TASCAM MX-2424
Does this change everything?

BROADCAST AUTOMATION: HANDS-OFF CONTROL
FUTURE COMPUTER: THE EVOLUTION OF THE PC
SURROUND SOUND: MIKING FOR SURROUND
HEADPHONES: CANS IN CLOSE-UP

The STEVE POWER Interview
Forgive and regret

OUR LOOK AT THE TASCAM MX-2424 serves to remind how delicate the relationship is between intermediate and milestone technologies. As someone who can remember DASH and ProDigi being dismissed in the 1980s as intermediate technology I could be forgiven for waiting for this machine for what has been an eternity.

Of course 'intermediate' is a word that is frequently confused and misused by those of shallow pocket who then proceed to power out of their personal handshake turns as soon as they can afford to. Digital tape was, and is, an intermediate, but it hasn’t stopped DASH representing the high-end interchange format in music recording or the DA-88 serving as an established post mix medium for long enough to pay off the financing deals for shrewed operators many times over.

What wide-scale technology adoption patterns all have in common is that they either facilitate efficiency and convenience or they are simply so cheap and simply make too much sense to ignore. The acid test of business plan justification and qualification generally throws up a lowish risk, no-brainer decision that slots in seamlessly alongside existing systems and work methods that don’t immediately have to be consigned to scrap.

The MX-2424 is not the first to offer affordable high track numbers on hard disk, but it remains significant in doing it quite so well for quite so little. We live in an increasingly disposable society and a major attribute of many recent ‘milestone’ developments has been a lack of emotional involvement and dependence for the user. It’s why we can recount and indulge ourselves in the finer points of analogue master tape machine ownership in this same issue despite the fact that we are unlikely to ever enjoy a similar attachment to the early incarnations of DTRs or, for that matter, early attempts at DAWs.

That change continues is not the surprise and not the issue, that is discards it predecessors, themselves milestones in their own day, quite so quickly takes an adaptation of attitude and a hardening of outlook. Increasingly we are being stripped of the technological crusts that have previously tied us to the world. We are left instead with our own transferable crafts and skills as the truly salable commodity. The most important thing you take into the room is what you have in your hand.

Zenon Schoepe, executive editor

Covering fire

COMPARING MUSICAL NOTES with Philip Newell in Spain recently threw up an interesting difference of opinion. Where I had always regarded the term ‘cover version’ to apply to any reworking of another artist’s song, no matter how greatly it differed from the original, Philip considered it applicable only to a fairly faithful copy. To illustrate his point he quoted Cilla Black’s ‘Anyone Who Had a Heart’ as being a legitimate (read ‘near identical’) cover of Dionne Warwick’s original, while Joe Cocker’s interpretation of the Beatles’ ‘With a Little Help from My Friends’ was another proposition entirely. I had to agree that he had a point.

Returning home, I chanced across a copy of Ann Harrison’s book Music: The Business (Virgin Publishing, ISBN 0 7535 0433 2), whose glossary contains the following: ‘Cover version: a song written and recorded by someone else, that an artist records a new version of.’ As the book is subtitled The Essential Guide to the Law and the Deals, I thought Philip should know...

My email prompted a fax reply that closed ‘Joe Cocker took a weak insipid ditty… and rearranged it into a monstrous rock classic which wiped the floor with the original. That, to me, is not a cover version but a huge development of the song—a new original interpretation’. Philip was still sticking to his guns.

So whose definition of ‘cover version’ were the scriptwriters of The Patriot and U-571 working to when they decided to rewrite history? For those not in the know, in telling the story of the American war of Independence, The Patriot (directed by Roland Emmerich and starring Mel Gibson) inaccurately casts the British as a murderous bunch, massacring their enemies in the manner of the Nazis. U-571 (directed by Jonathan Mostow and starring Matthew McConaughey, Bill Paxton, Harvey Keitel and Jon Bon Jovi), meanwhile, transfers the credit for the capture of the first German Enigma machine and code books from the British to the Americans during WWII.

In this case I defer unconditionally to Philip’s view; the films might be cast as the development of a theme, but a good, old-fashioned cover version would have been infinitely preferable.

Tim Goodyer, editor
Great Studios Of The World

Since installing the first SL 9000 J in France, in 1995, leading international studio Guillaume Tell in Paris has gone from strength to strength. "We are very proud that for five years now, we have had the pleasure and satisfaction of contributing to the success of the 9000" said owner Roland Guillotel.

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has just

Midge Ure

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

AMS Neve is the first customer for the new DPA
Type 4014 large-diaphragm mic. In the three
months since their arrival, Philip
Hurst has used the 4014s to complete
more than 20 classical recordings,
beginning with a little-known piece
by Beethoven for choral and orchestra.

With the 4014s as its main pair
the anticipated 90-minutes balance
was reduced to 15 minutes.

DPA, Denmark.

www.dynamicrophones.com

German postproduction facility
Voodoo Lounge Medienproductions
has installed two Martinsonsound
Multi-Max monitor controllers at its
Graminu premises. The controllers are being
used to control a Quested sound system
and Yamaha NS-10 close-field
in Studio A and Genelec 1032A
subwoofer surround system
in Studio B. Both studios use DiGiDesign
Pro Tools 24 workstations and Pro-
Control control surfaces. Voodoo
Lounge specialists in the creation of
on-air advertising and promotional
spots in surround sound. Two MultiMAX
units have also been installed in
identical film mixing studios at Soundtracks
in Lanz. Austria. Both rooms are centred
on Digidesign 24 Pro tools systems
and use Genelec 1037B surround sound
monitor systems with 1094A subwoofers.

MartinSound, US.

www.martinsound.com

Slovenian state broadcaster RTV Slovenia
has taken a fourth SSL Aysis Air
console for installation in a new TV
OB vehicle. The purchase follows the
earlier installation of an Aysis Air into
radio OB vehicle and Aysis Air in RTV
Slovenia’s television headquarters and the
radio control room at the Cankarjev
Dom international concert hall.

SSL, UK.
Tel:+44 1865 842300.

Baltimore-based facility Tonal Vision
Productions has purchased a Fairlight
MXF3plus workstation for use in the
creation of original sound for picture.
One of the Baltimore area’s most respected
and busiest sound-for-picture facilities,
Tonal Vision specializes in work for
television shows.

Fairlight, Australia.

www.fairlightsps.com

UK: Construction is under way on a new London
recording complex, Sphere Studios. The
ambitious music recording complex to be built in
the capital for many years’. The new studio hopes to
rival London’s finest and will offer three main
control rooms dedicated to
tracklaying, stereo mixing and 5.1 surround mixing. There
will also be six white rooms available for hire to
artists and independent producers, plus extensive
ISDN facilities, plenty of parking and a licensed bar.
Behind Sphere are musicIan and
producer Francesco Carmeli and Malcolm Atkin,
formerly general manager of Sir George Martin’s
Air Lyndhurst facility. The design of the main studios
and white rooms is being handled by Munro
Assoclates while Carmeli and Atkin, with
the assistance of Sphere’s technical manager Tom Schlum,
are currently finalising the technical specification for the
central suite. Giordana says: ‘In order to develop the
music industry on a global basis there is a genuine
requirement for recording facilities that can offer
the most up-to-date technical solutions.

With Sphere we are in a new-build situation and therefore
we have the opportunity to develop a facility that is truly
designed for the future. The construction of the complex and the
equipment we install will reflect our intention to offer
capabilities that far exceed those currently offered in the UK. By
September 2000, all six white rooms will be up and running,
with the main studios opening in Spring of 2001. Pictured are
(L-R) Francesco Carmeli, Tom Schlum, Malcolm Atkin. Sphere
Studies, UK. Tel: +44 20 7924 4550

www.prostudio.com/studiosound

Studio Sound
Studio Sound www.prostudio.com/studiosound

audio@bbc
THE IBC'S ORGANISERS have commissioned Studio Sound to produce an audio only supplement to raise the profile and branding of audio at its forthcoming Convention to be held at the RAII Amsterdam, 8-12 September.

"Audio has always been a key part of IBC," said IBC's Mike Crimp. "We look forward to working with Studio Sound to draw people to the audio hall to see the great technology now available to them."

The supplement, called audio@bbc, will contain audio news, comment and analysis alongside audio specific floorplans and listings and will follow the popular and successful formula established by Studio Sound in its production of the AES Daily News at the European AES Conventions.

"Audio@bbc represents something of a result for the cause of audio at this major international broadcasting event," said Studio Sound executive editor Zenon Schoepe, veteran of numerous AES and IBC Dailies who will edit the audio@bbc publication. "The IBC is demonstrating its understanding of the importance of audio in the future of new formats and delivery methods."

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Email: c.bastie@selectbusinessmedia.com

Webcasting
France: A collaboration between IDT, leading French manufacturer of broadcast sound-processing technology, Internet broadcaster TV-Radio.com subsidiary of Télédiffusion de France served the French International Radio Show with a broadcast quality webcast. Using a prototype of its forthcoming Digital Virtual Processor Net unit, IDT delivered around 50 radio channels over the Internet to the visitors at Paris Palais des Congres. When released the processor will feature functions specially adapted to high-quality webcasting and is claimed to be the first Internet processor to abandon the multiplex working in favour of FFT.

Custom 'fex
UK: Following design projects for the BBC World Service, Channel 4 TV, Danish Radio, Northern Telecom and the Finnish Parliament, Sonifex is offering bespoke design and manufacture services alongside its current portfolio of proprietary products. Julian Speed, export sales manager for Sonifex commented: "Much of this nature has generally been undertaken as a result of clients or distributors approaching the company direct. Some customers, therefore, are either not aware of the design and manufacturing services we can offer, or have looked elsewhere and found the process to be too time consuming and expensive. As we have already been successfully but informally offering this service for sometime, the logical next step is to formally market it alongside our current product portfolio."
Net: www.sonifex.co.uk

Super DVD
Germany: Syninx Music & Media, Audionet (www.audionet.de) and Frequentzwerkstatt have jointly announced the release of the Super DVD recording of Bertold Hamann and
t Quintet (Audionet Standards Nr. 1) at Frankfurt's High End 2000 show. This is claimed to be the first DVD to meet both the high-resolution DVD-A standard as well as the DVD-V standard. The disc can be read in an ordinary DVD-A player, with the consumer able to choose between (60kHz/24-bit stereo, 48kHz/16-bit stereo, or 5.0 Dolby Digital multichannel! The same disc in a DVD-A or universal player also offers 192kHz/24-bit stereo or 48kHz/24-bit 5.0 multichannel. The recordings were made in the Milchkekenmusk in Hamburg, with Syninx in charge of recording, audio post-production, screen design, DVD pre-mastering and providing the central elements of the production system. Overall production was in the hands of Frequentzwerkstatt. Since the new format is compatible with the old DVD-V standard, the audio market segment can be served without additional replication costs.
Net: www.syninx.de

Logical development
US: Dolby Laboratories has unveiled Pro-Logic II, narrowing the gap between the original Pro-Logic matrix developed during the mid eighities and digital 5.1-channel systems. Pro Logic II lets consumers enjoy Pro-Logic programmes with a convincing 5.1-like presentation. Claims Dolby, with its eye on the Pro-Log catalogue, Pro Logic II is intended to convert conventional stereo recordings into a natural, believable surround experience.

This system was invented by Jim Fosgate, who has been involved in surround decoding technologies since the late sixties. Fosgate said, "I have spent the past 25 years figuring out how to expose the hidden information in two-channel stereo recordings, both new and old. This breakthrough in matrix decoding technology allows users to enjoy all their existing 2-channel programmes, whether Dolby Surround encoded or not, with an enhanced level of spatiality and directivity! Pro-Logic II decoding can be implemented in analogue and digital circuitry intended for use in home cinema and new music surround products.

A Czech Republic: The last CEDAR Series 2 unit has been purchased by Sound Studio (pictured) at the Music Faculty at the Academy of Performing Arts in Prague to work alongside its DC-1 De-Clicker. The popular Series 2 line is being discontinued after six years in production as some of the hardware components are now obsolete, leaving CEDAR to regard it as no longer practical to build more systems but to retain remaining component stock to ensure support for existing units. Series 2 users include The US Library of Congress, the British Library National Sound Archive, Abbey Road Studios, Skywalker Sound, Warner Hollywood, Disney, and TV and radio stations in Russia, Indonesia, Egypt, and Botswana. CEDAR Audio, UK. Net: www.cedar-audio.com

> US-UK: A new western starring Jackie Chan is set to become a modern classic. The film mix for Shanghai Noon was undertaken by Andy D’Dario at Buena Vista Sound on an AMS Neve DFC console who took particular pride in the finale fight scene wanting 'to make the sound of every weapon distinctive'.

Randy Edeleman's score was engineered at London's Lansdowne studios by Steven Lewis also on an AMS Neve desk, this time the analogue VX572.

> August 2000 9
August
23–26
BIRT’2000
China International Exhibition Centre, Beijing, China.
Contact: P&O Events.
Tel: +44 (0)20 7370 8231.
Email: shanghai@eco.co.uk

September
4–7
ICEP 2000
Philippine International Convention Centre, CCP Complex, Roxas Boulevard, Manila, Philippines.
Contact: Overseas Exhibition Services Ltd.
Tel: +44 (0)20 7862 2090.
Email: philippines@montnet.com

8–12
IBC
Amsterdam. The Netherlands.
Contact: International Broadcasting Convention.
Tel: +31 20 710 7100.
Fax: +31 20 711 7530.
Email: show@IBC.org
Net: www.IBC.org

10–13
Plasa
London, UK.

16–20
Cinec 2000
MOE Events Centre, Munich-Freimann, Germany.
Contact: Messe München.
Tel: +49 89 94901.
Email: info@messe-muenchen.de
Net: www.messe-muenchen.de

22–25
109th AES
Los Angeles Convention Centre. Los Angeles, California, US.
Contact: Chris Plunkett.
Tel: +1 212 661 8528.
Email: 109th_exhibits@aes.org
Net: www.aes.org

October
17–18
Broadcast India 2000
Central Centre, 1 World Trade Centre, Mumbai, India.
Contact: Kavita Mehta, Saicom Trade Fairs and Exhibitions.
Tel: +91 22 215 1396.
Fax: +91 22 215 1269.
Email: saicom@bbm2.von.net.in
Net: www.saicom/bbm2/von.net.in
broadcastindia

November
7–10
Satis
Paris, France.

8–9
Sound Broadcasting Equipment Show
Hall 7, NEC Birmingham, UK.
Contact: Point promotions.
Tel: +44 (0)138 323 700.
Fax: +44 (0)138 323 780.
Email: info@pointpromos.co.uk.

8–10
Replicite
Hong Kong, China.

17–19
Reproduced Sound 16
Stratford Victoria Hotel, Stratford Upon Avon, UK.
Contact: Institute Of Acoustics.
Tel: +44 (0)1727 848195.
Fax: +44 (0)1727 855053.
Email: ioa@ioa.org.uk
Net: www.ioa.org.uk

24–27
21st Tommeister Tagung
VDT International Audio Convention
Hannover Congress Centre, Hannover, Germany.
Contact: Gesila Jungen, VDT.
Tel: +49 (0)203 323 598.
Fax: +49 (0)203 215884.
Email: vdt@tommeister.de
Net: www.tommeister.de

2001

February
3–6
Middle East
Broadcast 2001
Bahrain International Exhibition Centre, Bahrain.
Contact: Bahrain Exhibition Services.
Tel: +973 (0)207 8622 2046.
Fax: +973 (0)207 8622 2049.
Email: mebroadcast@montnet.com
Net: www.aeminfo.com.bh

13–16
Memrex 2001
Dubai World Trade Centre, United Arab Emirates.
Contact: Andy Drew.
International Conference and Exhibitions.
Tel: +44 (0)1442 878 222.
Fax: +44 (0)1442 879 999.
Email: andy@iec-ltd.denmon.co.uk

22–24
Broadcast Thailand 2001
Queen Sirikit National Convention Centre, Bangkok, Thailand.
Contact: Adam Ridley.
Overseas Exhibition Services.
Tel: +66 (0)207 7862 2069.
Fax: +66 (0)207 7862 2068.
Email: thailand@montnet.com
Net: www.besmontnet.com

May
12–15
110th AES
RAI Conference and Exhibition Centre, Amsterdam, The Netherlands.
Contact: AES.
Tel: +31 20 661 8528.
Email: 110th_exhibits@aes.org
Net: www.aes.org

June
19–22
Broadcast Asia and CommunicAsia 2001
Singapore Expo. Singapore.
Contact: Overseas Exhibition Services.
Tel: +65 (0)207 7862 2080.
Fax: +65 (0)207 7862 2088.
Email: broadcastasia@montnet.com
Net: www.broadcastasia.com

20–23
Musikmesse
ProLight & Sound
Lentexpo Exhibition Grounds, St. Petersburg, Russia.
Contact: Hubert Demmier, Messe Frankfurt Trade Fairs.
Tel: +372 2 570 726.
Fax: +372 2 233 800.
Email: hubert.demmier@messefrankfurt.de
Net: www.musikmesse.de

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United Business Media
AIR Studios, London, the brainchild of legendary Beatles Producer, Sir George Martin. The Focusrite ISA110 Microphone Preamplifier and Equaliser was developed in 1985 for the custom Neve console, which was reinstalled at AIR Lyndhurst Studios in 1992. It went on to become the most popular outboard mic-pre and equaliser in leading studios worldwide.

Focusrite have now reissued this industry reference product as a Limited Edition rack-mounting device and are proud to have presented Serial #001 of the Limited Edition to Sir George in recognition of this unique association with AIR Studios.

To check availability of the ISA 110 Limited Edition re-issue, please contact us, or call your local Focusrite Dealer.
Old gold

I especially enjoyed the June edition of Studio Sound because of the tribute to Dick Sweetenham and the excellent review of the Helios EQ1 by George Shilling.

In that article reference is made to Tony Arnold’s design being based on the early modules of Eric Clapton’s console. Interestingly, that particular console is alive and well and working in a busy ‘vintage’ studio in South Norfolk. We restored the console and it has been worked hard for four years now without a single fault.

I owned two Helios consoles with identical mic-EQ modules the other modules—there modules I sent to Tony Arnold could have belonged to either console—sorbid to an anorak. How about a follow-up article showing the old desk as it is now, working well and sounding fab!

Kevin van Green

Masterworks

In reply to the article ‘Masterclass—Tape Heads’ (Studio Sound, March 2000) and the remarks of Mr. Summers in Letters (Studio Sound, May 2000), we would like to point out that Studer still makes and delivers original heads for all its 2-inch, 24-track machines. Studer even offers original heads or replacement heads for most of the older machines and for all standard formats (from ¼-inch to 2-inch) like the Studer 162.

Unfortunately, the article and the chart in the article contained some misinformation that found its way into it due to a misinterpretation at Studer. We would like to clarify this situation and state the correct details. 2-inch, 24-track heads are available for all machines and their versions: from the Studer A80 Mk. II to Mk. IV, A800 Mk.J to Mk. III, A820 to the Studer A857, including the Gold Edition.

2-inch, 24-track heads:

<table>
<thead>
<tr>
<th>Machine Type</th>
<th>Record Head</th>
<th>Reproduce Head</th>
</tr>
</thead>
<tbody>
<tr>
<td>A80 Mk.II, Mk. II, III</td>
<td>1.317.184.00</td>
<td>1.317.185.00</td>
</tr>
<tr>
<td>A800 Mk.I, Mk.II, Mk.III</td>
<td>1.317.380.00</td>
<td>1.317.385.00</td>
</tr>
<tr>
<td>A820</td>
<td>1.318.780.00</td>
<td>1.317.785.00</td>
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<tr>
<td>A827</td>
<td>1.318.780.00</td>
<td>1.318.785.00</td>
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</tbody>
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Note: The Studer A80 Mk.I was never built in a 24-track version.

The article states that the exact replacement for 1-inch heads may not be available anymore. In fact, most of them have been replaced by the 1.317... series of heads offering much better performance and less wear. It is therefore possible to improve the performance of an older machine by mounting the new 1.317... heads. The cleaning fluid which should be used for the heads caused another question. Owners of original Studer heads are allowed to use both, methylated spirit as well as isopropyl alcohol (IPA), for cleaning heads. There are no restrictions like with some of the non-original head designs.

Any further queries are welcome at: Studer Professional Audio AG, Email: nicolas.boehmer@studer.ch Tel: +41 1 870 75 11.

Fairy stories?

I am the author of many books, several dealing with the unexplained such as Children that Time Forgot and Mystic Forces. I am now researching my new book that looks at Elementals, creatures from a different sphere of existence: commonly referred to as fairies, elves, and so on. I have collected some fascinating stories, for example David, a perfectly sober normal man, was walking along a riverbank one sunny afternoon, leading two horses back to his stables, when he heard a distinct voice coming from the stream. He looked down and to his amazement he saw a tiny man (about a foot high) standing on a stone in the water, dressed in trousers, boots and shirt, remarking in an ordinary manly voice that there had been no fishing that morning. He continued about the rain and how he hoped it would be better the next day and how were they expected to survive without good fishing. He then added that he shouldn’t complain as he wasn’t bad off as some of the others! David watched him turn round as if looking out for something but the moment he made eye contact with the entity it vanished. The horses were disturbed and remained agitated until they were safely back in their stables. I would love to hear from you if you have had any similar experience.

Mary Harrison, 12 Thristlestone Road, Northampton NH4 8HD.

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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.

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.

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.

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source code: RN22

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Tascam MX-2424

Offering possibly the first genuine step beyond the digital tape machine Tascam's new hard-disk recorder-editor attracts the attention of Rob James

The Pro Audio Landscape usually changes slowly and steadily. Incremental improvements mark progress that often only really becomes apparent over a period of years. Occasionally, however, something comes along that really changes the world, almost overnight, eclipsing other aspirants to the revolutionary product crown. Tascam has turned these out before: the DA-88 and DTRS format ushered in the age of affordable professional digital recording. Alesis ADAT had the same effect in some areas, but it is DTRS that became common currency for programme interchange in film and sound-for-picture. It is just possible Tascam can repeat the trick with the MX-2424. The machine alone is impressive enough to warrant close attention, but the implications inherent in the control and networking capabilities make it potentially astounding.

The MX-2424 is a 24-bit, 24-track hard-disk recorder-editor when run at 44.1kHz or 48kHz, 24-bit, 96kHz operation halves the number of tracks. The machine comes with an internal 9Gb SGS hard drive, 2-channel digital I/O, sensible audio interface options and a range of sync connections. Unlike the earlier MMR-8 Dubber, it uses a dedicated chipset rather than a PC motherboard and is ready to use less than 1s after switching on.

Two distinct modes of operation are supported. In TL-Tape mode, the MX-2424 functions like a tape recorder. Recordings are destructive and there is no undo function. When a project is created in TL-Tape mode, an empty audio file is created for each track for the entire project length. Recordings and punch-ins to pre-existing recordings edit the relevant audio file(s). If tape-style working suits the job in hand, there are several advantages: the disk space is fixed so that if there is enough space to create the required length of project, you know you won't run out. There is also virtually no waiting for housekeeping after intensive punching in and out on 24 tracks. Non-destructive mode allows for changes of mind via the undo and redo keys. All recordings are kept even where they are entirely overwritten and effectively orphaned. Repeatedly re-recording the same section over 24 tracks rapidly uses up disk space and multiple consecutive punch-in and out over 24 tracks will result in a subjectively lengthy wait for housekeeping once the transport is stopped.

There can be up to 999 tracks per project. Any 21 of these can be played back. Tracks can be freely loaded and unloaded even between projects. Present the files recorded by the machine are SDII (Digidesign Sound Designer II) on Macintosh formatted disks. The alternative of Broadcast WAV files onto PC FAT32 formatted disks should he available soon. OMF support for exporting projects to Pro Tools is also being investigated.

The EDL format used by the MX-2424 has been dubbed Open TL or Open Track List. Tascam is making this available to any manufacturer who wishes to use it. Projects can be backed up to a variety of supported media including DVD-RAM disks and Travan tape streamers.

Although the MX-2424 will function perfectly happily as a stand-alone machine, the ViewNet Audio MX software package offers easier setup, provides a moving track display, enhanced editing capabilities and can control up to 65,000 machines across a network. Upto 92 machines can be connected together via the TL bus for sample accurate sync and a possible total of 768 tracks. There is also a remote control, the RC-2424 that duplicates all the front panel keys and adds dedicated editing keys and a macro function.

There are ten banks of menus to navigate, although only nine are currently in use. Setup accesses the setup and configuration menus and banks may be accessed directly by pressing a number key. Individual menus are found by using the Up and Down keys or by scrolling with the jog wheel. To change a parameter you press 'F3' adjust the parameter using the Up and Down keys or jog wheel. The shuttle ring moves the cursor where appropriate. Changes are confirmed by pressing 'F4' - it sounds more complicated than it is. I found it reasonably intuitive, but much preferred using the pull-down menus in the ViewNet software.

Although MX-2424 is clearly intended to be used with a mixer there are useful input
Routing options. The 2-channel input can be routed to record tracks in blocks of eight. So for example, Tracks 17-24 might be fed in odd-even pairs from this input while Tracks 1-8 and 9-16 are fed from one of the multi-channel option cards. Each block of eight tracks can either take the same numbered analogue or digital inputs—Tracks 1-8 take keys in a small area the shape and spacing makes it easier to hit the ones you want. All the keys have white legends printed on them indicating the primary function. Where there is also a Shifted function its white legend is beside the key. No keys are internally lit, but there are adjacent LED indicators where appropriate.

Front Panel

For a 24-track machine, the front panel is not particularly busy, although many keys have Shifted functions. The big meter display is eye catching with 24 corresponding triangular track keys below it. Sixteen-segment bar graphs follow the usual Tascam pattern with only the 0dBFS segment coloured red. If three or more consecutive samples hit zero, the red blinks to indicate record ready, going solid in record. The triangular keys are not simply eye catching, they aid the ergonomics. With many

Four blocks of five LEDs indicate various parameters:

- Sample rate: 44100, 48000, 2X, pull down, NON Standard D.
- Time code: 30, 29.97, drop, 25, 24.
- Record mode: 24-16, tape mode, DigiTal InPut, 2 Channel - O, auto input.
- Sync: Sample rate, Chase Lock, TL-Bus Master, TL-Bus Slave, TC Chase.

A further block of four LEDs to the right of the meters indicates:

- Status of: Error, Busy, MIDI and Disk.
- The round Edit keys are (Shifted functions in brackets):
  - CUT (LOCAL CUT), COPY (SPLIT), CLEAR (DELETE), PASTE (SYNC PASTE), INSERT (SYNC INSERT), OPEN (IN/OUT), UNDO and REDO.
  - Two rows of transport related keys sit right of the drive slot, top row:
    - ON/NO-LOOP (FAST), TO (PREVIOUS), IN (OFFSET) and OUT (PLAYHEAD).
  - Larger, tape recorder type transport keys are below.
  - The concentric scrub-shuttle wheel has a semicircular halo of shaped keys.

Clockwise these switch:

- JOG-SHUTTLE, NUDGE (CAPTURE EVENT), TRIM, INCREMENT, DECREMENT, SETUP (TEMPO), PROJECT (NEW) and VIEW (UNLOAD).

Above scrub-shuttle the locator section has a numeric pad plus CLEAR-CANCEL, CAPTURE, SHIFT, and the all-important STORE-YES and RECALL-NO keys.

The 2-line, 40-character LED is used for operator and system messages. When menus are not being accessed it shows the playback position in the top line and entry register time values at the bottom.

The final item of interest is a "Smart Media" slot. This is used to update the operating system.
In today's rapidly evolving media landscape, confidence in new technology has to be earned. With the abundance of equipment being introduced, can you depend on your supplier, the product reliability, the life-span? Can you know if you've allowed for all the possibilities of new mix formats, digital input/output configurations and new standards of automation which may appear without warning?

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Apart from the normal tape recorder controls there are a number of autotlocator functions. Capture places the current location of the play head in a register. From here it can be used to load the In, Out or Offset registers or be stored in one of the 100 locator memories. Auto Punch uses the In and Out points to set punch-in and out points for rehearsal and recording. When Auto Unload is set to on each loop the record cycle creates a virtual track automatically.

From panel edit functions are all based around the concept of setting In and Out points, selecting the track(s), and then performing edit functions. Clear removes the selection to the clip-board without slipping subsequent audio. Cut does the same, but slips all subsequent audio on the track or tracks up. Local Cut is similar, but only audio from the same event is slipped up. Copy and Paste work as expected, with Paste overwriting any existing audio after the In point for the duration of the pasted audio. Sync Paste uses the position of the play head in the source to define a sync point. In this way the peak of a jet by or similar can be hit without calculation. In 10.SW moves the section between the In and Out points to the current play head position, overwriting any existing audio.

Insert slips all subsequent audio. Sync Insert works in the same way as Sync Paste, but slips subsequent audio. Open inserts silence, slipping subsequent audio. Reverse is an offline DSP function that reverses the selected audio. Render creates a single audio file from whatever multiple events are between the In and Out points. All edits, punch-in and out use are real-time crossfades. The length can be set from a menu anywhere between 0-90ms. Using Vienover more sophisticated editing is possible including level and fades in real time. There is a great deal in this machine and its accessories to discover and digest. It is also continually necessary to remind oneself of the asking price. Next month I shall be looking at some of the possible applications for the MX-2424 and how it performs in practice.

Connectivity

THE PLAIN-VANILLA MACHINE comes with a single stereo digital I/O in both AES-EBU and SPDIF flavours with input sample rate conversion. Most, if not all, purchasers will fit one more of the optional multichannel interfaces. Audio output slots allow simultaneous fitting of one analogue and one digital card. At present there is only one analogue option. Twenty-four channels of 24-bit 48kHz, (12 channels of 24-bit 96kHz A-D and D-A converters) is on one board. In fact calling it a board is almost an insult. It is housed in a metal screening box with only the top open (this is covered by the top of the machine casing) For the digital slot, as you might expect, one option is 24 channels of TDIIF, ADAT Lightpipe and AES-EBU are also available. The AES-EBU card features input sample rate conversion on all 24 channels. For 96kHz operation both 'bit split' high resolution two wire and single wire 'wide' are supported. Digital audio sync possibilities comprise wordclock In, Out and Thru on BNCs with the welcome alternative of video sync In and THRU BNCs. Audio sync may also be derived from the digital audio I/O on time-code In, Out and Thru plus MIDI In, Out and Thru and two TL-bus proprietary 9-pin D-sub connections.

Additional SCSI storage is taken care of by a half-height 5 25-inch drive slot on the front panel and a 68-pin high Density Fast-Wide connector for external storage devices. The slot is intended to be fitted with either a 'hot swappable' hard drive frame. (Kingston Rhino Junior and Data Express frames are recommended The Data Express has the advantage of being quieter in environments where this is an issue.) or supported DVD-RAM and tape drives. A list is supplied with updates on the Tascam website.

A punch in footswitch connects to a 1/4-inch jack. Polarity is automatically detected. It will also accept an Alesis LRC remote for basic transport remote control. More comprehensive control is available from the RC-2424 control surface which connects via a 9-pin sub-D, not to be confused with the TL-bus connectors. Apart from IEC mains input there is one further connector, an RJ-45 socket. This insignificant looking port is a 100 base T Ethernet network connector and is arguably the gateway to the machine's likely success.
DESIGNED FOR POST, broadcast and DJ applications the Marantz 330 series of CD players comprises three units. The entry level PMD330 offers a good portfolio of standard professional features which are enhanced in the PMD331 and PMD340 with the inclusion of a 10s anti-shock buffer, instant start and advanced pitch control and remote features. The top-of-the-range PMD340 distinguishes itself additionally with the incorporation of an industrial grade transport mechanism and optical pickup unit with an integrated preamp.

A bullet point hitlist of the range’s features includes CD-RW capability as well as CD Text—which for those that are unaware of its arrival employ album and track naming in much the same way as MD does. There’s also frame search (>50ps), index search, programmable cue-point memory, digital pitch control ±12%, end monitor, a 10-key pad, fader start, backlit LCD and adjustment free mechanisms with digital serve.

There can be no doubt whatsoever about the professional nature of these machines. They are built to a very high standard with a decidedly pro priority layout and dependable and programmable logic that can be customised by the user. Large transport controls are restricted only to play-pause, cue and view all of which are lit with smaller buttons provided for the less immediate tasks.

Because the range adopts a derivative approach by adding features to a core unit you go up the range I’ll major on the PMD330 and the PMD340 as representatives of the extremes of the series and point out that functionally the PMD331 spots all the features of the PMD340, but runs a PMD330 transport rather than the heavier duty one on the PMD340. Differences between these transports were not evident in the course of my use of these boxes. However, I think the inclusion of a befield transport in the flagship product is a significant one, not because it suggests that the ordinary transport is in anyway lacking, but because Marantz clearly thinks it will appeal to anyone looking to invest in such a machine on a truly long-term and extremely high mileage usage basis.

With the exception of the inclusion of a dial on the PMD331 and PMD340 (without which they would all look identical), all three machines look extremely similar with no outwardly visual clue to the existence of a different transport in the PMD340.

Rear panel arrangements also follow this theme with the entry-level machine being offered with phono analogue outputs and SPDIF on RCA plus a fader start jack and RCA compatible remote socket and output with accompanying selector switch. An infrared remote was not supplied with my review models. The PMD331 and PMD340 add balanced XLR analogue outputs with individual grub-screw trim pots, a balanced XLR SPDIF output and an optical digital port. Remote control possibilities are afforded by a 25-pin D-Sub that accommodates among most other functions the lack of dedicated fader start port as found on the PMD340.

As already alluded to, operation is identical across the range with the exception of the use of the dial. Time cycles between track and disc total, remaining and elapsed times while Test switches the display between time and text indication modes. Numeric buttons permit direct access to tracks and are used when programming playback which employ the play-pause, cue and stop transport keys which double as Enter, Clear and Exit buttons, respectively.

There are track and index increment buttons, and forward and reverse scan. The cue button permits the location and creation of a cue point that can be entered pretty much on the fly or can be fine tuned by using the traditional looped audio segment approach that is nudged up or >
voted the world's best sampler

twenty four

of the world's top musical instrument magazines recently had to decide which sampler should receive the prestigious Musikmesse International Press Award.
The E-MU E4XT walked it.
< down the programme by front panel buttons or the dial on those models equipped with it.

LED contrast is adjusted on a front panel grub screw and the machine can be set to play single track, programmed, random and repeat modes as well as 'power-on auto repeat playback'.

Pressing exit while paused auditions a preset selectable end chunk of the currently selected track. Pitch adjustment employs a pitch switch with up-down keys on the PMD330 to do what the dial does on the PMD331 and PMD340 while the more expensive machines also get the ability to bend pitch up or down by 8% instantaneously at the press of a key and which drops back to the original speed just as quickly.

There is a surprising amount of detail and functionality hidden in the 300 series for what are after all just CD players. There are enough options and custom configurations available to make a machine completely baffling for an outside user and, consequently, enough to configure it in almost every aspect to the specific requirements of say, a human driven radio station.

On the button

The PRESET button accesses the machine's customisable parameters while in Stop mode. You step through the parameters on repeated presses of the PRESET button or move forward and back through the menu on the index + switches with parameters altered on the track increment keys. Parameters available for adjustment include auto cue activation, auto cue level, end monitor, playing time, end warning flash time, and fade in and out times (up to 8s in 0.5s increments).

Other functions cover pause mode to playback delay, default time display, key lock, and a useful fader start lock mode which locks out all controls aside from the Text and Time buttons when playback has been triggered by fader start with full control returned in Pause.

There are also settings for locking the tray from opening while in play, for setting the auto close time of the tray, the default mode entered after loading a disc (stop, pause or play) and time for the machine to go in to auto standby mode. Control connectors can also be disabled along with a CD sync function and display alert to signify that a 'skip' has occurred. The PMD331 and PMD340 additionally get the facility to defeat the digital outputs, if selected instant start, shockproofing and pitch bend are themselves defeated, and to fire an output on the remote I-O pin 11 when selecting index 2 or 3. Clever stuff.
Things like how long you want the tray door to stay open before auto closing itself is just one example of the thought gone into these machines and how they have been made accessible. The remoting possibilities mean, in the case of the PMI)331 and PMD340, that they can also be really well integrated into the work environment while the PMI)331 does at least qualify with the essential entry requirement of fader start.

All models benefit from a rather good display that combines the usual sorts of things with a less usual rotary pitch display and numerical read-out plus an extremely useful 10-segment bar that moves from left to right to tell you where in the current track you are currently at —this is far more immediately meaningful than elapsed and remaining times, for example. The shockproofing available on the PMI)331 and PMD340 seemed to do the job despite my best attempts to upset it, but when the sturdy cases are locked in to a rack you’d probably have to go off road while on-air to catch it out.

I’m particularly impressed by the way Marantz has split the 300 series into its distinct price points. Rather than kick it off as many do with a substandard bottom of the range machine that progresses to a happier medium product and is followed by an over-the-top flagship complete with cup-holder, cigar clipper and floss dispenser, it’s weighed in with a base model that is extremely well speeded bettered by models that add what are a few fairly lax features.

All in all these are great packages all with Marantz high audio quality outputs that extend to excellent headphones circuits. The issue then becomes which one to choose and the decision here will depend to a large extent on your circumstances.

The PMD340 is probably adequate for most requirements for a dependable and trustworthy CD player with the PMI)331 cutting in for those who need to remote a machine more comprehensively, up its connectivity possibilities and like the comfort of the shockproofing. Those who in addition to this work ordinary machines to death on too frequent a basis and/or are perhaps located in the more far flung and more isolated reaches of our readership will gravitate most naturally to the PMD340. It’s a nice decision to be able to make and you won’t be disappointed.

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Drawmer 1969 Mercenary Edition

Adapting the successful 1960 dual vacuum tube compressor to suit all-American tastes has delivered the Mercenary Edition. Dave Foister signs up.

The Nine: Of Drawmer remains synonymous with dynamic processing. Fashions and techniques shift and develop, but Drawmer is always there somewhere, adding digital control, digital outputs, valves, and anything else the industry wants and needs, and always with a little something to make it own. Drawmer was at the heart of the retro movement, producing the 190X range of valve processors that included the Big 1960 compressor heavy on the British valve compression character. The 1960 was taken up enthusiastically around the world, yet Mercenary Audio's Fletcher still found himself reaching for the real vintage gear for periculiar sounds, and felt it would be useful to have a similar beast yet with a brighter, more American flavour. The result of his ideas and his input to Drawmer is the 1969 Mercenary Edition compressor.

At a casual glance the 1969 looks very similar to the 1960, but this is not a simple tweak of the original with a new badge; there are several differences in detail and operation. The basic package is essentially identical: two channels of valve-based compression, each with a fully specified microphone preamplifier as well as its line input, and a single instrument input with its own unusual features that can be routed to either compressor channel or even both. The compressor controls are simple and direct, the two channels can be linked for stereo with Channel 1's knobs controlling the compression, and a pair of traditional vu meters shows either output level or gain reduction.

Like the 1960, the compressor has a soft knee characteristic that makes it comfortable to control superfluous. Thus the only knobs on the compressor section itself are INPUT, ATTACK and RELEASE, and two of these are rotary switches rather than pots. Here we see the first clear difference from the other unit: the 1969 has six Attack settings where the 1960 only had three, for slow, medium and fast. These in turn do not have actual time values specified for them, partly perhaps to avoid mixing by numbers, but also because the times are varied according to what release time is chosen. Here too there is a change, in that of the six switched settings three are programme-dependent, operating over different ranges of times: position 4 gives 200ms to 2s, while 6 gives 1s to 10s.

Obviously with this many settings, this much programme dependence, and an interaction between the two adjustments, there is far more flexibility available than it might initially seem, and in the best traditions a bit of experimenting is usually rewarded. There is no automatic gain make-up, but a pair of output level controls allow it to be adjusted manually; unfortunately these are not ganged for stereo operation. Stereo linking has a new feature—a third position on the toggle switch marked inc. This contours the compressor side chains to make it less sensitive to low frequencies, avoiding the risk of heavy bass content pumping the rest of the signal up and down. I found this big benefit saving the need for patching an EQ into the side-chain inserts on the back.

The microphone preamps are valve-based, and now have switched gain controls and the addition of a phase reverse switch on each channel, which also operates on the line input along with the 2-position high-pass filter. The auxiliary instrument input takes the increasingly familiar idea of the front-panel DI jack to extremes, making it almost a complete guitar signal path needing only a speaker simulator on the end. It's got bass and treble boost EQ, a master switch to emulate the typical presence peak of a guitar amp, and its own gain control that allows its dedicated valve preamp section to be overdriven. This makes it a much more likely candidate than most to be the front end in a guitarist's rack, and adds useful facilities for bass and keyboards too.

There's something confidence-inspiring about a front panel like this, and it's very easy to get drawn in to what the 1969 can do and mess with the settings. There is a distinct character to its sound, which is wide open and clean with a hint of brightness that counteracts the tendency of some compressors to sound slightly dull effects are, of course, often brought about by injudicious adjustment, and the deceptive simplicity of the 1969's controls makes this easy to avoid. It has all the control you need to handle the expected compression tasks, and I found it particularly effective on an overall stereo mix, where it could be needle-bendingly extreme or unobtrusive as required, and always with its pristine clarity intact.

If there are some types of devices that should have no character, a compressor is definitely not one of them. The background to the 1969 shows this well -there's nothing wrong with the 1960, it just has a certain character that suits some applications better than others. The 1969 fills that gap, and could reduce still further the need to put up with the crackles and hums of forty-year-old equipment just to get that certain special something.
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Manley 120W Monoblock

HAVING TESTED a variety of fairly conventional, solid-state amplifiers over the last few months, we thought it timely, as the saying goes, to look at something completely different. Different in this instance, means valve-based amplification courtesy of Manley Labs. Based near LA in the States, Manley has footholds in the professional and domestic hi-fi markets with no less than 50 valve-orientated products. These range from microphone preamps, converters and preamplifiers accommodating both line-level and phonosources.

It is Manley’s wide range of valve power amplifiers, however, that really straddle the divide between pro and domestic scenes and here it is possible to choose anywhere between the 18W single-ended triode 300B ‘Retro’ to models as powerful as the 500W Monoblock. This test revolves around Manley’s 120W Monoblock, which weighs in at a not insubstantial £2,895 per pair but that employs the ‘technology’, including user-selectable feedback, associated with its bigger pro and power amps.

This sort of amplifier is not unknown in the control room, but is probably better suited to less intensive work in the editing and/or mastering suite where bomb-proof, 24-hour operation is rarely a pre-requisite. The engine-room is provided by a pair of KT90 pentodes, though 6550s, KT88s and even EL34s (with some small modification) may be loaded as alternatives. The single-ended input is serviced by a 5751 double-triode that, in turn, feeds a “044 double-triode in a phase-inverted configuration.

The amplifier may be operated in Ultralinear mode, where the feedback signal is derived from one of the transformer taps, or triode mode, where the feedback is taken from the point at which the transformers are connected to transformer primary. The lower-output triode mode ensures the tube is ostensibly more richly biased and therefore more linear, but maximum sound levels and, potentially, valve life will be forfeited. The output transformers themselves are wound in-house, promising a more consistent part that’s also tweaked very specifically for the product at hand. An output tap rating of some 5Ω provides a ‘halfway house’ between separate 8Ω and 1Ω taps commonly encountered on other valve power amplifiers.

Tested in the preferred Ultralinear mode, this 120W Monoblock succeeds in delivering some 135W into 8Ω at 1kHz with a THD limit of 3%. This represents a gain of +23.5dB. Unlike most solid-state designs, their valve cousins rarely clip so abruptly, but overload in a progressive and arguably more graceful fashion. Nevertheless, with transformer core saturation to cope with at very low frequencies and slew-limiting at high

For methodology see Studio Sound, June 1999, page 27.
See it on the web-site: www.prostudio.com/studiosound/index.html
frequencies, this same 3% limit is quickly reached at just 4W (20kHz) and 15W (20kHz), respectively (Fig.1).

It is possible to squeeze slightly more juice from the Manley's under dynamic conditions but, as depicted by Fig.2, the increase in THD with output follows an almost logarithmic path. Plotted here on a linear power scale, it's possible to see the anomalies decrease in distortion just before the notional clip point at 154W (red), 160W (blue trace) and 131W (green trace) into 8Ω and 2Ω loads, respectively. The amplifier's continuous output profile is traced in black. Clearly, the Manley lacks both the load tolerance (dynamic current - 8.1A) and headroom (just +0.6dB) of its solid-state competitors, so moderate-to-high sensitivity speakers with a minimum 8Ω impedance loads are recommended.

The nature of the speaker load takes on even greater significance in the light of this amplifier's high output impedance trend. Three levels of negative feedback (low 6dB, medium 9dB and high 12dB) are available and which inevitably effect this trend (see Fig. 3), but none can control the inductive rise at 1kHz which reaches nearly 5Ω at 20kHz (minimum feedback).

Even with maximum feedback (red trace), the overall amp-speaker response will depend heavily on the swings in impedance of the speaker load, especially at HF.

The linear losses are reflected on Fig.4 which shows the response into nonreactive 8Ω (green trace) and 4Ω (red trace) loads with minimum feedback. The responses may be compared with those obtained into no load (blue trace) and 8Ω with medium feedback (black trace). Note that all responses are normalised at 1kHz, concealing the uniform loss between 8 and 4Ω loading, for example. Either way, with the 1Ω impedance trend of many speakers dipping below 4Ω, the impact on the overall system response will certainly colour the final sound of the amp-speaker combination.

Tonal colour, of course, is also modified by the high and extended harmonic complement of the amplifier's distortion spectrum, traced out on Fig.5 with minimum (blue) medium (black) and maximum (red) levels of applied feedback. The lower spectrum clearly highlights the high 2nd, 3rd and 4th harmonics which typically persist at around 0.7%-1.0% through the midrange at modest power levels and medium (6dB) feedback. Higher-order harmonics are also clearly visible which do not enjoy the same degree of signal masking and, therefore, contribute to a greater subjective impact. It is also possible to see some supply modulation, visible as the cluster around the base of the 1kHz tone, which, typically, influences bass resolution and the crispness of stereo imagery.

All of which is reflected in the very rich, warm and undeniably colourful sound produced by the 120W Monoblocks. A sound that audiophiles would, with some justification, describe as 'musical' and thrilling if not especially neutral. So, as a transparent window on the musical event, this amp's slightly rose-tinted perspective is, perhaps, not ideal as a tool for mixing and production purposes. And yet, when the desks are powered down and the studio quiet of an evening. I can imagine the Manley's being brought out if only to enhance the simple pleasure of listening to the fruits of a day's labours.
THE DAS MONITOR 6 is a 2-way, passive loudspeaker comprising a 165mm woofer with a polypropylene cone and a 19mm aluminium-dome tweeter that radiates through a shallow axisymmetric horn. The cabinet has external dimensions of 455mm high, by 220mm wide, by 265mm deep, weighs 6.9kg and has a rear-facing port. The loudspeaker is supplied with a cloth grille that was removed for the tests. DAS recommend the use of amplifiers with a maximum power output of between 100W and 200W into 8Ω, and specify a maximum peak SPL at full power of 114dB.

The author regrets that, due to an equipment fault at the time of the measurements, the usual harmonic distortion plots are not presented in this review. Fig.1 shows the on-axis frequency response for the Monitor 6. The response is a bit uneven at high frequencies with a 5dB dip at 10kHz, but it lies within ±3dB from 70Hz to 2kHz and ±2dB from 651Hz to 2kHz. The low-frequency roll-off is approximately -15dB at around 8Hz. Figs 5 and 6 are the horizontal and vertical off-axis responses respectively. The horizontal directivity is well controlled, with no evidence of side-lobes or mid-frequency narrowing, but the vertical directivity shows the interference notch due to the spacing of the drivers appearing in the upward and downward directions. The 10kHz dip, evident in the on-axis response, is not present at 15° off-axis in any direction. The time-domain performance of the Monitor 6 is particularly good, with the step response (Fig.3) demonstrating excellent time alignment of the drivers, and the acoustic source position (Fig.2) showing a shift of less than 2m at low frequencies—an unusually low figure for a ported design. The waterfall plot (Fig.4) further demonstrates the rapid decay at low frequencies, although there is some evidence of ringing in the mid-band between 100Hz and 900Hz. The cause of the dip in response at 10kHz may be apparent from the power cepstrum which is shown in Fig.4. A strong echo can be seen after about 100μs, which corresponds to a path length difference of 3mm—equivalent to a wavelength at 10kHz. This suggests that the 10kHz dip is due to diffraction from an edge, such as the mouth of the tweeter horn, as diffracted waves are phase-inverted relative to the direct wave. As noted above, the 10kHz dip is not present at angles away from the axis, which also points towards diffraction. Overall, the DAS Monitor 6 is a good performer. Its has a particularly good time-domain response, with accurate time alignment and minimal low-frequency acoustic source position shift, both of which suggest an accurate response to transients. Horizontal directivity is well controlled, but the crossover interference notch is evident both above and below axis. The uneven high frequency response is not a problem if listening is carried out slightly off-axis in the horizontal plane (this applies to most loudspeakers which use axisymmetric horns). DAS specify harmonic distortion levels of around -5% at 1kHz at low frequencies for an input of 10% of the nominal power handling rating. This figure reduces to -50% at frequencies above 20kHz. These figures are typical for a loudspeaker of this type.

**Contact**

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Tel: +34 96 134 0860.
Europe: Sennheiser.
Tel: ++44 1494 551 571.

**Fig.5: Horizontal Directivity**

**Fig.6: Vertical Directivity**

**Fig.7: Waterfall**

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**Studio Sound's 'bench test' loudspeaker reviews continue with the Monitor 6. Keith Holland reports.**
Fig. 1: On-axis Frequency Response and Distortion

Fig. 2: Acoustic Source

Fig. 3: Step Response

Fig. 4: Power Cepstrum

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STUDIO SOUND

WITNESS THE EVOLUTION OF STUDIO SOUND, FROM SEPTEMBER 2000
Soundtracs nets
Soundtracs has released net.tracs as a network interface for users of their products that permits the interchange of project information between consoles in a multi-room facility and to provide a convenient facility-wide archival and restoring system. A 19-inch rackmount server controller connects via CAT3 ethernet and deals with Soundtracs session files for relocation and restoration.

Soundtracs, UK, Tel: +44 1372 845600.

SIENA PCI
SEK'D has introduced the SIENA PCI audio card which has eight analogue inputs and outputs and 24/96 resolution. Two separate MIDI I-O ports are included and multiple SIENA cards can be connected via a SyncBus. The card is Windows 95/98 compliant and comes with a complete driver set.

SEK'D, Germany, Tel: +49 7946 776 66.

Sea Sound Solo
The Solo EX recording system enables interactive monitoring of record and playback of signals with separate level controls. Essentially a PCI card supporting Windows and Mac, it has a separate break-out box that contains the A-D/A converters and all interconnection facilities including inputs for instruments and microphones with 24-96 conversion, channel inserts, headphone, and SP DIN and MIDI in, out and through with zero latency input monitoring.

Seasound, US, Tel: +1 415 485 3900.

Compact flexibility
The Altair DA410 is a 1U-high distribution amp, zone mixer and mic splitter able to process two stereo input signals to obtain five stereo outputs with independent level control. As a zone mixer it can control five zones in stereo mode and ten in mono mode. Individually selected mic-line input preamps allow mic or line signal splitting to additional destinations in 4 x 8 or 1 x 16. The unit has a monitoring system with built-in microphones, headphone output and selectable vu meter. I-Os are electronically balanced with transformers as are the two parametric EQ cards.

Altair, Spain, Tel: +34 918 043 265.

Domestic Interface
The Canford Pro-Interface Mk II is a bi-directional stereo interface for matching semi-pro or domestic equipment with many users of the original.

Studio Sound www.prostudio.com/studiosound

AEA R44C

Offering a real alternative to modern condenser microphones, this reissued ribbon brings the past up to date Dave Foister listens in.

‘Che Seasound, latency input monitoring. With 24/96 all outputs and card which has eight analogue inputs and outputs and 24/96 resolution. Two separate MIDI I-O ports are included and multiple SIENA cards can be connected via a SyncBus. The card is Windows 95/98 compliant and comes with a complete driver set.

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Sennheiser MKH800

With mics capable of feeding fast and wide digital records thin on the ground, the MKH800 is a welcome innovation writes Dave Foister

It is one of Life's Little Mysteries that the Sennheiser MKH800 has never truly become an industry standard, an image like the U87, the 414 or the 4000. It is comfortably one of the world's half-dozen or so most expensive microphones and with justified reason, as it is also one of the most flexible, high quality tools on the market. Yet outside a band of devotees it has not passed into the language. So, confident is Sennheiser of its innate qualities, however, that the microphone has been upgraded still further, as Sennheiser's first bid to offer a microphone that makes sense of 2:190 recording.

It does not take much thought to realise that one illogical aspect of the quest for ever higher sampling rates and the striving to deliver frequencies outside the conventional range of hearing, is the lack of those frequencies in much of the source material. There are some microphones whose small diaphragm orifice from the likes of DPA and Earthworks -that have uniform frequency responses extending to 40kHz and beyond, but most of the other models we use are starting to do well before they even reach the brickwall filters frequency of traditional CD sampling rates, and by the time we reach the area of the spectrum where the benefits of 96kHz sampling rates lie, there's really nothing much coming out of them to record. This is not the only point of higher sample rates, but if you accept the necessity to reproduce frequencies in that octave above 22kHz, the sad truth is that there often is not much there in the microphone output to begin with.

Hence the MKH800, which takes the design of the 80 and adds another octave. Since the original is not as familiar as perhaps it ought to be, a reminder of what it offers is in order. This is a side-firing condenser with an unusually small (half-inch) diaphragm, and a trademark set of four small rotary switches down the front of its body. It is impressively compact, with a body little wider than some simple end-fire models, and attaches happily to a stand with a conventional Sennheiser spring clip. Its construction is certainly not lightweight, but its size means that it is not the mass or awkward bulk to balance that bigger models do, and is very easy to position either singly or in pairs. There is an optional shock mount available, but this too is conveniently compact.

The switches are what lends the microphone its unusual versatility. To begin with, it has no less than five polar pattern settings, inserting supercardioid and wide cardioid between the normal set of cardioid, omni and figure-of-eight. This facility makes it a particularly flexible all-rounder for various stereo configurations, from spaced to crossed MS. In fact a pair of these makes an ideal MS coupling, partly because they are easy to set up physically, with close proximity between the capsules, and partly because of the increased HF response. Each microphone has such a good selection of patterns available to control the ambient pickup of the array.

Attention and bass roll-off are dealt with on two of the switches, again with extra flexibility as there are two levels of HF cut and two pad positions. The final switch adds the unusual facility of slight HF lift, again with two positions offering +3dB and +6dB at 8kHz. This degree of frequency contouring on the microphone itself is highly unusual, offering a useful alternative to EQ in tailoring the microphone to the acoustic while actually rigging.

This much is all there on the original version. The MKH800 adds only the extended top end, which is now claimed to be flat all the way to 50kHz while retaining a low end down to 30Hz. This is achieved by means of a redesigned capsule, revised electronics, and a fresh look at the enclosure as an acoustic effect in the upper reaches has become more significant.

Using the 80s, I quickly came to understand why the microphone's adherents are so enthusiastic about it. It delivers a top-flight sound in a package that makes it adaptable for any situation. The term all-rounder is often taken to mean jack-of-all-trades, with the implication that it is competent at all, but the MKH800 can be master of almost any of them. Superbly quiet yet with a maximum SPL handling ability of 142dB, it is equally at home as a single microphone on an acoustic guitar, recording a soaring classical voice, or as a stereo array over a drum kit. For this I set up an MS pair and the benefits were immediately apparent, with an ideal combination of consistency, control and quality. It also showed well the HF capabilities of the microphone, which appear to have no audible limits and next to no colouration.

Every development in audio requires parallel developments in associated equipment. Just as surround requires surround processing, high-sampling rate recording requires extended frequency response in its sources. The MKH800 is one of few mics that can meet that requirement, and perhaps the only all-purpose microphones on the market that can deliver the spectrum modern recorders are designed to capture.


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Lindos Electronics LSC24

Crossing the line between digital and analogue domains inevitably raises questions of quality. Rob James puts the LSC24 to the test.

Regardless of the inexorable onslaught of digital technology, audio still demands precision, analogue electronics. Successful converter design requires meticulous attention to engineering detail and some black art. Even in digital-to-digital conversion there is no single correct method, guaranteed to produce good results.

The name Lindos will be familiar to anyone who takes an interest in audio test equipment. The LSC24 Studio Converter is Lindos' first foray into the fiercely competitive world of outboard signal processing. And Lindos is not the first manufacturer to cross over from gamekeeping to poaching.

LSC24 is a stereo converter with 96kHz, 24-bit capabilities, offering a number of modes. Convert mode gives digital-to-digital sample rate, bit depth and format conversion. Record mode is intended for analogue-to-digital conversion with an analogue output for monitoring purposes. ADDA gives independent analogue-to-digital and digital-to-analogue conversion while HQDA mode offers high-quality D-A conversion.

Housed in the ubiquitous 1U-high metal case, the unit has a smooth membrane across the front panel. The rear panel has XLRs for analogue I-O, AES A and B I-O and AES-sync with IIGNs for N265 wordclock and AES 3-Id I-O. The last two double as SPDIF I-O. There is also a 9-pin sub-D data port and IEC mains input. There is no mains switch, when did you last turn off the power on a rack unit?

Nor are there any knobs or buttons as such as all switches are of the membrane type, clearly die-cast, grey keys—selection functions indicated by LEDs above. The panel is logically laid out. Panel lockout switch and indicator on the left followed by mode select, analogue input gain block, the other parameter blocks and finally output gain. Input and output meters are 11-segment LED ladders. The input pair has peak hold and reset key.

Analog level adjustments are made using the increment-decrement keys below the large bright numeric displays. Levels are set in discrete, 1dB steps. Input ranges from 8dBu to 26dBu for 0dBFS. Output from -9dBu to 26dBu at 0dBFS. D-A performance is optimised for the +18 and +26 settings. Lindos has taken the unusual course of using resistor arrays, switched by relays to set gain, achieving accuracy and solid tracking between channels. Lindos elect to use relays rather than solid-state switches to ensure minimum degradation. I think this is a better option than either knobs which get accidentally knocked or fiddly multi-turn pots which are only really suitable for set and forget purposes, although small clicks are sometimes audible between steps due to sudden gain changes and there is invariably a big one when you switch between 1kHz and 9kHz on the D-A. Since gain will normally only be adjusted for setting up this should not be problematic, but should be borne in mind for live work or high monitoring levels.

In A-D modes a soft limiter is available that begins to operate at -4dBFS. A red LED indicates limiting. Output word length is selectable between 16-bit, 18-bit, 20-bit or 24-bit. Noise shaping uses Lindos' own Minimally Audible filter algorithm with an ultrasonic noise-shaping filter option for double sampling rates. Analogue monitoring is switchable before or after noise shaping for comparison. Sync source (referred to as DARS, or digital audio reference signal) is selectable between the AES11 (XLR) input, external times 256 (super) clock or incoming audio. There is no indication of valid signal on the selected sync source which caught me out once or twice. The same selection of chip-sets is available to many manufacturers so the desirability of most designs is all down to implementation and factors such as user interface and design of noise shaping curves.

It is difficult, if not impossible, to achieve the full performance theoretically possible from 24 bits. The Lindos acquires itself honourably with quoted noise and THD figures well up to average for the current generation. The LSC24 combines three functions in one workstation-like unit. Setup is simple and logical for the operator and with 'stepped attenuator' gain adjustments there is little excuse for getting encoding levels wrong. Interfacing is flexible with the inclusion of both single and dual 96kHz moses welcome. The only strange omission is ±1 wordclock. At this level anything less than excellent performance would be surprising. The LSC24 does not disappoint. I liked the simple setup and subjective quality although: without similarly high-quality units to hand for comparison I cannot honestly say I noticed anything extraordinary about the sound. Performance is certainly superior to the converters built into many consoles and recorders. Either appears effective and smooth and I could not detect any obvious artifacts from the sample-rate conversion.

Analogue monitoring output in sample rate conversion mode adds to the flexibility. This combination of functions in one slim box

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*Virus availability scheduled for April 2000.

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Microboards StartREC

A CD-R duplicator that looks and acts like a piece of regular studio gear? Tim Frost takes a look.

This is where studio duplicators grow up. Gone is the industrial CD duplicator with its mini-tower approach, folded metal casings, low-cost pressure sensitive decal control panels and off-the-shelf innards. In comes the 19-inch rackmount that looks like a pro DAT machine with a control panel that is pure audio industry and a back panel that really was designed for the studio.

StartREC comes in a 4U-high rack package that comfortably holds the control panel and its five CD drives—a dedicated 40x Plextor read drive and four 8x CD-R burners. This panel has the normal—well, normal for a recorder—play, record, stop, forward, rewind, buttons, and level controls. The unit has level meters and a sector that is used for control menus and status information. There is also a headphone socket and level control—an other rarity on a CD duplicator.

The rear of the unit gives further evidence that the system has been designed to look the ground up as a studio copier. Replacing the normally obligatory RS232 ports and keyboard connector are balanced analogue XLR input/output and unbalanced phono input/outputs, an AES-EBU XLR, coax SPDIF input, optical SPDIF (TOSlink) input/outputs and a socket for a remote control—an optional extra.

When mounted, it looks, and acts, like any other bit of studio kit and has the huge advantage over other duplicators because it is capable of creating and editing together a master from external sources, as well as duplicating discs. StartREC is effectively a master CD recorder and duplicator in one package.

The system has three separate modes of operation. Copy for copying complete CDs, Track Extraction for loading individual tracks or a complete list of tracks from a CD to the hard disk and Audio for editing tracks on the internal 6Gb hard disk.

Each has its own control and menu system which is loaded from the internal hard disk separately. While this seems a long-winded way of going about things, it does simplify the way the system operates as you don’t have to page through endless numbers of options.

Making copies is as straightforward as you would expect from any copier system now. StartREC will identify the type of disc being copied (it doesn’t have to be audio disc) and
translated in to MTC and LTC. Providing a machine is transmitting constant time code the TimeMachine can be set to a Precision Follow Mode when connected to a sequencing or HD system in which it will follow fast forward and rewind. In jog-shuttle mode the system will show accurate machine position in its own timeline.

C-Lab, Germany. Tel: +49 406 944 000.

PostConform 2.0
Digidesign has released PostConform 2.0, an update of its autoconforming solution for Mac OS based computers, which is now fully compatible with Pro Tools 7 and Pro Tools 24 MXI systems. 2.0 is a stand-alone application that supports Pro Tools 4.x and 5.0 session file formats and HFS and HFS+ Mac OS disks formats. The software is compatible with the newer G3 and G4 Macintosh computers and can use USB connections for Machine Control and MIDI Time Code. It also supports the Digidesign Universal Slave Driver.

Digidesign, US. Tel: +1 650 842 7900.

Amp DSP module
QSC has introduced a digital signal processing module that offers two channels of DSP that attach to the back of most DataPort-equipped QSC amps. Each channel has high-pass and low-pass crossover filters, high-pass and low-pass shelf filters, signal delay, compression, peak limiting, and the ability to perform disc-to-disc copy which simply makes direct copies from the master to the four slaves. Alternatively it will copy the contents of the disc to the internal hard disk—important when you are creating a room of discs and don't want to keep reloading the master. Sub indexes can be copied and the user has the choice of record speed—1x to 8x.

The Track Extract function executes the simple transfer of existing CD music tracks either piecemeal or the entire contents of a disc. It can transfer the sub index information and ignores the SCMS settings on the incoming content, although it does offer full control over the SCMS settings for the copies. This all runs smoothly and the 8x drives mean that four copies can be made in well under ten minutes.

What makes StartRIP really useful in the studio environment is that you no longer need an audio CD-R recorder to create the master disc to copy from. Using the Audio menu: anything from an outside source—analogue or digital—can be recorded onto the hard disk and then edited to top and tail tracks, move, or erase them. Audio quality on the analogue inputs appears excellent, although users are most likely to be coming into StartRIP from one of the three digital inputs.

Like other CD-R audio recorders, the Record mode offers auto-mark to increment the tracks numbering when there is silence. Unlike consumer units, this unit offers full control of both the detection level (from 30dB to -4dB) and duration of the silence (2s to 8s) that will trigger the auto track numbering.

The logical process for creating a master is to use the Track Extract or Audio input menu to bring the tracks into the system and store them on the hard disk as separate tracks. Then any mistakes, wrong tracks, wrong orders of tracks, false starts or need for fine tuning dealt with using the edit functions. The start and finish points of a track can be altered by selecting an index point splitting the track, and deleting the unwanted bit—following standard MiniDisc editing procedures. StartRIP's editing functions also include track moving, erasing and fade-in and fade-out.

If there are any negatives it is the need to load the individual software segments separately rather than having an additional control pad. Also the system doesn't support CD-RW although given that all the possible recording and editing functions are carried out on the internal hard disk, this is hardly a practical limitation.

Combining the studio recording, basic editing and copy functions in the one rack-mounting box, with dedicated controls and input outputs creates a more relevant copying tool for studio applications than a simple duplicator. Microboards is to be highly commended for approaching studio duplication at last in the way-modified add-on-soundcard approach that has largely categorised the market so far.
WHEN DIGITAL AUDIO BEGAN its inevitable rise, a number of practical, operational problems became apparent. From the outset, several sampling rates were in common use. Of these, 32kHz has dropped out of favour in production since its only advantage is economy of storage. Higher up the scale, 44.1kHz was adopted for CDs while, for many reasons, 48kHz became the professional standard. Of late, the situation has been further complicated by the advent of double and quad-rate sampling. In sound-for-picture, life tends toward greater complexity largely due to the compromises inherent in NTSC television standards (like 29.97" drop frame) and the requirements of film running at 24 or 25 frames per second. To cut to the chase, if you wish to remain in the digital domain throughout a production, a means of converting between sample rates is a necessity.

Of course there is the alternative of converting to analogue and reconverting to digital at the required sample rate. In some cases this has been a preferred option since early converters were either seriously costly or not particularly transparent. One pro CD player actually used this technique as an economically acceptable method of avoiding wordclock sync with other equipment.

Lucid's SRC9624, then, is described as a high-definition sample-rate converter. It provides two independent channels of conversion between all the normal rates and, by using an appropriate external sync source, the pull-ups and downswings required for TV and film work. Alternatively it can function as a single-channel converter at the double sampling rates. It also offers dither to handle changes in bit rate and can convert between physical formats.

The shallow U-box has a finely sculpted brushed-alloy front panel. LED stacks indicate the selected functions and status while all parameter selection is via detented mini toggle switches allowing the shortest route to a desired parameter. Although the LEDs are laid out vertically they are circular in function, so if the required selection is at the top and the current one at the bottom it only takes one click down to move back to the top. The mains switch is a slightly incongruous horizontal plastic rocker.

On the rear, four blocks of connectors provide the audio connections (with TapLink optical, XLR for AES or SPIDIF (phono adaptors are supplied) and XLR for AES-EBU) All outputs are independent so may be used together. A further block has the sync connections, an XLR and two BNCs for wordclock in and out. The wordclock input is bridging and will require external termination if it is the last device in a chain. Wordclock output is always the same frequency as the selected sample rate of the audio output.

Four routings are possible. The independent routing option offers the A and B inputs to the A and B outputs respectively. The Distribution option routes the A input to both outputs. The 96kHz Dual AES (source) requires a 2-wire AES signal on inputs A and B and merges the signals to give a singlewire output at the chosen rate via Output A while the 96kHz Dual AES (receiver) does the converse, accepting a single wire input on Input A and converting to two wire on Outputs A and B.

Table indicates the settings of the status bits in the input signals. Signal sources, output sample rate and output dither are selected using toggle switches. The output sample rate can be derived from a single or two inputs. A word clock input or internally generated at 32kHz, 44.1kHz, 48kHz, 88.2kHz or 96kHz. Whenever an unlocked or invalid audio input or sync source is selected the relevant LED blinks. Dither is a flat triangular PDF (probability density function). This was chosen since if the resulting signal is subsequently re-sampled it causes fewer problems than noise-shaping types.

As a function of the converter chip employed (Crystal Semiconductor CS8320), the performance is relatively transparent at conversion rates of 1.71 or less. At ratios higher than this—like the 2.1 required when going from 96kHz to 48kHz—the distortion performance is slightly poorer. As a subjective test on the 96kHz to 48kHz conversion I used some sine wave tones generated in Samplitude. I could not detect any noticeable increase in distortion between a 48kHz source with no conversion and a 96kHz source down-converted to 48kHz. In operation, the SRC9624 is quick and convenient. The toggle switches are positive and intuitive. Whenever parameters are changed the unit appears to mute the outputs, neatly avoiding high-level spikes. Around half a second after any parameter change the panel LED blinks, indicating that the setup has been stored. The same setup will be restored on a subsequent power up.

Leaving aside the future proofing of the high-bit-rate options, the two channels proved very useful at normal sampling rates when used as an interface with a DAT recorder. This allows some material from tapes at 44.1kHz and 48kHz to be loaded into a DAW and the finished product to be recorded back to DAT at both sampling rates without any repatching.

The SRC9624 is an attractive package. Its versatility and good looks will find it many converts.
William Wittman is a multi-platinum Producer/Engineer, former Staff Producer/ A&R Vice President (RCA / BMG Records and Columbia / Sony Records), Musician and Songwriter. His career truly covers all the bases.

‘I’ll tell you a secret; I’ve always had a love-hate relationship with near-field monitors. But these LSR’s have changed all that. First, they’re just easy to listen to. They’ve got plenty of full, real bottom, great stereo imaging, and they go loud enough to feel right. Plus, they translate incredibly well to the rest of the world! They’re just musical. Wow; good sounding speakers I can trust! It’s love-love.’

‘The Three Best Performing THX® Monitoring Systems Are Also The World’s Most Applauded.

Since its introduction in 1997, the system-engineered JBL LSR Series has become a favorite choice of engineers, producers and performers, many of whom have also become its most loyal advocates. More important, this acceptance is found in every major geographic area of the recording industry; from Los Angeles and New York to Nashville and London.

LSR 32 LSR 28P LSR 12P

Monitors Whose Performance Profile Was Determined By Science, Not Opinion.

During a half century of building the most technically advanced studio monitors, JBL has developed a long list of working relationships with key recording professionals around the globe. As a direct result of this unique collaboration, these industry leaders have chosen JBL monitors more often than any other brand. Not once or twice, but consistently for decades. In fact, JBL monitors are a part of the history of recording itself. Consider as examples, the now fabled JBL 4200 and 4400 Series that, at their launch, actually defined an entirely new standard and new category of monitor. Such is the case now with the entire LSR line.

Joel Jaffe is an award winning Engineer/ Producer/Composer and co-owner of Studio D Recording, Inc., 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.

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Universal Audio 1176LN

While most reissued kit revives the past, little is as accurate as UA's comp-limiter: George Shilling travels back in time.

It is over two years since I reviewed the Purple Audio MC76, an enhanced recreation of the Urei 1176LN, my favourite desert-island compressor-limiter. Perhaps awakened by the continued demand for this design, the two sons of original designer, the late MT Bill Putnam, sprung into action to reissue the original model under its original brand name. And a good job they appear to have made of it.

Every last detail has been faithfully reproduced: even the manual cover has been copied, including the textured paper. The spec sheets make authentic reading, with virtually every figure identical, and the circuit diagrams are of a similar design, although they are no longer hand-drawn.

The front panel is black, recreating the look of the more favoured versions of the original 1176. The originals went through several revisions, designated A-G; this unit being based on the D/E black-faced models. Later units had a (D) output section, a class-AB push-pull stage, and the transformer input was replaced with a differential op-amp.

All controls feel similar, the unit featuring wonderfully large rubber and offset knobs, and attack and release pots, which are all smooth in operation; the attack retaining its defining high-sensitivity setting. This returns any compression, but leaves the gain controls active: turning the unit into a characterful line amplifier. The illuminated meter is similar in appearance to later Urei designs, with the recreated U logo featured, and a recessed trim pot across the other side of the panel for zero calibration. Old-fashioned chrome-plated side push-buttons all work exactly the same as on the originals, except that they feel a little stiff and new.

Ratios of 4:1, 1:1, 12:1 and 20:1 are available with the central row of buttons, and on the right are selections for meter. Gain Reduction Mode, Output Level at 4 or 8, and by pressing the bottom button the unit is powered off. The legending has been meticulously stickered with on-off symbols, no doubt to satisfy some pernickety modern safety rules. Although, talking of safety, upon removing the cover, I was surprised to see bare wires on the main connections—a nuisance for any service engineer. However, the approach is admirable; the circuit layout looks similar in every respect, with plug-in transistors inserted into sockets mounted on the board. On the rear panel are true connections, exactly like the original, with connections for signal and also remote metering, although I would imagine this to rarely be required. Apart from a fuse holding 150 mains socket—the only addition to the original design is a pair of XLR signal connectors. The previously available Model 301 bolt-on box attached to the tag strip to provide XLR outputs, but many owners made their own modifications and attached sockets directly to the case. The phone socket for stereo operation is retained. This is labelled 176LN, and the stereo adapter must be connected between two units for stereo operation. The timing capacitors are in parallel, so doing this doubles the fastest speed of the Attack time, and stereo operation is rigidly set up, especially in difference of different vintage, due to differing transistorbase speed.

Perhaps the 1176LN circuitry could have been included inside the case, as with the Purple model.

In use the unit feels very much like one of the originals. I was fortunate to have the luxury of a fine-known 1960s AU's CA3002 studio which housed several original black-faced and grey black-faces and grey grey HBC-boosted 1176s for comparison, to not mention excellent monitoring. The 1176 uses a FET as a variable resistor to control gain (there are no valves here), and this is the main reason for its unique character. All units were slightly different, but the new model's output gain pot was noticeably changed, with perhaps a slightly different value—I had to turn it further clockwise to match levels. However, in most other respects, operation is identical and the sound very similar, with perhaps a little extra grit from the new unit. Vocals sounded wonderfully enhanced by this compressor, and in normal use it was hard to hear any difference between the new and old models. Attack and release characteristics seem to have been accurately retained, with an attack range of 20-80µs and release times of 50ms-1000ms. There is a really subtle sound difference was in 'overdrive' mode—the legendary technique of pressing all four ratio buttons in simultaneously to give a really over-the-top distorted compression. The character of the new unit in this situation was quite different from any of the older models, with much brighter and sharper distortion, and less warmth, especially on fast settings. By backing off the release speed, a more similar sound could be achieved, but there was still an obvious difference, and although the older units may have been somewhat out of alignment, this was the sound I preferred.

The 1176LN sounds great but the lower-priced Purple competitor is sonically very similar, and features many improvements over the original design. However, as a strictly authentic official reissue with indispensable credentials, the UX is the box to have.

OMR8 gets controller

DAR has introduced an edit controller for its OMR8 24-bit 8-track system that includes a weighted action wheel and dedicated function keys. Two recently introduced features of the system include CD mastering and a new edit function. OMR8 mastering is accessible from within the OMR8 internal editing software and enables tracks to be prepared for mastering with the OMR8 controlling the CD-R burning process. Fine edit permits two alternative takes to be viewed side by side and edited at waveform level with hi-res zoom.

RealVerb 5.1 ships

RealVerb 5.1 is a multichannel reverb plug-in for Pro Tools with the ability to map reverb spatially for surround mixing and to map between room shapes. Rather than running parallel reverberators, the package uses a single reverberator and crossfades the controls producing intermediate states with no zipper noise.

Broadcast mic

Soundelux has released the R-1 cardioid condenser. It features a 1-inch diaphragm with gold on 6-micron Mylar and transformerless FET.

N/Dym wireless

The N/DY series of UF wireless mic system is enhanced by Secure Phase diversity while the receiver has an adjustable squelch level, beltpack, and hand-held transmitters include TX gain controls. Available in two hand-held versions, an omni lavaliier, cardioid lavaliier, a headset and guitar system.

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EADLY TRADING HIS SEAT at a keyboard rig for a seat behind a mixing console, Steve Power's recording career really began riding shotgun for Mick Glossop.

'We got a deal with Virgin, but instead of spending the money on renting studios we built one in Liverpool called The Pink.' he recalls. 'Mick Glossop, whose engineering skills are renowned, produced our album and I assisted him. I was just mad keen — I was 18 and I took everything I could from him, which is a lot, because he is very knowledgeable.

Before that I had done the band's 4-track demos and I was more into the studio than the keyboards. So when the band went on the road, I made a decision to leave in order to run the studio.

What had seemed a simple move was, in fact, only the start of an eventful learning curve for the young Englishman. 'I only ever worked one way — I wasn't used to 'You're the client and I do what you say',' he explains. 'I just thought people came in and organised them and got the best out of them. I didn't really think I was producing them but I suppose I was.

'People would come in and do six songs, record and mix in a day which is very good practice. I was doing demos for Dead or Alive, Frankie Goes to Hollywood, The Lotus Eaters, and several other acts that didn't get anywhere but did get signed up — it was the usual thing of the masters not being as good as the demos.

Power made the move to London and went straight into producing, working with Steve Lovell among others. Around 1986, when things went quiet for a few months, a chance meeting with another Liverpudlian put him back in the studio.

One day I was walking past Battery Studios,' he says, 'and out walked Mike Score from A Flock of Seagulls — I knew him from Liverpool and he needed someone to finish engineering his album, so he invited me in and I finished it.'

Another of the producers working at Battery at the time subsequently asked him to engineer a Billy Ocean album that he was producing and suddenly Power found himself with a management contract for Zomba.

I did The Railway Children, then it went quiet again for a bit around 1992-93 and so I studied classical guitar for two years. That really helped when I came back. I think that was part of the reason why I changed from wanting pristine multitracks to music that affected me. I suddenly realised that they don't buy the multitrack, they buy the stereo and as long as the stereo's okay... You can get a lot more work done if you're not so cautious with every click and pop on your multitrack. On my multitracks now there's clicks and buzzes all over the place, but there are more spontaneous performances there.'

As 1993 got underway, Power found himself working with Babylon Zoo on the album that spawned 'Spaceman'. 'Hard on its heels came Bahybird's 'You're Gorgeous' (Both records that everyone hated...). And then Guy Chambers asked him to produce...
Robbie Williams.

All those years ago Guy had replaced me in the band, explains a bemused Steve Power. 'I'd known him since he was 15. Rob wanted to be Oasis at the time and Chris Briggs, who had A&R'd World Party had the idea of bringing in Guy as someone who could work with Bobby and write with him. They had tried various people and it hadn't worked out, but they complement each other so well. Guy's much more of an artist, he's a bit scatterbrained, and he knew I'd be able to hold a net underneath him.

Often they don't know when they've done a good demo, he continues. 'I remember Strong and Bob Regrets for instance—I picked them out as singles straight away from the demos, and Guy didn't think Strong was very good. As regards demos for the new album, when they were touring in America, Rob and Guy got bored very quickly—they've really got no patience, either of them—so they had popped into studios and written songs. There were about 35 or more songs that came through which were quickly culled down to 12, and the last three were written because Rob decided he wanted to get more 'programmy' stuff, and they were just done about two weeks before recording started.

Power professes not to have had a standard way of approaching making master recordings out of these demos.

Because Bobby is such a large personality you don't have to do just one style of music, you can do anything and it's still him. Something like 'Angels' is approached very traditionally. The whole band went out and played—acoustic guitar, piano, drums, bass—and Rob sang while the backing track went down. Of course everything was replaced except the drums.

'I don't spend long on the drum sound though—I find it easy to get drum sounds now, if you don't mic drums too close, as long as they're tuned and played properly. Ambient mixing depends on the song, but I'll maybe put up one ambient mic, I very rarely put up stereo ambience and try and get a big sound that sounds great but is useless within the song. With electric guitars I don't put ambient mics on them either, just one as close as I can get, and that's it, because if you have two you get phase things going on, and it's not as present, and you can always add something after.

Mind you, I don't like reverb either. I've done almost complete mixes with no reverb for a long time. If I do use reverb, I'll use a plate. I don't like the sound of Lexicon plate very much. The new one I tried briefly and it sounded nicer, and I tried thete 6000 and I like the sound of that better, and I'm going to use that tomorrow doing surround mixes of 'Rock DJ' and a track not on the album, and six tracks from Robbie's Slane Castle gig for DVD. It has 5.1 reverbs and 5.1 compression, which I'm intrigued to try.

The album was recorded 48-track analogue.

For the last album we mixed to Genex 24-bit machine and half-inch tape, and compared every track at the mastering stage. The Genex won every time. The half-inch added warmth, but took away some of the clarity, which made you want to add some 5kHz to compensate, but with the Genex you didn't need to. So this time we didn't bother with the half-inch and just used the Genex.

As regards mixing, Power prefers SSL E-series consoles to later models.

When I tried the 9000, I found it too clean for this rock track I was doing,' he explains. 'So what I've settled on now is Neve and Focusrite, and the DPA 3511 preamp from sideharl. I record with as good-quality mic amps as I can, and then I kind of like the crunching up that happens on the old SSL for the mix. There's an SSL in Battery Studio 4 which has the black knob EQ, which is a slightly softer EQ. That's quite a special desk because it's got the crunch, but you don't find a lot of black-knob desks. You can use it without thinking, whereas when I went to the j—can't be bothered learning a new computer system that doesn't seem to be based on the same principles. They should make a desk that emulates a G-series computer.

Power's approach to mixing itself mixes work and play—a recipe he's found helps retain objectivity through the process.

'When I mix I go really quickly through it, and I can't go really quickly with the j-series, because I'm too busy picking the pen up and thinking, 'what do I have to say to it?' I like simplicity, nothing getting in the way of just getting done what you want to do quickly. Generally I'll come in, do a bit, have some lunch, read some Ceefax about the football, go and do a bit more. But I do a lot quickly, in half-an-hour or an hour, but then I'll want to not hear it, and I'll go and watch really for a bit, come back in, do another bit, have my dinner at seven and go home. And I'll take it home, and come in the next day and tweak it a bit, call Guy in and we'll put it down. So I seem to have spent two days on a mix, but I've actually not been doing a great deal most of the time.

I think it's so important to keep yourself separate from it and objective—if you sit there all day listening to the thing, you can't retain your objectivity.

Another thing I do to retain objectivity, which people who come to see me mix find very amusing is that I have every set of speakers in the world there. As soon as it sounds good on the KRAs, I'll put it on the AVR-8s that I've got on the floor, cos they were designed to be used on the floor. That's where they've got proper bass. So I switch >
<it over and go "Oh, I didn't notice that middle frequency sticking out a bit", and get rid of that. It sounds okay on them, so I'll switch it up to the PMCs that I borrowed to mix the album, and I'll think it could do with a bit more compression. It just makes me think of different things. If I switch to a different set it gives me a totally different viewpoint, like the situation where you take a mix somewhere else.

I've got literally about nine different sets of speakers. I'll return to the previous speakers after making a change, to see if I've overdone it. So I get that instant 'oh it doesn't sound like it did in the studio' straight away, before I've left the studio. I never use NS10s, by the way, because to me if it is right, it sounds terrible on NS10s. I can't mix to sound terrible. I spend a lot of time listening to other CDs on each set of speakers. A lot of what we do is going to be played on the radio. I do squash stuff a lot, and I don't like it, in a sense, but I know that the shop window is radio, and it does need squashing and it does need high end. I compress the mix with the NS, and a Prism Mas-elec, which is fantastic. On this album I had two sets of the 4-way mic amps, and I had the Prism EQ across my monitors, so I recorded a lot of the time without EQ, so I've never done anything that I'll regret. However, I'm screwing in a lot across the monitors. The quality of those things is absolutely incredible. I think the mic amps, the compressor and the EQ. I've never done anything wrong that I'll regret while I'm recording, however, it's loud and sounds great for the musicians who are working on it, so they can get off on it and get the excitement, but I haven't done it to tape. So when I'm mixing I'll get that. I've got a compressor here as well. Because if you mix and then you take it to the mastering room with a view to 'I'll put the top on then,' it can change the balance. I know what I'm doing when I'm mixing—I'd say it's mainly for radio, but then a lot of hog-standard hi-fi and ghetto blasters are quite like radios to me, and that is our selling point really, not to the hi-fi people. But there are tracks where you can get a bit more of that, generally the ballads. Robbie Williams is only the most recent notable artist to have come Power's way. Joe Cocker, Geri Halliwell (for Chris Briggs). The Dumb Dums and new artist Rebecca Ryan have all helped keep him busy.

"Chris is the only A&R man you can play a monitor mix to. Power enthuses: He once told me he'd like to sack all A&R men and bring in out-of-work record producers, because they understand what goes on.

Other notable achievements include mixes for various rock acts that have left Power hungry for more—in good time.

'I'd really like to work with Sine wavy,' he confi rms. They are a really talented Belgian band and I've mixed their next single. Also Feeder—I did some additional production and remix for two singles which got them on the radio. Then there's an artist called Rebecca Ryan on Zomba that I'm hopefully going to do after a break and after the DVD. But I'm definitely going to have a break first.'

With a career characterised by its stops and starts, a break might sound like a dangerous option.

Usually there are a few things queueing up, he says, and I can only do one of them. At the moment I'm in the good period but of course, I'll be at home for two weeks and then think, 'Oh my God...'.

WHEN I WAS AT METROPOLIS I was offered the DPA 3514 microphone kit to try out, and the thing I like most about it is the sound pressure handling. I like the sound of the [Neumann] 49, but you put it on a loud guitar and it can't handle it. And it can't actually handle Bobby when he's yelling. He's got a hell of a loud voice, and when he suddenly decides in the middle of a ballad that 'This is too down, I want a YEAH YEAH YEAH bit here,' and he'll suddenly do it completely out of the blue without warning anyone. However, I've found that even when I know he's gonna do it, on some mics like the 49, I'll have the mic amp all the way down, and he'll still distort the mic on the way in, because he's so bloody loud. The DPA can actually handle him and it sounds kind of similar to the 49, probably because of the treble lift. I thought, if it can handle the volume, it can sound like a condenser on very loud guitar amps and snare drums, which it can. Sometimes I like the sound of a 57 on the snare, but sometimes I would like to be able to get more like the sound of a 451, and with this thing you can, and you can get it close, which is a step forward, nobody else has managed that before. I ended up using it on acoustic guitars as well, and there its treble lift was very handy. And I like to use high quality mic amps, and the mic amp you get wish is very good and open.
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Head case

In most conversations, monitoring invariably means loudspeakers but headphones play are valuable too.

Derek Johnson & Debbie Poyser audition the options

Buying studio headphones seems an uncomplicated proposition—and typically people put rather less thought into choosing their cans than their monitors. Yet most studio musicians and engineers spend a fair amount of time wearing headphones, so selecting the right ones is worth some effort.

First, there’s the choice between enclosed and open types. Enclosed phones usually offer higher levels and increased bass perception, and as they allow little sound to escape they’re the most appropriate for providing cue mixes, where any leakage could make it onto tape (or disk). They also let in little external sound, so are especially suitable for live engineering in noisy environments. Good-open headphones are usually prone to less colouration than enclosed ones (making them more useful for mix checking) and are less fatiguing, but they allow more sound to escape and to enter. Semi-open phones provide a compromise between the strengths of both types.

Then there is impedance to consider: the best sound quality is produced by high-impedance headphones, but low-impedance phones offer higher listening levels (though they tend to more distortion) and are often preferred by location recordists. In the studio, probably the most important thing is to ensure everyone has access to the same type of phones. If there are a mixture of types, some listeners could experience problems with volume levels.

Quality headphones from major manufacturers span a wide price range (from under £50 to almost £200 exc VAT in the UK). It’s impossible to cover all models here, but we can at least provide a taster of what’s available. Starting at the beginning of the alphabet, AKG models are affordable right across the range, and are used in studios world wide. At the lower end is the K70 (£20), a light-weight, open-back, hi-fi-style phone with foam earpads, a 20Hz–20kHz frequency response and 100Ω impedance—a good bet if multiple sets are needed. Going up a step, the popular semi-open K 240 DT (£50) has a 600Ω impedance, 15Hz–20kHz frequency response, and a studio-grade build with deeply padded ear-cushions. If it is a quality fully enclosed phone you’re after, the K 270S (£110), with a 20Hz–20kHz frequency response and 75Ω impedance, could fit the bill.

Choosing between Audio Technica’s headphones is easy, as AT markets just three models in the UK, all enclosed—the 64Ω ATH140 and 64Ω ATHM40 Studiophones, and the 50Ω ATHF910 Pro. The D40 and M40 (£85) are identical in appearance, with padded headband, cushioned earcups, and a good upper frequency response of 28kHz. However, the D40 has LF extending down to 51Hz, compared to the M40’s 20Hz. Both Studiophones are, usefully, fully serviceable, with earcups, elements and cables being replaceable. The ATHF910 Pro model (20Hz–22kHz, £100, £200) is recommended for home listening and studio monitoring, but isn’t field-serviceable.

Many of Beyerdynamic’s headphones distinguish themselves by being modular, allowing defective parts to be removed and replaced individually. At the lower end of the range is the DT 220 (£70), an enclosed phone that produces a 20Hz–20kHz frequency response and 20Ω or 100Ω impedance. The fully serviceable, mid-priced DT 100 (£60) is an enclosed model widely used in the studio industry. It has a solid construction and a 50Hz–20kHz frequency response, and is also available with a variety of impedances: 8Ω, 400Ω or 200Ω. The DT 150 (£120) is similar to the DT 100 but has an extended frequency response of 51Hz–50kHz. Those looking for an open-backed phone might want to check out the DT 990 Pro (£190), which has a remarkable 51Hz–50kHz frequency response and a 600Ω impedance. Beyerd especially recommend the 990 for critical post and broadcast studio monitoring, but it and its enclosed version, the DT 770, are accepted as fine all-rounders.

At least two of Fostex’s four models of headphone are apparently gaining wide acceptance in US studios. Both the semi-open T20RP (£40) and fully enclosed T10RP (£90) are field-serviceable courtesy of a screwed assembly. The T20 boasts a 20Hz–20kHz frequency response, while the T10’s response is a slightly less extended 25Hz–20kHz. Both phones have a 50Ω impedance, deeply padded ear-cushions and padded headband. The T42 T5 (£40) and T02 T7 (£65), both with a 20Hz–20kHz frequency response, are described as good general-purpose phones, their main distinguishing characteristic being ultra-light construction for extended wearing comfort.
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Sennheiser markets a large range of headphones, including enclosed pro models and open-hucked hi-fi designs. The 522 HD500 (£60) is classed as a hi-fi phone, yet it has studio-friendly characteristics, including a more than respectable 10Hz-21kHz frequency response and unusually large circumaural earpieces which encircle the ear, with the phones sitting against the head rather than pressing on the ears. Open construction and low weight help ensure comfortable wear. From the studio range comes the HD250 Linear II (£120), an enclosed phone which is apparently one of Sennheiser's best sellers. Again, the cushioned earpads are designed to fit around the ears, and Sennheiser claim that though the phones are enclosed they offer the feel of an open design. Frequency response is a creditable 10Hz-25kHz, and impedance is 300Ω. Then there's the HD25 (£130), a rugged enclosed headphone recommended for studio and live sound applications, with a 10Hz-22kHz frequency response, a choice of 70Ω or 600Ω impedance, and a conventional 'on-ear' design.

Sony produces many headphones, a few of which are dedicated studio devices. The professional MDR7502 (£60) is an affordable enclosed phone described as general-purpose, with lightweight construction, 4Ω impedance and 60Hz-16kHz frequency response. The MDR7505 (£95) is another enclosed model with an improved 10Hz-25kHz response and a 4Ω impedance. It's recommended for recording, remix and DJ use—boasting a swivelling earpiece for single-sided monitoring—and has a build quality described by Sony as capable of withstanding years of daily handling. The MDR7509 (£185) is termed a professional reference monitoring headphone, with an appropriately wide frequency response (5Hz-30kHz), high power handling, an enclosed, circumaural earpiece design, and robust construction.

Studios on a tight budget needing many sets of identical phones, and prepared to be open-minded about brand-name may be interested to hear that Studiospares recommend the low-impedance Altai 1HVS22, at just £12. It has a 40Hz-18kHz frequency response and Studiospares say its performance comes close to that of models costing six times the price. Something for every pocket, then...
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With Apple back on its feet, the Mac-PC race is on once again. Martin Polon explores the likely impact of forthcoming computer technology.

HAVING FINALLY ENTERED the millennium, and having survived the Y2k miasma, we now face a new Millennium of computing and related technologies. We also find ourselves obsessed with processor speeds—Intel and AMD seem to be jumping over each other to reach gigahertz speeds and beyond for the PC.

By now, nearly every one is aware of the remarkable recovery of Apple Computer—ostensibly spurred on by the efforts of Apple founder Steve Jobs, returned to the fold for several years. The recovery was accentuated by the mid-April 2000 announcement of a two-for-one stock split, the first such action in over 13 years for the Cupertino-based company. The stock itself is selling at $120 per share pre-split (as of 20th April, 2000), or about ten times the asking price previously during the company’s near-death experience. Apple has even managed to maintain its share strength on the NASDAQ technology exchange, as that market place has sustained great wounds on many of the most significant technology sector and dot.com stocks listed.

Apple’s revival has been based on carefully-implemented technology innovations and strong sales of the i-Mac, professional desktop G3 and now G4 and PowerBook computers. The success of these technologies has surpassed even the fastest and most recent innovations available on PCs. Despite the fact that Apple desktop computers with G4 chips are still only rated at 450MHz-500MHz clock speeds (as opposed to the GHz range of Intel and AMD top-of-the-line chips), Apple computers consistently out perform their supposedly faster PC brethren.

Apple’s Power PC chips have profited from their ability to perform more computing tasks, during each clock cycle than a PC microprocessor in a similar time frame. In addition, the Power PC G4 chips use a “velocity engine” developed by Motorola and now adopted by IBM (Apple’s two captive chip foundries) which operates as a vector processing unit. This so-called Altivec feature set means that graphical or multimedia content-heavy computer programs run more quickly and smoothly. Obviously, this also means that audio programs specifically designed to take advantage of the Altivec feature will offer greater speed and functionality than those that don’t.

A further bonus is the cooperation finally restored between Apple, IBM and Motorola in terms of chip design, features, and chip production—a condition that had evaded the so-called ‘AIM’ triumvirate for several years, but now is finally operating again more like a finely tuned Swiss watch than not.

At any rate, when we measure the benefit of the G4-based Mac Power PC against that of an Intel or an AMD PC engine the difference is impressive. It is especially so when used to measure the performance difference between Macs and PCs operating in a power-hungry graphics processing environment, which is analogous to the demands of audio processing. In several recently published tests, the most important of which was Henry Norris in the San Francisco Chronicle, one method of testing between the two computing platforms was to use Adobe’s Photoshop (v5.5) as a common test medium. This was done specifically, because it is the one piece of power using software that is closer to a common denominator than almost any other computer program available. The importance of this is significant, because other specifically designed computer benchmarks are rather platform exact with various benchmarks available for the Mac platform and various benchmarks available for the PC platform. But...
there are none that are not common to only the one platform or the other. In other words, these measurement tools that work with the PC won't work with the Mac—making it very difficult to measure between the two systems. This was achieved so that by using the same amounts of random access memory in both PCs and the Mac to be tested (256k), and specifying the same tasks and the identical image to be operated upon within Photoshop, the processing speeds yielded would represent identical task outcomes. It was clear after the tests were completed, that IBM was using the computer, using a processor ostensibly half as fast as the chips on the PC, outperformed the PCs on the graphics heavy transactions in two-thirds of the time required by the PCs.

What is so fascinating about the direction indicated for the future of Macintosh computing, is the research advances and chip fabrication techniques that have recently been IBM's research labs. The IBM semiconductor research facilities have won five Nobel prizes for work done in semiconductor physics. That is an almost unbelievable number for the research output of one company. But IBM is not an ordinary company and the breakthroughs that have been recently announced, promise to increase the power and the speed of computers using IBM's microprocessors as well as reducing the heat generated and the size of the chips. The advances are as follows:

The future will see use of multiple local clocks on board IBM-designed microprocessor chips including the future range of Power PC, chip, as opposed to the current practice of using a single central clock to run the entire chip. This new technique is known as Interlocked Pipelined CMOS (IPC-MOS) technology. The use of multiple clocks will allow faster elements of a chip to run at their own speed—not having to wait for a system-wide clock cycle to proceed. This technique removes the current limitation of the fastest elements on a microprocessor having to wait for the slowest elements and could eventually yield operations in the 3bn to 4.5bn cycles second using conventional or unconventional silicon circuitry.

Second, it is the use of another method of insulating size-reduced chips from the internal interference caused by the closeness of connections known as silicon-on-insulator or SOI. This technology will have an immediate impact on chip fabrication and may influence succeeding generations of chips.

Third, is the continuation of the move from aluminium-based chips to copper-based chips. This advance too, came from IBM's research laboratories with the initial announcement made in 1997. The use of copper allows circuits to be made physically thinner without the loss of any ability to move electrons about the chip and to run the copper chips cooler and to use less electricity. IBM and the other Apple partner Motorola, pioneered this technology as well and other chip makers are also beginning to transition to copper this year. In fact, it is predicted that by the year 2000 will be known in the semiconductor industry as the "run to copper". It is clear that despite a history of using aluminium as a microprocessor and other chip substrate in some form for the last 30-40 years, the transition to copper will be complete for any new technology products sooner than anyone could have predicted. The current (non-computing) products that aluminium chip substrates will hang on in the short term. The multiple demands in computing and computer devices of speed, size, heat and reduced power consumption will guarantee the ascent of copper.

Fourth, IBM now proposes to build PowerPCs, using only 0.18 micron width in this is approximately 50% smaller than the current 0.18 micron width in use today's microprocessor chips. The new chips will also use an insulation spacing of only 0.1 micron between two individual circuits. Although the smaller space promises speed advances to as high a clock rate as 350MHz or better, the need to maintain inter-circuit isolation becomes that much more important. This is especially true, since the space between on-chip wiring and circuits could be measured in the thousands or even the hundreds of atoms. Inter-circuit electrical interference must be suppressed or else any chip using these micro dimensions will become unreliable. Current chips from all chipmakers use glass-like silicon dioxide-based insulation. That method of insulating is not appropriate to the incredibly small dimensions of the 0.13 micron chips.

Fifth, to insulate for the new chips. IBM has developed the use of an organic or polymer called low-k dielectric, or SiLK which is based on between layers of more conventional oxide materials in a sandwich. IBM has learned how to manage the incredible difficulty of using polymer material, so it can be layered and sculpted to ultra-precise specifications around the copper circuitry. All of this to shield the millions of individual copper circuits on the microprocessor chip, reducing the electrical cross-talk between circuits that could denigrate chip performance (speed) and waste electrical power on the chip. The low-k (or low capacitance) plastic material is a semiconductor dielectric material and Dow Chemical has, in co-operation with the IBM's proprietary specifications. Curiously, the Apple computer partner Motorola is the other major chip maker to be working with SiLK, although its plans for using the organic polymer material do not indicate chip production until 2002 at the earliest. IBM chips using this and some of, if not all of, the other new technologies discussed above will be available for commercial usage beginning in 2001.

The development of any of this extraordinary new technology is indeed challenging, but where does that leave those of us in the audio industry who have to make on-going decisions for the future as to platforms and functionality? The first issue that appears fairly clear that both IBM and Motorola will continue in one form or another to serve Apple's bidding for microprocessor fabrication. Whether the two companies exist in rigid logic-step or simply operate in relatively parallel positions, they both are married to the Power PC chip and that is not going to change soon or easily or even at all. It is also clear that the lead IBM has taken with these microprocessor fabrication developments will place it one generation ahead of the Intel-AMD camp in the development of PC microprocessors. And, as we have already seen of the lead held today by the supposedly inferior Power PC chips (inferior in terms of raw clock numbers rather than throughput), the future will be measured not in raw clocked numbers but by the use of multiple chip—cycles—the several insulation techniques used to reduce chip die size and increase speed—to the gains made in chip fabrication implementation.

In other words, the speed advantage held by Power PC chips over supposedly faster Gigahertz-range PC chips, will in all likelihood remain. And as to how long two generations of chip implementation will represent, it is possible to look at the current rule-of-thumb that says one chip generation equals one year. This may not hold true in the compressed universe of chip fabrication and development in the new millennium, but it is clear that whatever happens in the Intel-AMD camp, IBM will not be sitting idly by.

It is estimated by analysts in the microprocessor industry, that IBM's lead in technology should yield G1-Tz Power PC chips by the end of this year or the beginning of next and potentially two and three G1-Tz chips in forthcoming years. It is not unlikely that by 2005, the average through Power PC chip will be running at 5GHz or better. However, despite this very pretty picture there is one problem. Despite the advances that have been made and which have yielded powerful and significant improvements in video editing and manipulation technology, audio has lagged behind on both the hardware and software front. The next decade will see computer software have to be written to take advantage of these incredible future advances in chip design. That this will happen is very likely, but when and with what standards remains the huge question mark hanging over the audio industry.
SURROUND SOUND

Microphone Recording

The second part of our surround sound investigation highlights a study in microphone placement. Dave Foister follows a series of purist Swiss recording sessions.
dependent on a card fitted in one of two slots in the chassis. At the moment there is only one such card available, and this is therefore what was used. The fact that this card has the name Microphone Array Pattern (MAP) gives a strong hint as to what the approach is at this stage.

The conventional stereo output of a Soundfield is derived by mixing the B-format signals in a manner similar to MS techniques, but with considerably more flexibility. As a quick reminder, these B-format signals comprise four virtual microphone signals: three figure-of-eight points seeming forward, sideways and up, and an omni. A Soundfield's left and right channels are derived from the front omni and side (V) omni, and adding the omni (W) to these modifies the polar pattern to any first-order configuration you want. It is thus possible to synthesise any number of microphones of any pattern and pointing in any direction from a single Soundfield, and indeed the Mark III could do four simultaneously in a symmetrical array. The SP451-MAP card takes this a stage further, and creates five cardioid microphones pointing nominally in the directions of a standard 5-speaker surround layout. Controls are provided for altering the widths of the front LR and rear pairs, for altering the polar patterns of the rear-facing microphones, and for adjusting the relative levels.

It doesn't take much thought to work out that for music recording, the centre microphone is going to seriously reduce the apparent width of the frontal image the microphones are picking up: our feeling was that if it was used at all, it should be at least 10-12dB below the levels of the others, and this was confirmed by Soundfield as the intended use afterwards. Its purpose is for recordings where a strong front centre image is required, like a dialogue recording.

Having established this, the soundstage the microphone produced was impressively natural in a restrained sort of way. Those wanting particularly noticeable ping-pong effects would be disappointed, but the same is true of coincident cardioids in stereo, which never give as spacious a result as figure-of-eights. At the same time, the complete lack of any antiphase components or arrival-time comb filtering created a stable and convincing image, tolerant of moving around the listening space, and capable of being collapsed into stereo or mono without any ill effects. The adjustments were interesting and useful: the width controls did what one would have hoped, particularly once the centre microphone had been appropriately adjusted, and the polar pattern adjustment on the back pair had the expected effect of putting more signal on the rear channels. Turning it towards omni brought the frontal image forward into the room, while the other direction, by effectively putting front signal out of phase on the backs, exaggerated the front-back separation and enhanced the sense of space.

A major benefit of B-format work is that all this kind of adjustment can take place in postproduction, just as MS allows later adjustment of stereo images. Recording the straight B-format from a MkV, an ST350, or an Ambisonics mixing system requires only four channels (or even three if the vertical component, not used in the SP451, is discarded), and the entire range of adjustment and processing is available exactly as it would have been on the microphone live. This is useful enough in simple stereo, where a location B-format recording can have all the stereo configuration decisions made in a known control room after the event, and the additional power here to control the surround stage in such detail and so simply could become a huge advantage.

Those of us who have heard true Ambisonics derived from a single Soundfield will know that there is much more to be had from this microphone, including the facility to map an extraordinarily natural surround field onto to whatever loudspeaker layout is available. It is to be hoped that further developments of the SP451 will exploit Soundfield's technology to the full and include decoding to what has become known as G-format, which is five Ambisonically-decoded speaker feeds intended for the 'standard' layout in use here. Meanwhile the 5-cardinal MAP card is a significant step in the right direction and another useful addition to the small...
surround armoury. By the way guys, it's spatial, not spatial.

Having spent time playing with the Soundfield, we went on to review some of the other surround trials Sound Arts has undertaken, played over an extraordinary set of five Manger Zenboxes 109 loudspeakers, which use a new design of lightweight broadband transducer to minimise the transient problems associated with the mechanics of conventional drivers. These innovative speakers have attracted considerable attention in their native Germany and did an excellent job of conveying what we were hearing in a natural, transparent way, undoubtedly helping with the judgement of the spatial aspects. We began with the final surround mix of the Amos 5.1 recording, complete with surround-panned spot microphones, that I had heard in its raw state on my previous visit. This had used a large array of spots throughout the orchestra, and the care taken in the mix with the main SPL array had produced a very impressive and involving recording. Sometimes the balance between instrumental detail and a proper sense of the recording space is hard to achieve even in stereo, but this setup certainly worked well, with the SPL providing virtually all the rear information while the spots augmented the frontal space and depth.

Panning individual microphones into a convincing surround image was of course nothing new to Sound Arts, who had previously experimented with elaborate layouts of discrete microphones of various kinds. The very first such experiment, in December 1998, was on a recording of a piano concerto using a remarkably simple setup with only eight microphones in total. In a sense this was the surround equivalent of a semi-purist stereo recording that might have been done with a Decca tree plus some close-miking of the piano. Here five Neumann KM143s were used in a roughly pentagonal layout above the audience and orchestra; the rears were about 8m apart and 10m up, several rows back over the audience and facing backwards, while the three fronts were 3m above the stage and spaced by 2m or 3m. The detail of the piano was handled by three further 143s in a small tree arrangement. The result was a very vivid impression of the venue's acoustic, with excellent rendition of the localisation of the orchestral instruments and good clarity on the solo piano.

This basic idea was expanded on for a further session six months later, where two variants on a larger rig were tried. Here an out-front ORTF-style pair of Schoeps cardioids formed the basis of the front, augmented by a 3m-spaced pair of B&K omnis and multiple spot microphones. The back was handled by two rear-facing KM143s in line with the ORTF pair, and two Neumann KM130 omnis 5m or 6m further back. Two versions of this rig were used in the course of the recording; one as it stood with simple mixing, and one with delays added to all the closer microphones according to their carefully measured distances from the main pair. This technique of attempting to time-align the various sources is of course common practice in stereo, but in this case it actually seemed to work against the intended result. Both versions were very impressive, and both enveloped the listener in a persuasive surround image, but the one without the delays seemed more open, more natural, more clearly localised and generally more satisfying. In fact the very symptom the delays should have cured—muddying and smearing of the image—seemed to be worsened by their addition.

Sound Arts has been through many permutations of the idea of natural surround recordings, from simple specialised arrays to complex ad-hoc layouts, and two lessons appear to emerge. One is that experimentation is often rewarded by surprising results, the other is that there is no 'right' way to do it—different methods have different strengths and different weaknesses. Both reinforce the idea that the final decision as to how to approach each recording is a matter of taste, judgement, knowledge and experience. Bit like stereo then.

Thanks to all at Sound Arts for their enthusiastic co-operation in the preparation of this article, to Soundfield Research for the loan of the equipment, and to Josef Manger for providing and setting up five of his monitors for the sessions.

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The EQ Guide

Dividing its time between remedial and creative duties, the outboard equaliser is an essential studio tool. Derek Johnson & Debbie Poyser audition the options.

Equalisation has become a recording and mastering mainstay and the popularity of outboard units has only increased with the rise of digital recording and digital consoles. Studio models available range from the cost-effective and functional to the esoteric, where it's probably fair to say that if you have to ask the price you can't afford it. Design approaches vary equally widely. We can only provide a speedy tour, and we've had to exclude combination devices.

The United Kingdom has an illustrious history in EQ, and British companies offer a variety of equalisers at different price points and with different approaches. The idiosyncratic Joemeek proffers the VC5 Melcopasizer (at £325, UK exVAT), a dual-channel device aiming to offer a classic sound with simple facilities—shelving treble and bass bands, and a swept mid band. Cut or boost of 10-18dB is available, but there's no bandwidth control. Another model in a niche that spans the project and professional studio is the 5013 (£3,499) from T.A. Audio. This dual-channel, 1-band parametric uses hybrid tube/solid-state circuitry and offers overlapping bands with up to 15dB of cut or boost in each band and Q variable between 0.5 and 5. Drawnner's take on tube equalisation is the 33595 1961 dual-channel parametric, which augments its bass, low-mid, high-mid and treble bands with adjustable high-pass and low-pass filters. Each main band has six switchable, overlapping frequency choices, with Q variable between 0.5 and 3, and up to 18dB of cut or boost available. Klark Teknik is a venerable British company who made its name with graphic EQs, but are also famous for flexible, affordable and long-lived parametrics. The single-channel 105 (£595) is a 5-band device, with all bands fully parametric and covering a 20Hz–20kHz frequency range. Bandwidth is variable between 0.08 and 12, and cut/boost of -25dB to +15dB is available. It also features high- and low-pass filters. The DN10 (3999) is a dual-channel version of the 105.

Two Oram contenders are the 110-1EQ (£3,551 and the limited-edition 110-551 (£3,276). Electronically similar though the EQ2 has a sculpted front panel and more precise calibration, both are dual-channel devices offering six parametric bands plus HF and LF filters. Also available from Oram is the Octa EQ (£1,850), providing eight channels of parametric EQ. From Focusrite come the Red 2 ( £2,951) dual-channel 2-band EQ and the Red 2 ( £2,951) dual-channel 3-band EQ with a transformer-coupled design offering two parametric mid bands, shelving HF and LF bands, and high-pass and low-pass filters and the dual-channel Blue SL3 ( £895) isomorphic mastering equaliser. The latter is a 2-channel featuring stepped rotary switches and the same circuitry and components as the legendary Focusrite ISA 110. Yet another opportunity to buy British is offered by Trident-MTA, with the IX2 EQ module. Though this 16-channel EQ, featuring four overlapping frequency bands per channel, is part of the Internix modular mixing system, it could fit into any mixing situation.

A newer name to analogue signal processing is Carvin Master Series MEA-2 1-band mastering equaliser (£950). This 2-channel device has stepped controls for precise settings, and four bands grouped as two overlapping pairs, each pair being dedicated to low or high frequencies. Each frequency control offers 11 settings, making a total of 84 choices per channel. Gain in the range of ±8dB is adjustable in 10 steps, and the bandwidth control for each band selects between five preset curves and a shelving option.

Across the pond there’s once more a wealth of choice, including parametric EQs from the man who introduced the idea to the world. George Massenburg Labs markets two parametrics: the reference-standard, hand-built 8200 (£3,195), a dual-channel, 5-band model, and the 9500 program equaliser (£6,295) with detented controls. Both have discrete transistor circuitry, CMOS servo-stabilisation to correct DC offsets, top-quality passive components, and a transformerless, DC-coupled design. Another successful American export is Manley Labs, the company which has access to the passive EQ circuitry featured in the legendary Pultec EQs. The valve-equipped 5-band Pultec EQP-1A is available in single (£3,195) or dual-channel versions, and as mastering models featuring detented controls.

Manley even offers all-tube, dedicated mid-frequency EQ (£1,600) said to be ideal for mid-heavy instruments such as guitars. The latest EQ is the Massive Passive (£2,995), a stereo, 1-band mastering processor with additional high-pass and low-pass filters.

US designers certainly seem to favour valves: tubes are used in equalisers ranging from the Aphex 109 (£40), a cost-effective parametric featuring 'Tubesence' valve circuitry and dual 2-band mono and mono 1-band operation, right up to EQs costing £2,000 plus. The BQE-2001 (£1,795) from Sumit is an example of the latter type. It’s a 2-channel, 4-band parametric passive programmable utilising tubes as well as solid-state electronics, a transformerless signal path, and an unusual configuration. This comprises an HF shelf attenuator, a mid-high band offering two controllable stages, an LF band with boost and cut available simultaneously at five frequencies, and an LF shelving filter. The same company’s BQE-100 (£1,995), is a single-channel, full-range passive device, with four overlapping EQ bands followed by a valve amp stage. Millennium Media.
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< too, uses valves and solid-state electronics, in a rather unusual way. Its £1,950 NSEQ2 dual-channel, 4-band parametric EQ offers switchable valve and solid-state amplifier stages in one unit, for an instant choice of sound character, plus an impressively wide frequency response.

Over in the tubeless camp is California-based Avalon Designs, who offer two 4-band, dual-mono parametrics based around class-A amplifiers, a DC-coupled, transformerless design, and passive-active filter topologies. The 2055 (£2,635) has passive lower and upper bands with shelving or peak-dip characteristics, and two fully variable active mid bands. The substantial 2077 (£8,615) is recommended for mastering, and to that end it has switched controls. Even Summit, previously known for valve designs, is branching out, with the new Element 7R EQ-200 (£3,495). Summit used the talents of Rupert Neve in the design of the discrete Class A transformer-coupled circuitry of this solid-state 4-band parametric, and added MID, 25 memories, and support for Pro Tools with the Extension 7R software control panel.

Many ways to spend your Euros are offered by companies such as SPL, Z-Systems and Behringer. SPL's Qure parametric EQ (£999) features a mysterious Qure control which appears to be derived from some of their Vitalizer enhancement technology. The dual-channel, 3-band Qure also features variable-frequency high-pass and low-pass filters, and a valve stage.

There are not many digital equalisers in the world, yet Z-Systems, makes two. The Z-Q6 mastering EQ (£6,750) can operate at up to 96kHz with 24-bit resolution, and provides six channels of 4-band EQ—ideal for surround mixing—or two channels of 18-band EQ. The £2,495 Z-Q1 is aimed at mastering engineers, providing four parametric and two shelving bands, and can run in stereo or dual-channel mono. This 24-bit device is designed to behave like an analogue EQ, offering real-time adjustment without pops, clicks or digital artefacts.

Behringer addresses the lower-cost end of the market with the sub-£100 solid-state Ultra-Q Pro PEQ2200, but climb the price ladder somewhat with the valve-enhanced Tube Ultra-Q T1951 (£560). The PEQ2200 offers five fully parametric bands plus tunable high-pass and low-pass filters. The four overlapping bands of the stereo T1951 are fully parametric, though the high and low bands can be used as shelving filters. A quick trip north of the border brings us to Denmark, home of Tube Tech, whose valve processors share a retro-style, functional appearance and large precision rotary controls. The ME18 single-channel parametric offers three passive EQ bands, with five selectable centre frequencies in the LF band, 11 in the mid band, and five in the HF band. Also on the Tube-Tech roster are the PEIC 3-band passive mono program EQ, and the EQ1A 3-band single-channel EQ.

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Increased computing power has taken broadcast automation systems into the 21st century. Kevin Hilton explores the options and opportunities.

In the early days of computing, mechanical computers took up huge amounts of carefully air-conditioned space. Today, more powerful devices sit quietly in the corner of many living rooms. In the same way, time-controlled, mechanically driven radio automation systems were housed in special equipment rooms, away from the main studios. Their modern, server-based counterparts are also consigned to separate racks, but they take up considerably less space and could comfortably run the entire output of an station if necessary.

With the evolution of computer processors and soundcards, the radio station in a broom cupboard is now a reality. As is commonplace today, the Internet is adding a new dimension: systems developed for that medium have found their way into the mainstream and have forced the prices down, bringing automation to users who would have previously considered it a luxury. In a market that ranges from expensive, fully functioned, complete solutions to software packages for under £100, radio automation is a classic example of users getting exactly what they pay for.

During last year’s Sound Equipment Broadcasting Show (SBES) in Birmingham, UK, Jutel introduced a component version of its On Air-applications package. This can now be used for playing out schedules from the EPIS desktop, aside from its established jingle facilities. Included in this is the Carttel, plus AutoPlayer for automatic play-out of playlists, the Quick Edit audio editor. Matrix Controllers and the new DAB/WWW package, featuring the PAD-server. DAVID, at one time part of the Harman Group, is now owned by fellow German company Management Data Software Engineering. At the end of last year this software house completed a 2-year project to design and install a full digital production framework for all seven channels of the Hessischen Rundfunk network. Management Data has also been involved in initial pilot projects for the digitalisation of the BBC World Service’s Russian language channel and is collaborating with US company b-it-s Broadcasting Technologies. This year it expanded the scope of its automation activities by merging with OmniBus Systems, a British specialist in integrated TV studio control.

As much as manufacturers may want customers to use their system and their system only, end-users have become more sophisticated and shown that they do not necessarily want to be tied to one platform. Plug-in technology has made it easier for operators to pick and choose the applications or components they want. A classic example is the RCS Selectors music play-list package, which has become almost the generic term for such systems. Many stations will use this in conjunction with other automation systems, preferring not to use RCS’s own automation package or the scheduler of their chosen system.

A company banking heavily on plug-ins is Mediatron. Working on familiar drag-and-drop lines, Mediatron’s plug-ins work under either Windows NT 4.0 or Windows 2000. Current plug-ins include: HotControl, a hotkey and short-cut play-back module that replaces cart machines; X-FadeEdit for creating crossfades and segues; VoiceTrack, a module for recording and triggering linking announcements; and the Digital Radio control module, offering the potential of text and graphics on domestic receivers.

Most of these can be used as stand-alone applications and do not have to be used in conjunction with mediatron’s full automation systems, the AirControl NT and AirEdit NT. AirControl NT is a full studio software package for the digitisation and automation of radio stations and is designed to run programming on a 24-hour basis. Music, news, commercials and other sound elements are sourced in CD-quality from RAID5 hard-disk arrays, running at 32-bit on the Windows NT 4.0 platform.

All manufacturers of radio automation systems have built their reputation on one element, as with RCS and Selector, or have become the main supplier in either their home country or a specific part of the radio market. While aimed largely at the high-end, large scale sector, Netia Digital Audio and Dalet Digital Media Systems, coincidentally both French, are arguably two of the biggest names in automated radio.

Netia has consolidated its position in the last year by becoming part of the French/English owned company Management Data Software Engineering. This merger was regarded as prudent, as both systems work on the NT platform. Audio Follow’s products have now been absorbed into Netia’s Radio Assist range.

Netia has its own set of broadcasting modules, marketed under the Odyssey umbrella name. These incorporate Audio Follow’s ‘Air’ prefix and include the Air-DDO play-out system for network centres, the Air-PlayList for themed stations, which can be used in automated or manual modes; the Air-HotKey; Air-CartStack; and Air-Jingle. Users of Netia equipment include Radio France, BBC World Service and the Australian Broadcasting Corporation.

Dalet has an equally long list of users, among them the BBC, CNN, Radio Switzerland International and BIBS. Central to the company’s philosophy is the Dalet 5.1, which is able to acquire audio from a variety of sources, including satellites, CDs, digital files, and live recordings. All content is stored in a common database and assigned titles for correct identification.

One system that has developed from collaboration and licensing is the Sony BMS. Swedish software house BMS was looking for a hardware platform, which was supplied by the multinational in 1998. Latest developments for BMS include a new SQL database, based on ODBC (Open Database Connectivity) for Windows and a new edit function. ODBC acts as a translation device, which allows different data sources to communicate via SQL. The BMS editor...
has been able to open a range of file formats (MP3, MPEG Layer II, AVI) for some time, using ActiveX technology built into Windows. But this was only enabled it to ‘write’ (or mixdown) completed edits to a PCM WAV or Broadcast Wave file. This has now been expanded, giving the ability to use any file format supported in the Windows AC (Automation Command) system.

For production purposes, the system offers the Mix Editor and Surfer applications for editing and preparing material. Dalet also has an optional music scheduler, while the whole system can work in live or fully automated modes. For the future, Dalet has included a multimedia content database and a variety of optional broadcast modules. Webcasting is covered by the On-Air Now! function, enabling broadcasters to offer e-commerce and content data (for example, song titles, album covers) that complement the on-air programming.

A big name, yet a relative newcomer to radio automation is Fairlight On Air Systems. When the DAW manufacturer acquired Vamos Media Solutions, based in the Netherlands, and fellow Australian company Ogigen On Air Systems, Central to its range is GoStar, based around hard disk RAID storage, semi-online CD jukebox arrays or offline C13-ROM/DLT tape backups linked to a self-tuning database engine. As a modular system, users have a choice of various components: Quick Recorder, Multi Channel Recorder, Looping Recorder, Single Track Editor for journalists and editors. GoSTAR Multitrack Editor for special use, NewsWire third-party systems interface, GoSTAR internal wire reader and the GoSTAR NewsCollector. A recent major installation was at Radio New Zealand’s new Broadcasting House complex in Auckland.

Windows NT has become the operating system of choice for an increasing number of audio applications, so radio automation is shifting towards this platform as more new releases are purely software packages that work on any PC infrastructure. Although it can operate with servers running NoveLL or Unix, Studer’s DiGiMedia uses NT 4.0 as its main platform and, like a great many automation systems, employs Digigram professional audio soundboards, including the PCX9, PCX11, PCX11+, PCX80, PCX20 and PCXPocket.

Another convert to NT is Enco Systems, an American manufacturer that is apparently making a new push into Europe. It’s main contender is the DADpro32 digital audio delivery system, which will also run on Windows 98. The system can be used for live assist, full automation, on-air presentation, production and inventory management.

The array of radio automation software on the market is almost bewildering. There are those that are well known in most national markets, but there are plenty more that have carved out a niche on a local basis. Computer Concept Corporation is one of the longer serving developers, recognised for the DCS. This has evolved into Maestro 2.0, which uses apt-x, MPEG, or linear audio formats and can integrate with a user’s preferred music scheduler.

On Air Digital USA has introduced version 2.0 of its OnAir Digital Studio (UDS), which is designed to work with live presenters, but can also run as a completely automated device. At the core of the UDS II is the RS-HD (Radio Suite Hard Drive System), a Linux-based digital audio server. This store all events on hard drive or combines with external CD changers.

A venerable name in broadcast automation is Smart Broadcast Systems. Its radio system is the Smartcaster, a hardware rack and software combination that employs MPEG or Gen2000 files controlled by Windows 98, 95, NT or DOS. The software is currently in its second generation; a play-list scheduler, Digital Programme Director, is included in the package.

Both stand-alone and integrated products are produced by Zenon Media of Germany. Stand-alone includes the AirCheck hard-disk logger and Auto Recorder; integrated units are the Send 32 scheduler and the AudioCast complete one-channel broadcast system. VGCS has a family of automation products under the dira (digital radio) banner, including the OACTRL. On Air Controller and the On Air jingle machine, cart player and scheduler.

Two of the best known British manufacturers for small to medium radio stations are Barracade, which first came to note for the Brian editor, and ProCass, developer of the WAVcast. For wider automation, there is the RCS Gen2000 for stations, and the BCSCS (Barracade Storecasting System) and ProCass the SoundBox, an MPEG compression-based system that can run small stations on a 24-hour basis. Software house PSquared produces the Myriad V2.0 play-out and automation package. This comprises three components: Myriadin, Myriad Live and Q-NXT. A fourth element, Voice Custom, is also available. Another significant seller in the UK is RadioStation, which has a strong grip on its native Republic of Ireland market, being used by the country’s independent news provider to feed most commercial radio stations.

A great many software packages currently available for automation either began life aimed at Internet radio or are also intended for other music sequencing applications, like live DJing and music scheduling in clubs and bars. Web hobbyist FHYS caused a stir at last year’s SHS, where it was demonstrated by UK distributor Alice SoundTech. Retailing for around £70, Version 2 is due later in the year, retaining at approximately £600, and will offer more features for conventional radio.

Web jockey is not alone. In no particular order, and taking a very deep breath, also available are: Radio 2000 digital jukebox and Internet radio system; Raduga V3.0 low-cost automation; Keogh Software’s RMS WaveCast 3.0 for automated play-out of digital audio files; Supersonic software driven audio rack; Virtual DJ, using Microsoft Direct X; a range of systems from Group Two: AK Radio Suite, Horradio and Jazzer; Bass (Broadcast Automated Support Systems) Windows and Mac-based server and play-out tools; Discosoft MP3: DRS 2006: Eire Windows-based software that can interface with ‘leading components’; LAN International’s Sound Manager, with a range of components including OnAir and Cartwall; OZZ 2000, offering three modules for studio operations, planning departments and systems engineers; Pristine Systems’ Rapid Fire hard disk studio system and MusicPlus scheduler; Pro Studio Radio Host integrated software, including the Heavy Rotation scheduler; Tiesseki Broadcast Systems’ is-2000, a networked or stand-alone system for NT or Windows 2000; Win Antenna from France; Italy’s Winjaya; the Spanish Xsystem; SBZ AIRMIX sound automation PRO and AIRMIX sound automation DJ; and Radio Edition’s Megamix, equipped with Soundsoft MP3 TAG ID file labelling technology.

Unsurprisingly, some manufacturers are dismissive of the cheaper software solutions. Those involved with these packages admit they are limited in certain respects, but say it is a matter of what the user wants to do. In a perceived digital world, radio is neither as digit-based nor as automated as might be thought. Large-scale radio stations, particularly the cash-rich groups, might embrace the high-end systems in a big way. Many smaller stations either use low-cost software automation or still have not progressed that far. Technodreams must be tempered with the revelation that, apparently, 80% of American radio stations still run off cart.

Lower cost technology is becoming critical to radio automation. As off-the-shelf soundcards fall in price and rise in quality, there has been a move away from Digigram. MPEG and apt-X compression could see a similar migration to other formats, although compression in general could be a less common part of this technology.

This is due to a shift towards linear audio. Manufacturers are already addressing the issue, with RCS’s Master Control NT V14.3 supporting the format. Barcodex will introduce linear versions of existing products at SHS in November. The radio market has always been an idiosyncratic one and it seems that its future will once again be mapped out on a wet day in Birmingham.
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The pros know the difference.
Buenos Aires is more than the Tango, with two major recording facilities illustrating the future of the most European of Latin America cities reports Dan Daley.

Buenos Aires is quite proud of its European heritage. That's immediately evident in the architecture of the sprawling city's downtown section, where sleek stores sell even sleeker Euro couture and furniture designs. Where much of the rest of South America remains an adobe and glass-and-steel hybrid of structural design, part Castillian, part Indian, part Bauhaus, BA, as the locals call it, combines classical post-Renaissance, post-colonial and post-production harmoniously amid post-modern and twisty European lanes feeding into wide boulevards, including Avenue 9 de Julio, at 120m the widest avenue in the world. In short, BA takes some chances, but knows whence it came.

That sense of history seems to give it a good sense of where it's going, as well. Two facilities in BA illustrate how阿根廷 and possibly the rest of Latin America will handle the future. One facility is resting its hopes on DVD, using the recording business foundation it was once based on as a support mechanism for its launch into the world of digital media formats. The other is your straight-ahead rock 'n roll studio that is undergoing the increasingly common shift from private domain of a rock luminary to making a living in the day-to-day world of for-hire facilities.

Fingers Arts Group is rooted in the recording studio business, one started modestly enough by owner Sergio Molho back in 1986 out of a small space with a Soundcraft recording console and a lot of MIDI gear. However, Molho, a husky, gregarious BA native whose ear is glued to a smartphone that never seems to stop buzzing, found ways to follow the money in a business that was rapidly becoming the purview of everyone and anyone who could afford the increasingly affordable new generation of professional audio equipment. After MIDI had become a commodity, he moved into video editing and audio post-production, shifting the company's focus from music recording (rock, not Latino: another way that BA stays more Euro than South American) to corporate clients such as advertising agencies.

That, in turn, led to the addition of animation, graphics and digital audio editing, and later led to the creation of a joint venture with US media facility designer John Storyk's Walters-Storyk Design Group in 1994. After Molho had moved his expanded studio business into new digs in BA's Villa Crespo neighborhood.

"The studio business is changing all over the world," Molho says. "And even though this is Argentina, we're not immune from the forces of evolution in the business. We've had to adapt the facility to changes in the market and changes in technology. Yeah, we follow the money. But that's what business is about. When a lot of people can make records on their own equipment, then music studios have to move into areas where people can't do things at home."

That mind-set has now led Molho to turn yet another technological corner and establish what is South America's first independent DVD authoring facility last year. While the new format has barely a footprint in Argentina and not much more of a presence in all of the continent (only Brazil and Mexico thus far), Molho is betting that DVD will become a major force in this region, although not in quite the way it's doing so in Europe and North America.

"Argentina is what I would call Hollywood-dependent," observes Molho from behind his desk, where he sits framed by a Kandinsky print from the Guggenheim Museum. "I suppose it's understandable to some degree—with 35m people, Argentina is a relatively small market for movies. But I know that movies weren't the way to introduce an authoring business to Argentina."

What Fingers Arts Group—the name refers to the musical adroitness of Molho's cousin, a musician who lives in the States—did instead was to leverage its already large base of corporate clients, which it had developed during its transition to post-production, and introduce them to DVD as the interactive successor to VHS for corporate promotional videos. Although the awareness of DVD in the advertising and other corporate ranks was virtually nil in 1999, within six months Fingers Arts had created two DVD presentations, one for a real estate developer, the other for the BA branch of an Italian advertising firm. During that time, it also did the DVD authoring for the first domestically produced Argentine film to go to DVD, Mi Alma (De mi Alma).

However, Molho doesn't believe that the advent of DVD is going to be any sort of deliverance for the Argentine film industry or the audio post business which feeds it. The Argentine movie industry puts out only about a dozen new titles a year. Molho isn't kidding himself that Argentina is destined to replace Hollywood. Bombay or Hong Kong any time soon. Rather, he says, the theatrical titles that will start fill-
ing Argentina’s DVD pipelines will come from catalogue films, bringing domestic output from 12 to about 40 locally produced new films per year within a few years, still a very small number.

Fingers Art Group is fitted like a jigsaw puzzle into a 2-storey space above retail shops on the street level below. At the top of a steep staircase is a warren of small yet well-appointed studios and offices containing the DVD authoring suite and a multimedia presentation room back to back, with Molho’s and other offices on the other side of the reception area. The second floor, accessed via a spiral staircase, holds the audio studio that was the original business at this location, fitted with a Soundcraft analogue console and Panasonic digital mixer, as well as video editing and multimedia graphics suites using Adobe Photoshop and Quark Xpress, and a small café and lounge.

The DVD suite is based around a Sonic Solutions DVD Creator system residing on a Macintosh computer. This is fed by an array of media devices, including a Tascam DA-30 DAT deck and a DA-88 digital 8-track for audio tracks, and a Sony Beta SP video deck. Two Pioneer 244 DVD decks serves as playback for quality control and presentations in the adjacent presentation room. Both suites are fitted with Speakercraft audio monitors which, like the Denon power amps that run them, are decidedly quotidian as far as audio- philes go. “We decided that we wanted to be able to do the authoring and monitor the results under the same kind of playback environments that most of the work would be seen in,” explains Augustin Ignacio Alvarez, Fingers’ DVD manager and sole authoring engineer. (At the moment: Molho expects to add two more by year’s end in order to have the facility running a full three shifts. Currently, Fingers has a total of six technicians working here at various media disciplines.) The rooms sound good and tight, however, helped by the placement of shallow RPG diffusers on the rear walls, creating an articulate and accurate surround audio playback environment. And while most of the audio components of DVD productions are created out of house and brought in by the client, Fingers uses a Digidesign Pro Tools system to create its multichannel mixes. Despite its compactness, the entire facility has each media suite linked via Ethernet.

The authoring projects have been relatively simple and straightforward thus far. Mía Alma, for instance, has a middling degree of interactivity, including interviews with the producer and the cast, as well as cinema and broadcast trailers, a music video clip, and Portuguese and English subtitling. The subtitles are translated by another company in BA and delivered to Fingers as text files with time code embedded. The film’s audio came in on the DA-88, with tracks 1-6 holding the surround mix and stereo on track 7 and 8. The dialogue tracks are mixed in with the film score: there are no separate M&E tracks, which is standard operating procedure in Hollywood to facilitate foreign versioning dubs. They don’t do separate music and effects tracks in Argentina because the films don’t get out of the country much,” Alvarez explains. “That would change the entire culture of film audio post-production in Argentina. But that may happen at some point as the film industry here grows.”

And while Molho continues to pursue music recording to some extent, he believes that the region’s future will be based on a wider range of media, and he is acting accordingly. “We think corporate work is going to be the big growth area for audio,” he states.

Rock music has been the anarchistic muse to a few generations of social, political and cultural hellions, and it did not pass Argentina by.

ESTUDIOS CIRCO BEAT started as the private facility for Argentine rock legend Fito Paez, but in many ways it is the logical conclusion of how rock gained its foothold in Argentina, a narrative that has all the elements of made-for-TV movie. A war movie, actually, dating back to the 1982 Falklands War, better known here as La Guerra Malvinas, the name by which Argentina knows the disputed islands.

‘Rock ‘n’ roll had always been persecuted by the military governments of...
Argentine while I lived here,' recalls Alan Colbert, a native of Buenos Aires who
left for the States in the late eighties, working in the music business in New
York and Los Angeles and now back in BA as the label liaison between Warner
Music and Paez. 'If you went to a rock
concert here in the seventies, you could
easily be arrested. It wasn’t a place that
was very hospitable to rock ‘n’ roll.
Up until that point, says Colbert, the
local rock music industry was small and
based largely upon garage bands play-
ing cover songs of western artists
—especially British ones, since Buenos
Aires is culturally, architecturally and
emotionally a European city, and the
UK was the centre of much of rock’s
development in the seventies and eight-
ies. And though Spanish is the language
of Argentina, those bands sang Amer-
ican and British rock songs in English,
trying to get every note and inflection
perfect, like any adolescent garage band
of the time anywhere in the world.
The way things were then, with rock
essentially on a short leash and locked
in the cocoon of English that prevented
it from infusing itself into Argentine cul-
ture, was just where the ruling military
 junta wanted it, to the extent that it tol-
erated it at all. Besides, they had other
things to think about, including a fal-
tering economy that was producing quadruple-digit inflation on an annual
basis, and political unrest that was
inducing increasing violence—and in-
creasingly lethal—police crackdowns
on dissident.
It was a time in Argentine history
when the studio that Colbert and I were
sitting in and having this conversation
would not have been possible. Fito
Paez opened Estudios Circo Beat in Decem-
bcr, 1987, and named it after one of his
best-selling albums. A protege of Ar-
gentine rock legend Charly Isidro Garcia, with
whom he played keyboards and guitar
before stepping out to his own multi-
platinum career, Paez originally plan-
med it as purely a personal recording
facility. However, as has been the case
in many instances, building a complex
and expensive personal facility leads to
it being opened eventually for outside
rentals, which is the road that Circo Beat
is now taking, after a redesign of the
large space, located in the industrial
Paternal barrio of Buenos Aires, by the
Walter-Storvik Design Group.

The main recording space is 12m x
50m with a 7m ceiling rising to a small
skylight and surrounded by several iso
booths. Motorised acoustical cloud pan-
els can change the room’s acoustical
properties dramatically: from a decay of
0.4ms to 0.8s. The control room,
which houses a 56-input SSL G+ con-
sole with Massenberg automation and
a vintage 10.2 Neve sidecar mixer, is
heavily diffused, and loaded with Gen-
eric 1038 monitors.

The design was limited to some
degree by the building itself, which is
long and somewhat narrow—it was
originally meant to be a television pro-
duction studio before Paez purchased
it. Storvik recalls that the first move
was to get the control room lowered.
I needed the height,' he explains. Then
I angled the control room a little in the
space to give a new axis to the entire
facility, which also provided acoustic
high frequency ray reflection benefits
with the angled surfaces in the studio.
We also knew that at some point the
facility might start to be rented out, and
that was one of the reasons for getting
the control room sunken and made a
little bit larger than we might have done
if it was just a project studio.

The big issue in the studio was vari-
able acoustics. Fito needs live big band
recording potential—he uses a great
number of strings and acoustic instru-
ments on some of his recordings as well
as typical rock sets. We were able to
get a 100% RT60 (reverberation time)
swing from 0.4s to 0.8s with a system
of moving wall panels—hinged variable
surfaces as well as motorised ceiling
clouds. And all of the rooms are isolated
using a combination of floated and split
slabs. We actually made our own isolation
product in Argentina from materi-
als that we can get locally.

Power is always a concern in a Third
World environment and Circo Beat was
no exception. Power was a screw up,
that’s for sure, Storvik sighs. In gen-
eral, Third World power situations are
not that difficult; most of the problems
were and are political—getting it
installed and so on. Clearing up the
power was not that big a problem. It’s
getting the lines in properly that’s the
main issue in these situations.

In addition to work by regional
recording artists, including Los Fabu-
losos Cadillacs and Ilya Kuryakin & >
<the Valderramas, Circo Beat has also propelled Paez's film-scoring career, and a projector and drop-down screen are part of the design in the main studio. The facility is gaining an international reputation, helped by the fact that Paez' most recent record was produced by Phil Ramone and engineered by Frank Filipetti. Upstairs, the fittings for a prince of the musical realm are evident in a huge, modern house-like dining-conference room, next to a gourmet kitchen.

But back to the narrative, because it was in that dining room that the conversation with Colbert took place, and where, surrounded by racks of vintage Argentinian, Chilean, Californian and French wines, the story of how rock 'n roll came out of the closet in Argentina reached its conclusion.

'What changed it all was the Malvinas War,' says Colbert. 'The government was in trouble, economically, and politically, and war was a good way to deflect criticism and attention from those issues, as it often is for many authoritarian governments, and a way to rally the people around a central issue. So as part of the war effort, the government decreed that in the future, all broadcasting had to be in Spanish — no English. Especially not English, because the war was with England. And that turned out to be the best thing that ever happened to rock 'n roll here, because all of a sudden, radio and television needed a lot of Spanish music to fill up the airwaves. That meant that the musicians of Argentina — including the rock musicians — had to start generating new music. The bands began to write their own songs instead of covering English and American rock songs. And now that they could do that in Spanish and have an outlet for them, that really pushed the development of a rock culture here forward. And what had once been a business that stayed in small clubs was suddenly filling soccer stadiums. That was the beginnings of Argentine rock.'

And Argentine rock music never stopped to look back. After the military government fell in the wake of the Malvinas War disaster, Argentine rock artists found a wider reception for their music throughout South America, and several of them, such as Los Fabulosos Cadillacs, are making inroads into the North American music market. Rock's revolutionary roots run deeply in Buenos Aires. The walls of the buildings in Paez' working-class neighborhood are covered with political graffiti. And, up in his equipment storage loft, he keeps a portrait of Cuban revolutionary leader Che Guevara hung on the wall, overlooking the guitars and drums and amplifiers. Like some saintly icon, Che's visage confers a blessing on the equipment, reminding it that has a larger mission in the world than fraternity parties and Saturday night soirees.

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US: The musical bomb

Without some major reorganisation, the destruction of the established music market is at hand write Dan Daley

Imagine that the atomic bomb had been invented before the aeroplane. (Trust me, I know where I’m going with this.) You have a weapon of unimaginable destruction, but are incapable of delivering it. Along come the Wright brothers, who provide you with just the delivery system to whomsoever it is you feel needs whomping. You needn’t buy a bad Hollywood script to arrive at a similar scenario in the music business at the moment. Just think on your computer and watch the music business so we know it crumble around you.

If you look at the world from the perspective of the Recording Industry Association of America, you’d view MP3 as a slow of atomic bombs, going off randomly, chipping away at the fortress of your business. Then add Napster to the equation. If the RIAA worried over MP3, it went apoplectic over the delivery mechanism that is Napster, a software engine developed by a bright but bored 19-year-old college dropout that enables people to find exactly the music file they want, retrieve it from one of the estimated 10m hard drives of other Napster users across the US, and play it on their own hard drives, from which it can be retrieved by any of those users. This is precision bombing at its finest.

There is a twist to the MP3-Napster story that affects the professional audio world, and in doing so will perhaps hasten the erosion of the music industry. MP3 has caught on in the commercial music business. Audio post facilities across the States are using MP3 to send drafts and finals of commercials, promo bumpers, spots and other commercial audio bits to advertising agencies for reviews and approvals.

But it also seems that same record industry which so reviles MP3 for ripping music off the Internet is now also using the format in a similar fashion to the audio post business: they are taking in new releases, formatting them for MP3, and using the Net to send them to such destinations as radio programming directors to get feedback on which cuts should be the first single off the album, or sending them to prospective directors to bid on doing the music video, or distributing within the company preproduction tracks before an album is even finished. These files might have been conveniently labelled for internal use, such as ‘Great Idea’, ‘Demo Release’ or ‘String Single’ or ‘Oasis demo’.

Enter Napster. Go up there sometime and enter words like ‘demo’ or ‘advance’ and see what it pulls up. Could be anything from a garage band in Des Moines to the next Springsteen track, months before it was due. And once Napster get their hands on it, it begins to make the rounds of 10m hard drives.

According to a report in the Los Angeles Times, one major-label president reported that one of his artists recently made verbal threats to a company employee that she would be blamed if the act’s album turned up on Napster before the scheduled release date. Other artists, reportedly Radiohead among them, reportedly have advance copies of albums be sent to journalists or others in the promotional chain. The labels are reportedly scrambling to devise and implement security measures to avoid leaks.

It would be funny, if it wasn’t also so amazing. What’s being created out of this unstoppable use of new distribution technologies is a definitive disincentive to make music for commercial gain. At least, in the way that it has been done for the last 100 years or so. In the same way that personal recording technology created the horizon for the studio business the way it was since the 1940s. MP3 and Napster and Gnutella and other such software are putting nails in the coffin of the music business as we know it.

No longer will large conglomerated media entities have easy strangleholds on the way music is created, marketed and distributed. Combine this with the proposed sale of Universal’s music holdings, acquired and owned by Seagram’s, to a French corporation, and Sony’s long-

Europe: Listening and hearing

Listening tests carried out in London on the Verance watermarking system have done nothing to quell engineers’ fears writes Barry Fox

After a concerted campaign conducted over the Internet by UK-based engineer Tony Faulkner, the SDMI and Verance finally gave the first demonstration of their analogue watermarking system outside the US in early July. Around 30 engineers and a few journalists were invited to Sony’s studios in Wh United Field and given the chance to do comparative listening tests on four pieces of music, with and without watermarking.

Paul Jessop, technical director of the EPJ represented the Secure Digital Music Initiative; David Liebowitz, Chairman of Verance Corporation, was there too. But no-one was present from 4C. Enter the consortium of IBM, Intel, Matsushita-Parasonics and Toshiba which ran the US tests in the summer of 1999 on behalf of the SDMI, and on whose say-so DVD-Audio and probably also SA-CD will be watermarked. Parasonics has already launched DVD-A players in the US, over the 4th July weekend.

The testing for DVD-A is done and complete. Liebowitz insists. ‘From October all DVD-A players must have Verance circuitry built-in.’

The 4C tests which led to this decision were conducted last year in US studios, on 50 ‘golden ears’ who had been chosen by the five major record labels. So far only Denny Purcell of George Town Mastering, Dave Smith (Sony Music) and Mark Wilder (Sony Music) have been prepared to identify themselves. Although DVD-A provides a sampling rate of 192kHz, 4C’s test material was coded only at 96kHz/24-bit standard.

After 4C chose Verance as the winning system, 4C created a licensing corporation called LMI which will now administer the licences and collect huge royalties on behalf of Verance.

So why no 4C in London? ‘They were busy with other commitments, they send their apologies and have left me to carry the flag’, said Liebowitz. Yes, there were good personal reasons why one member of 4C could not be present. But why no-one? In the words of Pretty Woman, Bad Mistake.

Astonishingly, the testing process depended on a Compaq laptop PC in the listening room with a hard drive that whirled louder than quiet passages of the music and any subtle ambience. The drive could not be switched off during the tests. How can anyone who is serious about testing, organise a test with such an obvious distraction?

Faulkner flew back from Malaysia to be in London for the tests. ‘I’m 49 years old, I’ve got a cold and I had jet lag’, he said after participating the day after flying in. ‘I scored 75%, correctly identifying the watermarking in six out of eight cases. This system is not transparent, and anyone who says so just isn’t telling the truth.’

James Mallinson is a Grammy-winning producer for all the major record labels and independents. The tests were so badly handled that they are totally invalid. I don’t know whether it’s because they don’t know about sound quality or were trying to pull the wool over our eyes.

The first musical piece, Debussy piano was an old analogue tape with hiss and very constrained frequency range. Halfway through I refused to continue because I said there was no point in the test. The Petrussha recording was some of the worst recorded sound I have ever heard; analogue hiss and you could hear the azimuth flexing, as you did from a car radio.

It was quite impossible to judge a system with that kind of material and DVD audio and DSD are all about high quality. ‘Then they played some New World
rumoured desires to unload their entertainment properties. and you are watching the unstated reaction of the corporate mega-entities to this situation. divestiture of entertainment holdings due to a diminishing ability to wring revenues out of them. Not had for and afternoon's work by a disaffected teenager.

It was that further. Once the broadband pipe line becomes more widespread, this scenario begins to apply itself to films and television. Don't take my word for it: listen to Jack Valenti, president and CEO of the Motion Picture Association of America (MPAA), as quoted in the Miami Herald. If Napster can facilitate and encourage the distribution of pirated sound recordings, then what's to stop it from doing the same to movies, software, books, magazines, newspapers, television, photographs or video games? The answer: Jack is nothing.

It would be a cheap shot to say that the high-end marketers of music are playing a charade — decrying MP3 and Napster while simultaneously using it as an internal marketing tool, and crying again when it turns around and bites them in the shins. But the two points I will leave you with are that it was audio which established the pattern for what will likely become the most radical restructuring of the modern entertainment business since its inception. and that the field it was played out upon was the one delineated by American marketing concepts and technology. Keep an eye on how this plays out over here. It's how it's going to go everywhere else.

Music which was appallingly badly recorded, with no low end, no high end above around 10kHz, and heavily compressed. It was of magnificently low quality. Then they played some dreadfully recorded pop. It was a joke, a complete joke. I was really quite cross because I thought they had wasted my time. These tapes were totally invalid. Writing in Hi-Fi reviewer Martin Colloms had to invite himself. Says Colloms: 'I withdrew from the tests. The quality of the material was appalling. It was sub-Walkman standard. They were using very old analogue material, the Petrof of the dates back to 1962. The piano had a dynamic range of only around 40dB. It was awful music, lifeless and anaemic. There was a mess of hiss. The material had been copied from CD-R onto computer hard disk, and then processed through a 96-24 converter, so it didn't start as 96-24. The room acoustics was bad, with multiple reflections. The PC hard drive was whirring in my ear and a transformer was mechanically humming. I told Sony it was an embarrassment to their company and I couldn't avoid wondering whether they had sent the material together with a 72-bit full identifier payload?

Reframing the airwaves

From being a specific forum on broadcasting issues, Kevin Hilton signals a widening discussion of delivery matters.

This is the end of the broadcast as we have all known it since the mid-nineties. That much has been discussed as how 'traditional' broadcasting practices and companies will change through the influence of new technological flirtation. There will undoubtedly be a long-term evolution, but in the immediate future, long-established broadcast ways will form the basis of all activities.

What is changing is this column. From being a specific broadcast column forum to allow me to involve the events of current broadcasting issues and make irrelevant, obscure remarks about them, it will become a wider-ranging discussion of delivery matters. About which I will make irrelevant, obscure observations. It is reassuring to know that while things change, they don't change that much.

Some of the issues that will now be discussed regularly in this space have already been examined in past outings, whether in detail or just in passing. Digital technology has blurred the edges between many areas. where there was once a clear divide between production, post-production, delivery and other facilities. the core use of digitis has reduced the entire process to merely the movement of data.

Different front-end interfaces remind us that there are differences between the various stages, that they are different skills; but, ultimately, it is possible to imagine a methodology that combines all the functions, connected to a black box that will then direct where the finished product goes. Production and post-production techniques are highly developed and while they will continue to grow in sophistication, delivery of the programme or creative content is the area where most attention is being shifted.

With a plethora of outlets—digital terrestrial, cable and satellite television, digital radio, broadband communications and the Internet (including mobile 'phones, personal organisers, WAP devices et al)—delivery is arguably as important as the creation of what is carried. This is a massive turnaround in thinking. At one time, not very long ago, transmission was the broadcast journalism equivalent of the black spot. We all knew that it was absolutely necessary—otherwise the creative stuff wouldn't go anywhere—but that one man would do it for a fraction of the overall duties of the subject. It did not help that transmission looked like filing cabinets. The only commission more mind-numbing was one concerning routine and switching.

But things change and now those involved in transmission, routing, switching and distribution of technological meek—are very likely to inherit the earth. It would have been hard to imagine that NTI, once merely the engineering division of a now almost forgotten regulatory body, would become a brand name splashed over newspapers, TV and radio as 'the complete communications company'. But it has. Likewise, national telcos—France Telecom is a prime example of the distribution of exciting events like Euro 2000 and are poised to be crucial to webcasting.

Delivery is important; it is even, to use an eighties word, 'sexy'. Naturally, without good content, delivery is just a means to an end. Hence NTL taking shares in digital radio multiplex and others buying into programme delivery. In the light of all this, it would appear that the BBC's decision to privatised and sell off its transmission networks was hopelessly short-sighted. But if one considers that conventional terrestrial transmission is not the future, then the Corporation did not completely screw itself, particularly its retained the landline infrastructure between the transmitters.

In the spirit of free trade, the BBC made its studios and facilities business a wholly owned subsidiary—BBC Resources—with the implication that it would eventually be privatised. At the beginning of July, new director-general Greg Dyke announced the findings of his review of Resources. Privatisation is out and more efficient and cheaper working is in, with the loss of 200 jobs. Some elements of Resources will be returned to BBC Production, while a new subsidiary, BBC Technology, will be formed to protect the government.

'We want to change parts of Resources and make it more profitable,' Dyke told a strangely unattended press conference. Making a profit out of public service broadcasting was central to the approach of Dyke's predecessor. John Birt, while the new incumbent believes the BBC is to be set almost completely under the philosophy that became known as Britism. 'We are trying to get rid of the dafter parts of the internal market without losing the good parts,' Dyke stated bluntly.

One of the good parts is obviously delivery. BBC Technology will centre on 'BBC technological and distribution expertise', with the ultimate aim being to generate additional income to spend on programmes and services. (The subject is getting enough cash to win back sport from the commercial sector.) The new subsidiary will comprise approximately 1,300 personnel, experienced in broadcast, telephony, IT, on-line technology and communications. The BBC projects that Technology will generate between £150m and £200m over the next six years. Such projections emphasise the importance of delivery, and how much broadcasting is set to change. Delivery being used to underwrite programme making—who would have thought it?
Ampex Series

In the second part of his eulogy on the venerable ATR mastering machine, Tony Arnold gives more pointers and fixes.

A

LWAYS MAKE SURE that your ATR has tape lifter sleeves fitted over the tape lifter arms, otherwise a master can be severely damaged due to wear from the original arms. Machines are often updated from 1/2-inch to 3/4-inch without changing tape lifter sleeves or fitting them at all, and users then wonder why they have a severe scratch down the middle of a 1/2 inch tape or the edges are curled on 3/4-inch.-inch tapes.

Another further warning accompanies another tip. If you wish to run your capstan for adjustment or repair then you will have to place it in an extender card that is usually housed in slot No. 9. All you have to do is to switch the ATR off and remove PWA No. 9. This is the Red Servo Card and you will now have all transport functions without having to load the tape. That's the tip. Now for the warning. Ampex provided a clip on the extender card to prevent whichever PWA you are servicing from falling out, but now that the capstan PWA and the servo are removed from the machine it has been known for the capstan to be fitted accidentally into the wrong slot with a huge amount of resulting damage. To stop this happening we make up dummy cards from plastic and fit these into the empty slots not being serviced.

I will repeat again that you should never remove or refit cards while the machine is switched on and this includes the headblock. When removing or refitting the ATR headblock always turn the head release screw clockwise as this will prevent the allen screw from coming loose from the lower T section.

Users often ask for low-profile TOA transistors to replace Q5, Q6, Q7, Q8 and Q10 on the main heat sync chassis and the reason these were used was to prevent the collector from grounding through the on-off switch arm. We fix this by fitting thick sleeving over the arm to isolate this problem.

The 1-speed dual EQ Padnet assembly is designed as a plug-in compatible accessory for the ATR 100. It may interchanged with the standard Padnet assembly and requires no modification to the standard ATR 100 or to the existing main audio board.

The 1-speed Padnet can be configured in two modes by simple switch selections. In conjunction with the existing jumper options for master bias operation on the audio control PWA, it provides an extremely flexible equalising and biasing system.

In the 4-speed configuration each of the four speeds (30ips, 15ips, 7.5ips and 3.75ips) is individually biased, level controlled and equalised. The equalisation at each speed may be set to any required standard.

In the 2-speed configuration, as with the standard Padnet, any two speeds of the four available may be selected. However, each of the two speeds selected has two separate equalisers and level controls available. The selection of which equaliser is in use is made by the 1911 switch on the audio control PWA. In addition the audio control PWA also provides individual master bias adjustments for each speed equaliser.

With this Padnet it is thus possible to configure the full 4-speed capability of the ATR 100 or, having selected the 2-speed dual equaliser configuration, to align the two equalisers, levels and biasing for two major equalisation standards (that is NAB and IEC) or to align the system to use either of the two types of tape at each speed with no restrictions on operating levels, equalisation standards, or biasing. In addition to independent reproduce and record gain presets, and the normal repro HF, LF record HF, front panel equalisations, there is a second record equaliser, Record Shelf, which, in conjunction with the normal record HF equaliser, permits flatter overall response to be obtained with a wider range of tape and biasing conditions.

You can change the overlays on the appropriate PWA to suit 4-speed or 2-speed and if you have the wrong overlays showing you should find the correct one behind the existing one. You simply remove the allen screw, and slide the two overlays out vertically to change. On the Audio Control PWA you will also have to remove the printed board as the bias switch will prevent removal.

Some confusion does exist on how to set the DIL switches on the 4-speed Padnet. For the 2-speed setup (Fig. 1a) mount the appropriate front overlay and ensure the audio control PWA is set as required. Set either of the four position slide switches (S2 or S3) to the "4s" position. For each speed in turn, determine if record/low frequency boost is required (NAB). Switch S1 through S1 to select this, for 30ips through to 3.75ips respectively. Any switch in the on position provides no LF boost (IEC) as required by AES 30ips or IEC-CCIR 15ips and 7.5ips. For NAB 15, 7.5ips and IEC-NAB 7.5ips standards, set the appropriate switch to off and this will provide the standard IEC low frequency record boost.

For the 4-speed setup (Fig. 1b), mount the appropriate front overlay on the Padnet and the correct audio control PWA is jumpered correctly for the speed pair desired. Set S2 to the position corresponding to the higher speed of the required speed pair.

Set S3 to the position corresponding to the lower speed of the required speed pair. If S2 and S3 are set to the same speed, the illegal speed indicator next to the transport speed select switch will illuminate indicating improper operation. Set S1 through S4 to the correct positions determined by the equalisation standards to be used at each speed, and/or the EQ 1 or 2 selector. S1 1 and S2 are for EQ 1, high-speed and low-speed respectively and S1 3 and S4 are for EQ 2, high-speed and low-speed respectively. As before, the on position for each switch of S1 is for no LF record boost (AES 30ips, IEC-CCIR 15ips). The illegal speed set selects this.

www.americanradiohistory.com
These 4-speed Padnets are still available from Courthouse Facilities or M Tupper (Tel: +44 1729 552428).

Many second-hand ATRs are purchased without a service manual and most machines were shipped to only take 10½-inch reels, new owners often wonder which type of ATR accommodates 14-inch reels. There is not space in this article to explain how to modify your machine to take the larger reel-size, but all the relevant information is in the service manual. Photo copies of service manuals are available from us and in time these will be available on CD-ROM.

For the conversion from 10½-inch to 14-inch mastering operation, apart from the cost of the heads the other expensive items are the upgraded tape tension guides. If you have the early type of guide then you will have to change the whole assembly, but if you have the later type just the extra length shaft and fixing screw need to be fitted to your existing Ceramic guides and this brings the cost down dramatically.

When removing ICs, often the print itself is damaged, and with the print being on both sides of the PWA it makes it difficult to see if you have made good contact with the upper print which will sit below the new IC. We have had great success by using wire wound type DIL Sockets and when using this type of DIL you are able raise the socket and solder in place to see the print around the legs of the socket.

On the early version of the control panel switches the press-buttons shed a powder that gets into the switches causing intermittent malfunction. Ampex later made a protection sheet to trap the dust and it's worth checking to see if you have this update as it leads to longer switch life.

This is the completion of my comments on the ATR 100 Series. next month I'll be giving tips on the MM 200 series. If you have any questions or problems please email me as I hope I can be of some help.
Transforms and Duality

Exploiting the duality of digital signal analysis gives us an essential insight into the working of many audio processes. John Watkinson explains transforms and prediction.

A complete description of a signal in the frequency domain consists of frequency and phase information, whereas a complete description in the time domain consists of a waveform. Transforms allow either description to be used. In general, any phenomenon in audio that can be explained in the frequency domain can also be explained in the time domain and vice versa. Indeed if it cannot there is something wrong with our understanding.

The duality of transforms provides an interesting insight into what is happening in common processes. In general the product of equivalent parameters on either side of a transform remains constant, so that if one increases, the other must fall. Fig. 1 shows the example of a low-pass filter. As an ideal low-pass filter has a rectangular frequency response, the impulse response will be the transform of a rectangle which is a sinc/x curve. At Fig. 1a is shown a filter with a wider bandwidth, which has a narrow impulse response. At Fig. 1b is shown a filter of narrower bandwidth that has a wide impulse response. This is duality in action.

If we want to know the effect of a filter on a signal, we can use either the frequency or the time domain. In Fig. 1c the output spectrum is obtained by multiplying the input spectrum by the response of the filter. If these responses are measured logarithmically in dB, the multiplication is achieved by adding. However, in the time domain, the effect of a filter on the waveform can be calculated by convolving the waveform with the impulse response of the filter as in Fig. 1d. A good example of convolution is the tracking process of a vinyl disc player (anyone remember vinyl discs?) The output waveform of the pickup is the convolution of the shape of the stylus and the shape of the groove. As we might expect, narrowing the impulse response of the stylus, by making it elliptical instead of round, would extend the frequency response and reduce the distortion.

Fig. 2 shows some examples of duality. At Fig. 2a to increase the width of the frequency response of a tape head, the gap has to be made smaller. At Fig. 2b to improve the width of the directivity pattern of a loudspeaker or microphone, the diaphragm has to be made smaller. At Fig. 2c the narrower a pulse gets, the broader its spectrum becomes.

Duality is like paranoia, the more paranoid you get, the more threats you perceive. Thus the more familiar you become with duality, the more examples come to mind. The compact disc player has a storage medium that is a function of the spot size of the laser pickup. We can also apply transforms to light. Fourier optics holds that a lens is a filter to spatial frequencies. As detail in an image increases, the light goes further and further from the optical axis toward the perimeter of the lens. Thus the resolution is limited by the aperture in order to increase the capacity of CD to make DVD, the aperture of the lens has to go up. This has the effect of increasing the spatial bandwidth of the lens, so by duality it can focus to a smaller spot on the disc. The intensity function of the read-out spot then convolves with the track geometry to give the replay waveform.

Duality also holds for sampled systems. A sampling process is periodic in the time domain. This results in a spectrum that is periodic in the frequency domain. If the time between the samples is reduced, the bandwidth of the system rises. Fig. 3a shows that a continuous time signal has a continuous spectrum whereas at Fig. 3b the frequency transform of a sampled signal is also discrete. In other words sampled signals can only be analysed into a finite number of frequencies. The more accurate the frequency analysis has to be, the more samples are needed in the block. Making the block longer reduces the ability to locate a transient in time. This is the Heisenberg inequality. The ultimate case of duality because when infinite accuracy is achieved in one domain, there is no accuracy at all in the other.

Block length switching is common in audio coding. For example in MPEG Layer 2 coding, usually the blocks contain 1152 samples, corresponding to a time period of 2 ms. During transients, the block length drops to 384 samples, increasing the time resolution by a factor of three, but reducing the frequency resolution by the same factor.

Fig. 1a): Wide frequency response has narrower impulse response.
Fig. 1b): Narrow frequency response has wide impulse response.
Fig. 1c): In frequency domain, input spectrum is multiplied by filter response to give output spectrum.
Fig. 1d): In time domain input waveform is convolved with filter impulse response.
In compression, the transmitted bit rate can be reduced by the use of a number of tools. One of these is prediction. Fig. 2a shows that in a time domain predictor the system uses knowledge of earlier sample values to try to anticipate what the value of the next sample will be. In the real world it will always be in error and this is handled by subtracting the prediction from the actual value and sending the difference. The more accurate the prediction, the smaller will be the size of the error and the lower the bit rate can be. In the decoder an identical predictor is used and the addition of the prediction error results in the original value once more. Predictors can be simple or complex.

**Fig. 2:** Three examples of 'Less is more'

**Fig. 3:** A continuous signal a) has a continuous spectrum whereas a sampled b) has a discrete spectrum.
as long as the same predictor is present at both ends of the system. A stupid predictor might simply assume that the current sample has the same value as the previous one. This results in differential coding as in Fig.4b. A smarter coder might look at two previous samples to estimate the slope of the waveform as in Fig.4c. Using three previous samples allows the slope and the curvature of the waveform to be taken into consideration as in Fig.4d. Predictors of this kind work well on waveforms such as sine waves, but in the presence of transients the prediction fails. Essentially the predictor doesn’t see a transient coming, because it doesn’t follow from what went before.

Another type of prediction that can be used is in the frequency domain. Fig.5 shows that after a transform the result is a block of coefficients. By scanning the coefficients from one end of the block to the other, a predictor can be used to estimate the value of the current coefficient from the value of previous ones. Again any number of previous coefficients can be used to make the prediction more accurate. Predictors of this kind work best on broad or uniform spectra where many frequencies are present in the audio signal. Prediction in the frequency domain works poorly in the case of discrete tones because these make the spectrum spiky and the predictor fails to anticipate a discrete tone at one frequency because there is no energy in nearby frequencies.

Combining Figs.4 and 5 give another interesting example of transform duality shown in Fig.6. At Fig.6a in the time domain, a predictor fails to anticipate a transient because it is not a function of the earlier part of the waveform. At Fig.6b a time domain predictor works well on a sine wave because it contains no transients and the slope and curvature change smoothly. If we transform into the frequency domain we obtain an opposite characteristic. Fig.6c shows that in the frequency domain, the transient of Fig.6a has a broad spectrum, consistent with Fig.2c. A predictor working in the frequency domain finds the spectrum of Fig.6b easy to handle because it is reasonably smooth. In the same way Fig.6d shows that the spectrum of the sine wave of Fig.6b is a spike or singularity. This is difficult for a spectral predictor.

Thus the duality can be seen in predictors. A transient waveform that is difficult for a temporal predictor is easy for a frequency domain predictor, and a stationary waveform in the time domain that is easy for a temporal predictor is difficult for a frequency domain predictor.

This all sounds like a lot of abstract theory but it has a practical application. In MPEG Advanced Audio Coding (AAC) both temporal and frequency-domain prediction are used. The incoming audio is subject to a frequency transform, and the temporal predictor attempts to anticipate what coefficient values will be by using the values from earlier blocks. On stationary material this will work well. Additionally the frequency domain predictor operates within each transform block by trying to predict each coefficient from ones of lower frequency. This works well on transient material. The AAC coder is free to use either of these techniques and it may even try both to see which results in the most effective compression. AAC achieves a better subjective quality than Layers 1 and 2, which is no bad thing, and it does so by tangibly exploiting transform duality.
The Pro Audio A to Z is the first comprehensive directory combining information on the world’s leading manufacturers of professional audio equipment with a listing of the world’s principal pro audio distributors.

To be published in September 2000, this expansive volume will be an invaluable source of reference for anyone involved in the business of recording, broadcast, post-production, mastering or live sound. From manufacturers seeking new avenues of distribution to distributors looking to broaden their base of supplied lines, The Pro Audio A to Z will facilitate business-to-business communication in the most efficient way possible. There’s even a ‘Fast-Track’ section featuring industry and exhibition organisations, support services, PR agencies and more.

CD-ROM and online versions will be available after publication. Advertising opportunities start from as little as £500 – call +44 (0)20 7940 8531 or email rlawn@unmf.com to find out more!
Passing the buck

The mutterings of discontent over new audio distribution channels is becoming a roar. Peter Filieu ponders the position of the artist and producer.

Musicians are used to the fact that the big record companies in favour of (small bad) music distribution dotsoms and download directory services. The prospect of genuine commerce security is the straw to which we are still clinging but the speed with which new technologies arrive and are adopted casts a dark cloud over everyone's future. The SDMI standard seems to be the only game in town but there is still a measure of scepticism as to whether its implementation can stem the Napster terror. Much of the current confusion arises from the vast and disparate range of services and business models on offer. Sadly, many of the first players suffer from a Chicken and egg market-share dilemma — the need to acquire a critical mass of content before they can attract sufficient consumer attention to fulfill their promises to artists. This has led to there being little filtering of quality from dross. Currently, consumers are drowned in so much, usually substandard, choice that realistic profitable commercial volumes are hard to come by. This may change when the majors go on stream, but at the moment a new artist can expect to lose best tracks as free downloads and be sent looking for better venues to collect relatively insignificant mail-order revenue only for the first couple of quarters.

Maybe we should agree to reduce the quality of all 'free' downloads and apply intangible security to all high-quality distributions. Producers and artists are becoming increasingly concerned about how and at what stage security systems are applied. The old chestnut of inaudible watermarking still poses serious questions. Only recently have proprietary systems become commercially available and the very issue of competing regimes splitting up the total available catalogue can only add further problems for the consumer as well as causing nightmares for the producer. Listening tests have almost universally failed to satisfy the best expectations of audio professionals and the expression 'trade-off' is being bandied around. Maybe artists and producers will be forced to sacrifice some quality in favour of business but the idea that the imposition of a variety of packages, each depending upon particular commercial alliances, as an intrinsically post-production process is regarded by many discerning producers as untenable. In any event, the producers and the artists are not even in the loop.

Producer revenue streams are affected in parallel to artist revenues, but the situation for the producer is more critical. The question of performance revenues derived from related rights is still in the balance for producers and remixers, but there is an issue of even greater importance affecting all performers. While secure downloads can be compared to sales of physical products, the proliferation of users, some rebroadcasting from the horizon may, as was found with the standard SDMI, lead to a system where there is resistance to spending money on developing it in a cross-sector basis. As the studio producer is the one individual in the recording process who can provide accurate and real-source performance and asset protection data, facilitating the extra work should be a worthwhile investment. Sadly this seems not to be the case. Indeed, there are even been efforts to cynically transfer data collection responsibilities under take it or leave it contractual terms.

A true perspective of the current 'music on the Internet' situation would probably benefit from a regular reminder that, compared to say, commercial flight, the Web, even including broadband, is much closer to Lindburg's 'Spirit of America' than to Concorde. There may be new business models on the horizon that will place the studio producer in an altogether different position but our traditional role as bridge between inspiration and product will remain pivotal no matter how music is distributed. However, both producers and artists could find the quality of their recordings in jeopardy as a result of decisions made by company execs who in their enthusiasm to protect our business feel unable to trust those who make the music.

Producers and artists could find the quality of their recordings in jeopardy as a result of decisions made by company execs who in their enthusiasm to protect our business feel unable to trust those who make the music.

Different industry sectors have differing opinions on whether web-cast streaming is analogous to broadcast or whether as broadcasts they should benefit (or suffer) from related rights revenues. It seems the future of conventional broadcast is set to lie in web-casting. There are over 5,000 stations available all over the world, some rebroadcasting existing programmes and others providing new focused services. Web-casting organisations usually pay for music use, but not necessarily on a measured census basis even though on-line details of what is playing is available to the punter. Collection societies like PRS, ASCAP and BMI have historically relied upon measurement of usage to calculate distributions of licence income to composers and publishers. But the proliferation of users expected in the digital distribution environment threatens to expose an inability or possibly an unwillingness to afford the cost of the necessary data processing. The same applies to performance income to a greater extent. How PPL and PAMRA plan to handle web-casts is still up in the air. Even the question of whether web-casts qualify in the same way as conventional broadcast is a point at issue.

Another move in the US has the RIAA launching a royalty collection and distribution system of its own. Although at an early stage of development, fears are already being voiced over whether the US record companies are the right people to handle what is estimated to be revenue in excess of US$1bn in digital distribution sources and what or even whether performers will get any of it.

Key is metadata — the information about who played what, wrote what or owns what and how a recording was made — that feeds all commerce systems. Collecting and handling this information has always been a sector specific and piecemeal — record companies, publishers and the MII for example only wanting data that suits their own needs. While there is breadth acknowledgement of the need to develop a cost-effective and accurate metadata system there is resistance to spending money on developing it in a cross-sector basis. As the studio producer is the one individual in the recording process who can provide accurate and real-source performance and asset protection data, facilitating the extra work should be a worthwhile investment. Sadly this seems not to be the case. Indeed, there are even been efforts to cynically transfer data collection responsibilities under take it or leave it contractual terms.

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On balance, the marketing and career building added-value provided by experienced record companies (be they virtual or not) is likely to remain the most attractive option both for ambitious artists and discerning consumers. It remains to be seen whether the record company sector has the vision to include the studio producer community as full participants in future solutions. If they don't, the reduction in the cost of quality audio production and the convergence of the interests of producers and artists will encourage a vast increase in rights owning producers, management and production houses who licence their recordings to marketing and distribution entities that used to be called record companies.
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