June 2000

THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION AND BROADCAST

EXCLUSIVES

AKG C2000B
tc electronic Spark
Minnetonka MX51
Lab Gruppen 300
Stage Tec A–D
Mackie HR824
Helios EQ I
Wendt X2
dbx 386

Fostex-D1624
Fast, wide and hard recording

WIN A SONIFEX RB-DA6 DISTRIBUTION AMPLIFIER
THE MULTICHANNEL MIKING DEBATE
GARY KATZ: THE DAN MAN
POSTING THE GREEN MILE

RICHARD SWETTENHAM
A Tribute

www.americanradiohistory.com
AMS Neve
DFC

The New World Standard in Film Mixing

Digital First – Digital Best

The award winning DFC is an integral part of AMS Neve's solution for a connected world and forms a major part of AMS Neve's WorkFlow™ initiative.

The Academy of Motion Picture Arts and Sciences has recognised the AMS Neve DFC as the first fully digital audio mixing console specifically designed for post-production film mixing.

The Scientific and Engineering Award for 1999 – presented to Mark Crabtree, Huw Gwilym and Karl Lynch of AMS Neve plc for the design of the DFC.
Editorial
On success in the studio, on the radio and on the turntable

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Professional audio, post and broadcast news

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Lab Gruppen 300
The definitive amplifier review

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dbx 386
Exclusive: Good old microphone in, nice new digits out

Stage Tec Reference A–D
Exclusive: Mastering the conversion process

beyerdynamic DT150
Updating a standard in studio headphones

AKG C2000B
Exclusive: Latest Austrian studio condenser mic

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Special Focus:
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Comment
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When you next visit the Studio Sound web-site simply enter your email in the box and get our weekly news update.

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June 2000 Studio Sound

www.americanradiohistory.com
A question of outlook

A NUMBER OF INTERESTING POINTS are raised in our look at the prognosis of the DAW in this issue. While the devil’s advocate is clearly being played on page 57 it is the changing nature of the environment in which the DAW operates that surfaces as the issue rather than a shortcoming of power and application.

In too many people’s minds the cliched view of the DAW still remains a picture of an inclined and button-rich worksurface dominated by a green screen, and you could argue quite convincingly that the essential operational elements required to achieve DAW status have remained largely unchanged over the years. Bolt-ons and enhancements, such as larger and faster storage and processing which have led most significantly to increases in tracks, have only improved on the fundamental duties. Yet the stand-alone nature of these audio monoliths has slowly been eroded as you’ll find representations tied with sequencing, as widely available and profession-indiscriminate desktop utilities, and increasingly buried within larger picture editing motherships. It is the last of these that could be regarded as the fulfilment of what some see as the intermediate solution of our state-of-the-art combined and intergrated DAW-digital mixer-nonlinear, video-playback package.

When you create a DVD you are dealing in a variety of different media simultaneously with none of the artificial craft and discipline divisions that exist at other stages of its production chain. Elsewhere there are broadcast programmes produced in video-edit suites without resort to a dedicated audio professional and you will find many Soundscape and Yamaha cards working well outside the realm of audio rooms.

Not very many years ago I ploughed a furrow on the impending impact of picture systems with audio capability on the sound operator’s lot. Opinions were divided on whether such developments were a fuss about nothing or a threat. Economic pressures will cause the fuss that ensures that previously disparate processes will become combined. Whether this is a threat depends entirely on your outlook and whether you want to take advantage of it.

Zenon Schoepe, executive editor

Strings ’n’ things

WHETHER DANCE MUSIC was the catalyst for the rise of the European project studio, or whether the project studio was an essential ingredient in the rise and rise of European dance music, is a moot point. Either way, the two are inextri- cably linked as one of the Nemesis of major recording studios. The damage done to the lower end of the commercial recording market by Tascam’s first Portastudio is as nothing when compared to the sustained onslaught of successful records made in project rooms rather than bigger commercial facilities. But the swing back towards traditional recording venues seems to be gathering momentum.

It is easy to identify the elements that made dance a bedroom gift: drums sampled from vinyl or prepackaged in a drum machine, bass lines perfectly delivered by a sequencer, piano and strings from countless synthesiser modules packed with ‘dance’ patches… Only the topline vocal of the mainstream dance anthem promises anything traditional—maybe. But given dance’s success and proliferation, the question is: how to imbue such a formulaic sound with renewed appeal? And the current answer seems to be a return to the old-fashioned acoustic sounds that the genre had initially so readily discarded.

Let’s start with last year’s UK chart success of Touch and Go’s ‘Would You…’—an otherwise unremarkable single lifted in part by the plaintive female plea ‘Would you go to bed with me?’, but mostly by a trumpet lick light-years beyond the capabilities of any synth or sampler. More recently, Moloko’s ‘The Time is Now’ topped the UK charts—an unashamed dance track reliant on a token sequenced bass drum, acoustic piano and guitars, and lavish strings arranged by Audrey Riley. Since then, MJ Cole’s charting ‘Crazy Love’ has sought to further the cause of UK garage music with a captivating samples string lick that could only have been more beguiling had it been delivered by a real string section.

Admittedly, most of Moloko’s Things to Make and Do album (wasn’t that an Angletraxx single from the seventies?) is unashamedly electronic, as is William Orbit’s interpretation of Samuel Barber’s ‘Adagio for Strings’, but the message seems clear—dance music is offering to make peace over studio practice. And we all know that there’s no substitute for a good mic, a good preamp and the expertise to use them. Or, for that matter, for a talented violinist...

Tim Goodyer, editor
Focused on Film
Perfect for Post

Does your digital console have

A dedicated user-friendly control surface
with up to 96 fully-configured channels
and another 96 pre-dub channels?

6 EQ bands on the main channels
and 2 bands on pre-dub!

96 inputs (48 main, 48 pre-dub)
from 24 faders!

A flexible stereo channel option

Varispeed automation which can write
and replay at any speed down to zero!

Off-line editing for inserting, removing
and swapping sections?

The capability of mixing in surround
and stereo simultaneously?

A configurable, programmable
64 x 8 monitor matrix?

A comprehensive installation,
commissioning and training service
with unrivalled worldwide support?

If it does, relax, it's an SSL Avant.

If not, discover how Avant improves
productivity, enhances creativity and
expands your business.

Call to arrange a demonstration.

"We set out to build the most advanced
film dubbing theatre in Europe and
looked carefully at available digital film
consoles. We chose Avant for its sound
quality and advanced feature set - and
it has now become an integral part of
our film mixing facilities at Sun Studio."

Svend Christensen, Owner, Sun Studio, Copenhagen

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- 1 VS-8F2 effect board included, providing 4 mono/2 stereo effects. An additional board can be fitted if required, giving a total of 6 mono/4 stereo effects
- 24-bit A/D-D/A converters ensure optimum sound quality
- CD mastering is now even faster, and image files can be stored directly on the hard drive
- Emagic Logic VS software [PC/Mac] available free to registered VS users [excluding VS-840 series]

If music is a universal language, then this is where you write the story. These are the blank pages - you provide the pen and the inspiration.

Once you’re ready to record your music there are things you should look for which will help you achieve your vision: reliability, clarity, reputation, intuition, flexibility, power. Where do you start?

The Roland VS-1880 18-track hard-disk recorder catapults your music into the professional domain by delivering true 24-bit audio, via world-class effects processors, in a powerhouse package that has seen the V-studio concept sell over 150,000 units world-wide.
is set to become more popular

By combining power and ease of use with rock-solid stability, you can get on with the job of producing the most professional sounding material possible and then mastering it to CD with the optional CD recorder.

The message is one that's simple, yet powerful. The VS-1880 can make great music sound amazing... with a little help from you, of course.

For more information call (01792) 515 020
www.vstudio.co.uk • www.roland.co.uk
India: AMS Neve claimed collateral credit for Hitendra Ghosh’s success at this year’s Ux Zee Cine Awards in India. Honoured with Best Recording Award for his mix of the film January—starring Khayy Kumar, Karishma Kapoor and Shilpa Shetty—Ghosh (pictured seated) used the AMS Neve Logic 2 console installed at Gaurav Digital in Mumbai to secure his award which follows work on over 1,000 feature films and the honour of numerous other awards.

Sennheiser Scholars

UK: With the support of Sennheiser UK, the British National Film and Television School has inaugurated a student scholarship scheme for students of the 2-year NFTS Diploma specialising in screen sound. The award covers fees and provides an allowance towards living and travelling expenses. The first recipient is Robert Bourke from Dublin, whose final year studies will include shooting an animation, fiction and a documentary short.

Production on the Net

US: Euphonix has announced the launch of Listen-In, an internet-enabled remote monitoring service available to users of the System 5 digital console. The Listen-In Service allows customers of our System 5 facilities to offer a quality live, secure access to a Vision, Pro Tools and Fairlight systems. Sennheiser UK MD Paul Whiting commented: ‘We are extremely pleased to award the very first Sennheiser NFTS Scholarship to a student of Robert’s outstanding calibre. I think that his achievements at the NFTS are representative of the reasons for our funding initiative here.’

Contracts

Israel's leading studios have recently ordered a total of four Studer 928 analogue consoles for use in applications including TV broadcast and remote recording. The first 928 has been ordered by 36-input Globus Group's Israel and is the largest 928 in Israel. Jerusalem Capital Studios has ordered a 48-input 928 with modifications and customised functions specific to JCS while United Studios of Israel has purchased two 36-input 928s for broadcast transmission—one desk will be installed in a remote truck, the second console will be installed in Studio 6 and be used for TV broadcasting.

Sennheiser

Scholars

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control room mix via a high-speed Internet connection,' explains Scott Silfverst, Euphonix founder and chief product officer. 'Not everyone involved in the creative process of recording and mixing can be located physically in the control room at the same time. But a producer accessing the Internet from anywhere in the world—and through a secure link—can now listen to the mix as it’s happening live in the studio.'

John McGleenan, facility director at One Union Recording Studios adds, ‘Listen-In will give my clients a safe and secure way to hear a very high-quality audio mix of their TV and radio spots over the Internet and offer real-time direction from a remote location. They can monitor any System 5 studio live, to approve a mix or voice-over session as if they were in the studio.

‘Audio production is breaking out of its traditional boundaries,’ agrees Barry Margaretum, Euphonix CEO and president. 'No longer does creative talent need to travel to a central production facility, the Internet provides an exciting environment for collaborative efforts.'

**NBC on Dolby E**

**US:** American National Broadcaster NBC is to use Dolby E and associated technology for the distribution of multichannel and stereo for its with DTV programming. NBC will use Dolby E encoders at the Network to encode audio for which Dolby will provide technical assistance to NBC and its affiliates to ensure a smooth transition to multichannel sound. NBC began using Dolby Digital equipment for DTV beginning with the first high-definition broadcast of the Tonight Show with Jay Leno in April of 1999. New Dolby E products provide the technology we need to broadcast multichannel audio out to our audiences,’ said Jim Starzyński, principal engineer, technical planning and engineering, NBC.

Dolby E-encoded audio does not reach the consumer; it is re-encoded with Dolby Digital, the coding used for ATSC DTV broadcasts, DVD Video discs, and other consumer media, just prior to transmission. ‘Dolby’s help and technical expertise will be valuable in supporting NBC’s goal to provide the best technical programming to our viewers and listeners,’ Starzyński concluded.

**Fairlight firsts**

**US:** First sales of the Fairlight Prodigy integrated digital audio production system follow its North American launch at NAB
At Marantz we take the safety of your recorded material very seriously. With that in mind we would like to introduce you to the new PMD650 Professional Portable MiniDisc Recorder.

With the PMD650, Marantz has taken all of the performance advantages of a miniature disk-based digital recording format and built in all the features of a bullet-proof field recorder.

**SECURITY FEATURES**
- Dual Level Mono recording
- SP and LP modes
- One Touch recording
- Level Sync autostart recording
- Pre-Record memory cache and audio buffer
- Pre-UTOC Write (saves all recordings on power down or failure)
- Time & Date Stamp on all recordings
- Mechanical disk eject mechanism

**PERFORMANCE FEATURES**
- Full complement of analogue and digital I/Os
- Switchable 48V phantom power
- Integrated mic and speaker
- AA dry cell or rechargeable NiCd battery pack operation
- Automatic Noise Cancellation
- On-board editing

Award Winner 2000
"LOCATION PORTABLE EQUIPMENT" MARANTZ PMD650
A US: Manhattan's Photomag post house has added a second Soundtracs DPC-II digital console to its downtown Photomag@The Anx facility. The recently refitted Studio A will see the new DPC-II working alongside a Fairlight MFX3+ and is 5.1 equipped. Photomag 'Partners in Sound', US. Tel: +1 212 687 9030.

A UK: Sir George Martin has been presented with serial number 001 of the ISA 110 Limited Edition re-issue mic pre and EQ from Focusrite. The presentation acknowledged the relationship between London's AIR Studios and Focusrite, which dates back to 1985; 'I am delighted to see the new special edition of the wonderful Focusrite module, which Rupert Neve designed for AIR Studios many years ago,' said Sir George. 'It is something we use constantly today and its unique design is very popular with the leading engineers of the world.' AIR Studios boasts a large collection of Focusrite including the original 16 vertically racked Focusrite ISA 110 modules for the custom Neve console that were installed in 1988. The ISA 110 was modelled on Rupert Neve's 31106 EQ module designed for the A7971 by Rupert Neve, Sir George Martin and Geoff Emerick. AIR also owns four original ISA 110 modules which are used on outboard mic-ops and EQs.

Search for a star

Not: Two new websites intended to smooth the search for suitable studio and record company contacts have just taken to the air. Hobomusic.com allows record companies, personnel, and recording studies (on locations, price, availability, equipment) can be searched giving what is called an impartial family tree of the recording industry offering contacts with confidence and reliability. HitQuarters.com, meanwhile, claims to be 'the most detailed database of names, companies, addresses, numbers, recommendations, links, charts, contacts, classifieds and so on, ever made within the music business, free of charge'. HitQuarters' mission is to direct you to the right persons by providing information, advice and recommendations, thanks to our uniquely designed track-record search, the HitTracker, which reveals who is worth dealing with'.

www.hobomusic.com
www.hitquarters.com
High Gain

AFTER OVER 29 YEARS supplying professional audio equipment to studios, broadcasters, governments and the military, Sonifex thought the time right to offer Studio Sound readers a chance to win a selection from its Redbox range—an exclusive gold livery version of its distribution amplifier.

Redboxes began production in 1999 and are intended for use in various applications in radio stations, TV studios, video and recording suites. Having just passed the 1,000 sales mark, the range includes analogue units such as the RB-MA2 dual mic amp, RB-SM2 stereo-to-mono convertor, RB-SL2 2-channel limiter and RB-HD6 6-way headphone distribution amplifier, and digital boxes such as the RB-SC1 sample-rate convertor, RB-ADDA audio convertor, RB-DDA6A AES-EBU distribution amp and the new RB-MM1 mix minus generator. The complete list can be found at www.sonifex.co.uk

Studio Sound's competition prize is also a new unit called the RB-DA6, a high-performance 6-way stereo distribution amplifier that enables a stereo source to be directed to up to six stereo individually-buffered outputs. Best of all, we have three RB-DA6s—serial numbers 1000, 1001, and 1002—to give away to you.

In the time-honoured tradition, all you have to do is to answer three simple questions about Sonifex and its Redboxes. And for a little further entertainment, there's a tie-breaker:

1. In what year was Sonifex founded?
2. How many stereo audio outputs does the RB-DA6 have?
3. Sonifex also manufacture a portable hard-disk recorder—what is its name?

Tie-breaker: Suggest a new product for the Redbox series and the reason why.
World Wide Audio, without leaving your studio.

Rocket Network takes audio production beyond the boundaries of studio walls, making connections that let you work with anyone, anywhere, anytime. It's like a global multi-track.

On-line Flexibility.
Rocket Network uses the Internet to allow professionals to work together on audio productions without having to be in the same physical space. Instead of shipping tapes from place to place or renting high-capacity phone lines, you log into your Internet Recording Studio, where Rocket Network handles the details of passing your parts to others and vice versa. That leaves you free to concentrate on capturing the perfect take, using your own local system to record and edit. Whenever you're ready for others to hear your audio or MIDI parts, you simply post your work to the Internet Recording Studio, automatically updating everyone else's session.

Full Audio Fidelity.
With Rocket Network, there's no compromise in audio quality—the system handles files in a vast range of formats and compression levels, all the way up to uncompressed 24 bit/96kHz. And you don't need access to a super-fast connection; DSL or T1 is great, but you can also work productively over a humble 28.8 dial-up. The system supports multiple user-defined presets for posting and receiving, and handles all conversions, letting everyone participate in their own preferred format. That means you can conduct a session in a speedy, low bit-rate "draft" mode, then move on while the final parts are posted in the background at full-fidelity.

A Powerful Connection.
Rocket Network adds a new level of freedom to creative collaboration, allowing you to choose your team—singers, musicians, voice-talent, composers, engineers, producers—based on who's right for the project, wherever they happen to be. With full fidelity, plus anytime, anywhere productivity, Rocket Network is a powerful new connection to the world of audio production.

Escape the boundaries of your studio walls.
Register at www.rocketnetwork.com

source code: RN22

professional tools.
Through partnerships with leading audio developers, Rocket Network is bringing RocketPower to the professional tools you already use, starting with Steinberg Cubase VST and Emagic Logic Audio. A multi-level permission system lets you control access to your Internet Recording Studio, And our RocketControl client offers built-in chat capabilities, so everyone in the session can chime in with feedback as the project takes shape. The Rocket Network Web site offers additional resources and services for audio collaboration.
Ever since our founder, Rupert Neve, designed the first Focusrite EQ and Dynamics modules they have been the choice of many successful recording engineers and producers. The ISA 430 repackages these classic designs in one comprehensive unit. Versatile routing and access to the signal processing blocks means you can split the EQ and compressor into two separate signal paths for simultaneous use on two tracks. Or use them both together with the original classic transformer-coupled microphone preamp for the best Total Input Channel money can buy.

Cap it off with the AES/EBU 24 bit/96 kHz optional Digital Output Stage, which can also be used as an independent A/D Converter. This is the best of Focusrite signal processing packed into one unit. Hence we’ve dubbed the ISA 430 the “Producer Pack”.

Ask for a demonstration before your next session.

**ISA 430 Producer Pack**

*If you can’t get the sound you want through this unit, then you have chosen the wrong Mic (or the wrong performer).*

David Mellor – Audio Media(UK) October ’99

*This is a first rate piece of studio gear and is a perfect illustration of why high-end analogue is still the preferred choice for serious audio processing.*

Paul White – Sound On Sound Magazine October ’99

Ryan Neve does not design for, nor is affiliated with Focusrite Audio Engineering Ltd.

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Fostex D1624

Latest in Fostex' digital recorder line is the D1624 which continues a train of thought and comes complete with fast-and-wide recording capabilities. Rob James punches in

The D1624 continues the Fostex tradition of workmanlike, cost-effective recorders. In many ways this machine is a development of the D-160 and, like its predecessor, it operates as a 16-track digital recorder at 44.1kHz or 48kHz sampling rates but also offers eight tracks at 96kHz 24-bit. In keeping with this, the A-D and D-A converters have been suitably uprated to 24-bit sigma-delta 128x oversampling, over-sampling from the previous 18-bit A-Ds and 20-bit D-A. The chunky metal cased disk carrier is supplied empty. Fostex offers extra empty caddies at reasonable cost and a short list of tested IDE drives is in the manual although I have no doubt many alternatives would be satisfactory. The SCSI connection is purely for backing up or exporting data from the internal drive. On the D-160 a suitable SCSI drive could be used directly. Another change, definitely for the better is WAV file import and export. This is accomplished using a DOS (FAT16) formatted SCSI drive. If you wish to load files originated on other systems the restriction is the file name. This must be a maximum of six characters with a 2-digit track number and the WAV extension.

The internal drive is formatted to suit the bit depth and sampling rate required. Bit depths and sampling rates cannot be mixed on the same drive even in separate projects. The other major operational choice is whether to initiate the Multi-ple Undo function. The trade-off is a simple one: with the function on, all recordings and edits are retained taking up disk space, with it off only the last recording or edit is kept—giving one level of undo/redo.

The D1624 uses Fostex FDMS-3 v2 (Fostex Disk Management System) This is limited to a maximum of 99 program files per drive. Recording can take place anywhere in a 24-hour ‘window’ on any track up to the maximum capacity of the disk. Apart from the 8 or 16 ‘real' tracks there are 8 additional tracks. Complete ‘real' tracks may be swapped with additional tracks to free space for further recording, individually or in 8 track banks. The D1624 records individual files for each recorded event and it is possible to insert silence between existing events. Silence is counted as an event—this is important since the maximum number of events per track is 512, although recording is not possible after 507 events on a given track. When editing, each edit point splits existing events, further increasing the number. The number of events can be monitored using a menu. Fostex suggests a number of strategies to reduce the number of events.

At 44.1kHz or 48kHz recordings may be up to 16 tracks wide if you have either 16 digital sources or 8 analogue and 8 digital sources. At 96kHz the limit is 8. Punch-in recording can be either automatic or manual from the control panel or optional foot switch. It rehearse mode switches the monitoring from replay to source in the familiar manner. However, when you actually go for it, the track(s) monitoring switches to replay about 2s after the actual punch-out. After punching out the transport must be stopped before further recording is possible. There is a delay of a few seconds after stopping before you can do anything else while ‘housekeeping' takes place.

Monitor switching is otherwise accomplished by arming a track for recording and pressing the arm key. Alternatively, pressing the track's source and arm keys together switches all tracks to input monitor. Since the machine is intended for use with a mixer there are no internal mixing or level control options for track bouncing or monitoring. The scrub key gives access to the many initial...
The optional time-code board allows the based on each new program destructive options such Check (for settings memories are displayed through precaution master controller. 9-pin slave are required. Word clock and趋向 digital up and catering Four TOSlink optical connectors a block of 8 pin a and one large are for option boards. The big slot takes either the Model 5043 balanced analogue board that mirrors the phons and gives B ins and 16 outs or two Model 8350 AES-EBU digital I/O boards. The Model 8345 Time Code and Sync board uses the smaller slots and adds video sync and time-code connectivity via XLRs for Time Code I/O and BNCs for Video In and Thru Video has switchable 75Ω termination. On the control panel the connection cable to the main unit is at the top left corner. A 9/16-inch jack below connects a punch-in/punch-out footswitch. The large, bright display is divided into a block of 16 bar-graphs with 16,335 alphanumeric characters across the top which display parameter name messages, time or bar-beat-clock and timing remaining on disk. RECORD TRACK select buttons are below in two banks of four (plus TRACK SHUT OFF) to access the 16 tracks. Below the track select keys three sets indicate All Input, (Disk) Access and TC Ready. To the right of the display are the dedicated status and mode indicators, sample rate, -1% (dropout) COMPLETED, SURE, and so on. To the right of the display the row of keys begin with AUTOPLAY. This cycles through the autoplay, autoreturn and auto repeat options. Two associated LEDs indicate which, if any, of the functions is selected. Two keys store and Recall Clipboard In and Out times. The Autoreturn section has four keys for START, AUTOPUNCH IN, OUT and END. A separate Autopunch mode ON-OFF key sets above the RECORD key with two JOS to indicate rehearsal or take.

settings and submenus. These can be divided into three groups: Changing the initial settings, Check (for checking the number of events in a Program) and Execution (for the more destructive options such as disk formatting and erasing Programs). The jog wheel scrolls through the circular list of possibilities.

Time can be displayed as absolute which is based on each new program starting at zero hours and continuing to a maximum of 24 hours, as MTC which is an offset added to ABS or MIDI bar-beat-clock which corresponds to the MIDI Clock. Song Position pointer. Adding the optional time-code board allows the machine to chase and generate LTC and adds 9-pin slave ability. If 9-pin is required, it would be wise to check the D1624 works with your master controller. In fact this is a sensible precaution with any 9-pin equipment.

There are a variety of ways of navigating through Programs. The 99 numbered locator memories are displayed in time or bar-beat-clock depending on mode. The CLIPBOARD IN OUT, AUTO PUNCH IN OUT, AUTO KTN START-END and LOCATE keys plus the STOP, FIND and EDIT keys in combination all store locations.

The editing functions are basic but I don't think editing is the primary purpose of this machine. As it is, editing is more than adequate for a tape recorder replacement. Practically, in and out points are defined and a function chosen. Copy and Paste makes a copy of the data and pastes it to another location via the clipboard, even between Programs. Move and Paste does almost the same thing but erases the source data including the clipboard and can only be used in the current Program. Using Copy and Paste multiple copies can be pasted in one operation to extend a sound. Erase gives two methods of erasing data. Either erasing from a specified point to the end of program on a track or, erasing between two specified points. All edits come with a fixed fade of 10ms or 5ms at 96kHz. To aid location of suitable edit points, the Envelope scrub function is available. This is invoked by pressing the shift key and a TRACK RECORD select key with the transport stationary. The selected track is then "scrubbed" with the jog dial. A rather coarse display of the waves form envelope is shown on the bargraphs. Track Swap allows complete real or additional tracks to be exchanged singly or in 8 track groups.

Fostex has succeeded in producing a competitive 8-track, 96kHz 24-bit recorder with the bonus of 16-track capabilities at 44.1kHz or 48kHz. The D-1600 formula of providing a replacement for an analogue or digital multitrack with the added benefits of basic 'cut and paste' editing at an attractive price is maintained in the D1624. Connectivity has been enhanced with MMC, SCSI and numerous backup options. The user-interface has been tidied up somewhat.

Tape-style cut and paste editing is fairly painless and Fostex has made it relatively simple to move material requiring serious work to a DAW with WAV file compatibility. This adds a powerful argument to the case for the D1624. If the removable E-IDE drive could be read directly by a DAW the case would be stronger still. The cost per gigabyte of E-IDE drives has dropped dramatically in the last year to the point where you can now record 2596 for the same cost per minute as 168 a year ago. This is now a viable option for original recordings regardless of the intended distribution format. I could criticise this machine for its eccentric transport key arrangement, simple editing, repeating loop scrub and time spent housekeeping, but I think to do so would entirely miss the point. Truly professional kit concentrates on core purpose and does not rely on landing a product with unnecessary features in order to distinguish it from the herd. The D1624 is a useful step in the right direction and makes a lot of sense when considered as a tape recorder replacement.
Minnetonka MX51

Aimed squarely at users of Yamaha’s DSP Factory, MX51 is a virtual surround mixing package with a few surprises. Rob James goes into games mode.

A couple of years ago I looked at Minnetonka’s MxTrax, a multi-track DAW software package for PCs. At that time MxTrax worked with the DAL PC sound cards. Since then, the Minnesota-based software house has been busy — MX51 for DSP Factory is one result, building on the MxTrax interface but being specifically designed to take advantage of the massive capabilities and low cost of Yamaha’s DSP Factory with, as its name implies, 5.1 surround capabilities. Minnetonka has also added support for the Microsoft Sidewinder Force Feedback Pro joystick and several other, more conventional, hardware controllers.

Software installation is straightforward, but even if you are familiar with other PC audio packages, I would advise taking the time to work through the tutorial. It provides a relatively painless introduction to the Minnetonka way of doing things.

The default setup divides the screen into two main sections, track displays at the top and mixer at the bottom. The split point is movable with the mouse to reveal more tracks or more mixer according to the current task, or either area may be zoomed full screen. Transport control is conventional with keyboard shortcuts supplementing the on-screen buttons. The track display is of the moving-cursor type so the screen redraws when you reach the edge. Much as I like moving tracks with a static cursor, the processor overhead, even on fast machines, usually makes this a redundant option.

The real fun begins once you’ve found your way around. I have always liked build it yourself mixers on software DAWs. Some manufacturers make a virtue of the absence of this facility, citing ‘ease of use’ in mitigation. Well, nobody is forcing you to use the mixer design functions, there are plenty of library mixers available but I think there is more satisfaction and utility in rolling your own for specific tasks. The great joy of this particular software-hardware combination is you don’t really have to worry about running out of processing power. Providing your aspirations don’t exceed the maxima imposed by the DSP Factory card or cards there will be no problem. A mixer is ‘built’ by dragging and dropping components from a floating toolbar palette. Apart from blocks which control the kit of parts provided by Yamaha, Minnetonka has thoughtfully provided vertical and horizontal spacer blocks to help keep designs clear. To begin construction from scratch an input fader is dragged onto the blank sheet. Other channel components are added in whatever order suits you. Channels can be saved as objects, copied and pasted singly or multiply. Output buss controls are added in the same way and a complete 5.1 Output Monitor is added as a single component. Once defined, default connections are made to the physical studio monitoring outputs whenever a surround monitor is placed.

An easy way to patch is to left click on, say, the track control area of a track and drag down to the fader-meter area of the input channel you wish to connect it to. The cursor changes to a representation of a jack plug to show you what you are doing. The record, playback and editing functions are much as before which is to say workmanlike with most of the things you would expect to find. Editing is quick and easy to use without too many frills.

I would like to have seen some more stereo functions. Tracks or segments can be moved as stereo pairs by grouping them but it is too easy to forget and get things out of sync, although the 100 levels of undo come in handy here. It would also be nice to be able to link adjacent mixer strips for stereo.
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Reid Hyams, President Chicago Trax

When only the best will do. When no compromises are acceptable. When sonic performance rules. These are some of the criteria in selecting a 9098i. Along with its sonic integrity, the feature set is also equally impressive. Recall, dual moving fader automation, built-in dynamics and indisputably superior mic preamps and equalisers. The 9098i combines the best characteristics of vintage consoles with features demanded in today’s mix environment. We invite you to audition a 9098i and experience the finest mixing console ever created.
The mixer automation is simple and easy to use. All functions except solo are automated. The three possible modes, Read, Write, and Update are selected globally from a pull-down menu or on a per-channel basis using the A keys which show green, red or green-red to indicate the respective modes. There is no direct way of saving mix passes but a workaround is to save the project under different names. The other luxury I missed was the lack of a "slide" function. Dropping out of automation record results in all the parameters currently in write immediately reverting to the previous value.

For surround work you simply drop a Surround Panner and Subduss filter into the channel strip. The panner graphic uses a red ball to indicate the current position and a light coloured V shape to show the divergence or spread of a centre source to left and right. The surround panner block can be zoomed to facilitate fine adjustment. To pan you simply grab the red ball and move it with the mouse, or you could use a force feedback joystick. This seems a curious choice of controller for audio use, then consider the cost of the alternatives and don't knock it until you try it. Apart from left right, X-Y axis, panning control this joystick has several buttons and can swivel (used for rudder control on flight simulators). On the top of the stick the big button on the left toggles transport stop-start and a mini 4-axis switch is used to select which mixer channel is controlled. Two more buttons toggle solo and zoom. The fire button is used to write automation to the selected channel. The channel automation mode is toggled by the A button at the base of the stick. The rudder control is used to position the playback cursor. Prior to pressing the fire button the force feedback is used to give an indication of the pan position at any given moment. Once the trigger is pulled to update the automation, the feedback can give you the location of the pre-existing panning movement. This is accompanied by a grey ball in the panner display. The object of all this is to enable a seamless transition to the previous position. The amount of force feedback is adjustable. In practice, I found it rather crude but fun. It also allows for fast working. If the force feedback joystick doesn't appeal there are other options.

The Yamaha DSP factory has failed to attract the level of support among writers of non-sequencer-based software front ends I expected when it was launched. The merits of the approach should be obvious to anyone who has spent frustrating hours juggling scarce resources with software which uses native processing or even some of the DSP card alternatives which try to address the problem. The only real potential downside of the DSP factory is the timbre of the effects, dynamics and EQ are hard coded by Yamaha. In fact, they are the same as the algorithms used in the estimable 02R. For many purposes, this and the reliability and ease of use confered, is well worth while. And of course, if you are not a fan of the 'Yamaha Sound', there is nothing to prevent the use of software supporting other plug-ins. Current versions of software such as MX51 and C-Mexx C-Console with SEK D Samplers as the editor make the case for DSP factory even clearer.

By developing MX51, Minnetonka has provided a plausible alternative to some of the more obvious names. The surround features are a good demonstration of just how commonplace mixing in 5.1 is becoming in the US. For forthcoming attractions are software encoders for Dolby Digital and DTS. MX51 is also available for use with DAL hardware. On the Yamaha-PC platform the software appears robust and a good number of tracks are possible with PCs which usually struggle to keep up. If juggling resources is becoming tedious MX51 may provide the answer to a prayer.
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Wendt X2

being small has its obvious and less apparent advantages, as in the case of Wendt's 2-channel location mixer Neil Hillman finds it perfectly formed.

There is a school of thought that believes that if it's big, it's good — big car, big guy, big hair, big fish. Big hill. Wendt have a product in the X2 2-channel mixer that's not big, but it is clever. Small size, small weight, small footprint, small outlay. Big performance.

All necessary adjustments and settings can be made easily, with accessibility obviously carefully thought out. The left-hand side of the mixer takes the microphone inputs on two electronically balanced female XLR sockets, with 12V Tonader, 48V phantom powering or dynamic settings selected via a 5-position slider switch located underneath and slightly to the left of each XLR input. Adjacent and to the right of each powering switch is a similar 3-position slider switch for amplifier gain, marked \(-10\), \(-20\) and 0. By ear and comparison rather than instrument, I assessed 0 to correspond to a gain of 80dB, with the \(-10\) position providing 10dB of attenuation to reduce amplifier gain to 70dB, and similarly the \(-20\) position providing 20dB of attenuation resulting in a net gain of 60dB. Below the inputs is the substantial battery compartment lid, indicative of the hugely robust feel of the Wendt X2, secured through a large-headed screw. The mixer takes six 1.5V AA batteries, which with care can turn the stated 12 hours endurance into a sensible two days location use.

Above the inputs on this left-hand side is a \(1/4\)-inch stereo headphone socket with a level control knob alongside, with bags of gain available for my 250\Omega DT150 headphones. Below the headphone socket is a 2-position slider, to switch the highly efficient limiter in or out. Only the very largest of input levels caused any discernible recovery slump in its operation, and working with it 'in' and riding the gain to ensure only occasional peaks just pushed the +6dB meter LEDs resulted in controlled use of the maximum available dynamic range. Alongside the headphone volume knob is a further 3-position slider switch to adjust the meter LED display intensity between Hi, Low and Off.

The meter LEDs are sited on the top of the front face above the SQN-esque rotary faders. Bright in the normal operating conditions of a low setting, an intense laser-blue impression reminiscent of a trace oscilloscope is burnt into your retina should you ever be so foolhardy to consider the possibility of using the Hi setting. For a good 10 minutes after I tested on your behalf the meter's brightness, colleagues appeared to be walking around with a large \(\cdot\cdot\cdot\) in the middle of their foreheads. The vu meter scales are calibrated by four blue LEDs corresponding to \(-9\), \(-6\), \(-3\) and 0, with \(-6\) signified by a red LED. In the top left-hand corner of the front face, and alongside the row of LEDs is a toggle switch marked TAPESX that enables the monitoring to change between the mixer output and the tape return signal. There is no level adjustment pot to trim the return signal on the mixer, so this needs to be adjusted on the headphone output of the recording device be it a DVCam or MiniDisc. A second headphone output on a 3.5mm socket is sited beneath the return selector switch, to the left of channel one's fader, and this is a parallel feed of the \(1/4\)-inch socket on the left-hand side. The mixer has an in-built 400Hz, 0dBvu oscillator. It is engaged by means of a latching push-button sandwiched between the right-hand channel row of meter LEDs and in the top right-hand corner, the green power monitor LED. This flashes on and off, in a ratio of about 1:2, to signify normal operating voltage and remains lit when the power is running low. When it doesn't come on at all is a sure-fire clue that either the batteries have completely expired, or the circuitry has walked off with your external NP-1 battery. The mixer is switched on through a 3-way slider beneath the power monitor LED, and is marked \(\text{in}\) (internal), \(\text{off}\) and \(\text{ex}\) (external). In between the two fader knobs is a hold-to-make, battery-test push-button that shows on the right hand row of meter LEDs, with full voltage corresponding to \(0\) on the meter scale. A slide-microphone, denoted 'mc', is situated behind a grille in the bottom right-hand side of the front face and is operated by another white nylon hold-to-make push-button, identical to the battery-test switch, and located underneath the power switch. Beneath each fader is an associated pan-control that hard switches through a 3-way slider the channel output to either the \(L\) (left), \(R\) (right) or, when the centre 'PAN' position is selected, across both outputs. Also beneath the faders is their respective 3-way toggle-switch for the selection of low-cut filters, these being 12dB octave with turn-over frequencies of 140Hz, 20Hz and 100Hz.

The right-hand side of the mixer is the most uncluttered of the three control surfaces, carrying the two outputs on male XLR sockets. Above the two outputs are associated slider switches that set the output signal to Mic or Line levels. An unbalanced Mic Out on a 3.5mm socket provides a stereo feed at microphone level for unbalanced equipment, and is situated below the disappointingly flimsy external power connector. This accepts a 2.1mm coaxial power plug of the kind popular with manufacturers of transistor radios and really is a poor choice of connector given the otherwise impeccable build quality of this mixer. The 3.5mm stereo socket for the Tape Return feed is closest to the operator on this right-hand panel, and sits alongside the preset pot for the gain adjustment of the line microphone. On location the X2 sat comfortably in the front pocket of the delightful Marantz PMD-60, and I found them to be ideally suited. Good things come in little packages. Small is the new big.
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Audio editing and mastering software with powerful plug-in effects capability are a first for TC. Simon Trask goes in search of illumination...

COMING FROM PLUG-IN specialists TC Works, the Spark Realtime Digital Audio Editor program for the Mac has an impressive line in plug-in effects—four lines, to be exact. The program’s FXMachine effects routing matrix, accessed via the Master View window, supports up to 4 x 5 effects plug-ins, that is four parallel audio streams with up to five effect plug-ins on each stream—computer processing power and available RAM permitting. TC Works specifies a minimum 166MHz Power Mac 604 machine (Power Mac G3 recommended) running Mac OS 8.1 or higher with 64MB RAM.

Spark comes in two versions: Spark (currently at v1.5) and Spark XL (v1.6). By the time you read this, Spark | Modular should also be available, this is an add-on component that brings modular synthesis into the FXMachine environment—or into any other VST environment. Spark supports VST native plug-ins, while XL adds support for TDM plug-ins and comes with two additional TC Works plug-ins for audio restoration work. Spark XL running only in native mode (TDM support disabled) has the same system requirements as Spark. However, in order to run TDM plug-ins within Spark, you need to have as minimum a Pro Tools 24 Mix system, and consequently a computer setup as specified by Digidesign. You’ll also need DigiSystem INIT v5.0 or higher and the Direct Connect plug-in.

TDM plug-ins get their own Master View window, but in this case with a single serial arrangement of up to five plug-ins. However, you don’t have to give up native VST processing to use TDM. Spark lets you use VST and TDM plug-ins at the same time, with the output of the VST FXMachine being routed into the TDM Master View chain.

The FXMachine effects matrix complements the Browser View and associated Cue Editor, which are where recording, editing and assembling is done. The mono or stereo audio file is fed into the FXMachine section, which at its simplest need use no effects at all but can provide a variety of effects routing configurations.

FXMachine is not only an adjunct to the editing process, however. Select Realtime FX-Machine from the program’s Options menu and it takes a feed from wherever audio source you’ve specified in its I/O Preferences.

A feature of Spark and Spark XL is that the FXMachine matrix is itself available as a VST plug-in. This means that it can be assigned as a plug-in effect within Record mode or to any of the plug-in slots in the FXMachine matrix in Realtime FXMachine and playback modes.

The FXMachine functionality can also be used outside of Spark as a VST or MAS plug-in, which means that it can be used in programs such as Cubase VST and Digital Performer. More accurately, FXMachine is a...
Sparks comes with 10 Spark VST plug-in effects (11 if you include the FXMachine in its plugin form) and two additional TC plug-ins (Native CL and Modx), while Spark XL adds the TC DeClick and TC DeNoise plugins. What comes as standard, then, are bandpass, cutoffter, delay, expander, fuzzsat, gainlifier, oneshot EQ, resfilter, reverber, and 3-hand EQ Spark VST effects, plus the FXMachine plug-in and the two additional TC plugins mentioned.

To make the VST plugins available to the program you have to select the plug-ins folder within the Preferences window. Spark (all references include XL) supports the Steinberg-originated ASIO 2.0 audio streaming protocol for getting audio in and out, as well as the standard SoundManager and Digidesign Direct I/O.

Spark’s Browser View window (the Worksheet) contains the project file database (File View), audio editor (Wave Editor), and the file playlist (Playlist). Spark also has a Transport window that provides the usual controls plus a Jog Shuttle wheel for scrumming (up to ±200%) or for setting realtime timestretching or varispeed amounts (up to ±25%). In addition, non-realtime (offline) timestretching and time-shifting are available, via the Process menu, with associated parameters.

A Scroll Bar within the Transport window lets you move to any location in your audio file, and a Cyclic button lets you turn looping of the file or selected region on or off. You can record from this window, or you can select the Record window from the Options menu, where you can select two effects to be added at the time of recording. The Record window is also where you select mono or stereo inputs, the record format (AIFF, SDII or WAV), and sample rate and resolution (11 options ranging from 8kHz-96kHz, and 8-bit, 16-bit or 24-bit).

In addition to recording via your audio card’s inputs, you can also import audio from your Mac’s CD-ROM drive, or import existing audio files off hard disk—including MP3 files, which are converted to AIFF. Spark can also export to MP3 format.

TC Works has certainly maximised the flexibility of getting material in and out of Spark. The program lets you import samples from Akai sample CD-ROMs, and it supports import and export of samples from Akai S1000, S9000 and S9000 ranges, E-mu ESI and F4 series, Roland S760 and Yamaha A3600.

With an audio file recorded or imported into the Browser View window you can set to work editing, creating regions and compiling playlists. The Wave Editor section of the window provides you with a familiar waveform display. To quickly loop any section of the file, you just drag through it with the mouse in the edit view section of the Wave Editor; Spark automatically switches to looping this section. To adjust the loop start or end point in either direction, you just hold down the shift key and drag the left or right edge of the highlighted section accordingly. Extremely simple and quick. Regions appears as subfiles of the actual audio data file in the File View pane, where you can name them and save the audio data contained within a Region as an audio file in its own right. From this pane you can also add into the Playlist to create a new track or to create a playlist of whole tracks for a CD. Spark comes complete with Adaptec's Toast.

The program provides an array of processing functions for the audio data, with fade, change gain, normalise, reverse, invert, CD removal, resample, alongside with the aforementioned pitchshifting and timestretching, which work very well. You can also select a plug-in to process the data. All the effects are editable via graphical ‘front panel’ displays.

I was very impressed by Spark’s combination of power, versatility, depth, and yet ease of use. Some might say it goes above and beyond the call of duty as an editor with its plug-in effects capability, notably with the FXMachine matrix and the ability to use the FXMachine as plug-in shelf in its own right—plus of course the XL version’s ability to use VST and/or TDM effects in the same environment. However, these elements are an important part of what makes Spark special and creatively intriguing. The effects that TC Works have provided as standard are very capable, and provide a good basic set of effects that can be put to good creative use if you wish. It’s also very easy to integrate into the Spark environment any plug-ins that you might already have.

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Lab Gruppen 300

Studio Sound's bench test amplifier reviews continue with the Lab Gruppen 300. Paul Miller reports...
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Fig 3

< from typically 0.05% to 0.6% by 60W/4Ω. This is compounded by Gruppen's VHF limiter that holds the single-channel output to 105W/8Ω at 15kHz, progressing to just 60W/8Ω at 20kHz (see Fig.4). Autograph Sales, the distribution and technical support arm of Lab Gruppen in the UK, was inclined to think that the VHF limit was somewhat higher than 20kHz, which may indeed be the case in its costlier, higher power designs. It is difficult to escape the in-band limiting of the little Lab 300, however, which also shows a 0.5dB loss in power output at 20Hz compared to the healthy 110W available through the midrange.

The practical handlimiting of this amplifier is equally evident from its frequency response (Fig.5) which, sensibly in my view, is attenuated beyond 30kHz. This may fly in the face of recent developments like SA-CD and DVD-A which offer the potential for extended, 100kHz bandwidths, but there seems little point in aggravating metal-domed HF units with the ultrasonic output of current CD sources.

### Power Amplifier: Lab Gruppen 300

(Rated Spec. in brackets where given):

<table>
<thead>
<tr>
<th></th>
<th>20Hz</th>
<th>1kHz</th>
<th>20kHz</th>
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<tbody>
<tr>
<td>Max Continuous Power Output, 1% THD into 8Ω (1 channel)</td>
<td>120W</td>
<td>130W</td>
<td>60W*</td>
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<tr>
<td>Max Continuous Power Output, 1% THD into 8Ω (2 channels)</td>
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<td>Max Continuous Power Output, 1% THD into 4Ω (2 channels)</td>
<td>130W</td>
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<tr>
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<tr>
<td>Dynamic Headroom (±HF)</td>
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<tr>
<td>Maximum Current (1Ω, 1% THD)</td>
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<td>Damping Factor</td>
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#### Unbalanced Input

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<td>Total Harmonic Distortion</td>
<td>-91dB</td>
<td>-57dB (-62dB)</td>
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<tr>
<td>CCIR Intermodulation Distortion</td>
<td>-83dB</td>
<td>-44dB (-62dB)</td>
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</tr>
<tr>
<td>Noise (A wtd, re. 0dBW)</td>
<td>-94dB</td>
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<tr>
<td>(re. 2/3 power)</td>
<td>-90dB</td>
<td></td>
<td></td>
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<tr>
<td>Residual noise (unwtd)</td>
<td>-83.5dB</td>
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<tr>
<td>Input Sensitivity (for 0dBW)</td>
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<tr>
<td>(for full output)</td>
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<tr>
<td>DC offset, left/right</td>
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<td>-1mV</td>
<td></td>
</tr>
<tr>
<td>Serial Number</td>
<td>911-116</td>
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</tr>
<tr>
<td>Retail Price</td>
<td>£695 (ex-VAT)</td>
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</table>

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The full benefit of future digital sources will only be realised by amplifiers whose compensation extends at least two octaves beyond 20kHz and by loudspeakers free of sharp VHF break-up modes. Otherwise, VHF intermodulation will only cause potentially audible distortion to fold back into the audio range. If it were not for the intended bandlimiting, this fate might well have befallen the Lab 300 where, for example, at a steady 10W/8Ω (where the lower voltage rail is still firmly in command) there remains a ten-fold increase in THD over its final decade (see Fig.6).

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Two balanced power rack systems from Equi-Tech have been designed to power the needs of larger sound and video applications. The new Models ETT-9R and ET10R are large capacity systems designed to furnish continuous power for systems that require between 60A and 85A. Rack system features include a magnetically shielded toroid isolation transformer, surge protection, overcurrent circuit protection, and GFCI protected power distribution. For 3-phase location power, the company offers Dube Series systems which employ 3 phases of balanced power for 120, 185 and 250A.


**APT telecom codecs**
APT has launched an apt-X-based audio codec family targeted at the telecoms market. Complementing advances, such as DSL, SDN, E1, T1, Fractional E1 and Fractional T1, the codecs offer a new range of network audio solutions for broadcast circuits which claim savings in network capacity, improved network efficiencies and reduced network costs.

APT, UK. Tel: +44 26 9037 1110.

**Maselec mic pres**
The MMAS is a 4-channel mic preamp in the Prism Sound Maselec Master Series which already includes the MEA2 stereo EQ and MLA2 stereo compressor. Used with the Prism Sound AD2 224-96 A-D converter the company claims it represents the ultimate machine for recording. prism sound, UK. Tel: +44 12 234 24998.

**V multiformat upgrade**
MAD Labs' CP8 and CP4 upgrades for the AMS Neve V Series consoles are multiformat center-section replacement packages that provide sonic and functional upgrades. The CP8 includes advanced signal routing, switching and monitoring functions required for post, broadcast and music studios working on 5.1 projects. Features include bus and monitor mode buttons, status buttons, master meters and selector buttons, monitoring level controls, talkback facilities, studio loudspeaker controls, monitor select buttons, user-defined trims, paddles and tape control, fold-down mixer and a dialogue and EFX monitor path. The 4-bus CP4 is a stripped-down version designed for use on the desk-frame size and other features. The upgrade package starts at $65,000 US. Similar packages are expected for the V3 and V35 systems.

MAD Labs, US. Tel: +1 818 351 6199.

**LTO Ultrium cartridge**
EMTEC Magnetics is developing a high-capacity Ultrium cartridge for use in the Studio Sound [www.prostudio.com/studiosound](http://www.prostudio.com/studiosound)

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**Helios EQI Lunchbox**

Resurrected and patiently brought into line with modern expectation, the EQI is history in the remaking.

**A BRIEF HISTORY LESSON:** The late Dick Swettenham was a maintenance engineer at Olympic Studios in its sixties heyday. In the early seventies he built a rather wonderful desk for Olympic and subsequently a number of Olympic clients' private studios. When Island Records wanted Swettenham to build a desk for Basing Street Studios he was told that his Olympic employers were none too happy, so he left and went into business with the Helios name. As time went by, gear fanatic and music producer Tony Arnold started to buy old Helios consoles and acquired seven of them for export to the US. One of his early tape decks built for Eric Clapton as his reference, Arnold saw no replicating Helios modules carefully including all the faults of the original to begin with. In the course of his research he found that the original desks' 20dB pad used a centre-tap from the microphone input transformer. These sounded better than later Helios desks, which used Beyer transformers with resistors rather than a centre-tap for the pad. Arnold approached Soaver to copy the Lustraphone transformers. He then built a console for Elvis Costello and Chris Difford's Helioscopic Studios and put in 20 original Helios modules and 18 new modules. Over a period of over two years incremental modifications to correct problems with the original modules were carried out on the new ones. Each time these were carefully compared to the originals to make sure none of the modifications had any detrimental effects. Cyril Jones of Rummik FAME helped with some aspects of problem solving. Having acquired the Helios name, Arnold set about perfecting the EQI modules we have here.

The original's unreliability was always a bugbear for Arnold, so the new units are built to military specifications by CLJ who among other things makes the Formula One racing teams. The front panels are dipped in paint and the discrete circuitry is butt-joint. Modules from subsequent production runs will feature phantom power (which is already included in the rack version).

The Lunchbox comprises a flightcase with two vertically mounted Input EQ modules alongside a panel with six bantam sockets for mic and line inputs and outputs — very convenient if you encounter bantam patchbays, a nuisance when you occasion-ally do not — and an IEC mains socket. There are no LEDs or lights anywhere. And with everything on the front, there is no need to open the back of the box.

The input section features a 3-position toggle switch for mic-padded micline input and a stepped control for gain. I had a couple of the original modules in a Lunchbox for comparison: the old ones sometimes sounded slightly smoother. This is probably due to the new ones' extended high-frequency response, which if audible, is not necessarily preferable to my ears. But any difference is extremely subtle. One vast improvement over the originals is in the build quality and reliability — crackly knots are understandable on something almost 30 years old, but the original was notoriously unreliable.

The stepped gain fixed 10kHz high frequency band is gentle and open, wonderfully enhancing clarity without introducing harshness. It goes in 2dB steps from -10dB to +10dB, although I have an inkling which way it will generally be turned. The Mid band comprises a selection of eight switched frequencies from 0.7kHz to 6kHz. A toggle switch on and off, with an uncalibrated pot giving a roughly 10dB range. Gain for the Low frequency band is also controlled by an uncalibrated rotary pot giving a maximum boost of approximately 10dB at 30Hz, 80Hz, 120Hz or 240Hz. These are boost 'only' frequencies. The frequency selector becomes an attenuation switch below zero, with results in dBV kept down to -15dB at a fixed 75kHz rate. If I thought I might miss the ability to cut low-mid frequencies, being something of a fan of sub-tractive EQ, but with the Helios, boosting always sounds so good that I quickly adapted to this way of working.

One Helios aficionado I spoke to testified to the glory of 'F-IAAF' (Full Boost At All Frequencies) on the Helios desk he had used. On the low frequency gain pot there is a click at the off-position. The originals had a low-frequency boost of 1dB when the EQ was bypassed. For purists this has been retained, but the click position gives you the option of switching it off. A high-pass filter gives 15dB octave cut at 40kHz or 80kHz, which is bypassed if the EQ is bypassed. A proper phase switch circuit is incorporated, the original design for this being something of a hodge.

By modern standards, the features and control are limited. But here the sound is everything. You know you're in heaven when just about any setting sounds good, not just one compromise setting, which can be the case with many desk EQs. I recorded and processed all sorts of sources and the Helios always impressed with its wonderful openness and clarity. It went down to just one input module and the EQ1 would certainly be near the top of the list.

A giant from a previous era returns in style, and in a portable package to take wherever you go.
Adding its own mark of quality to the choice of combined mic pres and converters, dbx 386 catches Dave Foister ear.

The must-have outboard in the analogue studio is a pair of posh microphone preamps. The must-have outboard, increasingly, in the digital studio is a pair of posh microphone preamps with 24-96 digital output. There are few digital desks that can't be improved by the addition of a couple of high-grade input paths, and few projects that can't benefit from good mic output formats, and very nice it is too.

The 386 is a 1U high box that manages to pack in a tremendous amount of user-control thanks to clever use of indicators. Its front panel is not used quite as efficiently as it might have been, because even now, when we're all a bit blasé about the valve stuff, dbx has been unable to resist incorporating grilles to show off the glowing valves within. This scarcely matters, as the rest of the functions rely heavily and imaginatively on multi-coloured lamps to show us what they're doing without hogging too much space.

The basic preamp functions are precisely what you would expect. Each channel has the expected pair of inputs, microphones on XLR and line on TRS jack, a TRS insert point, plus the now-familiar sight of a 2-pole jack on the front for direct connection of an instrument. Analogue outputs are available on XLR and TRS jack, and in between comes a useful and typically simple set of controls. The only rotary ones are for gains, the input gain is actually referred to as Drive as it allows for creative use of the valve distortion characteristic if new tape storage architecture Linear Tape Open (LTO). The medium is designed for backup and archiving applications in the mid-range and network server sectors, and can store up to 200Gb of data at a compression ratio of 2.1. The first-generation Ultrium drives allow data-transfer rates of between 20Mb and 40Mb per second for compressed data. Ultrium is one of two tape formats used in LTO storage technology. The open standard is designed to replace Quantum's DLT system and is available to interested licensees. The new LTO system is an open standard and precise format specifications and compulsory compatibility tests will ensure that a single cartridge can be used in drives manufactured by various producers.

EMTEC, Germany. Tel: +49 621 59 20 406.

Hard sync
TimeLock Pro is a dedicated hardware synchroniser for Steinberg's Nuendo DAW. It acts as a wordclock interface and reads VITC and LTC time code, and generates MTC and a phase-locked, low-jitter wordclock reference. The interface supports LTC, VITC, LTC-VITC audio switching, LTC+video sync and video sync only. Reverse time code and time-code loops are translated into MTC-MMC locate commands and MTC slaves cue up during jog shuttle.

Steinberg, Germany. Tel: +49 45 2 10 330.
The valve thing has settled down now, with its advantages in certain areas generally accepted expected psychoacoustic lines, with the second being a more pronounced version of the first. The sample rate can be +1.1kHz, -48kHz, or double these, and this is where a tri-colour LED shows which is chosen. The only disadvantage of the indication method is the necessity for very small type on the front panel to show what the various colours mean. The output format can be SPDIF or AES-EBU, and the chosen format appears on both output connectors—it can’t do both simultaneously. There are word clock in and out BNCs on the back, and the sample rate LED shows what frequency it’s locked to.

The conversion uses dbx’s proprietary Type-IV processing to maximise the use of the converters’ dynamic range. In essence this is gentle soft-knee compression of the top few dB of the analogue input signal, but the fact that half the manual is devoted to a white paper on it suggests it’s not quite that simple.

The preamps themselves are good examples of modern valve circuits—very clean, very quiet and smoothly musical. Coupled with dbx’s now-established digital expertise this makes the 366 a powerful and useful combination of technologies, with the flexibility to slot into a wide range of applications.

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Studio Sound www.prostudio.com/studiosound

June 2000 35
Stage Tec Reference Mastering A-D

High end, high quality, Dave Foister has high hopes for Stage Tec's flagship conversion package

Whatever black arts apply in the design of some types of equipment, there will always remain areas where good science and engineering are the only things that will get you what you want. Perhaps the best example is conversion between analogue and digital domains, where the technology belongs in a laboratory and there is no room for fairy dust. This can make convertors quite dull objects, but no doubt the designers would take that as a compliment.

But the Stage Tec Reference Mastering A-D manages to rise above the obvious, principally because of its scope. Here we have a big modular rack system that when full provides 24 channels of A-D conversion, and when fitted with the appropriate options has microphone preamps on the front of all 24. This gives it a hugely impressive front panel, with rugged practical and cosmetic touches and a carefully designed, simple control interface. The review rack was full of the complete modules, giving me 24 mic amps delivering their signals solely on 12 AES-EBU outputs.

The complement of controls is surprisingly small: one rotary encoder on each channel, turning out to be a push-button as well for function selection. Along with it in a small panel for each channel is an alphanumeric display and an LED meter; no ordinate meter this, but an arc-shaped array of small lights offering unusually high precision. If the input modules have only line inputs, there is nothing to adjust, with this meter showing the level present in the channel. With the more elaborate microphone modules, there are the expected functions to adjust individually on each.

Although each channel has its own control, the entire front panel is globally switched to each function, which makes for much greater ease of use than having them all separate. The neat trick is that the current function is selected from any one of the 24 controls: pushing any knob will cycle through the parameters after which any control is live to adjust that parameter for its channel. In line with this, all the associated displays show the currently selected parameter value for their channels.

Gain is digitally adjusted in 1dB increments that are entirely click free, with a maximum gain of 70dB. The LF filter is labelled eqi, which suggests more than it actually does, in fact it offers a useful selection of cut-off frequencies up to 80Hz, but no equalisation beyond that. Phantom power and phase are simple toggles.

That's it for the preamp controls, and, as it turns out, for any controls at all. The very high-end specification of the analogue path is matched by an advanced conversion stage using Stage Tec's proprietary True-Match sigma-delta system. The specs claim 28-bit resolution at 64x oversampling, which will no doubt raise a few eyebrows, this signal can be transmitted intact via AES-EBU or digitised to 24 bits on a channel by channel basis. After that, it's up to you what you do with it, as there is no facility for reducing the wordlength below 24 bits, elegantly or otherwise.

Conversely, and surprisingly in this day and age, the sample rates only cover the range from 32kHz to 48kHz. Perhaps current mastering requirements are met by this selection, but I doubt if that will be the case for long. I was also left scratching my head for quite some time as to how to change the sample rate. The unit fixed up at 48kHz on its internal clock, and my initial impression that there was no onboard way of changing this turned out to be correct. In fact the unit locks to incoming clocks on its rear BNC, and when this is removed its internal clock stays locked to that frequency until something else gets connected. House sync would mean this would not be an issue.

Output formats are determined by the cards fitted, and encompass AES-EBU on XLRs and SPDIF on either phonos or optical terminals. A fourth option gives eight channels on a 15-pin D, but there’s no facility for delivering ADAT or TDF signals.

On the same card as the BNC word clock in and out is a 9-pin D for hooking up an internal PC. This gives almost no extra facilities, other than a more comprehensive graphic display of system status and control of everything from the computer; there are however one or two extra bits like an on-board analogue signal generator that can only be run from the PC. To go back to the beginning, the Stage Tec system looks, behaves and is specified like a laboratory instrument with attitude. There can’t be many more elegant ways of getting microphone signals on to a 24-bit recording medium or into a digital mixer, although there are few systems that can interface directly without adding optional cards. This is a small additional price to pay for access to this much class.

Stage Tec, Valentinerstrasse 43, DE-96103 Hallstadt, Germany. Tel: +49 951 97225 15, Fax: +49 951 97225 32.

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Beyerdynamic DT150

Attempting to improve on a well-established and popular headphone model is a tall order. Zenon Schoepe admits it has come to pass.

F
ew items of audio equipment can make a stronger claim to the tag of industry standard than Beyerdynamic's long-lived DT100 headphone. The appeal can be attributed to a peculiar combination of durability, dependability, serviceability and flexibility, although the sonic performance, while solid and consistent, has never been the major selling point. People love DT100s because they immediately know where they are with them, because they know they can fix them because they were made to be fixed, because they come in a wide variety of configurations that include mic boom options and because no other headphone can be clamped as only a DT100 can clump, to the head of a drummer and feed him all he wants before his eyebrows explode. I still remember an instance where an engineer was holding a pair (DT100s not drummer's eyebrows) when the playback was rolled and he dropped them in fright as they sprang in to life but stuff in all SPL conscious days, but then the customer is always right.

You'll see the unmistakable items on the heads of location recordists - on FOH desks, draped on the arms of chairs in film mix theatres, instructing camera men on the studio floor and in the field, on the table in the voice booth and somewhere in every studio you visit.

These enduring qualities have made the DT100s a strong seller for decades but it has been a little known fact that Beyerdynamic addressed the model's single weakness of lackluster sonic performance with the introduction of the DT150 some time ago. Here is a headphone that carries over the original's features and appeal, but better it and adds a far more distinguished reproduction circuit.

The range encompasses the standard DT150 headphone with microphone cable in straight or curly leads, DT180/DT190 with boom DM190 dynamic hypercardioid; DT181/DT191 with Em191 electro condenser omni (can be used for wireless applications); and the DT182/DT192 that can also be run wirelessly, but with the EM192 electro condenser for high noise environments. The microphone equipped models can additionally be supplied as V.11 version with integrated equaliser/matching preamplifiers for use with TV cameras.

Visually the thing that differentiates the DT150 is the use of dark grey plastic on the ear piece outlets - the moulds and other components look identical, although the newer model does have a less bulky headband cushion. The feel is identical, but the specs start to tell a slightly different story with a claimed frequency response stated as 5Hz-30kHz compared to the DT100's 50Hz-20kHz with nominal SPL of 97dB compared to 96dB on the older design. Impedance is 250Ω compared to the DT100's 400Ω.

The reality is that the DT150 is a far more pleasant listening experience with a marked extended bass response and a smoothness to the remainder of the range with none of the sibilance, and, if we're honest about it, lumpiness of the DT100 which while it adds intelligibility doesn't amount to a discerning occasion. It's an uplifting performance that is a light year and several decades removed from that of the original. A winner and better all round performer.

I do prefer to have the cord running to the left lug-hole for historical reasons rather than Beyer's arrangement, however the signal comes in on a cable of heavy-duty guitar lead proportions and one of the advantages of the multipin connector is that if you do inadvertently step on the lead while you stand up then the whole assembly pulls out quite cleanly without ripping the ear-piece off of your ear or bending the connector if they haven't employed a grub-screw permanent fix. Reconnection in these circumstances is fast and positive.

I've never been a fan of the vice-like grip that the venerable DT100 exerts on the cranium, but I've now ascertained that the reason for this is that I have never actually managed to wear a set for any length of time - it's always been a case of being helped in to a pair, realising that they still sound a tad basic and pulling them off before the pressure builds up and the nose starts to run. Because the DT150 sounds so much better I persisted and found that the clamping actually doesn't feel that bad for extended periods thanks mostly to the plushness of the ear cushions.

There is an involving and engrossing quality to these headphones that open lock models never achieve and, while they do get hot, isolation is exceptional which makes them ideal for voice-overs and vocals. They remain the definitive article for applications where operators need a high degree of coothing from their surroundings. However, positioning is critical as the grip is strong enough to pull up the eye brows in a permanently surprised look or serve as a cheaper alternative to face-lift.

So there you have it, an industry standard has been improved on in every respect. Beyerdynamic continues to manufacture the DT100 for all the traditionalists, but for the record, and for their information, a better one does exist. I'm converted. I just hope that nobody thinks I'm using DT100s. They've taken ten years off me.

Contact

Beyerdynamic, Germany
Tel: +49 71 31 6170
UK: Tel: +44 1444 258258
US: Tel: +1 516 293 3200

Schoeps tubes

Schoeps is now offering its Colette tubes as a mechanical accessory for the CCM compact condenser microphone series. With the Colette tube an exchangeable microphone capsule was screwed onto the front end whereas now a CCM microphone takes its place. At the lower end there only a miniature Lemo connector is left instead of the CCM microphone amplifier that has to be screwed on here with the Colette version.

There is also a version available that carries two microphones one above the other for MS recordings. The VMS 5/4 high-mic preamp with MS matrix is a successor to the VMS 02 IB and VMS 52 UB and features phantom powering and an input impedance of 10kΩ.

Schoeps, Germany, Tel:+49 721 943 200.

Tascam by TimeLine sync

TL-Sync is a stand-alone synchroniser for use with Tascam digital recorders.
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Joel Jaffe is an award winning Engineer/Producer/Composer and owner of Studio D Recording, n.c., home to a long list of platinum and Grammy Award winning albums and artists. Currently, Joel is working on DVD surround mixes for some of the industry’s top touring acts. LSR surround systems are his choice for stereo and 5.1 channel multimedia projects.

“The THX Approved 5.1 JBL LSR28P with the JBL LSR12P subwoofer provide an extremely linear response, great transients and full-frequency monitoring in a near-field set up. In addition, the LSR speakers allow us to go between stereo mixing and multi-speaker formats, which is absolutely necessary today in a state-of-the-art studio.”

ALL NEW

LSR 25P

The Only Workstation Monitor Good Enough To Be Called LSR.
AKG C2000B

The latest variant on AKG's landmark SolidTube-fronted range is the C2000 B. Dave Foister finds it suits all tastes.

All the big microphone people have had to respond quickly and positively to the influx of cheaper alternatives from various points around the globe. They've all done a good job, maintaining the status of their flagship models while bringing us scaled-down versions that compete well with the alternatives. Fundamentally the attributes that put them where they are are rarely compromised, making sure that when AKG brings us an entry-level version of an established model, it will be well-built and it will behave itself.

The C2000, with its obvious links to the 414, because quite a bit when launched for these very reasons. Now there's a new AKG style, begun by the SolidTube, and the new C2000B follows those lines. Just over a year ago I looked at the SolidTube's first relation, the C400B, and as the name would suggest the 2000 is in many respects a scaled down 400.

The first and most obvious respect is its size. The curved cylindrical shape is very similar, with its protruding connector shaft that mounts the microphone in its holder, and the champagne gold-colour and close mesh grille are the same, but the whole thing is about half the size. Where the SolidTube and C400 appear to be designed to catch the eye and make a statement, the 2000 tends itself more to general use in the thick of it, allowing easier access and reduced visibility.

Its flexibility is also scaled down in that it is a single-diaphragm cardioid microphone where the 4000 has twin diaphragms and three polar patterns. It is not totally devoid of controls, however: there is both a 10dB pad and a low cut filter. The filter is very unusual: where most turn over quite low and drop off steeply, this is 6dB per octave below 500Hz. This may in fact be better suited to dealing with proximity effect than the more familiar characteristic, although for removing rumble it may be a bit intrusive on the wanted spectrum.

Although it is small and relatively light, it is no lightweight in terms of its construction. The body material is ambiguous to the touch, and could almost be plastic, in fact it is all die-cast metal to ensure adequate screening as well as robustness. The diaphragm is plastic foil, gold sputtered on the front only to reduce the risk of internal shorts to the back electrode when subjected to very high SPLs. It claims to be able to handle 150dB with the pad in, almost dynamic territory. The capsule is shock mounted internally, and there is a built-in pop screen in addition to the optional foam wind-shield, which is the same as the one supplied with the C4000B. The accompanying literature makes it clear that AKG see a role for the C2000B on stage as well as in the studio.

The stand mount is also the same as that supplied with the C400B, and as I recall thinking that was a little over the top, it's certainly more than adequate for the 2000. It is a full-blown suspension mount, with a clever rotating clamp that locks around the connector shaft at the microphone's base and engages with a small lip for added safety. A slight twisting of the clamp allows the microphone to be rotated, and the vertical position is locked with a typical AKG handle. So big is the mount that this little microphone almost disappears when resting in it, and in fact it becomes a bit awkward to get at the switches.

This is more than compensated for by its manœuvrability, poking a 4000 in amongst a section is tricky, but the 2000 is much easier even with its huge mount. And this is the kind of situation on the microphone is going to be happiest in. It doesn't have that big fat present sound of the classic vocal mic, but it lacks nothing in clarity and ability to pick up the real sound of the instrument. At the same time there's nothing small or thin about the sound, which in many ways is closer to that of a big diaphragm than a small one—it looks to be around 1/4 of an inch. It's an all-rounder in the best sense of the word, not the one that's left over when you've stuck your favourites up on the important things, but the one you know you can use anywhere with confidence. It's clean, quiet and natural, and its SPL and transient handling capabilities were well demonstrated on a percussion session.

RTW's 5.1 metering

RTW's Surround Monitor 1080 is designed for use in surround sound and multichannel recording, post-production and mastering. Based on a dual-screen design, with one screen being used as an 8-channel peak programme meter and the other as a displays. The 1080 is fully equipped with analogue and digital inputs and is able to display different domains on one screen.

The eight PPMs are multi-standard types that can also calculate and display levels derived from the LCR/SR signals, as well as loudness. RTW claims that the audio vector-scope offers selectable grids for different surround modes such as 2.0, 3.1, 3.2 (5.1) and four phase meters for the LC, CR, LR and SR channel pairs. These additional phase meters are said to be especially useful during surround sound recording or monitoring to check the setting of multi-microphone arrays or to check phase relations between the channels. Additional features of the 1080 are a real-time third-band filter.
De-everything
The world's most effective range of tools for noise removal and audio restoration
Clean it up
Away from the entertaining aspects of workstations is a host of equally essential functions. Rob James checks the list.

**Soundscape EDL v2**

Just as there are glamorous pieces of hardware and less exciting but essential boxes, so it is with software. In the arena of sound-for-picture, no matter how well a system performs in the headline tasks of recording, editing and mixing, it will be useless for many applications if it cannot perform certain fundamental tasks. Some of these are immediately obvious such as synchronization to picture and the requirement for a variety of frame rates, others may be less so. Autoconforming is just such a function.

The need for autoconforming arises from the frequent use of NVL (nonlinear video) editing systems with `guide' audio. Autoconforming, then, is taken to mean the entire process of semi-automated re-recording of the source audio into a workstation and arranging the resulting cues on the correct tracks and timecode positions within the picture editor.

Soundscape has carved a secure niche in sound-for-picture and have offered an autoconforming package for some years.

There are several ways of presenting autoconforming to the operator. Some manufacturers integrate it into the main user interface. Others use third-party packages for the job. Soundscape has chosen to separate autoconforming into a stand-alone software application that addresses its standard hardware. The workstation interface does not become more cluttered and fewer compromises are necessary.

At the simple level autoconform can convert an imported EDL into the native EDI format of the workstation and control the source machine to fill in the blanks. The essential information contained within an EDL is source and destination in and out times, source and destination tracks and the source reel number.

Machine control may be simple time-code chase or 9-pin 185422 serial. This works well enough with straightforward projects where the source material has nice tidy continuous code. The fun begins where there is a requirement to merge lists or the source code is discontinuous. An alternative is to simply play entire source reels feeding time code into the EDL processor which records only the required sections.

Soundscape has obviously learned a great deal from experience with the first version of the Soundscape system and the new package contains a wealth of features which will save a great deal of time and trouble to those who need them. Rushes recording is now supported—including automatic recording and identification of audio clips with broken code or time-of-day. Also, handles (extensions of audio beyond what the EDL specifies) and crossfades can be independently set for each reel. Multiple EDLs and Audio Projects can be contained in one EDL processor project which allows both guide and source audio to be conformed. The Scene feature enables a project to be split into several Soundscape arrangements to allow several workstations to work on parts of the project. Conforming may be undertaken from existing sound files even across a network. Additionally, I would have liked to be able to use reel order lists and to be able to directly edit EDLs but neither of these are essential.

EDL source files must be a `A' mode—that is, in finished product order. As the processor loads the EDL it sorts into source reel order. The original EDL is displayed on the right of the main screen. The left-hand side shows the EDL audio sources (reels) and or source files. Clicking on a source reel brings up a list of Takes to Record, in blue from the selected reel. Entries with errors are shown in red. Tracks and source reel mapping allows for lists containing mono events to be conformed to stereo or for where the source reel numbers have changed. Video edits can be converted into markers for swift location of video events. As recording proceeds, takes will disappear from the Takes to Record list to be replaced by entries in green on the Recorded Takes list. When all the material has been recorded the final stage is to create the Soundscape arrangement. This will usually be created from `source reels' however it is also possible to create from `destination reels' where the audio has been recorded from the NVL workstation. Manufacturers are always pulled in certain directions in part by client feedback, but also by alliances formed with manufacturers of other kit. To a casual observer of the NVL market, a couple of names may seem all pervasive, but there are several other serious players. So I am delighted to find support for Tektronix Lightworks and Discreet Logic/IDMX Vision formats in addition to the standard variations on CMX, GVG, Sony, Avid and OMF (OMF1 composition import/export and audio import is also available for the Soundscape editor.)

With v2, Soundscape has rounded out the autoconform capabilities of its system to rival anything on offer from other manufacturers. All the expected functionality is present together with the added helping of useful extras. Autoconforming may not be the most exciting of applications but this example makes the process as painless as possible if not actually pleasurable.

**NEW TECHNOLOGIES**

**Eela Audio’s Anniversary**

Eela Audio, celebrating its 25th anniversary this year, has the IDMX Integrated Digital Mixer and Xpoint system components. The system offers integration of audio, communication and machine control. The mixer can easily be configured for a broad variety of applications. The IDMX system components comprise the On Air console, the ICom panel. The Producer panel and the Presenter panel. Targeted at broadcast journalists and producers, Eela’s logos is a flexible audio matrix with incorporated matrix functionality. It provides extensive control of source machines and DAW editing software. More complex facilities such as telephone hybrids and ISDN codecs can be addressed from a phonebook or manually by the dial keypad. Via the Logos Window’s Menu, complicated settings can be pre-configured. Eela reports interest in its Reportable 530 DSP PCMIA recorder for ENG applications due to reduced prices for the flashcards and file compatibility with PC automation systems and less maintenance than tape media according to the company.

**Eela, The Netherlands.**

Tel: +31 40 251 0484.

**CR-10S update**

The popular BPM CR-10 condenser has been updated to CR-10S status with the addition of switches for Pad and Bass roll-off. The 1-inch gold sputtered 2m diaphragm mic comes with a suspension, cable, windscreen and aluminum flight case for a retail price of US$299.

**Net:** www.bpm-studiospeichark.de

**SPL launches channel strip**

The new SPL Channel One is a complete channel strip based largely on processing elements already developed and proven by SPL within its existing range of separate processors. It incorporates a tube mic line preamplifier, de-eisser, compressor-limiter based on SPL’s Double-VA-Drive circuitry, noise gate and 3-band EQ section. The incorporated headphone monitor can be used to pass an individual mix directly to the singer-speaker without latency. To enable immediate digital recording, a 24-96 A-D-D-A converter can be provided optionally. Further options are sound finishing Lundahl transformers for the inputs and outputs.

**SPL, Germany.** Tel: +49 21 63 8761.

**www.americanradiohistory.com**

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**June 2000**

www.prostudio.com/studiosound

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**Analogue and the AES/EUR in status monitor**

The latter is a helpful tool when digital interface problems occur. It allows checking of all bits and bytes of channel, user and audio data. The status monitor detects the real word length of faulty bits on a digital signal.
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No endorsements implied
Mackie HR824

Studio Sound's 'bench test' loudspeaker reviews continue with the HR824. Keith Holland reports

The Mackie HR824 is a 2-way, active loudspeaker comprising a 220mm mineral-filled polypropylene cone woofer and a 25mm aluminium dome tweeter mounted on a shallow, die-cast axisymmetric horn. The electronic crossover, protection circuitry and power amplifiers are all built-in. The cabinet has overall dimensions of 254mm wide by 400mm high by 310mm deep, and is fitted with a large, elliptical, rear-firing passive radiator with an aluminium honeycomb piston. The front edges of the cabinet are finished with a radius to reduce diffraction effects. Mackie specify the power amplifier ratings as 150W for the woofer and 100W for the tweeter giving a maximum peak SPL of 120dB and a maximum short-term SPL of 111dB at 1m for a loudspeaker pair. The crossover is described as a modified Linkwitz-Riley having 4th-order slopes and a crossover frequency of 2kHz. The rear panel has controls for input sensitivity, acoustic space, low-frequency cut-off and high-frequency shelving. The input sensitivity is continuously variable, the acoustic space control is switchable between quarter-, half- and whole-space giving -4dB, -2dB and 0dB low-frequency shelving respectively, the low-frequency cut-off is switchable between 80Hz, 47Hz and 37Hz and the high-frequency shelving is switchable between -2dB, 0dB and +2dB. This review was conducted with the switches set to whole-space, 37Hz and 0dB. The mains and input signal sockets exit downwards at the bottom of the rear panel allowing the loudspeaker to be placed with its back against a wall if desired. The cabinet assembly has an overall weight of 15.2kg.

Fig.1 shows the on-axis frequency response and harmonic distortion for the Mackie HR824. The response is kept within ±2.5dB from 38Hz to 20kHz, a remarkable result, and among the best of the loudspeakers tested to date. The rapid low-frequency roll-off is 6th-order with -10dB at about 32Hz which is very low for a loudspeaker of this size. Harmonic distortion is also very low, having a maximum in-band level of -46dB (0.9%) at 50Hz. There is, however, evidence of a sharp peak in distortion at 300Hz indicative of a rattle. The horizontal and vertical off-axis responses are shown in Fig.5 and Fig.6 respectively. The horizontal directivity exhibits a gradual narrowing with increasing frequency from about 500Hz upwards demonstrating good matching between the drivers. The interference notch at the crossover frequency of 2kHz, due to the spacing of the drivers, can clearly be seen in the vertical off-axis results. Time domain performance is presented in Fig.1, Fig.2, Fig.7 and Fig.6 which show the acoustic source position, power cepstrum, step response (Fig.5) and waterfall plot respectively. The movement of the acoustic source position to about 3m behind the loudspeaker at low frequencies is due to the 6th-order roll-off, and is typical for loudspeakers of this type. The power cepstrum plot (Fig.4) shows very little evidence of reflections indicating good control of diffraction and horn termination. The step response is fairly accurate with the high frequencies leading the mid frequencies by about 0.5ms; the woofer and tweeter are in phase with each other. The waterfall plot (Fig.7) shows little evidence of ringing, but the effect of the rapid low-frequency cut-off is clear. Overall, the Mackie HR824 is a remarkable loudspeaker. The measured performance is the equal of any loudspeaker tested to date and it demonstrates no particular weaknesses. The on-axis frequency response is commendably flat, the off-axis response is very well controlled and the harmonic distortion is very low. The rapid low-frequency roll-off may offer some cause for concern, but this is perhaps a worthwhile compromise given the impressive low frequency performance for a loudspeaker of this size. The provision of acoustic space selection means that this loudspeaker should work well under a wide variety of acoustic conditions.
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Producer Gary Katz worked on all seven Steely Dan albums but these days he’s looking out for the hottest hip hop, as he explains to Simon Croft.

The phone rang. 'You like Steely Dan, don’t you? Oh, the power of the rhetorical question. Like them? Like them! No, I bought all those albums to hide an embarrassing gap in my record collection somewhere between the letters R and T. Actually, all I said was “Yes, why” and so I found myself heading for north London to meet producer Gary Katz.

Since the last Steely Dan album was recorded in 1980, I was curious to know what Katz had been doing in the last two decades. His later credits include album tracks with Diana Ross, Civilised Man with Joe Cocker, Zazu with Rosie Vela and Laura Nyro’s final album Walk the Dog and Light the Night—as well as numerous ‘Dan-related’ projects, including Donald Fagen’s The Nightfly album. Among more recent Katz credits were the 1996 Eggs & Arms album with Lole—otherwise known as Inara George, daughter of Little Feat’s late Lowell. Katz work of late has also included artists I hadn’t come across; Raw Stylus (1995), Repercussions (1995) and Digital Underground (1998).

As I was soon to find out, Katz is a man of quite eclectic tastes.

Although the early Steely Dan albums were recorded in California, Katz is actually from New York, where he still lives. It was a fortuitous start for a would-be producer. I had some friends called Jay and the Americans when I was growing up and they were having a lot of hits in the States,” he recalls. ‘They were in the studio. I was about 18. I was always listening to music, it was a passion of mine. So when they went to the studio I went along with them.

‘They were being produced by Leiber & Stoller, who were my favourite producers at the time and made all my favourite records with The Drifters and The Coasters. I watched them produce and I saw that one of them was really good and musically trained. And one of them really knew nothing about music—and basically worked with his ears. I said, “I can do that”. That’s really what kicked me off,” says Katz.

Luckily for the aspiring Katz, Leiber & Stoller were happy enough to have him around. ‘They befriended me and let me come to the studio to watch them work,’ Katz acknowledges. ‘That was a great thrill.

Music is clearly very important to Katz and he later describes himself as Steely Dan’s ‘biggest fan’. He certainly shows little interest in the technicalities of recording. ‘I’m not an engineer-producer. I don’t want to be involved with that,’ he affirms. ‘I work with people who are really good and I try to concentrate on the music with the artist. I try and bring the most out of them performance-wise.

‘Because most of the artists I work with write their own material, I know a good deal about that insulated feeling. I can take a step back and have my suggestions, help them to structure the tunes and realise their full potential.

But isn’t he to some extent a project manager, contracted by the record company to deliver on budget? ‘I don’t look at it that way,’ Katz responds. ‘No—I’m working for the artist, always. I never consider myself as working for the company. I mean, it’s not like I don’t have a sense of responsibility with regard to budgets and whatever, but in the actual process, I’m always working for the artist.’

Although Katz feels that this philosophy shouldn’t create ‘any sense of contradiction’ for the record company, he becomes uncharacteristically animated if you ask him to what extent to accommodate the artist’s way of working.

‘You’re at the mercy of the artist,’ says Katz. ‘What are you going to do, say to the artist, “What do you mean you don’t want to sing at 2 o’clock? We’re gonna sing at 2 o’clock.” So you work at the mercy of the artist’s likes and dislikes.’

For the interview, I brought along eight Steely Dan albums, which included one I doubted Katz had even seen before. Entitled Sun Mountain, it features Donald Fagen and Walter Becker ploughing through some of their early songs. No guitar solos, no brass, no drums to speak of—indeed I’d hesitate to say it was recorded in a studio as such. God knows how it ever got released. It certainly doesn’t do much for the duo’s reputation as perfectionists.

www.prostudio.com/studiosound Studio Sound
There are definitely demons," says Katz. "Some of these are songs that never appeared. 'Android Warehouse', 'Sun Mountain' and 'Ida Lee' are tunes that we had worked with prior to it being Steely Dan. They were just two guys writing songs — there was no Steely Dan.

The album also appears to have been packaged by someone who has no idea what they have on their hands. Hence 'Hurry Town' is rendered as 'Berry Town', although there are some startling contrasts between this and the version that ended up on the third Dan album, Pretzel Logic.

'There's no relationship whatsoever,' opines Katz on the two versions. 'When we went to the studio, the song was in the form it was going to be and there was never any reference to the one that had been done previously.'

It is instructive to discover that Sun Mountain also includes 'Caves of Altamarea', a song which was not to surface until the sixth album, The Royal Scam. So the team were obviously not short of material in the early days — but how prepared were they when they came into the studio?

'Donald and Walter came to the studio with what they thought were finished arrangements,' Katz attests. 'I don't know if I can count on my hand the number of times that they got changed once we got to the studio, so far as the structure goes. As a matter of fact, I can't think of one. They were very prepared.'

Katz met Fagen and Becker through a mutual friend and was struck by the duo's talent. Although they were all living in New York, Katz secured a job as producer for ABC in California. Once he had his feet under the desk, he soon put in a word for his friends.

'I told them that I was working with some people in New York that I thought were just terrific and please let me bring them out. You know, unheard,' Katz recalls. 'They had a lot of faith in me and they let me bring them out, and they reaped the rewards.'

At first, the pair were working purely as songwriters, but Katz saw more potential for them. They were writing songs for a little bit for other artists and we realised that was a waste of their talent. So we decided to put a band together. The drummer was in a band that I had produced earlier and Jeff Baxter was a guitar player that I had worked with earlier. They knew Denny Dias, so we flew the three of them out.

The first fruit of their labours was the seminal Can't Buy a Thrill, which was low-key radio stations and listening public on both sides of the Atlantic when it was released in 1972. Although its success would be eclipsed by later Steely Dan albums, its popularity caught almost everyone by surprise. "There wasn't anything like that music," Katz explains. "Therefore there was nothing against which they could judge its potential popularity, apart from Katz' gut instinct. 'We didn't expect it to be popular, we just liked it. Long solos, long songs, sophisticated chords... we made the record and liked it a lot, but no-one thought it would be popular. Except for me.'

I always thought it was great. I couldn't think how anyone wouldn't like it. But Donald and Walter didn't think it was commercial and it wasn't by virtue of what was on the radio in those days. So I got proven to be more right than them, which wasn't often believe me!

While Steely Dan arrangements were sophisticated and the performances tight, the sound from the album's drum kit was obvious. "While the studio was the best, they had to change some of the stranger and more memorable guitar solos. Walter was the bass player and he's a great bass player and very facile with it," Katz explains. 'He wasn't as facile at that time, playing guitar, as he was when he played. Donald and I were just huge fans of Walter and we were willing to take the time that it took Walter to accomplish what he wanted to.

By the time album four, Katy Lied came round, Steely Dan's musical pool had grown to more than 16 musicians, who were carefully selected for their suitability in certain contexts. For instance, Becker's contributions on bass were supplemented by Chuck Rainey and Walter Felder, while the roster of guitarists now included session legend Larry Carlton and one-time Johnny Winter sideman Rick Derringer. The emergence of the guest artists we loved alone would account for the increasingly slick, jazzy sound Steely Dan were making. The sleeve notes to Katy Lied also reveal a singular approach to the recording process. 'This is a high fidelity recording. Steely Dan uses a specially constructed 24-channel tape recorder, a state of the art which was not computarised mixdown console and some very expensive German microphones. Individual equalisation is frowned upon. The sound created by musicians and singers is reproduced as faithfully as possible... This little treatise goes on to wax lyrical about the Neumann>
One of the benefits Katz will acknowledge is freedom from gradual deterioration of the multitrack as successive passes of the tape wear off the oxide, but it’s not a debate he cares to enter too deeply. ‘It’s just a matter of taste,’ he says. ‘I never have listened to music in those terms. Maybe because I’m not technically inclined. If I can tap my foot, I’m liking it.’

In order to increase the continuity in sound from track to track, Gaucho also saw the use of triggered drum sounds, a real innovation at the time.

‘Roger Nichols built a piece of equipment called the Wendel,’ Katz recalls. ‘And it was the first high-tech, fast trigger music machine to work with drum tracks.’ By the mid-eighties, replacing the ‘live’ drums with samples was epidemic in modern recording and bands would spend hours agonising over snare sounds. Or you could use a sequencer and cut out the middle man altogether. How did Katz feel about these techniques?

‘I personally like live drums, but that’s not to say that the technology these days has gotten to the point where you can’t create good drum tracks without them. I still prefer in fashion to use real drums, even if you record a drum track, capture two bars and loop them, or whatever. But the actual feel would be live.’

Katz carries a complex set of preferences when it comes to recording. He prefers live drumming and prefers to capture the percussion hand in one room, even if he subsequently only uses the drum tracks. Guitarists, he prefers in the control room where he can talk to them. Sequencers have a place he acknowledges, but he’s not the kind of producer who is going to cut up someone’s performance in Pro Tools. If he has to pick favourite equipment and studios, it’s Neve desks and Bearsville New York, for the quality of the room.

So, Steely Dan apart, who are the artists he admires? ‘Van Morrison. I just love his music,’ offers Katz. ‘I’ve been in the studio with him and I love him and the band. I guess I like more of the old stuff than the newer stuff these days. I used to love Police records. They had a good drummer. It helps a lot. I’m not a particular fan of Sting records, but I love Police records. Jimi Hendrix, I listen to Jimi Hendrix still.

‘Today? Do I like anything today? Katz muses. ‘I like hip hop mainly today. I like Dr Dre records and I like Snoop records—West Coast hip hop I like a lot, it’s my favourite.’

There is an unwritten, but generally understood, rule that it is bad manners for an interviewer to adopt a look of astonishment, or repeat odd parts of the interviewee’s last sentence in an incredulous voice. Too late. ‘Hip hop?’

‘I know, it’s the antithesis of what I used to do, but I like it,’ Katz responds evenly. Digital Underground, with whom Katz recorded the Who Got The Gravy album in 1998 are indeed a rap act, I am informed.

Katz says he would also like to work with Snoop Dog or Dr Dre: ‘I like what they wind up with a lot. I like the grooves and I think it’s the only form of music these days where you are listening to things that are inventive. Not all the time, of course, but when it’s good it’s good.’

‘I’m a listener like everyone else and when you hear a record that works well, you like it,’ Katz explains. ‘There is another role Katz plays as well—talent scout. ‘It’s part of the process. I find artists through word of mouth or people find me. But I look for artists all the time,’ he affirms.

It turns out that Katz was running a record company for a while and one of the reasons he is in London is to check out new people he might be able to work with. ‘That’s what I do,’ says Katz. ‘I’m looking for artists. I’m looking for talent, which is what I really like to do: find talent and make records.’


But thinking about it, hip hop is American black music. Its lineage—however convoluted—can be traced back to those Drifters and Coasters which led the young Katz to a career in music, and to the R&B records which influenced Becker and Fagen. It’s just well, if you want to flip a disc onto the turntable and create a sound that goes something like, ‘RikkiRikkiRikki don’t lose that number’, please use someone else’s copy of Pretzel Logic. Mine’s still in use.
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Continuing from last month's oeuvre, Chris Woolf completes his study of digital radio microphone systems.

<1W~1.5W much of which goes to running the audio processing. The RF output of analogue radio microphones is rated at anywhere from 2mW~250mW. At the extreme lower end range is poor, at the higher end the problem of intermodulation between multiple channels becomes very serious and the range improvement is far from pro rata so most professional devices fall into the 30mW~50mW category. The power rating is rather nominal because absorption by the body is highly significant and increases with frequency. At 1800MHz there is no doubt that a great deal of the available power will disappear. Raising the output is to some extent self-defeating since the battery consumption increases and the intermodulation problems become more severe. Worries about excessive exposure to RF also mitigate against this policy. For most manufacturers improving reception, perhaps with increased diversity is more attractive.

Space diversity—spreading at least two aerials apart by >1/3 wavelength and using the strongest signal at any one moment—greatly reduces the fading caused by multipath signals which create deep nulls where the radio waves interfere destructively and hence improves usable range. Digital systems may ease their troubles by using multiple diversity techniques with three, four or more aerials to give good coverage and, with the extremely short wavelength 1800MHz band, to ameliorate the problems of severe shadowing. As a transmitter moves further from its receiver so the carrier-to-noise ratio becomes poorer until at <50dB (for analogue units) the noise performance of the channel starts to deteriorate. The practical range is usually governed by the point at which a multipath null is produced that is significantly worse than this—the familiar drop-out.

As with so many analogue systems, this degradation is relatively benign. A good many users will have allowed...
Although but problems of radio rowing broadcast sometimes less friendly, so European casting share there systems may 'mixed economy' fill centre band, but tends also means more requires or allows more RF schemes advantages of simple modulation and amplitude, sometimes take many forms mal grain for of put display of a nal to graceful.

Another aspect of this is the energy density of the signal which is just barely distinguishable. Of course a manufacturer could provide an output display of the error-correction activity that should be a clue to the proximity of disaster, but it might go against the grain for them to show how much repair work was going on during normal operation.

Another aspect of power is how the spectrum is used, digital modulation can take many forms—frequency, phase and amplitude, sometimes in combination and often in more than single steps. The advantages of simple modulation schemes are that they take relatively low RF power and are easy to detect. The disadvantage is that they cannot code very much data onto an RF carrier. Increasing the modulation complexity allows more bits to be coded but the signal becomes increasingly like an analogue one where several different levels or phases must be distinguished and this requires a higher data-to-noise ratio. This means more power or a shorter range.

The energy in these dense signals is also no longer concentrated in a narrow centre band, but tends to spread out and fill the entire allocated bandwidth. In a 'mixed economy' digital and analogue systems may make poor bedfellows and there is some resistance to letting them share RF spectrum. Digital Video Broadcasting—DVB—using COFDM (the European terrestrial system soon to be used in the US too) is proving to be even less friendly, so the accepted, even if sometimes unofficial, practice of borrowing broadcast TV channel space for radio microphones may become impossible as well as illegal.

So far we have concentrated on the problems of digital radio microphones, but there are undoubtedly advantages. Although FM signals are resistant to AM interference they still suffer when it is severe. Electrical switch 'splits' cause impulsive noise and hum fields from electrical equipment may cause audible effects. It should be possible to gain immunity with digital techniques, and, within the defined range, get greater reliability. The component faults that beset cheap analogue systems and occasionally trip up the better ones should disappear completely too. Overall quality could also improve. We have only discussed 16-bit systems since the data rates threaten to stretch current techniques to the limit. The practical dynamic range of these is only about 90dB (10dB lower than good analogue) but the limitation is not fundamental as it is with analogue. Given clever modulation schemes or improved bit-rate reduction a 20-bit system could be developed that would give a satisfactory mastering margin.

Many manufacturers have prototype digital radio microphones on their drafting screens and workshop benches. During this year we can expect to see a few being delivered for beta test even though the RF spectrum regulators have yet to sort out a long-term home for them. This indecision is certainly unhelpful to development. For users it is undoubtedly time to think very hard about how a digital radio microphone—which is unlikely to be as a simple, painless plug-in swap for its analogue brother—will affect the way that they work. They might use the short time available to evaluate and voice their needs clearly.

they're almost here
The Green Mile

Taking the hot seat, Richard Buskin talks with the production sound mixer, dialogue, sound effects rerecording and music rerecording mixers about their work on the Oscar-nominated story of Death Row

The TIME is 1935 and America is in the depths of the Great Depression. The place is Gold Mountain Penitentiary, a prison in the Deep South, and as for the minutes... Well, they’re on Death Row.

Superintendent Paul Edgecomb (Tom Hanks) is a decent, peaceable guy who takes a sympathetic approach towards his work. The governor’s nephew, on the other hand, is his polar opposite; a sadistic guard named Percy Wetmore (Doug Hutchison) who gets his kicks by watching the sentenced men being fried in the electric chair. Among the new arrivals in this cemetery-aways-from-home are both a white-trash murderer and a seven-foot black man named John Coffey (Michael Clarke Duncan), who has been charged with the rape and murder of two little girls. Sir Crazy this is not. When Coffey cures Edgecomb of a painful bladder infection by wringing it out with his hands and releasing the venom through his own mouth in a stream of mini-locusts, the kindly guard starts to perceive him in a different light. Indeed, the childlike hulk with the symbolic initials and miraculous powers just might not be guilty of the crime he’s been convicted of after all. So begins the real drama.

Based on a Stephen King novel, The Green Mile is written and directed by Frank Darabont who previously directed King’s The Shawshank Redemption. The title refers to the pristine green floors of the penitentiary itself, and it was this environment which presented the film’s dialogue mixer Robert Litt with something of a challenge when he had to determine whether or not reverberation should be applied to the characters’ voices.

“We were looking at a 20-reel picture and asking just how much a person could listen to in terms of reverberation,” says Litt who, along with his fellow dubbing crew members, had first befriended Frank Darabont when they all worked together on The Shawshank Redemption. “We wrestled with that problem for a while, and what we ended up using is a very soft, subtle reverb. If you were to listen to the picture—or a reel or a scene—without the reverb in there you would definitely miss it.

‘In a true jail cell you would have hard surfaces and plenty of reverb, whereas on a set it doesn’t really exist. You’ve got a ceiling that’s 40-feet high and so all of that sound gets sucked up. It felt very dry and not at all like the atmosphere which we were looking at, but then on the other hand there was a real fear that we would put on some reverb and, many reels down the line, we would screen the picture and say, ‘Oh my God, it’s too much.’ Then we would be trapped and faced with how to get rid of it. So, we actually did a couple of reels without Frank being very involved, and when we asked him how he felt about the reverb he said, ‘Well, it feels okay to me.’

Indeed, what is refreshing about The Green Mile’s Oscar nomination in the Best Sound category is the fact that this isn’t a special effects extravaganza that attempts to bombard the audience with dazzling sounds. Rather, it uses an atmospheric score, crisp dialogue and subtle background effects to support the story line.

Willie D Burton was the production sound mixer on the movie, recording the dialogue and any simultaneous effects, such as the sparks that were actually flying when scenes were being filmed in the execution chamber, the cracking of the guards’ shoes and leather belts, and the sound of the Tennessee winds. Burton was involved in all of the main location work that took place in Nashville over the course of three months, although not that in North Carolina which was utilised for Reels 1 and 20—the film’s beginning and ending—and which were shot a year later in just a few days.

‘I use a Foster T22 digital recorder,’ he explains, ‘and for the dialogue most of the time we try to mic overhead with conventional microphones. However, from time to time we do have to put the wireless radio mics on, so on this film I would say it was a combination of 85 to 90 per cent miking overhead and the rest of the time using wireless mics.'
postproduction

The biggest problem that we ran into on location in Nashville was the 30 to 40 miles-per-hour wind, and so in that situation we would have to decide on whether we were going to use wireless or the conventional microphones. Finally we went with the overheads because we were able to get them close to the camera shot, whereas with radio mics we would be able to bury them out of sight and get a good sound, but it would also be muffled.

Audio postproduction work on The Green Mile took place at the Warner Hollywood Studios in West Hollywood, using a 96-channel Harrison PPI analogue console equipped with Flying Faders, Fairlight to record sound effects and dialogue, and mag tape for the music department which elected to sit on the digital stage.

'The console itself is probably 12 years old,' says effects recording mixer Elliot Tyson, 'so I had to bring in three additional automated outboard consoles because I didn't have enough faders. They were actually Harrison PP's which had chopped up from bigger consoles, and so, thankfully, the automation matched. As a result I had 96 faders just to myself, with everything wrapped around me—in fact, the way the room was configured, there was a back bar behind us, and then elevated above that was where the director and producers sat, so I had to climb up on the back bar to get to the back console. Still, as it was all automated, once you moved the move hopefully you were kind of done with it.

In the light of the aforementioned in-house technology being a throwback to the time when a dialogue mixer didn't necessarily require more than 15 faders, and given the fact that these days said engineer would usually adhere to the adage that 'you can never have enough', both Robert Litt and music mixer Michael Herbick also had outboard consoles to help expedite their work.

'We all sat and worked together on this stage, while but sound designer-supervisor Mark Mangini tended to work closely with Elliot, I worked closer with Julie Evershade,' says Litt. She cut all of the dialogue and all of the ADR.

Meanwhile, receiving the music material on 24-bit Pro Tools, Michael Herbick was using anywhere from 14 to 28 channels on the Harrison console. To a director like Frank Darabont the little things mean a lot, and for the postproduction crew this translates into spending plenty of time making minute adjustments. In Herbick's case, this often amounted to playing down certain big orchestral pieces according to the director's wishes, yet the trick was to achieve this without losing any of the intended flavour. Sample the scene in which John Coffey is discovered with the two dead girls and people start rushing towards him.

'There's a big orchestral piece that swells and builds, and I wanted to go with that, but Frank also wanted to hear the agony of the father screaming, "I'll kill you" at the guy,' Herbick explains. 'So I couldn't go as big as I might have wanted to, but it worked out okay, because we filled in those gaps with the people yelling and screaming.'

For their part, as all of the predubs done by Bob Litt and Elliot Tyson were recorded to the Fairlight, they were working with a completely digital path. This included the sound effects which Mark Mangini brought in on Pro Tools, meaning that there was no mag tape involved in any of the sound effects or dialogue work.

'Mark went out and recorded specific backgrounds in prisons around Los Angeles, and then did some semi-mixing of those and brought me about 48 tracks of what we call 'air candy', recalls Tyson. 'Ninety per cent of the film takes place on The Green Mile, and the film-makers didn't want to have the standard off-stage "wallab" of other prisoners. They wanted it to be fairly confined even though it wasn't that far away, and so, at least as far as backgrounds go, we had to make the soundtrack be more than just air sitting there. It couldn't be dull. There were air tracks that had pipes breathing and the corrugated roof breathing—metallic things that came and went—and even though you might not notice them there were also voices, way off in the background. All of these sounds provided texture, like a little pepper sprinkled into the air so that the whole thing wouldn't be too monotonous.

'Mark and I were picking out things of these air tracks and going, 'That voice is great, but maybe it's in the wrong spot. Let's leave this one out and move it down about 20 feet into that little hole where it will work better'. We were doing that all day long, even during the final mix, and we'd sometimes find things that we hadn't seen in the picture. For instance, the exterior picnic scene featuring everybody—I'd just watched this like a thousand times, and suddenly I said, "Hey Mark, there are two flies that go by. I've never seen them until just now". So, we put in these two ever-so-subtle flies buzzes, and we did a lot of stuff like that which doesn't go over the top, it's right on the edge and makes it real. Of course, there's always the temptation to get too busy, but that's exactly what Mark and I don't want to do. We want the film to speak for itself instead of trying to "Watch me, listen to me, listen to me". There are a lot of other pictures that do that.'

Of course, one way of unabashedly attracting attention is by way of the surrounding channels, with supplementary background voices, noises and musical pieces cropping up from behind and, in some cases, panning all over the place. However, as noted before, The Green Mile was not the type of film to overtly go down that road.

'If it isn't already there, I usually just try to put some sort of delay on the orchestral score so that it'll fill the room,' says Michael Herbick. 'It will be anywhere from 30 to 90ms on the front channels and in the surrounds, so you'll hear most of it coming off the screen but there will also be a short delay to the back of the room and when it's all playing together you don't notice that it's coming out at different times.'
"Surrounds are not to be taken lightly," adds Elliot Tyson. "They are there to create the mood, but they're not there to distract people from what's really happening on the screen. Still, on The Green Mile we actually did a lot of surround work with the insects. Mark and I had the idea to fly them up over our heads, so that the audience felt them coming out of the screen. If we got them, they're definitely moving. We did that a little bit with the air and temperature accents, but then sometimes those things could also be distracting, and Frank made a specific point with regard to a scene where the corrugated roof was sweating and stretching and contracting. We had to excise a lot of those noises on one particular reel, and they were in the surrounds.

Tyson and Mangini did, in fact, pre-mix the insect scenes without Frank Darabont even hearing them, in the hope that there would be enough separation to allow for adjustments later on if necessary. They made about five predubs each on those scenes," Tyson recalls. "So we could raise, lower, take out or add anything that Frank liked or disliked. Mark would bring me 25 or 30 tracks of each different food group, as it were, and we would pre-mix those together and then, of course, premix everything in context against those. Consequently, when we got to the end, the balances were basically there the way that Mark and I liked them, and then we could show them to Frank.

We worked the same way on the execution scenes. They were made up of lots and lots of different elements, and for some of those scenes there were as many as nine predubs. We wanted to give the film some edge without making it super-gory, because that's one of the things we were really well那就 about; those three and one of them was fairly long, so it could prove to be a little much. You know, it got tough to watch over and over again, so we had to strike the right balance where we kind of put ourselves in the movie-goers' seats and said, 'What would they want to hear and see without just hanging them over the head?'

In the case of the last execution — John Coffey's execution — we elected to go with more music and less effects, and, although we predubbed it to make it real, we kind of knew that was the direction in which we would be going. However, at least it gave us the opportunity to present our vision and then pull it all back, rather than play it all down and not know in which direction the music was going, because they didn't score that scene until the last minute.

Trying to avoid too many characteristic Foley sounds, at the inception of his work Mark Mangini decided to record all of the Foley on location. Then, because the film's sets were still available, he and the sound department formulated an idea whereby he would use the actual The Green Mile set with fibre optics, and send the results back to the Foley stage where they would be laid down on tracks.

They did some experiments and the stuff sounded incredible," says Elliot Tyson. "When we took away the production track while we were listening on the stage, it sounded like these guys were actually walking on The Mile. It was almost scary. You played the production track and then you took it out, and it was like these guys continued walking, but where did their words go? They were just talking a second ago. It was so seamless. One thing that we had feared was noise build-up, because basically you're shooting one track at a time on the set. However, when we put the tracks up, with a little bit of filtering they just melded seamlessly.

With that, we added the squeak of the footsteps. Percy's squeak was added in Foley — or in Mangini, as we now call it. All of the web belt sounds, all of their Mangini movements, were added afterwards, and it gave this kind of heightened realism that these guys were walking on The Mile. After all, they didn't make that much sound. We had a separate key track for everybody, and we had a separate web belt, because Frank wanted that real leathery kind of guard sound. He wanted these guys to be a little larger than life whenever they moved, and the same applied to the mouse.'

That's right, the mouse. Mr Jingles, John Coffey's sidekick, and a veritable scene-stealer if ever there was one.

Mark did mice vocals separately and those were then processed, explains Tyson. "There was Foley for the mouse footsteps and anything that he does within the box with the bobbin and the little spool. Practically everything was added.'

Elliot Tyson, Robert Litt and Michael Herbick all worked for three weeks on the predubs and then followed up with four weeks of final mixing before returning after the beginning and end of the film (the book-ends) had finally been shot. In total, the post schedule ran to a total of about 12 weeks.

'We had the nicest schedule on this dam picture," says Litt. 'For many weeks it was like nine in the morning, then in the evening and then we'd go home, although when those first and last reels became available we were already onto Magnolia and The Three Kings. Still, we really didn't want to farm The Green Mile project out to anybody else, and neither did Frank, so we managed to squeeze in a couple of days here and there and finish it up.

'It's not unusual to go from one project directly onto another, but it is extremely unusual to only complete a portion of a picture, totally jump ship on that, get completely immersed in another project, and then get re-immersed in the original film... before jumping back into yet another one. Within an hour you're back into the project and you can do it — you just have to cancel one out of your head and go into the other, and I have to admit that it does allow you to see it in another frame of light. Then again, at times it can also get pretty crazy.'
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DIGITAL AUDIO RESEARCH

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WE REGRET to announce the death of Digital Audio Workstation at the tragically early age of 15. DAW was born in 1985 and survived to see the new millennium. Although a sickly infant and remembered by many for fits of sulking and absentmindedness as a child, DAW grew into a robust, and trustworthy teenager. Always an individual there was a period of experimentation with rock‘n roll and speed enhancement. The rise to fame was nonetheless meteoric and DAW energetically bound elder relations into early retirement. Many of us were initially lost in wonder and admiration at the achievements. Unfortunately, this later gave way to a rather blasé and cynical acceptance of virtues and vices. Recently DAW had begun to show distinct signs of middle age spread and loss of direction and focus. Too late these were recognised as symptoms of a terminal decline. Despite this, DAW remained useful and productive to the end.

If this report seems a little premature, please bear with me. I believe the DAW has almost had its day as an identifiable single product category. At its best as a finely honed tool for a specific job it changed the world, and perhaps in doing so, signed its own death warrant.

Consider the areas where the DAW is found in large numbers: Broadcast television was an early adopter. Here it enabled downsizing of whole departments and improved quality. To put it rather more bluntly, it enabled broadcasters to make substantial numbers of operators and engineers redundant and saved a fortune in rent on the real estate occupied by the people, and the bulky machinery they operated and maintained.

Storage capacity and processing speed have arrived at the point where real-time video workstations are available at a fraction of the cost of an early DAW.

Highly significant for several reasons. Along with the video a number of audio tracks are thrown in along with mixing and effects unheard of on dedicated DAWs just a few years ago. Even if individual specialists continue to perform picture editing and sound-editing mixing there will soon be little justification for using different kit. There may be specific user-interfaces for sound and picture, but all will address the same underlying hardware. Networking is an integral part of this. While the basic craft competencies will remain almost the same as they always were, the user-interfaces will be immediately familiar to the computer generation and require only limited training.

At the zenith of hardware complexity and scope, the movie business, change is also afoot. Picture is now almost universally edited on NLE (Non Linear Video) workstations and sound on DAWs. Mixing takes place on huge (and hugely expensive) stages with massive automated consoles. However, the industry is seeking to reduce costs and the focus is shifting from the shooting end to post and one way of improving efficiency is to use highly sophisticated networking. In editorial, potentially this can provide tools for tightly managing projects and reduce or eliminate whole stages of copying.

A generation has grown up with computers. They also understand enough of the rudiments of sound and picture grammar from constant exposure to enable them to be rapidly trained. More importantly, they are available far more cheaply than the established operators.

This may be all very well for truck laying and sound design, but surely a large sophisticated console is still required for the final mix? For any of the elite dubbing (rerecording) mixers who still think they are safe, think again. At least one respected Hollywood sound facility is using the same arguments seen in television ten years ago. These basically boil down to: If the same or comparable results are achievable on cheaper kit with fewer, and more cost-effective operators, then why not do it? If this principle is established similar arguments will apply as in television and we will see networks carrying video, audio and metadata with remote processing addressed by various control surfaces. What we won’t see are individual DAWs.

What, then, of music? The dedicated DAW has long been vastly outnumbered in this area by MIDI sequencers and audio (and often video) bolted on. Recently, the instruments themselves have begun to be seen as extensions of the package.

The best of these are highly effective and creative tools. At worst they can be jacks of all trades and masters of none. One of the most useful aspects is the ability to run the computer front ended systems. Digidesign’s Pro Tools, is rapidly evolving from pure DAW into a combined sequencer, audio generator, editor and mixer with pictures attached.

So the last bastion of the true DAW is radio, right? Well wrong actually. Radio broadcasters are just as keen on finding viable networking answers to problems such as how to arrange multiple access to material and word processor-style editing of audio. Multiple access allows source material to be used in many different products—like news bulletins and in-depth analysis. Text editing of audio enables journalists to edit without technical knowledge. The logical end-point is source material physically recorded in only one place. Edit decision lists then determine how this sounds in the final products. In the hi, tri and quad media-world of film, TV, radio and video, the single point of entry will be manipulated by operators on different continents before contributing to all these destinations.

The reports of its demise may be exaggerated, but the essence of the DAW is certainly transubstantiating into a far more morphous creature. A few early examples may survive a while longer in a few habitats, but I confidently expect the rest to be part of a remarkable evolution. Expect key manufacturers to be announcing the conception of their new children in short order.
The most critical part of any audio production has always been the final mix and in today's world that means being able to work with surround sound. Until recently this type of mixing power was available only to a select few. But not any longer...

Nuendo's integrated 5.1 surround mixing means you don't have to buy expensive plug-ins or hardware to get the perfect mix. Nuendo lets you work in any surround format you choose with support for up to eight channels and includes presets for 5.1, 6.1, 7.1, LCPS and more.

Nuendo includes LCPS Encoder and Decoder plug-ins and works with surround encoded material right out of the box. A dedicated SurroundPan plug-in lets you freely position your audio within the surround field and features two modes: Position, which lets you work in classic cinematic situations with variable distances from the center, and Angle which defines an equal distance from the listen position. Nuendo's Master setup window offers total freedom to modify your studio's speaker placement for creating your own custom surround presets, plus any VST compatible stereo plug-in can be inserted into a surround configuration creating a whole new dimension in real life audio production.

Nuendo supports the VST 2.0 Surround protocol and using the optional Surround Edition collection of plug-ins from Steinberg Spectral Design you'll get up to eight channels of audio processing all in real-time.

Six plug-ins are included featuring a single band compressor, Loudness Maximizer, 7-Band Parametric Equalizer, a natural sounding reverb, plus an LFE Splitter and Combiner for working with low frequency channels.

When you are ready to mix down you can choose mono, stereo or 6-channel split or interleaved files for total compatibility at the final mastering stage.

Nuendo delivers the professional features that make placing your product a breeze.
Beyond the familiar and laborious use of cassette duplication systems, there are now short-run CD copiers that can revolutionise the operation of a variety of facilities. Beth Arnold investigates Virgin Records' Nashville operation.

As an A&R Co-ordinator, at Virgin Records' Nashville outpost, an important part of Jason Krupek's job is creating compilation albums. This usually involves culling large numbers of songs to give the 8 to 15 songs that the artist, manager, producer and the A&R department feel best represent a lifetime's work. In order to ensure that the selection is all it should be, every week or two the selection is reined in with the weakest songs swapped out for newer, stronger ones until the compilation is complete. At each stage a working compilation is produced so that everyone can evaluate not only the selection, every aspect of the finished record.

In his previous position with another record company, Krupek created those comps using an overworked, interlinked system of 24 cassette decks—two banks of six and one set of 12—which required him to make separate modifications and adjustments for each job on each cassette deck. The setup made the process tedious, time consuming and prone to error, which resulted in recordings—done in real time—of varying and inconsistent quality.

'Someone could accidentally bump the controls, causing decks to record hotter and possibly distort the sound, and others to be recorded at a lower level making songs sounding much quieter on the cassette,' he recalls. 'And if one cassette decided to click off in the middle of the song, it was very time consuming to stop all cassette decks, rewind each one individually and re-record the song. Basically, you had to tend to each deck one at a time.'

Clearly, there was a better way. They called it short-run CD copying.

In Krupek's case, MediaFORM's CD-5900 CD2CD stand-alone CD duplicator offered everything he needed. And when one was installed at Virgin Records' new offices in the Fall of 1998, Krupek cured one of the worst headaches of his working life.

Capable of burning multiple CDs in a single pass, the CD-5900 tower houses eight recording bays and a reader; and will deliver eight copies with a minimum of attention. In full flight, the system will dupes...@ www.focalpress.com

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Other benefits of having an in-house CD duplicator include not only the time and money the company saves by freeing up Krupek's time for other A&R responsibilities, but also the resources Virgin Records Nashville would spend on materials. It is cheaper to stock CDs at $1 per blank CD, than cassettes, which can cost roughly $3 for top-quality audio-cassettes.

Krupek has also used the CD-9800 to duplicate photo and art files created by the art department for those occasions when visual promotional material need to be forwarded to the media. Other benefits of having an in-house CD duplicator include not only the time and money the company saves by freeing up Krupek's time for other A&R responsibilities, but also the resources Virgin Records Nashville would spend on materials. It is cheaper to stock CDs at $1 per blank CD, than cassettes, which can cost roughly $3 for top-quality audio-cassettes.

Technology in CD-R duplication equipment is constantly developing making in-house duplication an even more cost-efficient and sensible alternative to outsourcing duplication needs. In fact, newer models of CD copier are capable of duplicating up to 130 CDs (300Mb) per hour with 12x technology. Antipiracy is another persuasive argument for keeping CD duplication operations in-house, regardless of the kind of facility being operated. With the advanced antipiracy features inherent in systems such as MediaFORM's SmartDRIVE2—like watermarking and copy protection—studios can maintain greater control of content and distribution. The MediaFORM SmartSTAMP codes each CD with a unique serial code, allowing CDs to be tracked to the original source of duplication.

Krupek admits to outsourcing when he needs need 500 copies in two days, but for the most part, a short-run machine has met all the CD duplication challenges thrown at it. The bottom line is that, for short runs, in-house CD duplication is invariably saves time and money on both materials and human resources.
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TASCAM’s range of latest digital recording products provides 24-bit high resolution signal quality for multitrack tape, hard disk, DAT and digital mixing.
The death of Dick Swettenham, in early April, will have caused many in the widest domains of the recording industry to stop and reflect. He was one of the ‘names’ of the UK studio business, and through that, both important and influential on a far wider scale. Although most widely known by the reputation he gained in the seventies, when his company, Helios, was one of the major forces in mixing console development, his is a far more diverse story.

To understand, you have to look back, perhaps 40 years, to a far, far smaller industry, where the few studios around built their own equipment and the ‘rules’ of the business had yet to be laid down. Multitrack was years away and even the small consoles then needed for recording were largely a technology under development. Dick was one of those who helped make that happen, pushing at the available technology without forgetting that a human being had to use the console, and that the purpose behind the whole exercise was to capture music.

After leaving the RAF in 1948, Dick spent three years studying Physics and Telecommunications Engineering at the London Northern Polytechnic before joining the maintenance department of Abbey Road Studios in 1951. Five years later he moved to Argo Records, a small independent record label where he learnt classical recording and disc mastering, as well as spending a period at La Scala in 1956. There was also an enduring parallel interest in folk music, being a director of Topic Records, and even performing himself.

He joined Olympic Sound Studios on their formation in 1958, as technical director. This was a position he held for 11 years, during which time Olympic became one of London’s leading independent studios. In 1961 he built and designed what is widely considered to be the first professional transistorised mixing console—a 12-channel desk that went on to make hits for Dusty Springfield, Troggs, Rolling Stones and Bee Gees.

Following the studio’s relocation to Barnes in 1966, a new generation of consoles was required. Dick was able to integrate the operational expertise of Olympic’s chief engineer, Keith Grant, into the design and build of the new studios following Grant’s detailed analysis of the recording process. It is here that many of the basic ideas that characterised Dick’s later console designs were formed—a fusion of his technical ability, and skill in realising the operational ideas of Grant, his high profile staff, and producers passing through the studios. Certainly the early equalisers were the result of this collaborative process, while the ergonomic approach to console layout also had its origins here.

In 1969, Dick was, in his words, ‘head-hunted’ by Chris Blackwell’s fledgling Island Records to design consoles for their new studios. This opportunity led to the formation of Helios Electronics to provide consoles for Island and in turn, a long succession of further clients. Helios actively marketed itself as a bespoke console company—‘We start with a blank sheet of paper…’ Virtually every single order was custom designed and built. Only Island Records who ordered four consoles had the same modules, and even those were in only three of them. In total he personally designed over 50 custom consoles.

Consoles went to such UK studios as The Manor, The Townhouse, the Beatles’ Apple Studios, Maison Rouge, Strawberry, the Who’s Ramport, and all Island Studios. The custom building and lightweight construction techniques found favour in mobiles with consoles going to the Stones Mobile, Manor Mobiles, Maison Rouge and Island, while international sales grew steadily—to Europe and the US. The seventies also saw the beginnings of personal studios and Helios’ flexibility in design gave them an enviable client list—a virtual who’s who of rock—an area he liked to handle himself. An observer of Helios of the time noted ‘Studios bought Neve, musician’s buy Helios’, which had more than an element of truth to it.

While Helios consoles were notable for frequently being of a wrap-around style, they had more conventional approaches and the company successfully moved into broadcast consoles through the seventies. One notable console was for Mainos TV which had perhaps the first integrated video monitor for level display and the introduction of ‘free grouping’ of faders.

However, a combination of factors led the company to cease trading in 1979, the publicly quoted reason being increasing competition in the form of standard off-the-shelf consoles. Dick had always run Richard Swettenham Associates, a specialised consultancy company, in parallel with Helios, and from this time onwards it was here that he focussed all his activities. Aside from studio design, planning, manufacturer consultancy and up-grading clients’ consoles, he continued developing ideas on console design and digital control, particularly in relation to mobile recording. His unmistakable form was frequently to be seen at AES Conventions worldwide, usually deep in discussion with fellow designers.

Dick actually liked old ideas but had long wished to return to manufacturing products. The growing demand for his classic equaliser designs presented an opportunity. He was about to become a director and shareholder of Helios Professional Audio, a company established by close colleagues to develop both his old and future audio designs. Unfortunately this was not to happen.

Dick Swettenham died of cancer on 9th April. He was a major influence on mixing console design, one of an elite band of ‘names’. He should be remembered for introducing the world to the idea that a console should be an active combination of audio quality and ergonomics.

In his words, ‘It is the control surface under the hands of creative engineers which enables them to achieve wonders, and the quality of the resulting sound is their, and our, primary objective.’

I would like to acknowledge all who contributed, with special thanks to Neil Adams, George Chibnall, Keith Grant and Crispin Hopfield. Dick Swettenham spoke from short autobiography to be found on www.heliosaudio.com
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I DID NOT OCCUR TO ME that there was any particular reason why so many Beatles songs were being played on the radio that morning. I was late for school, so it was all a blurry soundtrack to my haste. Besides, the Beatles were having something of a resurgence in 1990; there was nothing strange about hearing them anymore. Many people appeared subdued in class: when a friend said ‘Isn’t it terrible?,’ the shocking reality was revealed to me.

That evening the television was logged-jammed with tributes. The news, magazine shows and quickly assembled programmes interspersed the thoughts of friends and contemporaries with reunions of old interviews with John Lennon in his various incarnations: chirpy, wise-cracking Moptop, spaced out high priest of psychedelia, serious spokesman of a generation, drink-fuelled rocker, domesticated house husband. The abiding image was the white room where a white-clad John played and sang at a white piano while a similarly attired Yoko opened white window shutters.

Despite now being criticised for its art college philosophising and latent hypocrisy, ‘Imagine’ is undoubtedly one of the great songs of the late 20th Century. Last year it got to No.3 in the UK singles charts after being voted the British public’s favourite lyric. The same titled album, recorded in 1971, is arguably more important: it demonstrated Lennon’s development as a solo performer and writer, spawning songs that veered between melancholy and anger, ‘Jealous Guy’ and the vitriolic ‘How Do You Sleep?’ being the obvious cornerstones. Its importance as a piece of production should not be underestimated: melding what he had learned in the early days of the time: rooftop and Abbey Road, Lennon worked with Yoko Ono, Phil Spector, a couple of other ex-Beatles and a group of leading session players to create a sparsey beautiful collection of songs.

In the twentieth year since John Lennon’s murder, both album and man can be examined and reappraised through the release of a remastered, remixed audio CD and the DVD of a documentary mixing 16mm home movies and footage shot by independent film-maker and film historian Jonas Meier. ‘Gimme Some Truth—The Making of John Lennon’s Imagine album is a logical progression from Let It Be, with every aspect of a recording, and the time-off that helped drive it along, being documented to give an insight into the creative process. John and Yoko filmed themselves and musos including George Harrison, Ringo Starr and Nicky Hopkins, while Meier recorded visits by the enigmatic Jack Nicholson, the even more enigmatic Miles Davis and the completely enigmatic Andy Warhol.

The BBC screened a shortened version of the film earlier this year to much acclaim. This DVD version is a 63-minute cut and, as is now expected from all releases on the format, features added extras. In addition to 40 minutes of interviews filmed at the time, the release offers a stereo soundtrack, a 5.1 surround mix and a full Lennon discography with album-track listings, cover artwork and clips from the songs. Yoko Ono has been criticised, along with the surviving Beatles, for grave-robbing the works Lennon deemed not good enough for public consumption—‘Free as a Bird’ being the obvious, heinous example—but ‘Gimme Some Truth’ goes some way to justify her policy of issuing previously unreleased material. As one newspaper critic put it, ‘Why haven’t we seen most of this before is a mystery…It is a fascinating and historic rock document.’

Ono is a constant presence in the film, not just in the private moments with John but during the recording sessions as well. There is fun to be had watching her making suggestions to the supposedly dictatorial Phil Spector, who produced Imagine with Lennon. For this new version she visited London last autumn to oversee the album’s digital remix and to executive produce the movie. Remixing and production of the DVD took place at Abbey Road Studios, where most of the Beatles’ recordings were made. Now, as Abbey Road Interactive, the studio has become a leader in DVD authoring.

The stereo and 5.1 soundtracks for this new DVD release of Gimme Some Truth were engineered by Peter Cobbin, who has been a senior recording engineer at Abbey Road since 1995 and was instrumental in the remixing of the Yellow Submarine soundtrack for 5.1 release last year. Trained at EMI in Sydney, he worked as an engineer and producer in Australia before coming to Britain. Cobbin’s credits include Texas, Air and Janet Jackson; at the moment he is specialising in 5.1 and has created discrete surround mixes for Bruce Johnson of the Beach Boys.

‘Gimme Some Truth’ is about a moment in time: how an iconic musician, at the height of his creative powers, came together with a group of like-minded artists in a specially created environment to record what is, 29 years on, regarded as a classic album. Lennon is dead, Phil Spector mad, by all accounts, Yoko difficult to get hold of at short notice; but behind the star names were the engineers who worked on the sessions. They may only be bit players in the movie who sometimes get the rough edge of Lennon’s acerbic nature, but they can still claim to be part of the story.

Eddie Veale is one. Now a studio designer from the Veale Associates, he was surprised to be asked for his reminiscences, but agreed that it might be fun. Veale’s main role at the time was engineering support. He had been a staffer at Advision, involved in the studio’s move from Bond Street to Great Portland Street. ‘After that I was looking around for new things, for new projects for people that would help me progress,’ he recalls. Veale had met the Beatles by dint of the fact that he was looking after Moog synthesisers in the UK; he had then worked at Abbey Road while George was recording ‘Here Comes The Sun’. Through that the Quiet One asked Veale to help out at his private studio in Esher, Surrey; the Beatles connection continued when he became involved with Apple Studios in Saville Row.

In the wake of the Beatles’ break-up >
and speculation over who would do what, John Lennon released his acclaimed eponymous solo album. In 1971, Eddie Veale was invited down to Lennon’s country home, Tittenhurst Park just outside the town of Ascot. ‘He commissioned me to build a studio for him there,’ Veale says. ‘The intention was for what became Imagine to be the first thing to be recorded there. After I finished the design and build, I sat in on the sessions, mainly providing engineering support but also doing some tape editing.’

Veale designed the acoustics and specified the equipment for the studio, which was constructed in a single storey wing of the manor house. The console was specially designed for Lennon’s needs and custom built using mostly Cadac modules; the B-track was a 3M machine. ‘The equipment and the studio embodied various ideas John took from Abbey Road and Apple studios,’ Veale explains. ‘It was always intended to be a full-blown studio, not just somewhere to record demos.’

During the build, Veale and his engineering colleagues did not appreciate that Lennon had specific projects in mind for the studio. ‘John became keen for us to finish the work so that he could start recording,’ Veale remembers. ‘The pace was forced when Lennon brought Phil Spector over from the States, partly with the intention of him working on the new album, partly to get his input on what the studio needed.’ Phil came in, made comments about what was already in and requested equipment and features that he wanted,’ says Veale.

John and Yoko are the focus of Gimme Some Truth, while Imagine the album is centred on John, his angst, his love for Yoko and his anger towards Paul McCartney. But one-time wunderkind and certified mad genius Phil Spector is the archetypal special guest star. For all his reputation as a task master, Spector appears vaguely bewildered by proceedings through the whole film. ‘That was just Phil’s way,’ Veale says in the producer’s defence. ‘He was very laid back but when it was required, he took a strong lead. Most of the time he was quiet but things always got sorted out when he was around. He was very positive, an exciting guy to be around and work with.’

Lennon too, it appears, took a strong lead, setting a brisk pace for the sessions. ‘John had very clear ideas about how he wanted everything to proceed,’ Veale recollects. ‘They recorded every day of the week, but there was no work at weekends, apart from some maintenance on Saturdays.’ With Lennon and Spector setting the pace, the album was recorded in around two months.

Veale attributes much of this to the experience of Spector and the discipline that had been instilled in Lennon, Harrison and Starr during their earliest recordings at Abbey Road. ‘Most bands today are fairly leisurely in how they

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**The basics**

Although sound delivery methods have evolved dramatically over the years, musicians still perform in front of microphones and our ears are still the final interface between the delivered audio and the human brain.

The preservation of sonic integrity within modern working methods has never been more important. No matter what the final format may be, the challenge for SSL is to capture, mix, enhance, re-mix, and route the performance to whatever delivery system is desired. SSL is unique amongst console manufacturers in offering a choice of large format multitrack consoles in either analogue or digital technology at the highest possible sonic integrity with reliability, repeatability and a world wide back-up which is second to none.

The SL 9000 J SuperAnalogue™ console is the analogue choice for the world's great studios and scoring stages. The Axiom-MT digital multitrack console has amazed engineers and producers with its accurate, warm and musical sound, and has further enhanced SSL's reputation as the leader in integrated console automation. The new MixTrack 96-channel hard disk option is an additional development in this total integration.

With Avant for Film and Post and Aysis Air for Broadcast, Solid State Logic provides consoles at the highest specification levels for installation around the world wherever serious sound is the issue – whatever the delivery system.

No matter whether your choice is analogue or digital – the future is SSL.

**John Andrews, Marketing Director**

Front cover: kd lang and Damian le Gassick at O'Henry Sound Studios

Photo credit: David Goggin

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**Hit console for The Hit Factory**

New York's The Hit Factory, one of the world's premier recording facilities, recently installed a 48-fader, 96-channel Solid State Logic Axiom-MT digital multitrack console in Studio 4. The MT joins the seven SL 9000 J Series consoles already in service in both The Hit Factory, New York and The Hit Factory Criteria, Miami.

"Given the overwhelming success we have with the 9000, the move into the MT was a natural extension of our ongoing relationship with SSL," says Troy Germano, vice president of The Hit Factory. "The best engineers in the music business are already producing world-class results on our 9000 consoles. The MT shares the same ergonomic design as the 9K, so the learning curve for our clients and staff has been very short, allowing us to produce the same world class results immediately."

The MT, with its superb automation system and extensive channel dynamics processing, offers all the power of the 9000 with the advantages of an all-digital workflow. This combination of features serves The Hit Factory well, considering its stellar and diverse client base, which includes such artists as Billy Joel, Whitney Houston, Wyclef Jean, Paul Simon, Aerosmith, U2, Tony Bennett, Jennifer Lopez, Nas, Marc Anthony and Jim Steinman, among a host of others.
Invincible kd lang in 5.1

The latest project from kd lang, 'Invincible Summer,' was mixed on an SL 9000 J Series console with the SL 956 surround monitoring section at O'Henry Sound Studios in Burbank, Ca if.

Produced by Damian le Gassick, the record was mixed entirely in stereo and then in 5.1 surround sound. The 5.1 version will be released on the Warner Brothers record label as part of a collection of releases produced in the DVD audio format.

"I'm extremely happy with the 5.1 mixes that I've done on the 9000," says le Gassick, who likes the console for its sonic integrity and familiarity. "This is the first mix I've done in 5.1 surround sound and working on a board that was already familiar to me was a definite plus. I didn't have to learn anything from scratch. The SL 9000 is the only board I could imagine doing this on." le Gassick worked with engineer John Smith on the production for lang's 'Invincible Summer.'

Avant moves ahead in Paris

The versatility of the Avant digital console for both film and post-production is clearly demonstrated in Paris - home to the most buoyant national film industry in Europe, including established Avant users SIS, and leading international post-production company Dubbing Brothers, who use two Avant consoles, the second in their Italian facility.

Two recent installations confirm the Avant's continued success. The first, for Jackson, is the second Avant console to be installed by this leading facility in St. Ouen, Paris. Jackson were one of the first film studios in France to equip for digital, and the latest Avant now positions them as the only film studio in Europe with four digital consoles at the same location.

The 80-channel Avant is designed to accommodate two operators for film mixing but the versatility of the console will enable the facility to mix TV dramas with a single operator. Jackson's film output is prodigious with over 35 feature films mixed each year — in addition to TV drama production.

The second Avant, also an 80-channel console, is to be installed at Copra of Boulogne, Billancourt, Paris — a company that has grown steadily throughout its long association with SSL. Copra's business has evolved through video editing and long form TV work and the decision has now been taken to expand with a fully equipped 5.1 surround theatre — with the new Avant as its centrepiece.

At home in Media City

SSL are to supply six consoles for installation in Media Production City, dubbed 'Hollywood on the Nile,' the £200m state-of-the-art production, dubbing and music recording complex under construction on the outskirts of Cairo.

The 420,000m² studio complex is being built for the Egyptian State broadcaster (ERTU) and will be the focal point for future television and film production in the region.

Four SL 8032 GB analogue multitrack consoles with stereo channels will be deployed for broadcast production and a 64-channel Avant digital console will be used in a post-production role. A 96-channel Axiom-MT digital multitrack console for music recording and mixing completes the initial order.

SSL consoles were selected for the project for a combination of audio quality, ease of use and marketability — as the centre plans to attract clients from around the world.
Sun Studio of Copenhagen has an enviable record and an outstanding international reputation. With 120 staff, 35 studios, and facilities in Norway, Sweden and The Netherlands in addition to the main studio in Denmark, they are arguably the most successful audio post production company in the region.

A long-established SSL customer with numerous ScreenSounds and two OmniMix rooms, Sun Studio's latest phase of expansion incorporates a 96-channel Avant digital console which takes pride of place in Stage 1 — a new all-digital 60-seat dubbing theatre.

The new room is fully THX™ approved and is fully equipped for surround with 5.1 for Dolby SRD™, DTS™ and the newer Surround EX™ format, plus eight-channel SDDS™ — one of the few studios in the world to have such comprehensive capabilities.

Sun Studio has a long and successful history in feature film foreign language dubbing for the Hollywood majors including Disney, Warner Brothers, DreamWorks, 20th Century Fox, and Columbia TriStar. A considerable amount of work is also provided from Scandinavian TV stations and, more recently, the company has utilised its in-house project management expertise and become involved with original local film production in both sound design and mixing capacities.

Benni Christiansen, who mixes at Sun with his brother Brian and father Svend — Sun's founder — concludes, "We tried other consoles but preferred the Avant's layout and knob-per-function operation. The automation is a very powerful tool."

Aysis Air Mobile for Turner Studios

Turner Studios in Atlanta, Georgia has purchased an Aysis Air Mobile for installation in its new 53-foot production truck. The trailer, scheduled to go on-line in August, will be used primarily for live sports production.

Bob McGee, director of technical operations at Turner Studios Field Operations, sees the design philosophy SSL employs with Aysis Air Mobile as critical for the move into digital.

"The Aysis Air Mobile console control design ensures a smooth and comfortable transition from analogue to digital for our audio engineers," McGee explains. "This 'Mobile' configuration of the Aysis Air affords greater flexibility in room design due to the reduced console footprint. The SSL 6000 we are retiring was the first 6000 in a network truck 18 years ago and has outlasted two trailers. Our second 6000 purchased 14 years ago is now in its second trailer. That's a combined 32 years of continuous remote service and we have never had an on-air console failure. We believe replacing our first 6000 with the first Aysis Air Mobile is in keeping with Turner Studios' commitment to provide the highest level of dependability possible to our clients. With the Aysis Air Mobile, SSL has designed a console with the needs of the mobile broadcaster in mind."

Rainer Oleak's large private studio in East Berlin now boasts an SL 9056 J series SuperAnalogue™ console which will be used in conjunction with the owner's impressive collection of vintage outboard gear collected over many years.

Oleak is a well-known composer and producer from former East Germany who describes himself as an 'audio-junkie.' Trained in electronics and acoustics, Oleak became a sound engineer in 1978 before turning to professional composition.

The main body of his work is music for German TV programmes and films — he's scored over 250 in a 20 year career — and his compositions from pop to rap have reached a wider audience since reunification, with artists such as Ute Lemper and German band Puhdys. The choice of the SL 9000 J followed lengthy evaluation but eventually the decision was made. "My budget didn't originally stretch to a 9K," Oleak candidly admits, "but as sound quality was my absolute priority, the choice became inevitable. Also I was looking for a console as an investment over the next 15 years. The UK provides that and gives me the flexibility and features I need — and it gives me a lot of fun!"
M A J O R  T V  s e r i e s  g e t s  f i l m - s t y l e  m i x

Randall and Hopkirk (Deceased), produced by Working Title TV for BBC1, is a million-pounds-per episode remake of a classic 60s series which now stars comedians Bob Mortimer and Vic Reeves.

With high production values incorporating a considerable amount of computer-generated effects, Working Title also invested substantially in the audio quality of the series which was mixed on one of the two SSL Avant digital consoles at Anvil Post Production in Denham, England.

Anvil's Alan Snelling explains, "The new series makes considerable use of complex sound effects and music to complement the visual trickery. We mixed in Dolby Surround™ in a style more commonly used for feature films. The Avant greatly simplified that task."

[Left to right] Anvil's Sound Mixer Alan Snelling with Charlie Higson, Producer and Colin Chapman, Supervising Sound Editor on Randall and Hopkirk (Deceased)
**technofile**

### The Numbers Game

The characteristics of analogue consoles can be compared at a basic level by measuring various parameters and setting out tables of input noise, gain, overload margins and the like. Comparisons between digital consoles are not so easy, as digits out should equal digits in, and any differences are usually attributable to the performance of the converters.

Some pundits enjoy quoting processing speed, and figures are bandied about in much the same way as enthusiasts discuss the 0-60 acceleration times for cars and motor bikes, and whether the engine has four or five valves per cylinder. Most digital audio processing uses floating point arithmetic, and speeds are quoted in millions of operations per second — MFLOPS. The latest SHARC (Super Harvard Architecture) chips are quoted by their manufacturers as offering hundreds of MFLOPS, but this is only half the story. The ASICs (Application Specific Integrated Circuits) used in SSL's digital consoles can perform up to 7 FLOPS per clock cycle, and are very efficient at parallel processing. A further advantage of the ASIC philosophy is that ASICs do not use software and therefore, unlike SHARC devices, can't crash when a software bug bites.

SHARC and ASICs each have their advantages, but comparing consoles is best done, as with motor vehicles, by asking what they do, not how they do it.

**Matthew Willis, Technical Director**

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**Artist DJ Quik recently worked on his latest album for Arista Records using the new Axiom-MT digital multitrack console at Skip Saylor Recording in L.A. Pictured here at the console, he has also produced artists Tupac, Snoop Dogg, R Kelly, Tony! Toni! Tone! and Ice Cube. Photo credit: David Goggin**

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**KABC opts for Aysis Air**

When KABC, the ABC owned-and-operated television station in Los Angeles, broadcasts the news from its state-of-the-art Glendale studios in September, it will be with the assistance of an Aysis Air digital broadcast console. Designed for on-air studios, production studios and OB trucks, Aysis Air will be used by KABC primarily for live news production at its new facility, currently under construction.

"We wanted an all-digital system that had a proven, on-air track record," explained Michael Englehaupt, KABC’s Director of Engineering. "The Aysis Air has been on the air successfully in other television markets and we expect to continue that success in Los Angeles."

"The console provides an easy-to-learn user interface for operators," Englehaupt continued. "SSL really puts the power of the multiplexed environment into routing capabilities with an interface that is workable and makes sense."

"When you've been in a situation where you've had to pick up the phone and reach somebody, SSL's reputation is put into action. They have a very strong track record of delivering excellent customer service."
Solid State Logic's Axiom MT is the digital console of choice...

For Phoenix, Maryland—based Sheffield Remote Recordings, a leader in remote recording, broadcast and production, Sheffield has already found great success with the console in the nation's capital, where it made its debut at the CBS broadcast of America's Millennium Celebration.

The 96-channel MT was used to mix and broadcast an 80-piece orchestra, which was part of the live show. The broadcast was produced by Quincy Jones productions and mixed by Tom Vicari. "The MT performed beautifully," says Tom Kline, engineer-in-charge for Sheffield Remote Recordings. "The MT is perfect for complex shows such as the America's Millennium Celebration. It allows for virtually instant reset of settings, which is vital when broadcasting a live show with numerous groups or sections performing. The MT's flexibility and ease of use made it unbeatable for an event of this magnitude."

Following a recent order, Visual Arts, Japan's premier music engineering school, will have a total of five SL 9000 J series SuperAnalogue™ consoles installed in colleges throughout the country.

The original SL 9000, delivered to Visual Arts in Kyushu in 1997 will now be joined by four other consoles for their engineering schools in Osaka, Tokyo, Nagoya and Sapporo.

As Takeo Asano, Managing Director of SSL KK, Japan, explains, "The adoption of the console by leading music engineering schools such as Visual Arts will help to ensure an important source of new talent for the recording industry in our country."

Both the SL 9000 J SuperAnalogue™ console and the MT digital multitrack console are gaining plenty of northern exposure with new Canadian installations announced in Ontario and Montreal.

Metalworks Recording Studios, which has been host to superstars including David Bowie, The Cranberries, Anne Murray, Tom Cochrane and Def Jam hip-hop star Ja Rule, is celebrating its twentieth year in business by building Studio 6, a new 5.1 surround mix room featuring an 80-channel 9000 J Series console. The console, which includes the SL 959 eight-track multi-format surround monitor system option, joins two other SL 4000 consoles in Studios 2 and 3.

When Montreal-based video game designers Ubi Soft sought to enhance its audio capacity, the company decided to create a specialized multimedia studio based around the MT. The new studio ensures improved quality control and optimises audio production, allowing the company to achieve a level of sound quality unparalleled in the field of computer games.

In an industry that values both technological and artistic achievements, Ubi Soft manages all stages of sound production of its products. "I strongly believe 5.1 is the immediate future of video game sound," says Didier Lord, vice president of Ubi Soft. "In order to compete in the fast-changing industry of video games, one needs quick flexibility, upgrade ability and pristine sonic quality. The MT met these tough requirements and became the obvious choice. The MT will help us lead the way in video game sound."

Ubi Soft Entertainment

Didier Lord, VP Ubi Soft

www.americanradiohistory.com
MT 'Made me a fan of digital'

When top record producer Tony Brown booted time with Chuck Argyr at Backstage Studio at Sound Stage in Nashville, the experience he had on the MT convinced him of the benefits of digital mixing. The MT's flexibility, automation and exceptional sound made all the difference for Brown. "I used the MT to mix a hit album for Wynonna's new album, and it may be the best sounding mix I have ever done," says Brown. "Chuck is always the first person in town to get the latest gear, but because I am so busy, I didn't go into the session thinking I was going to create the console. I just listened to the mix and immediately was very impressed with the clarity."

The 96-channel MT at Sound Stage was also recently employed by rock legend Peter Frampton for a major recent present set for multi-format stereo including HDTV - with multi-channel sound, audio CD and DVD.

New German Office

Solid State Logic has established a Regional Sales Office in Germany, based in Dusseldorf. The founding of SSL's German Office follows a successful pattern of European regional offices already established in France and Italy - areas where the company has experienced substantial growth in recent years. The German market is similarly expanding, particularly in the broadcast sector where national German broadcaster NDR, for example, has already taken delivery of six SSL digital consoles. SSL's German Office may be contacted on 0211-332 2986.

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MT made me a fan of digital'
record,' he says, 'but the Beatles were pretty well organised in all respects. That was reflected in the pace of their recordings and performances. Their style was to get things done.'

Despite the best plans and intentions, Phil Spector's approach to the recording process was an experimental mix of sonic tricks and technical wizardry. This was evident in the use of the CEDAR system, which was used for restoration and de-clicking, and the CEDAR's ability to eliminate noise without affecting the dialogue. This was a major contribution to the sound quality of the album, and it was a true testament to the dedication of the recording team.

As Eddie Veale observes, 'The whole project was lavished on Yoko Ono, and the sound work was never acrimonious.' This was a reflection of the high-quality sound that was achieved on the album, which was a result of the careful planning and attention to detail that was put into the recording process.

Despite the best plans and intentions, plus engineers from Abbey Road, the problem was solved by people travelling up to Abbey Road to see how it was done, then returning to Tittenhurst Park to do it for themselves.

Veale observes that the facilities and support industry almost taken for granted today did not exist to the same extent in the seventies. 'John was coming up with new ideas and they required a bit of thought because it was a new studio,' he says. 'It was inevitable that we would make errors because of the pressure. John obviously got frustrated and let off steam, but it was never acrimonious.'

It can be fatuous to ask someone who worked on such a significant project whether they knew at the time that they were working on something special, but the temptation is too strong. 'During the formative stages of the album, I didn't from the view that it was going to be as special as it turned out,' Eddie Veale replies honestly. 'It started to become evident in the later stages of recording.'

The fact that, nearly 30 years on, Imagine has been remastered and a home movie of its making transferred to DVD tells you everything you need to know about how special this recording became.
Wherever multi-channel digital audio is being recorded or mastered, chances are you'll find a Genex recorder at the heart of the process. The new GX8500 is equipped to record at sample rates up to 192kHz and bit rates up to 24-bit, either to internal MO or any external SCSI drive.

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The Genex GX8500.
Multichannel mayhem

Surround sound's additional delivery channels challenge accepted microphone techniques. Terry Nelson takes in a multichannel mic workshop

The advent of multichannel audio for music—whether Dolby Surround, DTS, DVD-Audio or SACD—is prompting experimentation in microphone techniques. It could be argued that this is not particularly new and that the seventies' B-format Ambisonics recordings have already paved the way. However, these recordings were matrixed over four channels and the current requirement is for 5.1 channels.

The Paris AES Convention presented a workshop entitled The Comparative Study of Multichannel Sound Recording Arrays of the 40 Vocal Motet by Tallis at the Paris National Conservatory of Music, chaired by Christian Hugonnnet of the Conseil Supérieure Technique, a prominent exponent of multichannel techniques. The aim of the workshop was to present recordings of the same piece using various mic techniques. Two pieces of music were chosen, the motet and a jazz ensemble using percussion instruments. The workshop featured presentations of the different arrays and techniques used, together with debate among the panel and audience.

The aim of the experiment was laudable in that it set out to provide comparative studies of different systems recording the same work under identical acoustical conditions in order to provide a realistic basis for reference. The playback system was laid out according to the ITU recommendation 775-1 and used Cabasse loudspeakers.

The hall at the Paris Conservatory where the recordings were made has a total surface of 400m² with a usable surface of 311m². The room volume is 3500m³ and the RT is 1.5s. The critical radius is 2m for an omnidirectional source. All of the various microphone systems were mounted in the same vertical plane with the exception of the array from Radio France and the system from the Conservatory itself.

The motet, Spem in alium by Thomas Tallis (16th century) was performed by the Madrigal de Paris under the direction of Pierre Calmelet. The piece is exceptional in that it was written for 40 voices, divided into eight 'sub-choirs' of five voices. Even more interesting is that at its original performance, the composer was thinking in terms of surround sound. The premier of the motet was in the chapel of Arundel castle, home of the Duke of Norfolk, who commissioned the work. The chapel is octagonal in shape and this explains the eight internal choirs, which were placed around the perimeter of the room with the audience in the centre. This allowed the polyphonic effects to come from the different parts of the room. The recording saw the 40-piece choir placed in a circle around the microphone arrays at a radius of 4.77m. The second recording consisted of five percussion instruments with the musicians improvising on a theme by Al Jarreau.

Before opening the workshop Christian Hugonnnet posed various points to ponder by the panel and the audience: are we looking for a virtual or coherent signal? How important is spatialisation? What coherence do we expect from the LCR speakers and what phantom imaging is possible with the rear or surround speakers? Is the 'sweet spot' a vital ingredient?

The ASM 5 system is an array of five Brauner VM1 microphones with double membranes and controlled by the SPL Atmos processor. The microphones are arranged in a star with the three front microphones placed on arms of 17.5cm at 0° and ±90° and the two rear microphones placed on arms of 59.6cm and angled at ±150°. The position of the microphones can be adjusted on the arms and the capsules can be pivoted over ±45° should this be desired. The polar characteristics of the individual capsules are continuously variable from omnidirectional through to Figure-of-8, which also provides panning functions for LCR and front-to-back. In addition, the distance between the front and rear arrays can be 'enlarged' electronically.

The array proposed by Mike Williams and Guillaume Le Dû places the microphone array in relation to the loudspeaker array, which means that the location of the microphones can be optimised. The other idea is that the pickup pattern represents segments to be captured and reproduced by the microphones and speakers respectively. The array itself consists of five non-coincident pairs of Schoeps cardioid microphones. Each pair possesses a coverage angle that depends on the distance and the angle between the two microphones, these being determined by the curves established by Williams in his research work on stereophony. The five pairs are then arranged to provide a coverage angle of 580° but without the individual pairs overlapping one another. The overall pickup performance of the array is adjusted either by manual offsets of each pair of microphones or by introducing >
Jean-Marie Porcher of Radio France is one of the early pioneers in multichannel recording. He opened by stating: 'spectral fidelity is the first priority, followed by localisation'.
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IN THE STUDIO..

London’s Town House Studios is home to some of the most memorable hit records to come out of the UK. It’s also home to 7 HHB CD Recorders.

Pictured: Virgin Studios Director Ian Davidson.

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I cannot help wondering whether we are chasing shadows with multichannel microphone arrays. If the aim is to reproduce the original listening experience, then it seems logical that the microphone system needs to work on the same principle as the ear-brain combination.

Given the level of debate prompted by 2-mic stereo recording, the possibilities for argument over multichannel appear endless. I must admit to being disappointed at the amount of academic discussion because this often seems to miss the point. Theory is good, but psychoacoustics have a mischievous tendency to distance real life from our beliefs. For example, back-to-back M&S microphone deployment is theoretically suspect yet Danmarks Radio’s use eclipses everything else in terms of localisation and spatialisation. Whereas no sensible person would dismiss that research and theory is extremely valuable, our ears (and brain) still remain the final judge and psychoacoustics are still really in their infancy.

I also cannot help wondering whether we are chasing shadows with multichannel microphone arrays. If the aim is to reproduce the original listening experience (which would appear to be the case in this instance), then it seems logical that the microphone system needs to work on the same principle as the ear-brain combination. We already have the ‘ears’ in the form of microphones so why not use the power of modern DSP to provide the brainpower? Over to you, microphone manufacturers... This said, some of the systems presented were using a certain amount of processing so may be this does bode well for the future.

What of the recordings themselves? Whereas this could be considered subjective, more than one person was surprised by the weakness of the frontal information and it appeared to me that the rear speakers dominated the image of the seating area. Again, I was disappointed by the results and they were hardly of the sort to send the average consumer stampeding for the shops. The recordings sounded dry and image had an annoying tendency to collapse into the nearest speaker. In general, the percussion recording came over best and this begs the question of the actual placement of the singers for the motet and the acoustics of the room.

Having heard John Eargle’s recording of the 1812 Overture which was recorded in 5.1 (and with very little time to put the microphones into position), the artistic and aesthetic possibilities are definitely there and this also applies to other recordings that I have heard. Interestingly enough, it is the recordings made more by intention and less with the slide rule that win out every time—is this telling us something?
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Of the many mixing systems available to the post-production and film industry, Elite+ clearly offers more flexibility and performance at a price that will surprise you.

Contact your Otari dealer now to arrange a personal demonstration.
THERE'S THE STEADY and often exponential evolution of audio studio technology — especially during the last decade — that has influenced many aspects of studio life. The visible effects range from what a given budget will buy, through changes in working practices, to the nature of the kit itself and associated shifts in workspace ergonomics. Fundamentally, the tasks are the same — sound still has to be recorded, edited, mastered, matched to picture and delivered to its audience. But the tools have become smaller and more multipurpose, control surfaces have shrunk and become multi-layered, with fewer functions needing — or offering — simultaneous direct access by touch. Armed with a mouse, monitor screen and QWERTY keyboard, you could accomplish as many tasks, apart from positioning a microphone or plugging the cable into the wall box. The kit has evolved, the software responsible for much of the action is invisible, and so rooms are changing as a result.

The scale of recording, mixing and editing equipment has been tending towards smaller footprints for some time, with compact-format digital consoles becoming relatively uniform in size. Mixing consoles, workstations and edit controllers, along with computer keyboards and pointing devices, can all fit into similar space envelopes. The comparative ergonomics of the equipment are also similar, allowing a single, seated operator to be within an arm's reach of all regularly used controls. Given this trend, you could argue that there is now far less of a requirement for custom-made studio furniture, as the prerequisite to fit furniture around a disparate set of equipment shapes and sizes no longer exists to the same extent.

Commercial aspects have also evolved. Size used to be more important for many music studios, but nowadays, most operations would not invest in a console simply because it was the largest, most visually impressive and client-wooing option available. Profitability is paramount. Kit needs to be chosen and positioned carefully, such that it will accomplish a given task as swiftly and efficiently as possible — so that jobs can be completed and dispatched in the optimum timeframe, closely followed by the invoice. Speed is also of the essence when it comes to the initial installation, as most companies cannot afford the revenue lost through any unnecessary downtime.

THE CHANGING face of technology is changing the face of interior design. Andrew Sutcliffe fits furniture to function in the modern studio.

The evolution of storage media also has an influence on the workspace, especially in the postproduction environment. Audio storage on reel-to-reel tape has evolved into the use of S-VHS or Hi-8 videocassettes for many applications. DAWs typically extend the choice of media to mag-optical discs and to fully networked systems, in which only the movement of data by cable needs to be accommodated. Physical media are therefore often only required at the entry and exit points of the process, with less need to store media at each workstation, further contributing to the downsizing of workspace furniture.

The brief would therefore seem to be for a design solution that allows a facility to arrange its equipment to meet the workspace requirements, from an ergonomic point of view, while ensuring that the quality of presentation supports a company's image. Future updates and re-arrangements have to be accommodated, with the overriding requirement being one of flexibility. A large percentage of audio equipment may well be similar in shape and scale, but the pace of technical development is still rapid. Most facilities will therefore require the option to re-arrange their workspace to cater for new developments and customer demands, and possibly at frequent intervals. The logical route is therefore to look for an off-the-shelf modular solution, and preferably one that embodies a similar attention to detail as could be provided by custom-build suppliers. A modular approach would answer the obvious questions of future-proof adaptability, provided that approach could also meet a facility's aesthetic expectations.

There is a relatively small number of modular furniture systems on the market that address the needs of audio recording and postproduction. Some have grown more out of the Musical Instrument (MI) end of the market, while others have developed from the television and broadcast side of the industry. A prime example of the genre is a furniture range produced by US manufacturer Middle Atlantic Products, based in Riverdale, NJ. Called the Edit Centre range, it was launched at the NAB Convention in 1999, and comprises a choice of modules including central desks and side-bay racks, as well as overbridge components that incorporate swivel mounts for close-field monitor speakers. The design of the units' work surfaces is based on interlocking curves, allowing a flush surface to be achieved, yet with the angle between the units freely adjustable through a wide range. The desk units include cable-feed-through holes and cable management facilities, with power distribution strips, while the side bays are available in single and twin-rack options. Reflecting good design principles, the Edit Centre's rack units are offered with a built-in ultra-low-noise cooling fan, plus gasketed front and rear doors for noise control, making them eminently suitable for housing hard-disk arrays. A brush grommet on the cable exit maintains the integrity of the seal for noise and air-flow purposes, highlighting the design's attention to detail. A choice of hard-wearing laminate finishes for the work surfaces should satisfy most studios' aesthetic requirements.

The manufacturer cites the explosion of PC and Mac-based audio and video editing systems, and the growth of small-format digital mixing consoles, as the main drivers behind its introduction of the Edit Centre. The principal design requirements were for a system configurable by the end user to provide the free-choice benefits of a custom-built design, but without the production delays and budget levels.
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dominantly at video applications. The selection encompasses equipment consoles, rack enclosures, file-server housings and media storage—including the Multimedia furniture range. This modular furniture can be supplied with main desk units in a choice of widths, with straight or curved desk-tops, and the desks can be fitted with height-adjustable overbridge risers. Side racks are included in the system, and the design incorporates multiple cable raceways to manage interconnections, as well as below-desk papertrays and CPU support bays.

Preproduction rooms and project studios may find an appropriate solution from the Raxxess products. Located in the US at Patterson, NJ, Raxxess markets a range of studio furniture based around four alternative centre-desk configurations, with free-standing side-rack options. Side wings are available to extend the basic systems in straight runs, or at an angle through the use of wedge-shaped connecting panels. The systems use powder-coated steel for the support structures, with laminate-finished desk-tops edged with a solid oak trim. Cable management facilities are provided, and the main desk systems incorporate retractable keyboard shelves.

Pacific Research & Engineering, of Carlsbad, CA—part of the Harris Corporation—manufactures PrimeLine Modular Studio Furniture, which incorporates design features from the company’s custom-built furniture. It features hardwood trims and a wide choice of laminate finishes. Designed primarily for radio and television broadcast studios, rather than being directly suited to music recording or postproduction applications, the PrimeLine is built up from centre-desk sections, with side racks and various heights of above-desk rack turret. PrimeLine can also be supplied in a choice of heights for seated or stand-up operation. The same manufacturer also markets the QuickBlit II modular system, which is designed to provide a fast-assembly alternative for radio stations.

In Europe, Swiss manufacturer Media Engineering, based in Weiningen, offers the MEFUR extendable furniture system. The modular system has been designed principally for applications in radio broadcast and preproduction studios, and enables straight or angled desk runs to be assembled. Options include overbridge shelving and rack housings.

The choice of furniture systems highlighted above serves to underline the validity of the modular approach. Custom-building studio furniture for aesthetic reasons, to fit with a particular interior design, will always remain the optimum option—given sufficient time and budget. Modular off-the-shelf systems, however, offer most of the benefits of custom-built studio furniture in a more accessible way and with far greater flexibility. Longer term, the benefits will be accumulative, as each subsequent change or adaptation can be accommodated with the minimum of expenditure or delay. Changing a room needs to be a swift and economically viable process. Modular systems can make this a practical reality, while ensuring that new hardware can always be located in the optimum position for comfortable and efficient operation.

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Above (left to right): HHB Circle 5 midfield monitor (active and passive versions available), HHB Circle 1 powered sub-woofer, HHB Circle 3 nearfield monitor (active and passive versions available).

Left: Mick Fleetwood with HHB Circle 5 active midfield monitors and Circle 1 powered sub-woofer.
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JOINVILLE LE PONT is regarded as the birthplace of French cinema as the site, some ten miles east of Paris, was chosen as the location for Charles Pathé’s operations at the beginning of the 20th century. It also serves as the location for the Audis de Joinville facilities complex established in the mid-eleveties. Changes in company ownership and direction more recently saw the creation of the Duran Duboi group which combined Joinville’s audio facilities with the Duboi digital special-effects operation and Duboi film studios, creating a potent force on the international and local film-making scene.

Having made increasingly ingenious use of available space at the Joinville plot the decision was made to build the Audis Boulogne facility complex in the Boulogne region of Paris. The original Joinville complex has seven rooms of which three are mixing rooms with two SSL 5000S and a Harrison Series 12 plus Foley rooms, recording studios, and a full complement of editorial suites and offices. The initial plan was for the Boulogne site to serve to handle spill over work but the opportunity to grasp true expansion by combining even more audio facilities with the advanced picture capabilities of Duboi will create a complex that will amount to a full service audio and picture facility.

Still being built and nearing completion it represents one of the grandest and most ambitious film-oriented builds on the planet.

Occupying some 5400m² of floor space over several floors, many of the audio facilities were already completed at the time of my visit including a huge mix theatre quite unlike any other in Europe. Measuring 14m wide and a 2500m² capacity it houses just over 8m of AMS Neve DFC in what is popularly regarded as the largest digital desk ever built.

Technical director and general manager John Rutledge laughs when asked if size really does make a difference. ‘I think that it is more a question of comfort for the engineers,’ he says. ‘We have big ambitions for this room and what we wanted was the comfort of having access to as many sounds as possible which is the way that it is all going. We’ve gone from 3-track 35mm a few years ago to 32-track workstations very quickly. The ability to have access to all the sounds at the top of the board with the possibility to go back to a previous layer is a great advantage.’

The desk has 120 physical faders in a 160 chassis with Rutledge stipulating in his original spec that the board should be the equivalent to 150 channels of analogue plus the requisite monitoring and all the other bells and whistles required for film. You could say that the desk could be made in to a 4-layer deep 480-fader console, he adds. ‘It’s a big beast and it has been very well received by the engineers.

This may seem like a surprising observation given that working film practices in France dictate that the producer brings his own engineer in tow, and editorial staff and talent are taken care of directly by the production company. Despite the fact that a number of other...
DFFs have recently also been placed in France; the odds would seem to be stacked against freelance engineers gaining early experience with the board. Rutledge counters by pointing out that he is in the 'service' business.

'The way I explain it to people is that we are not Renault we are Avis,' he continues. 'Joinville and I have always been technical facilities not a producer. It's a terrible glamorous analogy, but we are in effect a technical hotel. People check in and check out and we give them a service.'
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US: What's in a title?

As technology evolves, empowers and destroys established boundaries, new terms and titles need to be defined, write William Shakespeare and Dan Daley.

AND YOU THOUGHT the British were obsessed with titles? Think again. The country that brought you the Internet has developed its own obsession about titles, and it goes right to the root of who we are as Americans. In a WASP country the few words that define you to the world of business carry tremendous import.

It also has some significant implications for those of us in professional audio, as well. During a recent conversation, an old friend was trying to figure out what to call himself for a panel discussion he would soon participate in: 'I'm the president of my music production company, but that sounds generic and maybe even boastful, considering that I'm the only employee,' he said. I'm also a bass player, but musician sounds too thin. There's got to be something in the middle that doesn't sound too ridiculous or obscure, and that conveys a sense that my company and I mean something.

Filling in that blank in the middle has become, seemingly, a full-time job for a lot of people in audio. And the proliferation of technology has hindered more than helped, because if the Digital Revolution is anything, it is amorphous. Here are just a few examples of titular madness that has gripped the States:

Chief technology officer, chief technologist, chief webmaster-audio, senior vice-president of new media and related business development, chief web-site systems architect, audio & effects designer, director of online & video game audio, and (my personal favourite) media wrangler. The latter goes on, such one straining human memory and the capacity of a standard-sized business card.

This is explainable in two ways. First, the increasing complexity and amount of technologies—audio and otherwise—require more precise definitions as to what individuals actually do in regards to those technologies. Incredibly specific titles, as ponderous, pompous or pretentious as they may sound, do serve a purpose in that anyone who has to find exactly the right person in an organisation has a better chance of doing so. Just as we never needed the term 'analogue' until digital came along, titles define our sense of place in an increasingly fragmented and specialised universe.

Secondly, and less scientifically, the title becomes part of the person's identity, a somewhat sad commentary on the state of the American psyche, but an accurate one I think. When you are absolutely obsessed with your work—and we're famous for that—your title can often be viewed as an end in itself, defining someone not only professionally, but personally, as well, a surrogate for self-esteem.

But what this also underscores is the changing nature of the human asset in the pro-audio picture here. The simple term 'engineer' seems less and less valued by those who aspire to the position. At some of the larger professional audio schools in the US, as much as a third of the graduates never go into conventional recording studios, going off instead into bedrooms and basements with their own recording gear. They never have the chance to assign them a title—they invent their own, the anthropic 'producer' being the initial default setting for most of them.

I have a new favourite story about titles. I was recently talking with Brian Pollack, the manager of Trent Reznor's (founder of Nine Inch Nails) studio in New Orleans. When he arrived in New Orleans five years ago, he needed to set up a bank account. He asked Reznor's management company in New York which of the several holding companies that Reznor

Europe: Audio feat. The Tamperer

Simultaneously assuring the consumer of audio fidelity and the record companies of copyright security is proving increasingly tricky writes Barry Fox.

TAKING IT AS READ that no-one who cares about music condones counterfeiting, piracy, or home-dubbing over legitimate purchase. Take it as read, too, that it makes sense to make audio recordings to the highest possible standard, either in quad-rate PCM (176kHz or 192kHz and 24-bit coding) or Direct Stream Digital one-bit coding (at 2.8224MHz).

But the music and electronics industries are going to have the Devil's own job trying to sell the idea of a new quad-rate or DDS version of DVD, when CD already keeps most listeners happy and DVD already offers 5.1 surround. The idea of competing on the open market with two rival and incompatible systems, the 'official' DVD-Audio PCM system driven by Panasonic and the Sony-Philips Super Audio CD Bistream system, is simply certifiable. Mere mortals can only beggyle when the companies behind this madness talk of doctoring the music waveform to control copying, by using technology developed for copying Internet with mid-tier Audio CD.

But this is exactly what is now happening. I have already catalogued the extraordinary way in which the Secure Digital Music Initiative, 4C Entity test group and DVD Forum's Copyright Working Group chose the watermarking system developed by US company Ars—now known as Verance after merger with another US company Solana. As a reminder, though, Verance codes modifications to the analogue music waveform, which are so robust that they will withstand transmission by AM radio and over the Internet, to identify the source and trigger copy control in future players and recorders, just like SCMIS. But Verance was not tested with quad-rate audio and the tests at lower quality were run by record companies using anonymous golden ears. Talk of further tests, to be run by the UK's Phonographic Performance Ltd, has evaporated in silence.

Panasonic and Technics are now promoting '100kHz ready' hi-fi hardware, in the run-up to the DVD-Audio launch now scheduled for this autumn. Because DVD-A is the 'official' DVD Forum standard there has never been any doubt that the 100kHz system would be used with music that has a targeted waveform. Crazy, but true.

Last August, when Sony and Philips were showing SA-CD at the IFA exhibition in Berlin, Paul Reynolds of Philips pledged: 'We do not want to tamper with the waveform!' He reminded that data pits on the SA-CD disc are both visibly and invisibly patterned so that a player can reject a counterfeit disc. There was no such assurance from Sony because the company launched SA-CD at the Philharmonic Hall in Berlin with grand speeches from executives who then ducked the usual question and answer session. Foolishly we assumed that Philips and partner Sony were singing from the same hymn sheet.

Foolishly we assumed that Philips and partner Sony were singing from the same hymn sheet.

All of the five major world-wide record companies, BMG Entertainment, EMI Recorded Music, Sony Music
B EING A TECHNOLOGY writer is often like being caught in a time warp. I am privy to news of the latest developments sooner than others are, so by the time new technologies become generally available, it's like I'm going back in time. A number of issues are giving me this feeling at the moment, but none more so than digital radio. Despite pilot schemes, prelaunches and launches, this increasingly beleaguered technology is still struggling to assert itself.

In recent weeks I have noticed DAB and radio presenters using the words 'digital radio' as a kind of marketing tag-line. 'In stereo on FM, on the world-wide web and in digital, this... has become the newest mantra spouted by people desperately trying to talk up the news. (It may be significant that webcasting gets second billing above digital radio, but that's probably for another discussion.)

There is the argument that such buzzwording is superficial and superfluous—regrettably exactly the kind of thing that appeals in today's radio market. In reality, many radio transmissions have been digital, or at least had a digital source, for several years, regardless of DAB. CDs, MiniDisc and the increasing use of both DAWs and automation servers have created a digital infrastructure, the implementation of DAB multiplexes is the logical continuation and conclusion of the process.

There is still confusion about what digital radio exactly is and what it means for the listener. It has been strange hearing technology pundit being interviewed on the subject, because I feel slighted at not being one of them, but because I was giving similar interviews five years ago. Back then there was definite aggression towards the format, to the point where I had to tell one particularly obstreperous presenter that if he had a problem with digital radio he should take it up with his employers, not me.

Digital radio's cause has not been helped by the BBC charging ahead with launching pilot services at a time when the means of receiving their services was not available. There is still a problem regarding digital radio receivers: while units are now on the market, the majority are in-car models, home units are still only produced by a small number of manufacturers at the minute. As ever there is the issue of cost, although, as we always say, this will decrease as the technology becomes established.

More worrying is the format's quality has been questioned, not least by my esteemed colleague to the bottom left. When first demonstrated DAB was claimed to be of 'CD quality', this crept back to near CD quality and has stayed there. People listening to test transmissions were always disappointed by what they heard: shaking off my aloof technology writer status I understand why. It's new, it's digital, we're all going to have to buy new radios so why doesn't it sound absolutely amazing?

Audophiles have made much of this, augmenting their case with arguments used in the CD versus vinyl debate of the eighties: the sound lacks warmth and is too real. The case for the defence uses much the same logic. DAB, like CD, is robust and convenient, going some way towards eliminating interference and doing away with the need for jiggling the dial. It is also consistent, something that can only be said of FM on very good days with the best equipment.

The source of the problem can be traced back down the distribution chain: one of the things digital is doing for radio is making the transmission process less of a villain. Presenters, operators and engineers—the ones who are left—can no longer use the venerable radio excuse 'It's almost leaving me' because as complaints are beginning to show, it isn't.

Glyn Jones, managing editor of BBC Digital radio, explains: 'Because digital radio tends to be more transparent, we have to be more careful in the production process. Any faults show through more clearly, so there are several areas where care has to be taken now. We're aware of some problems in the use of SDI and where microphones aren't right for the job. There's a lot of slippage being heard at the moment and it's not pleasant. Unfortunately DAB is not forgiving of the production process.'

Jones comments that many radio producers and engineers working with analogue transmission have had to consider two formats, either FM and MW or FM and LW. Now, because there is no processing on digital signals, there is an extra chain to take into consideration. 'It is a case of throwing long-established production methods away and starting from scratch,' he says. 'It's a matter of altering the approach and taking more care.'

This may explain problems with speech radio, but not the instances cited by Gary Fox, where certain types of music sound tinny or muffled. The obvious explanation is the transparency claimed for DAB, that transfers the blame to the source material. I can hear the audiophiles saying 'Tell us all about it.'

Attacking the sibulant

Establishing acceptable levels of sibilation for digital radio may be more difficult than technology promised hisses

Kevin Hilton

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T he AMPEX ATR124 is, arguably, the most misunderstood tape machine in existence. Only 50 were manufactured and engineers feel that it is extremely complicated because it certainly was years ahead of its time with awesome specifications and facilities. I regard the 124 as the finest sounding tape machine ever built and today, with the correct updates and modern component availability, it can also be made to be one of the most reliable. I have had seven ATR124s go through my workshop—a few have kept—but all now give good reliable service. When transporting a machine, it is important to make sure that the centre of the machine is supported by a wooden block, placed between the wheels from front to rear because, if it is pushed down a step, or is dropped by just a few inches, the weight of its heavy transformer distorts the base of the machine.

The first task when refurbishing a 124 is to upgrade the two relays on the Power Failure Braking System (our Part Relays 1 and 2). The new relays are much larger than the existing models; so space has to be created to enable room to mount them. Please note that only one of the relays needs to be a 3-pole device; we supply one 3-pole and one 2-pole.

To make the modification, remove the panel from the power supply and strip the old relays, making careful note of the connections—note the path through the relay is not the solder position (take care to wire the new relays, the connection layout is not the same as the old). The top mounting lug has to be removed from the new relays. We do not give a template on new component relocation, but with the old relays off the panel, you can offer up the new relays and see how much space has to be made. The resistors and wire mounting post can be moved to new locations, the mounting holes redrilled, it is then possible to mount both the new relays.

Check with a low ohm meter all the wire wound resistors, paying particular attention to the heat sync type, also check both bridge rectifiers under the power supply [A1 and A2], inspect all the fuses on top of the transformer, look for any oxidising on the fuse holders. The fuse rating is 25A, and you should look for any blackening around the fuse holders.

There are two earth rails on the transformer PSU assembly, one is digital, the second is common ground, if these are not at the correct connection then you otherwise will get severe crosstalk problems effecting both transport and the audio functions. The correct complete part No Assy 405107-02 for this cable, is printed on to black sleeving on the cable itself, the spec for this cable is on page 8-14 in the updated service manual.

For a reliable 124, then, we advise that you check that the offset card assembly is changed to the latest auto offset card assembly 4051221 to reduce extreme capstan motor work by sensing capstan motor current and proportionately adjusting reel motor torques. (Note that modification to both the reel servo and capstan PWAs is required.)

The life expectancy of an EPROM is about 8-10 years if they are the originals, and so they should be changed as they are likely to be fragmented. We do blow new EPROMs and also supply the programmes on diskette for DIY clients. We always fit new EPROMs to fully refurbished ATR124s.

The current EPROMs are CPU PWA = U27 = 4570029-05, U26 = 4570028-05 and U25 = 4570030-05. The Control Panel = 4570026-05. The 100 Point Search to Cue = U14 = 133271-01B; U15 = 133271-02B; U16 = 133271-03B.

Before switching the machine on, remove all PWAs (Cards), the only PWA’s that should be installed are the Power Supply Regulating PWA, and the CPU PWA, switch on and check all voltages on the front of the regulator PWA with a scope and voltmeter, checking for any ripple. The next move is to install the reel servo PWA on an extender card. You should only add further PWAs to the machine’s control bay, making sure that the regulator PWA does not change its voltage at its various test points. Do not add any audio cards or audio control cards at this stage. If the power supply has not changed under the load of the PWAs being added to the control bay, or the resistors on the reel servo board do not show any >
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Then the pages will be Page 5–42 through to Page 5–54 inclusive.

It is important that the transport is running perfectly because the audio will not work correctly until the transport is perfect. Any problems in the transport will cause the audio to come out of record mode. Other problems with the transport will be caused through poor adjustment, fragmented EPROMs, faulty tachos, faulty LM381 on capstan tach PWA, faulty LED TIL31 on capstan LED PWA, faulty Photo Transistor on Capstan Tach Sensor PWA, faulty DS1 on Position Adjust PWA, and faulty Offset Adjust PWA.

When you are satisfied with the transport, you should now check your Audio Control PWAs. Install your extension PWA into the first Audio Control Position without the Audio Control PWA, check for 8 Volts on pin Nos 99 or 100 using pin Nos 95, 96, 97, and 98 for ground. If you have 8 Volts then fit the first Audio Control Card into the Extension Card. There are three bays, carry out this test on each. (Warning do not fit or remove PWAs while the machine is switched on.)

Next check that the 5 Volt regulators U49 and U50 are not faulty, also make sure there are no oscillations, the two taut decoupling caps C5 and C32, replace with new items. Carry this out on all three audio control cards. Some engineers recap everything, but we agree with the recapping in the right places as a complete recap is a long and expensive task. Before fitting the audio cards check, or replace C303, C310, C604 and C605 as these caps will most likely cause severe problems.

If any of the inductors L602, L603, L304 or L305 have been changed, chances are they have been changed to a higher current version. When any of the above listed caps goes short, the older type of inductor would go open circuit due to excess current to ground but any higher current inductors would place a short on your + or - rail causing the PSU (on that rail only) to go into shutdown mode. This then causes the transport to stay in your last transport command until the tape reaches the end or you switch the machine off.

Check that the relay screening plate on the side of each audio card has not come unstuck and made contact with its adjacent audio card causing a short. Only fit one bay of audio cards at a time, checking volts at the regulator PWA after you have added the 8 Audio PWAs. When you have completed the 24 cards if you have any transport problems such as the machine going into Record illumination only then when not requested then this problem can be fragmented EPROMs or a faulty U201 = TL084CN on the Record Timing part of the audio PWA.

Here are a few tips and warnings on various updates. The thing that concerns most owners is the lo-crosstalk heads.

The play head was never updated, although I do prefer the Flux Magnetic Replay Head for its extended frequency response (30ips 20Hz-50kHz ±2dB). The record head is the cause of the crosstalk problem and if you have the old type record-sync head Part No 4350372-01 or 4350372-02, then change it, because this was later replaced with an improved crosstalk head, Part No.4350752-01 or the new Flux Magnetic Type Record Head. We supply Flux Magnetic 124 heads on a 6-week turnround, ATR102 ⅜ or ⅙-inch heads are ex-stock. You can recognise the crosstalk head without removing it from the machine.

Common fault diagnosis can be found on the website: www.prostudio.com/studiosound

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Transforms and Spectra 3

In the third study of transforms and spectra, we note that square waves do not occur in nature and wonder why we test audio equipment with them. John Watkinson explains.

The Square Wave is a useful signal because it has some interesting properties. Fig. 1a shows an ideal square wave which is a waveform spending 50% of its time at each of two fixed voltages. Fig. 1b shows the spectrum of an ideal square wave that has a sin(x)/x shape. The spectrum passes through zero periodically at all even multiples of the fundamental, showing that a square wave has no even harmonics. The remaining odd harmonics alternate in phase.

Note that the spectrum of an ideal square wave is infinite. This is consistent with the time domain waveform of Fig. 1a in which the vertical transitions have a rise time of zero, requiring circuitry with an infinite slewing rate. As a result it is impossible to generate an ideal square wave and equally impossible to display one on a scope.

In the real world all we can do is to produce a band-limited square wave to display on our band-limited test gear. This is just fine because the whole object of the audio industry is to excite band-limited human ears. The band limit of human hearing has been established beyond question by countless scientific experiments, but periodically the limit is questioned by people who perform experiments which suggest otherwise. These results only show that the experiments concerned are unscientific, but finding the inevitable flaw in the experimental process is good sport.

A band-limited square wave lacks the frequency components that make it square and as a result it will contain what looks like damped oscillation leading to the term 'ringing'. Ringing is often said to be a defect of filters, but this is incorrect. In fact the ringing proves that the filter is doing its job of band limiting. It is possible to estimate the frequency response of a low-pass filter by examining the ringing.

Fig. 2 shows a system with an ideal filter cutting off at 20kHz bandwidth and passing square waves (or what is left of them) of various frequencies. With an 8kHz square wave input, the output (Fig. 2a) is a sine wave because only the fundamental is below the cut-off frequency. The third harmonic is at 24kHz and isn’t passed. With a 5kHz square wave (Fig. 2b), the third harmonic and the fundamental pass, whereas with a 3kHz square wave (Fig. 2c) the fifth harmonic can also pass. Thus by counting the number of bumps in each wave, the order of the highest harmonic to pass can be established.

From the above it should be clear that in an ideal 20kHz band-limited system the same output will result whether the input is an 8kHz sine wave or an 8kHz square wave, and this can be demonstrated with a suitably well-engineered filter. Repeating the experiment with a non-ideal device such as a mixing console or an audio power amplifier won’t give the same result. The high harmonics of the square wave will not necessarily be handled well by analogue circuitry designed for audio bandwidths. The thirteenth harmonic of 8kHz is 104kHz—approaching radio frequency. Audio devices are notoriously nonlinear at such frequencies because they have run out of loop gain without which the linearising effect of feedback does not work. Fig. 3 shows that in nonlinear circuitry, adjacent harmonics of the square wave, which are 16kHz apart, will intermodulate with the fundamental to produce a series of harmonics at 8kHz spacing, and one of these is at an absolute frequency of 16kHz and is clearly audible, giving the impression of second harmonic distortion.

Of course in the absence of the explanation of Fig. 3, the unscientists will claim that the audible difference between a sine and a square wave ‘proves’ that we have hot-line hearing.

The proper approach to high-quality audio is to ensure that no piece of equipment ever receives a signal it can’t handle precisely. As all real mixing consoles and power amplifiers perform poorly with radio frequency inputs, professionalism demands that at some point prior to such a device there is a low-pass filter that defines the system bandwidth.

The presence of this filter, with a frequency response such that it only

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affects signals well beyond 20kHz, it may actually improve sound quality because out-of-band input signals are unable to excite high-frequency nonlinearities. Of course, the filter must be of high quality, or the whole object is defeated. A number of high-quality power amplifiers use this bandwidth definition approach, and, of course, digital audio by definition has a very high bandwidth to prevent aliasing. Again, provided the filter is good enough, the fact that the bandwidth is defined is a further reason digital audio equipment sounds so good, where in comparison most analogue systems undoubtedly lack a defined bandwidth.

Square wave testing can also tell us about the low frequency response of a system. The flat parts of the waveform contain low frequency components and if the response is lacking the waveform will ‘droop’. Fig.4 shows the result of passing a square wave through a system with a high-pass characteristic. The constant voltage at the extremes of the square wave cannot be maintained. In practice a response down to DC is unnecessary. Loudspeakers can’t reproduce it and we can’t hear it. DC components reduce dynamic range, try voice-coils and cause thumps at edr points, so we can well do without them. In practice, a lower cut-off frequency of 6kHz seems to be a good compromise for analogue circuitry. A number of stages with such a response can be cascaded without audible detriment.

I have commented before that the audio industry’s worst failing is its fixation with frequency response, and I am not about to make the same mistake. Again the square wave is a very simple test for phase linearity and time response.

Audio systems should always display linear phase because this is the only way to avoid timing errors between different frequency components which damage realism. A system that is non-phase-linear causes what is called linear distortion of the waveform. In other words the waveform is changed without any harmonics being generated. As a result the spectrum remains unchanged in the presence of linear distortion because the same frequency content is present in both input and output. However, when a system is not phase-linear, different frequency components experience different delays.

The square wave shows up these different delays very clearly, whether or not there is any band limiting. Fig.5 shows that when, say, the third harmonic doesn’t line-up with the fundamental, the waveform is quite different. Returning to transform fundamentals, any circuit having a certain frequency and phase response will have a corresponding impulse response in the time domain. It is fundamental that when a system displays linear phase, the impulse response must be symmetrical. Thus when a square wave is band limited by a phase linear filter, the pre-echoing before the transition looks like a double-improved version of the post ringing after the transition. If this isn’t the case, the filter is not phase linear. If we can’t achieve phase linearity, at least the phase should change smoothly across the band. In conclusion, the humble square wave may not occur in nature, but it has a lot going for it as an audio test signal. As square waves contain a rich spectrum, they can test for frequency response at both extremes of the frequency range. They can also test for nonlinear distortion that shows up as the presence of even harmonics as well as being able to display at a glance any lack of phase linearity—not bad for one signal.

It would be misleading to suggest that square-wave testing is all we need, far from it. The audio industry still lacks a comprehensive set of audio test signals which will reveal what a device really sounds like, although there is no fundamental reason why this should be so. What can be stated with confidence is that if a device fails a square wave test, it will impair the audio quality. Whereas if it passes a square wave test it may or may not have other deficiencies.

That said, many everyday devices such as mic mics, mixing consoles and power amplifiers have quite respectable square wave response, unlike many microphones and loudspeakers which generally don’t. The reason generally given is that it’s not possible. This sounds to me like giving up before starting. In fact mics and loudspeakers with an excellent square wave response are perfectly feasible and only require some good engineering.

When two systems have the same frequency and phase response and the same percentage of THD, why can one sound different? In my view that’s putting the right way round. Why should they sound the same when THD only measures the amount of unwanted stuff in the spectrum and not its distortion?

Put it another way. Would you prefer to be hit on the head with a brick or the same weight of sand? Both weigh the same on the scales, but the subjective effect is quite different.

Fig.4: Droop shows poor LF response.

Fig.5: Linear distortion.
As managing director of CEDAR Audio, Gordon Reid has no objections whatsoever to increased audio quality.

Scene 1: THE NIGHTMARE

You drive a nice, sensible car. Nothing too flash, but it is well designed, reliable, not too greedy for fuel, and it gets you from here to there with a minimum of fuss.

The larger car manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require one of their newer and more expensive models. Sleek and sexy, these are capable of double the speed of previous cars, although they require a great deal more petrol to achieve this. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable vehicle (which has years of life left in it) and replace it with something newer and more expensive that, in essence, performs exactly the same task. Unfortunately, just as you fork out thousands of pounds-dollars-euros-shekels on your new car, the maximum speed limit in your neighbourhood plunges from 70mph to 7mph. It's a nightmare, but fortunately, it hasn't happened.

Scene 2: The Nightmare (Part 2)

You work in the pro-audio industry. You have a studio with a good selection of equipment. Nothing too flash, but it's well designed, reliable, and it gets the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require several of their newer and more expensive units. Sleek and sexy, these handle double the amount of data as previous models, although they require a great deal more processing power to achieve this. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable equipment (which has years of life left in it) and replace it with something newer and more expensive that, in essence, performs exactly the same task.

Unfortunately, just as you fork out thousands of pounds-dollars-euros-shekels on your new equipment, the audio delivery format in your neighbourhood plunges from 16-bit, 44.1kHz to 128kbit MP3. It's a nightmare, unfortunately, it's happening right now.

Scene 3: A Dream Within A Dream

You are an engineer within a pro-audio equipment manufacturer. You have developed a good selection of equipment. Nothing too flash, but it's well designed, reliable, and it gets the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade them that they require a new audio data format. Sleek and sexy, this format requires double the amount of data space as the previous format, and it requires many times the processing power to manipulate it in the same ways as before. Your customers would prefer to spend their money on something else, but they think that they have no choice. So they dump their dependable, reliable equipment (which has years of life left in it) and demand that you provide significantly more powerful hardware platforms that in essence perform exactly the same task.

Unfortunately, just as you fork out thousands of pounds-dollars-euros-shekels on your new development projects, the audio delivery format in your neck of the woods plunges from 16-bit, 44.1kHz to 128kbit MP3. Never mind. Your customers will still cough up for the replacements whether they need them or not. It's a dream. And it's happening right now.

Scene 4: Real Audio

You are a consumer who listens to music. You have a number of music systems comprising a good selection of equipment. Nothing too flash, but they're well designed, reliable, and it gets the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require a new audio data format. Sleek and sexy, this format is incompatible with the previous format, and it requires new development to play it in the same environments as before. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable hi-fi equipment (which has years of life left in it) and replace it with newer, more expensive systems that in essence perform exactly the same task.

Unfortunately, just as you fork out thousands of pounds-dollars-euros-shekels on your new equipment, the audio delivery format in your neck of the woods plunges from 16-bit, 44.1kHz to 128kbit MP3. Never mind. You're buying the latest things whether you need it or not. And you're doing it right now.

Scene 5: Unreal Audio

You are a Director of Really Quite Big Audio Inc. You have a sleek, sexy new car funded by the sales of new, high-sample-rate audio equipment. While driving, you listen to MP3s of your favourite music. As you accelerate past 100mph, you grin. You are well aware of the benefits of the 24-96 masters from which they were derived.
"REASONS NOT TO BUY A MACKIE D8B...ZERO."
—Roger Nichols, EQ Magazine

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1 FREE UPGRADE! NEW OS 3.0 ADDS OVER 30 NEW FEATURES!

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3 1999 TEC AWARD WINNER!

Normally we don’t name competitors in our ads. But in this case, Mix Magazine published the other nominees for the 1999 TEC Award for Outstanding Technical Achievement in Small Format Consoles: Allen & Heath’s G3-3000, Digidesign’s ProControl, Panasonic’s WR-DA7, Spirit’s Digital 328 and Yamaha’s O1V. Thanks to all who helped us win this prestigious award.
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- Third Stage Reflected Plate Amplifier Tube Circuit Discrete Class A Impedance Balanced Output Stage

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