hormel soon subordinated his own use of The Village Recorder—named after the neighbourhood—to that of a growing coterie of young artists that gravitated to the studio, including Becker and Fagen of Steely Dan, who went on to make classics such as Pretzel Logic, Can't Buy a Thrill and Aja in that studio. Today, it is equipped with a vintage Neve 8048 (all of Village's four present rooms are Neve-equipped) and a pair of Studer A800s (one of which still has a plaque affixed that indicates the machine was built with Willi Studer expressly for Phil Ramone) and has made records for Dr Dre, Snoop Doggy Dog, Counting Crows and the Eagles, who mixed 1996's Hell Freezes Over in Studio A. In his own words, Hormel explains his philosophy by stating, 'When someone rents the studio, it's theirs, not mine.'
They own it lock, stock and barrel. Making the artist comfortable in a good positive atmosphere has a lot to do with our success over the years.' Brewer built the three succeeding studios over the next decade, and they reflected their era: none took advantage of the 20ft ceilings of the original Masonic Temple, and all still have an acoustic that features wood paneling and trim. While Van Haaf's work has brought them up to contemporary standards today, the Brewer designs were perfect for the time and the studio continued adding studios to accommodate the new business as both rock 'n' roll and The Village grew in tandem. These artists tended to hunker down and stay long-term in the studios, such as Sly Stone who spent well over a year making 'There's A Riot Goin' On' in Studio A. The rooms seemed to take on the character of the artists using them after a while, as Hornell, ever willing to make his facility as protean as his clientele, listened to their music with as much a mind for the business as he did for their art. Studio A's tight recording room was perfect for the close-in-miking of the Steely Dan Records. It was during those years that Jeff 'Skunk' Baxter, who played so many of the solos on those records, eventually decided to move in and set up his own studio in the building.

A suite of rooms on the third floor is snuggled up with wires and its walls lined with guitars. Baxter's private digs are mirrored by those of guitarist-autour Robbie Robertson, whose rooms have allegedly been seen by fewer than a dozen people over the years—no one really knows quite how many—he's been in the building. A quick peek—authorised by a call by Greenberg to the sometimes elusive Robertson—reveals a relaxed parlor-like anteroom whose walls are lined with his guitars dating back to The Band, and an equally informal studio room where much of his stunning, hypnotic Blue Train album was conceived and fleshed out on a vintage Neve SSL desk, a Roland sideward mixer and an array of gear and instruments as eclectic as their owner. The two artists are joined by six-time Grammy winning engineer Al Schmitt, who was brought in as consultant for the renovation period and who also continues to maintain an office there. Studio D has a 2-input VR legend console fitted with one of the first 8x8 VST surround matrices, which has been used on films including Shoutbank Redemption, Good Will Hunting and The X-Files, but its large recording space still resonates with the beats of Fleetwood Mac's seminal Tusk (for whom Hornell again modified a studio's design for the artists and the Rolling Stone's classic 'Angie' from Goat's Head Soup). One large iso booth also serves as a massive echo chamber, adjustable by how far the door is left open, and whose

zebra wood-lined piano room Stevie Nicks favoured for her vocals and which later gave Bruce Hornsby & The Range's The Way It Is a unique sparkle.

In Studio B, built in 1971, where the recording of Smashing Pumpkins' Mellon Collie & The Infinite Sadness was one of the factors in helping revive Village's fortunes in 1996, Oscar Petersen's 1921 Steinway grand sits like an honoured elder in the small recording room on the opposite side of the control room from the lounge, whose ceiling is studded with fibre-optic stars. Second Floor Studio E's lounge, built in 1986, is reputedly where Eric Clapton penned 'Tears In Heaven' on an acoustic guitar for the soundtrack to the film Rock. Since so much of the original acoustic remains in place, there is an otherworldly ambience to the facility, a sense reinforced by the reputed presence of a bass-playing ghost whom a few long-time clients have reported.

The renovation of The Village was undertaken in 1994. Greenberg brought in studio designer Vincent Van Haaf, whose mandate was to modernise the control rooms but retain their acoustical je ne sais quoi. 'Some of the rooms are not very deep, about 13 feet or so,' says Van Haaf of the original designs; although Studio D's control room has a depth of 30 feet. 'It would have been difficult to have staged most of them. For instance, Studio A's control room shares an external wall with the post office next door.' Van Haaf placed a Helmholtz resonator on the rear wall, tuned to 100Hz, to add a sense of depth to the room. His most radical change in all the control rooms was to shift low-frequency absorption from cavities in the floor made by Brewer to new ones in the ceilings, thus maximising the floor space of the control rooms.

'The facility has a certain attitude that affects the artists and producers who work there,' observes Van Haaf. 'You can hear in the records I think it's an overriding tension created by music and the environment of the Masonic temple. I've always noticed that records coming out of there had a feel like no others. I don't think you could give a studio a greater compliment.'

Adds Greenberg, 'The business had changed and we needed to re-establish The Village as one of the premier studios in LA. I looked at the physical presentation of the building and decided it needed a transformation from how the outside of the building looked to how clients felt when they first walked through the door. The lobby had been cool in the sixties, but with the mix of clients we were looking for in the nineties we needed something different. We went through the complex and literally ripped up the carpets, cleaning and renovating, then listened to every amp, speaker, wiring combination and component available to get the most acoustically perfect room.

Geordie Hornell was always a fan of technology: The Village was one of the first studios ever equipped for 24-track recording, was the first studio in Los Angeles to offer 24-track Dolby capability, and was one of the first facilities to commit to digital recording with the purchase, in 1984, of three Sony 3524 digital multitracks. He was also instrumental in establishing Fairlight sampling engines to the US during the period.

Greenberg is committed to continuing that plan. We are already in the drawing stages to turn the second-floor games lounge into a dedicated 5.1 surround mixing room designed by Van Haaf, slated to open in October featuring a Capricorn console with wings loaded with 1081 EQ modules. Greenberg intimates that the development of this room is taking place with more than simply the purchase of a Neve digital console. 'It has the potential to become,' he says, a showcase room for 5.1 audio mixing technologies, including Neve's. If so, it would fall in line with a current trend in the industry that sees surround audio systems manufacturers teaming with engineers, producers and facilities to showcase their wares, such as the pending arrangement in Nashville between surround sound DTS on one hand and mastering engineer Denny Purcell...
and engineer Chuck Ainlay on the other to build such a facility jointly. 'This room has a lot of potential for new technologies,' says Greenberg. 'And the way it's envisioned, it wouldn't look like most studios. We're keeping the windows in place, for one thing, to change the feel of the room from a studio into something more versatile.'

Across the hall is the Temple's original movie theatre—a huge, unfinished room with a 20ft ceiling—which has been used over the decades as a recording annex, including for the Eurythmics drum tracks. This room is in the research stages of a complex redesign plan that would make it a large film-scoring stage that would be used for music sessions, as well. (Village has had significant success in film and commercial audio for video, including sound elements for films including Phenomenon, Con Air, Sharebank Redemption, The Man in the Iron Mask, Dead Poets Society, Fried Green Tomatoes, The Prince of Tides and The People vs Larry Flynt, and commercial spots for Reebok, AT&T and Jack in the Box.)

This has been an amazing recording space over the years,' says Greenberg, clapping his hands to illustrate as he walks through the room.

'The key to making it workable is how to isolate it, which is something we're researching right now with [Van Haaf]. We'd have to create a room within a room, extending new support columns into the bedrock beneath the building. What makes it a bit more tricky is that we have to be able to isolate the room from the other studios even during the construction phase.

While the technology is being updated and the aesthetics polished up, Greenberg and studio manager Robin Bulla (after five years' tenure, he is one of only two staffers to have been there longer than Greenberg), whose petite form underscores her earlier career as a professional jockey, have worked on the facility's marketing plan.

'We needed to recreate the culture of The Village the way it was when Geordie first started it,' says Bulla. 'If you were to be here for a few days you'd see that it truly exists again and it continues to be the basis for how well the studio had done for all those years and how well it's doing again.'

But there are some twists to the studio marketing strategy implemented by Greenberg, whose ability to think in Hollywood terms again comes to the fore. Under his direction, the studio is a sponsor of new talent festivals South by Southwest (SxSW) and North by Northwest (NWW), both of which have grown into sizable industry attractions. As a result, the Village Recorder name gets significant exposure to a large cross-section of an increasingly fragmented industry. Greenberg has also made the facility a daily sponsor of LA's National Public Radio affiliate station, one known for taking musical chances and promoting new and underpromoted talent on a national basis. (Greenberg's relationship with the station was evidenced on a cruise down Santa Monica Boulevard when Greenberg waxed enthusiastically about a cut off the new Rufus Wainwright record, which gets him so excited he pulls out the cell phone, hits a speed dial button with the station's control room number pre-programmed in, and finds out the producer's name within two minutes.)

'It's all about interactivity,' explains Greenberg. 'The same thinking goes into getting the ISDN lines and making connections with other facilities, like Howard Schwartz Recording in New York and doing projects together. LA is now the centre of service for the communications industry in the middle of the Age of Communications. It's all part of The Village mind-set: make it comfortable and easy for people to create."

All this is combined with a restaffing program that infuses new hires with a service-oriented mindset and a rededication to Hornel's notion that music and the client come first. (Greenberg prefers graduates of professional audio educational programs because he wants to focus on training them in the studio culture, not its theory, he says.) 'It took 30 years to build the culture of Village Recorders,' observes Greenberg. 'It took us three years of hard work to restore it. I'm ready for the next 30 years.'

September 1998 Studio Sound

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MARK GRIF""
US: Nashville Redux

Whether it is nature or nurture, certain family lines seem more suited to running recording studios than others, writes Dan Daley.

It's difficult to see individual trees when in the middle of the forest, it's equally hard to see individual studios from the centre of such a dense environment as Nashville. And any introspection is hampered by the fact that this is no peaceful glen here but rather a swirling vortex where the wind can and does rip trees out by the roots and, like the tornado that passed through downtown Nashville in late spring this year, leave those on the ground wondering what the heck happened.

The biggest problem with Nashville recording studios is that there are simply too many of them in too small a place. Being the hub of an entire genre of music is a double-edged sword: there is a massive supply of music projects generated within Nashville; even with the pared-down rosters of artists of the mid-1990s, the eight major labels and subsidiaries and independents located here churn out close to 800 country records a year, plus one-off projects such as the tribute records that have remained so popular ever since the country-does-The Eagles recording in 1993, despite the fact that most fade dismally in the market.

That's just country. Nashville's very good altrock scene is one of its best-kept secrets, understandable when it must exist in country's large shadow.

All this work would make one think that Nashville is studio heaven. Well, it was once. But the delicately balanced ecosystem of studios and records was upset — probably forever — by the wild and ultimately unsustainable success country music experienced in the early 1990s. In response, the number of studios increased dramatically, as did the level of their capabilities. An entire new generation of facilities was launched in late 1993 when Masterfonics opened The Tracking Room, a large 24-track studio heaven with Nashville's first SSL 9000 and the first major new facility to hit town in over a decade.

That debut was followed in short order by a cascade of other upscale rooms, including three brought in by one joint venture between Ocean Way owner Allen Sides and House of Blues Studios owner Gary Belz. The city is up to five 9000 desks and now has a growing population of new Neves and Euphonix boards. But for every palace that opened, there were a dozen half-dozen less dazzling, but quite credible new studios hosing on their tails.

Then the bubble burst. As country music's sales began to decline in 1994, so did label rosters and thus followed studio bookings. In short, the studio building and upgrade momentum hit its stride just as the very phenomenon which sparked it was running out of steam.

Four years later, both the music and the studios find themselves on mixed trajectories.

For starters, the number of studios continued to increase throughout those years, despite plenty of evidence that the fields of Middle Tennessee were not as fertile as they once were. A friend who is an executive at the Music Row branch of a large regional bank continues to be astounded at how many loan applications he continues to get for studio projects here. Applications come from everywhere including Nashville itself, where many studio people prove they are as capable of self-delusion as the would-be country music recording artists who still flood Nashville on a daily basis.

Many new facilities are project studios: many of those are owned by major producers in Nashville, which two pulled thousands of hours and millions of dollars a year out of existing for-hire facilities. Some, such as Loud Recording, owned by producer James Stroud, former head of Giant Records, now head of the DreamWorks label and who has always negotiated the right to produce records outside of these deals, is probably responsible for upwards of two dozen records a year, as a producer, co-producer or executive producer.

That is a substantial number of major-label projects that are no longer feeding a studio community that continues to grow.

T

The MATH is incapable. Yet when studios do go down and they are beginning to do so with increasing regularity — their sites are often occupied by another studio venture before the first one is cold. Despite appearances, these are not technologically advanced lemmings. Close inspection of many of the new ordering for DI are being in-house production, music.

When pushed on the cost to Sky of converting 3m existing satellite dish homes and winning the several million new subscribers Sky hopes for, Booth said something meaningless about 'if people aren't subscribers it won't cost us anything' and called for the next question. 'We have one chance to make a first impression,' he said.

Digital terrestrial is analogue TV masquerading as digital. Sky will be able to offer New Video on Demand, with one movie running on several channels, at staggered start times. ONdigital does not have the channel capacity to do this. But ONdigital's more limited and simple choice may appeal to some viewers. Sky promises comprehensive interactive services, such as teleshopping. But it is not yet clear how many people want this from a living room TV.

The big advantage of digital terrestrial TV is that existing UHF aerials will often be adequate for its reception. Even setup rabbit-ears may deliver perfect pictures without snow or ghosting. But where an aerial is misaligned or blocked by an obstacle, and currently delivering very snowy or ghosted pictures, a digital box may give no pictures at all. If the new digital transmissions are at widely differing frequencies from existing analogue services, viewers may need a new wide-band aerial.

In Europe analogue terrestrial TV has nationwide cover. But it won't be the same with digital. Initial cover will be less than 70%. This is why the BBC will simulcast from satel-

Europe: Digital duelling

The evolution of television promises a bloody struggle between providers of digital services — with the usual costly penalties for the consumer, writes Barry Fox.

Europe IS DREAMING of a digital Christmas. France and Germany already have digital satellite transmissions and Rupert Murdoch's BSkyB goes digital into the UK on October 1st. The BBC and ONdigital, previously called British Digital Broadcasting, will start the first full-blown digital terrestrial services in the world soon after.

In theory, the mass of new channels should create more work for sound and vision studios. In reality there will be a bitter standards battle that will sap the rivals' resources. The British government has now admitted that it has failed to ensure that first-generation satellite and terrestrial boxes would be 'interoperable' (convertible). The pay services' money will go on sport and movies. Only the BBC, with a guaranteed income from the compulsory licence fees, and, perhaps, the existing analogue terrestrial channels, will have money to spend on original programming. The depressing model is Channel 5, Britain's fifth analogue service, which already has a very cheap look.

The war of words has already begun. Issue 1 of Sky's Digital broadcast newsletter blames ONdigital for incompatibility between terrestrial and satellite receivers. ONdigital sees Sky as trying to kill it at birth. Digital service will come from Astra's new satellites at 28° East. All existing Sky dishes point at 19° East and very few have LNBFs which can handle the high-frequency digital band. So all existing dishes will need modifying or replacing.

Sky has now offered free installation. The Sky set-top-box receiver contains a modem that must be connected to a phone line, to control billing and impulse pay per view. European Union legislators have insisted that Sky pay the-box subsidy (of at least £200, UK) as soon as the viewer connects the set to a phone line. Viewers are not obliged to sign up to pay a subscription.

Sky's chief executive Mark Booth (who once ran Robert Maxwell's satellite division) admits that aerial installation will cost Sky £1 a home. This means that Sky will be buying subscribers at a price of around £300 each.

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publishing and new media such as DVD are at least a partial basis for a business foundation. That’s radically different for Nashville, where the separation of such categories as creative and technical had generally been observed in years past.

What does the future hold for this studio community? One thing is certain: that the fortunes of the studio business in Nashville will no longer be so closely and inexorably linked with those of country music. Between more aggressive marketing to non-country clients and more reliance upon in-house revenue as their financial basis, the current generation of Nashville studios has learned not to trust their eggs in a single basket.

A good thing, too, because what’s just as inevitable, I think, is that the tendency towards in-house revenue streams is going to open up Nashville’s horizons well beyond country music. Just as independent record labels have collectively superseded all but the largest one or two major distributors, the fragmentation of the music market is going to ultimately leave less market share for the big genres of music.

It’s interesting to sit back and watch the very fabric of the studio environment in cities change. It is Darwinism at work, and an organism is responding to external stimuli. It’s a painful change, too, for many who have been here a long time and who have made the commitments that built Nashville to the status of Third Coast. But come 2000, things are going to look even more differently than they do now. And who knows, Music Row might just stretch all the way to Memphis by then.

Who wants wide-screen?

One of the qualities of leadership is to make decisions, right or wrong—but it is better if they are right, writes Kevin Hilton

T’S ALWAYS disheartening when one is forced to find fault with something or somebody that you once admired, promoted or believed in. From an operational respect, it’s probably just as well that we are forced into this kind of re-examination where technology is concerned because hype can be a distraction. Long ago and far away, I contributed to a home cinema magazine. Its twin preoccupations were sound and wide-screen, everybody who worked on the title at whatever level and the readers followed our evangelicalism and were similarly convinced by our vision of a multichannel, true aspect ratio future. Most of them, anyway.

There were the few who did not care whether film directors had decided to shoot their masterpieces in CinemaScope, VisiVision or Panavision. All they wanted was ordinary, vanilla TV pictures, the ones they had been used to ever since their parents had first propped them up in front of the set when they were children.

What they knew, probably subconsciously, was that the programmes shown on TV were made for the particular dimensions of the screen that were dictated by the shape of the original cathode ray tubes, which letter-boxed material was not. Geneflies, like the people who won on the magazine and the readers who knew that their opinions were more into the aesthetics, inspired by such masterpieces of wide-screen cinema as Lawrence of Arabia and 2001—A Space Odyssey. With a big enough television screen, and a smart card needed, we could recreate the Roxy in our living rooms and see everything that David Lean and Stanley Kubrick intended us to see.

Wide-screen fans knew that the coming of digital TV would make it the standard aspect ratio, so all other types of programming that we saw on the little Cyclopic monster in the corner of the room would have to be broader as well. Some production companies became pioneers, shooting new dramas on Super 16 stock that would give the 16:9 wide-screen picture, which could be shown in the compromise 14:9 frame size to appease the bulk of viewers. Acclaimed programmes like Cracker and Prime Suspect in the UK looked good in this style but, watching them again, one can see that the constraints of shot and protect—where the image is concentrated on the portion that will form the 14:9 frame—meant that directors could not be as adventurous as their feature film counterparts.

Broadcasters deem 14:9 to be a successful compromise, avoiding the complaints where viewers are convinced that there is something wrong with their set because all they are seeing is a narrow picture across the middle of their screens. The thin black bands of this intermediate format are becoming a familiar sight on many TV stations; more movies are being screened in something close to their original ratios. But there is now doubt within the industry as to why it is being done.

At a recent Moving Image Society seminar designed to update producers and technicians on the progress of the technology, there was obvious concern about the transition from black and white to colour, the viewer got something for his money. But with wide-screen, the picture is misrepresented for both 14:9 and 4:3. Who are we doing this for? Us or the viewers? Producers are also concerned that wide-screen increases costs, particularly as manufacturers and hire companies are not automatically charging more for wide-screen equipment, at a time when commissioning TV stations are expecting the same, if not better, quality and content for the same, if not less, money.

The coming of digital TV, and therefore of wide-screen, is written in the law. The broadcasters are following the lead given to them by government, which is no doubt considering how it can spend the money it makes from selling channel frequencies to mobile phone companies. As much as people think that this is a conspiracy designed to allow manufacturers to sell new TV sets, it is going to happen anyway and everybody is going pulled along in some form.

Some producers feel that wide-screen is changing how TV is made, moving away from the screen shape and style that has suited it for so long. There are others who believe that it poses new challenges. Both views are correct and things will only become clear once the compromise situation caused by simulcasting analogue and digital is over, although it will probably go on for some time.

There are issues here because, despite the number of wide-screen movies and the majority of new programmes being made in the format, there is still a vast library of material made for the old rectangular picture. This is not only vintage programming but also news footage and the huge number of movies that were not shot for the wider frame. All of this will have to go through some kind of aspect ratio conversion. Good news for manufacturers of such equipment but it means that some form of compromise will always exist.

Once again, TV parallels real life.
Space: the final frontier

With multichannel considerations forcing their way into almost every discussion and development of audio technology, Francis Rumsey questions our preconceptions of stereo.

After 30 or 40 years of devaluing 2-channel stereo recording and reproduction, it is possible that we have become relatively entrenched in our views about what 'stereo' is intended to deliver. This short article is designed to get recording engineers thinking again about the spatial aspects of reproduced sound, especially as we grow in understanding leads from conventional 2-channel to multichannel stereo. It is suggested here that some concepts of what may be called 'spatiality' mean different things to recording engineers than they mean to the average listener or concert-goer, and that the industry may have dug itself into a position concerning what is 'correct' in sound reproduction. Of course, nothing is likely to spark more argument than stereo-aspectiveness, but it is possible that there may be some features relating aspects of spatiality to listener preference, as has been found in the field of concert-hall acoustics.

In my limited experience, one encounters two rather polarised groups in the industry, with regard to the aim of sound reproduction. There is what we may loosely term the 'acoustic holography' or 'soundfield reconstruction' school of thought, usually populated by the more scientifically-minded of the community, who might claim that our aim should be to get as close as possible to a reconstruction of some natural acoustic experience. On the other hand, there is the group which might claim that reproduced sound is a totally different experience to natural listening, and should not be concerned with any attempt to recreate 'reality' but that entertainment value can be achieved to be manipulated at will. Somewhere between these extremes lies a useful compromise that will enable us to deliver some of the desirable spatial cues apparent in natural listening using reproduced sound.

This is not meant to be an exhaustive review of the development of stereo (what a task!), but it is noted that most of the historical research work relating to loudspeaker stereo reproduction concerns itself almost exclusively with the idea of recreating some of the directional cues needed for sound localisation. It seems natural that this should be the case, since an ability to deliver directionally accurate sound images is normally considered to be one of the main reasons for using stereo as opposed to mono.

Various models of spatial hearing have been used as the basis for the design of stereo systems, some based on mainly interaural time differences, some on amplitude, some principally on low-frequency mechanisms, some on spectral cues, some on various forms of precedence cues. The most successful with regard to directional localisation of individual sound sources have tended to be those that satisfy more than one localisation mechanism, thereby leading to less potential confusion of the hearing mechanism. But the concentration on source localisation, or 'imaging accuracy', has tended to obscure equally important aspects of spatial reproduction, such as perspective, envelopment, and those many attributes of reflected or ambient sound. Even the late Michael Gerzon, who attempted a 'General Metatheory of Auditory Localisation' in 1992 as a basis for designing future stereo sound systems, had to admit that the theory did not yet 'take into account ambient sound.' Real spaces, unless they are anechoic, give rise to involved patterns of ambient or reflected sound, and our day-to-day subjective experience of spatiality is as much governed by these features as it is by our awareness of the directionality of direct sound sources. When we attempt the simulation of acoustic spatiality in reproduced sound we are simultaneously willing to deliver some of the 'acoustic cues' that influence subjective impressions of 'space', recognising the vital fact that unless we adopt a system involving a very large number of channels (millions have been suggested) we are unlikely to be in a position to undertake truly accurate 3-dimensional soundfield reconstruction that allows for the listener to adopt a wide range of different positions within the soundfield. This limitation rules out the accurate directional imaging of sound reflections in a diffuse soundfield, requiring us to find other ways of delivering impressions of space or ambience to listeners. The idea that reproduced sound might somehow be able to render the acoustics of natural environments accurately has fascinated audio specialists for years, but it is contended that for most practical purposes relating to consumer audio this idea is Holy Grail not worth pursuing. The best we can expect to achieve is a limited compromise solution to the problem of delivering spatial sound to the living room (or kitchen, or wherever), and it is both in recognising the compromises we are adopting and then optimising compromise solutions to maximise listener satisfaction that our responsibility lies.

At this point I should perhaps say that I specifically exclude implementations of binaural and transaural 3-D audio from this discussion, since I believe they are not suitable for general purpose, multi-listener consumer sound reproduction, where listeners cannot be expected to adhere to a very limited range of physical positions or wear particular headphones. Such approaches are capable of good results particularly when using individualised transfer functions, when the listener's position with relation to the transducers is known, and this has been used successfully in virtual reality, computer applications and simulators.

As we make the gradual move from 2-channel stereo to some form of multichannel surround there will be more flexibility available from a spatial point of view, but we should not kid ourselves that suddenly we are going to be free of the various spatial compromises inherent in 2-channel stereo. It is nonetheless possible that the priorities may change, and aspects of spatiality considered vital in 2-channel situations may be lower down the pecking order in multichannel environments. We have to discover what aspects of spatiality in different modes of reproduced sound relate most closely to listener preference, and then design the recording and reproduction approaches to deliver appropriate signals.

One of the terms used most widely in discussions about spatial reproduction is 'imaging.' It is well known that 2-channel stereo reproduction is capable of delivering phantom images of individual sources in positions between the loudspeakers. The precise location of these phantom images depends on a variety of factors, including where you are sitting, the arrangement of the loudspeakers, how the images were sourced, the room you are in, and so on. Although I am quite willing to be shouted down on this matter, I would contend that many sound engineers and hi-fi buffs regard precise imaging as a positive factor in sound reproduction. The more precisely defined the phantom image the better, they might say.

A second point to consider is that any ambient-reflected room sound present in a recording is reproduced from the same loudspeakers as the direct sound sources. Consequently the majority of the
ambient sound, that in natural environments would have been widely spatially distributed, is constrained to a relatively limited reproduction angle with two-channel stereo. Now this does not necessarily mean that one cannot perceive an enveloping spatial impression from such a system, since, as David Griesinger points out in his recent AES paper 'Spatial Impression and Envelopment in Small Rooms', a variety of mechanisms can still result in degrees of spaciousness, including interaural fluctuations resulting from the direct sound at mid frequencies, the decorrelation of recorded reverberation, and interaural delay modes in the listening room. Nonetheless, it is acknowledged that the spatial impression from only two loudspeakers is quite limited. A more satisfactory spatial result can be obtained if the reverberant component of the signal is reproduced through laterally decorrelated multiple loudspeakers. Considering early meters to be almost entirely decorrelated only two loudspeakers in a multichannel arrangement, but that greater LEV (the listener's sense of envelopment by the ambient sound) required the presence of the rear loudspeakers. He also found that ASW did not alter even if the front-back ratio of the surround sound setup was doubled, provided that the IACC (Corrected to the early IACC mentioned earlier) was kept constant.

Informal discussions with people who have experimented with making 5-channel music balances raise the question of what to do with the central channel. The view is often expressed that pandering a center channel source with a vocal solo, for example, to just the center loudspeaker creates an unnatural spatially constricted effect. Spreading it over a slightly wider angle, using the left and right channels, leads to a more pleasing effect. This is perhaps anecdotal evidence of the value of a broader source in multi-channel reproduction. It also raises questions about differences between phantom and real centre images in multichannel reproduction.

One often hears audio enthusiasts talking about certain converters or systems giving rise to a greater sense of depth 'as if they were there' or 'as if you were there'. These are all spatial terms, with meanings that are more or less understood by the people who use them, and they imply that there are subjective aspects of sound reproduction that are not characterised well by currently available measurements relating to auditory responses or distortions. In the ongoing debate about the merits of high-resolution audio systems with greater bandwidth and dynamic range, the subjective effects that are noted by people when discussing differences are often those subtle spatial ones, rather than timbral, noise or distortion issues.
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Mixing for surround

The rising profile of 5.1-channel surround systems is causing increasing concern over their use in music mixing, Dolby's Tony Spath offers some insights and suggestions.

For the first time, DVD offers us a practical means to deliver discrete high-quality multichannel audio to the consumer. The concept of 5.1-channel audio originated in the film and cinema world so it is inevitable that some of the terms used will be carried over. These guidelines are an attempt to give some basic starting points for mixing music in 5.1, by explaining the relevance of these terms and pointing up some of the areas where there are alternative courses of action where the outcomes may not be immediately apparent.

Unlike virtually any other recording and playback format for consumers, film sound is mixed in the same environment as it is reproduced. All aspects have been standardised and calibrated to maintain uniformity between what the mixers create on the dubbing stage and what is heard in the cinema. For purposes of this discussion, this includes the recording levels on the film soundtrack itself and the overall loudness during playback.

When it became important to improve the sound in cinemas years ago, the level and calibration standards had to be maintained to ensure uniformity and interchangeability. Even though it became possible to produce a higher quality of sound in the cinema due to the continuous evolution of power amplifier and loudspeaker technologies, strong bass was still not easy to deliver or reproduce. The best soundtracks of the day ("7mm magnetic") were already recorded to their maximum capability, so it was not possible to significantly increase the amount of bass they carried without causing overload. Additionally, the main screen speakers used in cinemas do not reproduce much below 40Hz even today, so if the main soundtracks carried more bass to the amplifiers, the speaker systems would not necessarily reproduce it. To supplement the bass capabilities of the screen speakers, subwoofers were installed. To get the bass signals a separate channel was added in the soundtrack. This is known as the LFE (Low Frequency Effects) channel and handles bass created especially for subwoofer 'boom' effects and may also be used to carry bass derived from the other channels in order to help enrich the overall soundtrack presentation.

Consumer delivery formats like CD differ in some significant ways from a cinema system. CDs do not have a calibrated loudness when played - the consumer decides where to set the volume control. Nor do CDs have a calibrated recording level - if a music producer wants to add more bass to a recording, the overall levels are simply adjusted accordingly to ensure the bass can fit without causing overload. The CD and LD digital audio formats have proven very capable of delivering substantial bass when desired for a particular programme, whether it be music or a movie soundtrack.

Each main channel in the Dolby Digital system is likewise able to carry substantial bass content. So why is there an LFE channel in a consumer audio delivery format? Quite simply, it allows movie soundtracks to be transcribed directly and without alteration to the home video format. Does this mean the LFE channel should not be used for music recordings? No. But it does suggest that the LFE channel may not be the only, or even the best way to provide loud, deep bass in a music recording. This point will become clear as you begin mixing multichannel music in a studio with a properly configured and calibrated monitor system.

There are some aspects of multichannel studio monitor setup that are well understood and accepted, but others that are still under discussion and debate as to the best approach. Here are some basic suggestions.

Instead of one pair of stereo speakers, there are now three speakers across the front. They should be identical in the same way the LR pair in conventional stereo must be matched to promote good imaging. If they cannot all be the same model, the centre speaker could be a smaller speaker from the same product line.

The speakers ideally should all be at the same height and distance from the listener. If the centre speaker is not at the same distance as the LR pair, time delay may be used to obtain coincident arrivals. The front speakers must exhibit the same acoustic polarity through the entire monitor-amplifier chain.

Ideally, the surround speakers will sound the same as the front speakers. This may be most easily achieved by using the same speakers all around. If this is not possible, the surround speakers may be smaller than the front speakers but should maintain the same character - they could be smaller speakers made by the same manufacturer.

The surround speakers should again achieve coincident arrival with the front speakers either as a result of equal path lengths or through alignment with time delays. The surround speakers must be in phase with each other and with the front speakers. Assuming some or all of the speakers used do not adequately cover the deepest bass found in recordings, it is important to include one or more subwoofers and proper bass management in the monitor system. The bass from each and every channel that is not reproduced in the main speaker for that channel must be redirected to the subwoofer(s). There are now various products that handle bass management (crossover filters, bass mixing, and combining with the LFE channel in the proper mixing ratio) that can be used to achieve a proper monitor setup in the studio.

The ITU-R has a specification for listening room layout intended for critical evaluation of multichannel programmes. These recommendations also appear to be a good starting point for mixing room setup, and have been informally adopted as such. Aside from time alignment, a specific geometry is described. With the centre speaker directly in front, the LR speakers are each positioned 30° away from centre, forming a 60° angle. This angle may be reduced somewhat (say, 15° to 30°) and give equally successful results. The surround speakers are each positioned 110° off centre. This puts the surround speakers to the sides and somewhat behind the listener, which is not only what often happens in typical homes, but has proved to be a good way to achieve overall front-back soundfield integration and envelopment. If the surrounds are too far to the rear, the listener finds himself lost somewhere between two separate soundfields, rather than wrapped inside one cohesive soundfield.

All 5 main speakers are calibrated in the same way. Once each speaker is in position and equalised (if necessary),

Using the centre speaker alone creates a stable centre image for every listener no matter where they sit. To prevent the centre image from sounding too focused or narrow, its reverb can be spread to the LR channels.
the balance among the channels is adjusted such that equal signal levels in each channel at the output of the console results in equal loudness at the listening position. There are no dictates regarding the SPL on a system must deliver, but the power amplifiers and speakers should be chosen to allow all of them to play as loud as and as cleanly as is required by the mixers. The subwoofer system and its power amplifier must be capable of the same result.

Use of a surround setup presents music mixers with opportunities and problems. Here are some insights and suggestions concerning both.

In a stereo programme there has previously been only one way to obtain a centrally placed sound image — mix the signal equally to the LR channels. In a multichannel system, there are three ways to achieve this. The first is to create a phantom centre just as with stereo, the second is to use the centre channel alone, and the third to use all three front channels equally or in some varied proportion.

Reasonably enough, each approach offers advantages and drawbacks. The phantom centre is well understood, as it has been used since stereo began. Its main disadvantage is that the centre image is dependent on the listener being seated equidistant from the LR speakers. This is not always the case in domestic setups and is never possible in an automotive environment. One other disadvantage is that the timbre is not the same as from a direct speaker source due to cross-channel correlation effects.

Using the centre speaker alone creates a stable centre image for every listener no matter where they sit. To prevent the centre image from sounding too focused or narrow, its reverb can be spread to the LR channels, for instance. Using all three front speakers in combination can allow the range of spatial depth and width to be controlled. A phantom centre can be reinforced by some additional signal in the centre channel, or a centre channel signal can be enhanced with some additional signal spread into the LR pair. As more channels are used to carry the same signal, the more likely side-effects caused by these signals interacting with each other, or the phantom image conflicting with the true centre image, become. In systems using dissimilar speakers or in cases where the listeners are seated off the central axis, the sound arrivals from all three speakers may not blend well. Differences in arrival time can lead to a comb-filtering effect, shifts in tone colour, or a smearing of the image, for example. It is advisable to consider this when placing identical signals in all three front channels. To counteract this, the additional signals may first be processed somewhat to change their spatial character, timbre, or prominence relative to the main centre signal.

Whereas centre signals were always part of mixing for stereo, the surrounds are a completely new dimension to consider. There is lots of room for experimentation, so no one knows where multichannel music will ultimately lead. One thing we can say is that it does not take too much to enhance the sense of depth over what conventional stereo could achieve. The subtle ambience and room reflections of a concert hall delivered from the surround speakers can change the listener’s perspective from that of peering through a window to actually sitting in the concert hall.

Popular music often benefits from a more creative use of the surrounds, such as with background singer or instruments, but as with any new tool or effect, it can be overdone and become tiresome if used to excess. Maybe the principle that has served so well for the film industry can also work for music mixing: don’t use the surrounds to distract the listener’s attention away from the story.

What’s the difference between the LFE channel and the subwoofer signal? The LFE channel is a separate limited frequency range signal created by the mixing engineer and delivered alongside the main channels in the mix. A brick-wall filter at 120Hz in the Dolby Digital encoder limits its use to handling the bottom two audible octaves. Dolby generally recommends limiting the signal to 80Hz in the con-
A question of balance

Keeping a clear vision of your dream installation can be hard once the retail arena is entered. Philip Newell, gives advice on withstanding the assault.

Readers may wonder why I occasionally make some rather blunt attacks on studio equipment manufacturers. Did I get some sense of satisfaction by pointing out their mistakes or is it some Freudian way of compensating for my own errors? I hope that neither is the case. Almost anyone who knows me well will surely testify that I am neither cynical, sarcastic, nor nit-picking by nature, and that I rarely speak badly even about those who may have unjustly unaligned me. I hope that I have too much of an unbiased sense of honour to allow myself to fall into such ways.

One thing that does rile me, though, is seeing others suffer. I am a vegetarian, solely by virtue of the fact that I cannot countenance the thought of animals going through any suffering just so that I can enjoy the taste of certain foods. I am not in the least a religious person, but perhaps some of my character is because I was brought up by caring parents. I care about rats and worms, but especially, I care about my friends. Around the world, on average, I design and usually supervise the building of about six or eight studios every year. I am in no stretch of the imagination a businessman, so clients, in almost all cases, become my good friends, and their frustrations become my frustrations.

The face of the recording industry is changing rapidly, and the speed of expansion is ensuring that the number of people with in-depth knowledge of studio design are becoming ever more thinly spread throughout the industry. I'm a very lucky person to have been able to spend over 50 years working in an industry which I find fascinating, and with some really special people. One aspect of the expansion that I find rather sad, though, is the way that the manufacturing has changed from cottage industries, run by small groups of dedicated people, into what is largely a mass production industry which is more or less solely governed by financial motives. In itself, this is not necessarily a bad thing, but it becomes a bad thing when customers are seen to be an irritation, rather than the valued clients on whom the industry is ultimately dependent.

Twenty years ago I built studios almost exclusively for clients with a good knowledge of the industry, and who usually had a staff of well-educated personnel. Now, I would say that around 70% of the studios I build are for owners of little technical education, and who often do have staff at all, and who don't entirely rely on freelance (or dubious abilities), are asking me to build a dream. They have child-like faith in the abilities and the integrity of manufacturers and dealers, yet time after time, I see them being led like lambs to the slaughter.

My writing of books, technical papers, and countless articles tends to be driven out of either sheer enthusiasm, or in order to post the areas of quicksand, to try to prevent people from inadvertently falling into them. Unfortunately, in so many instances, the unwary are led directly into those quicksands by the hype of the half-truths of many glossy advertisements, and the unscrupulous profit seeking of many equipment dealers.

When clients commission me to design a studio, then they engage my services as an adviser on how to achieve a complete recording system. Dealers, on the other hand, are often desperately trying to influence them in other directions. There are reputable dealers out there, but unfortunately, on a worldwide scale, they tend to be a minority. The majority, it would seem, do not care one jot about my problems of building the integrated system which I believe to be in the best interests of my clients. Head-on conflicts between designer and dealers seem to be on the increase. Beware of the dealer who six months ago told you that the products of manufacturer A were the best, and now that they have either lost the dealership, or changed their allegiance to the more profitable range of manufacturer B, say that manufacturer A's products were really not very good. Beware also the dealers who always want to sell the next most expensive unit to the one which the designer has specified. The designer will most likely be thinking about the best overall and end result, the dealers will more likely be thinking only of the profits.

It's a tough world out there. Manufacturers put pressure on dealers to sell a minimum amount of product each year, or risk losing their dealerships. Remember that fact if you find yourself confronted by dealers who try to sell you not what you were asking for, but what they want to sell you. This is especially true if they seem to be treating your initial requests with disinterest, or trying to make you feel ignorant. Rarely take the advice of dealers over that of designers. Designers are probably considering a balanced system. It seems to be intuitively obvious to Ferrari owners that if a Porsche agent tries to sell them a new-style Porsche gearbox, it is unlikely that it will fit comfortably into a Ferrari, for which it was not designed. Unfortunately, however, it does not seem so obvious to many studio owners that much equipment may not be suitable for an environment for which it was not designed. In the recording world, the question of balance is an issue of musical instrument relationships.

Inequalities happen due to the insecurity of many studio owners and operators about their beliefs in their own ears and musical judgements, and it is on these insecurities that many parasitic operators thrive. The only real solution to the insecurity problem is experience and education, and hopefully, via the pages of Studio Sound, more of this can be shared out, which will hopefully lead to a healthier recording industry; and that is something else that I care about greatly.
mum have lower capacity than primary cells so many radio mics use alkaline dry batteries. In live performances a battery running down can be a disaster so the engineer may fit a fresh battery whenever possible irrespective of the estimated state of the previous battery.

Newer rechargeable technologies such as lithium may be useful for radio mics because in addition to much greater capacity than nickel cadmium or nickel metal hydride they are extremely light for their capacity. Rechargeables do require rigid discipline in battery management so the state of every battery is always known. In a short-term rental scenario, charging time may not be available and dry batteries will be the best bet.

There are two physical approaches to radio microphone design. In the first, everything is packed into a single handheld unit. This may resemble a conventional hand-held microphone except that the cable coming out of the end doesn't go anywhere because it's the antenna. This is an advantage where production constraints require the microphone to be passed from one user to the next, although getting microphone, battery and electronics inside an enclosure that is robust enough is quite a challenge.

In the second approach the microphone itself is on a short lead and everything else is in a small box that can be pocketed or clipped to the wearer's belt. This technique means that almost any self-powered microphone can be turned into a radio microphone. Another advantage is that if the wiring to a miniature lapel microphone is routed inside the wearer's clothing a very discreet installation results. Miniature microphones can be affixed to the forehead above the hair line with sticking plaster and a very slim cable can be routed to the control box resulting in an installation which is all but invisible.

Conventional condenser microphones are out because they need a polarising power supply. The moving coil microphone is robust, but the magnet makes it heavy. The electret principle answered the radio mic designer's prayer because it has many of the qualities of the condenser mic, but needs no polarising supply, is very small, and weighs next to nothing. A simple FET amplifier drawing very low current is enough to interface the capsule to the audio circuit.

Radio microphones have been built to work on a number of wavebands, including VHF and UHF. VHF units require a fairly long antenna which is usually a trailing wire. In UHF a short flexible rod is sufficient. The precise frequencies and licensing requirements vary from country to country and present a huge administrative problem for international artists. Clearly where several microphones are to be used in the same venue each must work on a different radio frequency.

There is very little control of the radio frequency environment and radio microphones are forced to use low power to allow reasonable battery life. The transmitter has to be omnidirectional because the receiver cannot be expected to be pointed at a directional antenna in the receiver. Consequently multipath reception and shading are major problems and it is very difficult to avoid dead spots where there is so little signal that the reception is noisy.

Fig 3 shows how multipath reception causes fading. A reflected signal from some metallic object arrives later than the direct signal, and at some time the delay will be half a cycle at the radio frequency in use. This means the receiver sees a pair of antiphase signals which suffer cancellation.

One solution is to use diversity reception in which two or more receiving antennae are used. When the signal picked up by one antenna fades the chances are that the other antenna is still receiving a strong signal. In low-cost equipment, the two antennae may be at opposite ends of the receiver box. Better performance will be obtained if the two antennae are several metres apart. Suitable stands can be used to elevate the antennae. In general, the antennae should be at least a metre and preferably two metres from any other object as this helps to reduce the intensity of reflections and reduces fading.
sole to ensure best uniformity no matter how the program is delivered. The subwoofer gain is centered in the decoder product based on the use of bass management crossover filters as needed for the particular speaker complement in use. A subwoofer is present, the bass (including the LFE channel if present) will be redirected to the speakers best able to handle it, usually the main stereo pair.

In most music productions, there is unlikely to be a technical need to use the LFE channel. Since the overall programme level may be adjusted to allow for any proportion of bass to be perfectly rendered, the LFE channel might only be an advantage for something like the famous cannon shots in the DTL2 overture. In such a case, the overall programme level might have to be reduced several dB just so the last few minutes can make the desired impact without overload. By using the LFE channel, the orchestra can be recorded at a more normal level, with some of the loudest, deepest bass of the cannons carried in the LFE channel. Of course, the main channels must still carry the cannon shots so that they will be heard from the appropriate locations.

There is another benefit to using the LFE channel when carrying explosive bass signals, in that smaller stereo systems may not be able to handle such high levels of deep bass without significant stress. Since the LFE signal is discarded in the Dolby Digital downmix, these bass signals will not present any difficulty. The remaining portions of the cannon shots mixed into the main channels will convey the essential aspects of the performance for the downmix listener. While it may be of no particular consequence for cannon shots, the fact that the LFE channel is separate from the others means that its ability to blend seamlessly with the higher frequencies from the other channels can be affected by filters used to generate the LFE signal, for example. The best way to ensure a cohesive audio signal across the entire audible spectrum is to keep the entire signal together in the main channel or channels.

Even with the popularity of the 5.1-channel systems, there will always be a need to address stereo reproduction. There are three basic ways to do this: prepare a new stereo mix from the original multitrack elements, prepare a studio adjusted downmix from the multichannel mix, or let the decoder derive a stereo downmix.

The first option is different from today’s conventional stereo mixing sessions. The second option takes advantage of all the work that has gone into the mixing of the 5.1-channel version, and allows the mixing engineer to quickly derive a stereo version while retaining flexibility in the exact proportions of each channel represented in the final stereo mix. The third option is to make no separate stereo mix at all. In this case, the decoder will derive a stereo downmix based on preset formulas in the decoder. Consumer decoders can apply a degree of dynamic range reduction during the downmix process if necessary to prevent overload. The downmix options are all able to be previewed in the production studio, and a range of adjustments are possible.

One other aspect of previewing how the production will sound to the consumer has to do with checking the mix on a ‘budget’ surround system, in the same way that small speakers have traditionally been used in stereo mixing to check TV and small stereo playback compatibility. Having a ‘home style’ surround system on hand may prove useful in evaluating how well the mix translates to more modest playback systems.

Further information

For more information see the Dolby Digital Encoding Manual available from Dolby Laboratories or from the Dolby web site at www.dolby.com. Details include Room Layout, Speaker Placement, Channel Calibration, Subwoofer Calibration, and Setting Centre & Surround Delays.

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Radio microphones

Modern production values often preclude the use of a microphone requiring a cable link to a preamplifier. John Watkinson looks at the radio microphone.

In most recording applications, the microphone is stand mounted and the presence of a microphone cable is no great problem. The same is true for the outdoor recordist who will be carrying microphone and recorder. However, in many applications the microphone cable is an unwanted constraint and microphones are used in conjunction with radio transmission. An example of use might be a single camera ENG team consisting of camerade operator and reporter. The reporter has a radio mic and the receiver is mounted on the camcorder so that reporter and cameraman don’t need to be connected by a cable as this is fraught with difficulties in public spaces.

In many modern musicals the artists are too mobile to trail cables. Artists don roller skates and swing on trapezes, yet the audience demands clear amplification. The musical could not exist in its modern form without the radio mic.

It is not easy to obtain high quality with radio microphones—not least because of the potential for noise and interference in the radio link. In general radio microphones should never be used without good reason simply because a wired microphone is always going to sound better. In order to minimise noise, techniques such as pre-emphasis and companding are used in the audio frequency domain and FM frequency modulation is used in the radio frequency domain.

Fig.1 shows that in frequency modulation, the frequency of the carrier is driven up and down by the audio waveform. The receiver contains a device called a discriminator that produces an output proportional to frequency to recover the audio waveform. The advantage of FM is that interference is additive, and so it may change the amplitude of a signal, but it cannot change the frequency. As a result, FM gives better signal-to-noise ratio than AM; although it requires more RF bandwidth than AM because the sideband structure is wider. This requires wireless microphones to use VHF bands or above.

It is a characteristic of FM systems that the noise in the channel is proportional to frequency. Stated differently, the signal-to-noise ratio is triangular and so the upper sideband contributes more noise than the lower sideband. Fig.1b shows the reasoning behind the problem. An ideal demodulator removes amplitude disturbances by clipping the received signal. This eliminates amplitude modulation, but it doesn’t remove all forms of noise. The discriminator works by analysing the position of zero-crossings in the FM signal with respect to the time axis.

In the RF domain the transmitted signal has finite bandwidth so the transitions in the signal cannot be vertical. Instead they are sloping. Fig.1b shows that noise added to a sloping signal can change the position of a zero crossing, and cause the estimated frequency to be in error, resulting in noise in the demodulated baseband signal. Fig.1c shows that as frequency rises, the period of a signal becomes smaller. As a result, a given disturbance on the time axis assumes a greater proportion of the signal period at high frequencies than it does at low frequencies. As a result, the noise is proportional to frequency. Stated in audio industry terms, the SNR deteriorates at 6db per octave.

Pre-emphasis takes advantage of the fact that most audio-programme material has lower signal levels at the extremes of the audible range. High frequencies can be boosted prior to transmission and given a corresponding reduction at the receiver which will reduce hiss. This is particularly useful against the triangular noise spectrum of FM radio. In some cases audio signals have exceptional levels at frequency extremes. In these cases pre-emphasis is not beneficial as it could result in overloading. This is unlikely to arise in radio mics used for vocals, but should be borne in mind in other applications.

Companding is another useful technique to cut down noise. Fig.2 shows the details. The transmitter is preceded by a recursive compressor—that is, one which takes gain control decisions based on its output. This approach means that the encoder and decoder see the same signal so that the decoder can reverse the compression. Companding of this kind cannot be used to excess because, although the noise that the microphone and receiver will be supplied as a matched pair.

Designing a radio microphone is challenging because there are limits on size and weight yet complex processing at AF and RF is necessary. Surface-mount technology was a godsend because it allowed more complex processing at the microphone without an increase in size. Power is limited because it has to be supplied from a battery. Traditional rechargeable batteries such as nickel-cad-

Figure 1

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