Studio sound

EXCLUSIVES

Drawmer DC2476
CEDAR BRX+
Sony MDS-E11
Westlake BBSM-5
TOA IX-3000/5000
Roland VM3100 Pro
Quantec Yardstick
Merging Mykerinos
Alesis ML9600

STAR WARS

Sci-fi sound for the phantom channel

HISTORY OF THE MIXING CONSOLE
5.1 RECORDING IN THE MAKING
MP3 IMPACT AND IMPLICATION
WIN AN A&H GS3000

The
NORBERT PUTNAM
Interview

REVIEWS

FEATURES

COMMENT

TECHNOLOGY

78

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Buying into the method

PASSED ON AND INTEGRATED into our DNA with the first sip of Coca-Cola, even when it is used as a mixer, and fixed for generations when you accept those courtesy fries from someone else's MacDonalds, as a species we have become extremely good at being consumers.

You know it is the case when you wander out of the audio sphere and your area of expertise into the realm where all are equal in the eyes of the preying salesman and the saturating awareness campaigns.

Buy a washing machine and feel how you are swayed by the 'designer' layout of the controls and the easy-to-compare bullet point feature list. Let the mobile phone nestle in your palm and allow yourself to be persuaded that the smaller one is by far the most popular choice for those who can appreciate it. Catch yourself comparing car spec lists and electric gadgets and underlining BHP figures.

I cannot remember who said it but 'BHP sells cars, torque wins races' has always appealed to me as by extrapolation it condemns the his 'n' hers dash-mounted cup holders to the bottom of the list of important things.

When was the last time you appraised a selection of potential purchase equipment on their audio quality? Was the dealer able to arrange such a comparison or did you just base your decision on operational experiences and detailed analysis of the competing feature lists?

One of the many incredible things that digital has allowed the audio industry to achieve is to largely drive around the audio quality issue. In too many instances we take the audio quality of digital equipment as read and largely similar despite the fact that common sense suggests that they certainly can not be identical. And we have emerged from the reign of analogue, where audio quality was everything as the features were relatively few.

We are buying into this new method of appraisal and it bothers me that audio quality is just not high enough up that feature list. This has to be because the manufacturers seem to think that potential purchasers don't expect to see it there.

Surely this cannot be because we are not bothered by it anymore?

Zenon Schoepe, executive editor

Reading the past

CONSISTENCY IS EVERYTHING. Read my bedside table: *The Recording Angel* (Evan Eisenberg). *The Tears of an Angel* (Yukito Kishiro), *Burning Angel* (James Lee Burke)… If you are still unconvinced, you might look back to C16 Russia where, after a fire badly damaged Moscow, the authorities set out to predefine what an 'artist' should be in order to participate in its reconstruction.

The document of concern, penned over 300 years ago, was the work of a handful of authoritarians who believed that to leave the reconstruction of the city to circumstance was to court disaster. Instead, they chose to identify qualities and characteristics of architecture, painting, sculpture and the other arts that would ensure the regeneration of a city consistent with the cinders that were then Moscow. They did this obsessively—to the point of defining appropriate influences, listing artists and works that should be studied as references.

The question they had set out to answer was what should replace the old —a reconstruction of what went before or a new, modern Moscow? Their fears were that the new would obscure the old. That the new would be undeserving of the city's earlier accolades. They could not escape their fears.

Until recently, the document had not been translated out of archaic Russian. And even now, the only translation is into more modern Russian. Today it represents an important opportunity to compare and contrast conservatism and modernism. At the time it precluded the opportunity of a cultural paradigm shift.

Certainly, the nature of a paradigm shift is more closely allied to the objective doctrines of science than the subjective ones of art. But did the Russians miss an opportunity? I'd have to say that the British did when they rebuilt London after the Blitz…

There's nothing as simple nor as complex as progress. Comparable questions that pro-audio currently needs to answer concern what we are building with digital technology on the ashes of analogue. If we are embracing a genuine paradigm shift (as most believe we are), what is the potential cost in terms of yesterday's culture? The various, specialised groups protecting the past need more attention than most of us has time to give. Are we simply waiting to regret what we did not do? I invite you to think very carefully.

Tim Goodyer, editor

■ Kitchen Mastering, now at a new facility, designed by the Arthur Design Group, in Chapel Hill, NC has chosen Chevin A3000 amps to power its Duntech Sovereign speakers. Also in use are the Pacific Microsonics' HDCD convertor and a custom ATR-102. The facility has attracted artists and labels including Mammoth-Disney Records, The Squirrel Nut Zippers, Far Too Jones and Backsliders.

Chevin Research, US.
Tel: +1 508 429 6881.
Pacific Microsonics, US.
Tel: +1 510 644 2442.

■ London's new Wave post facility has installed Spendor active monitoring throughout for its stereo and 5.1 studios, in line with West London's Master Room 777 Productions whose recent refurbishment has seen SA500 and SA300 in all four studios.

Spendor, UK. Tel: +44 181 388 5000.

■ Belgian national broadcaster, VRT, has installed several 360 Systems DigiCart II Plus hard-disk recorders and Short/cut 99 'personal audio editors' in its Brussels TV studios. The installation is part of VRT's move towards digital working and will see the systems in use jingles, musical intros and sound effects, using 360's D-Net transfer network for broadcast.

360 Systems, US. Tel: +1 818 249 8885.

■ Florida-based WFLA TV has ordered two 36-channel Calrec S2 analogue production consoles for its new media centre. The facility, which will be shared between WFLA and the Tampa Tribune's newsroom, will provide both news and commercial production, one S2 being assigned to each.

Calrec, UK. Tel: +44 1422 842159.

■ Italian commercial television network, VideoTime, has ordered a second SSL Aysis Air digital broadcast console to replace a Soundcraft 3200 in Studio 11. Part of the Berlusconi Media Group that already boasts six Aysis Airs, the new console will be used for live entertainment and games shows.

SSL, UK. Tel: +44 1865 842300.

■ Canadian postpro house Deluxe Toronto has taken 12 Tascam MMR-8 MDMs and a MMP-16 modular player for use on feature film productions and network cable series. They are currently providing 24-bit audio for *The Taste of Sunshine*, starring Ralph Fiennes.

Tascam, US. Tel: +1 323 726 0303.

■ French all-digital private radio station, Nostalgie, has purchased two Klotz Digital 20-fader Spherion digital on-air consoles for two studios and a main control room with three Vadis mainframes. The station serves both national and regional programming. A Vadis main control room, and 12-fader and 20-fader versions of the Spherion have also been delivered to RMC Moyent Orient's Paris facility from where they will deliver French-language programmes to the Middle East.

Klotz, Germany. Tel: +49 89 46 23 38 75.

■ Among British hire companies, Birmingham-based SSE has ordered a second 48-channel Soundcraft Series Five front of house console and a pair of Soundcraft SM12 monitor consoles, while

Wigwam has taken its first SM20 to replace an older SM12. London's Gig Hire, meanwhile, has also ordered Europe's largest SM20 specifically for the Sidmouth Folk Festival, while The Sound Company has added to its collection of Denon DN-C630 CD players. A novel hire option is offered by Dreamhire whose range of vintage and rare outboard comes with the collaboration of Strongroom studios.

SSE, UK. Tel: +44 121 766 7170.
Wigwam, UK. Tel: +44 1706 363800.
Gig Hire, UK. Tel: +44 181 655 9435.
The Sound Company, UK.
Tel: +44 171 286 7477.
Dreamhire, UK. Tel: +44 181 451 4454.

■ ITN's refurbished London news facilities have taken a 14-channel WL-8000 wireless mic system along with the new WRT-805 belt transmitter and ECM-77 miniature lavalier mics. Supplied by Total Audio Systems, the UHF system will be divided between the C4 news studio and ITV production facility. The new system joins the Sony WRT-860A/WRT-810A/WRR-801A Freedom system in C4 News' Mobile Production Unit. The BBC's World Service, meanwhile, has bought 125 beyerdynamic M58 interview mics in readiness for a busy summer OB schedule. The new mics will be used primarily for news gathering. Further north, Chrysalis Radio's new Galaxy dance radio station has taken two Audionics on-air studio consoles. The desks are part of a complete studio package supplied by Audionics.

Sony, UK. Tel: +44 1256 355011.
Total Audio Solutions, UK.
Tel: +44 7000 456000.
Beyerdynamic, UK. Tel: +44 1444 258258.
Audionics, UK. Tel: +44 114 242 2333.

■ Tokyo's Bernie Grundman mastering studios has fallen in line with its LA sister in installing Pacific Microsonics' Model One HDCD processors in all mastering rooms. Model Two HDCD processors are on order for DVD-Audio services at both facilities and for Grundman's forthcoming European facilities.

Pacific Microsonics, US.
Tel: +1 510 475 8000.

■ Australian pay-TV giants Foxtel and Austar have invested in 23 Orban Optimod 6200s in preparation for the 2001 launch of digital broadcast services. Austar offers 13 channels of varied programming while Foxtel has 31 including locally produced material. Australian education advances through SAE Perth's upgrade. The new facility includes a 32-channel Soundtracs Jade console, Otari MX80 multitrack, Pro Tools system and Alesis ADAT XT20s.

Orban, US. Tel: +1 510 351 3500.
Soundtracs, US. Tel: +44 181 388 5000.

■ Scottish Radio Clyde has taken Audix Broadcast's ADD7000 series of digital audio mixers to accompany ARM matrix and DATH telephone hybrids. The 8-channel desks all have dual input switching, channel limiters and stereo PPM metering.

Radio Clyde, UK. Tel: 44 141 306 2272.
Audix Broadcast, UK.
Tel: +44 1799 542220.

▲ **US:** After 15 years as one of Avatar's workhorses, the **NYC** studio's SSL 6000E console has been moved aside for a new 9000j. 'We wanted to keep the room analogue, that was the first thing,' reflects President and General Manager **Zoe Thrall** (pictured with engineer **Dan Gellert).** 'Let's just say that we looked at every major recording console out there before choosing the 9k. We looked, we listened... Keep in mind that the board of directors are all engineers and producers, and we rely very heavily on their input because they're out there making records. Who better to ask?

'There were a couple of vintage consoles that came up—one was in Europe and I think the other was in Australia—over the last year. One was an 8078 the other was an 8068, both fully tweaked out, fully modded with GML. It was a serious consideration because this studio is known for the flexibility we have in tracking and the great sound that these rooms have, so it seemed like a good option to put in a nice vintage console because they sound so great. The SSL came up and, you know what, it really challenges the vintage consoles. And being new, you avoid all the inherent problems of an old desk and you have modern mixing capabilities. We ended up with the 9000. Now that it's in, I can't imagine any other console in there.'

Until the console arrived, the studio monitors had been **Urei 813s.** After measuring the room response, **Sam Kinoshita** returned to Japan to build a custom monitor system based on a horn and four 10-inch drivers to replace them. The **Bryston 7B** amplifiers will remain the driving force behind the.

'None of our control room monitors are soffit mounted and, to my understanding, Sam's never made a pair of monitors that weren't soffit mounted so it was unique for him to tackle it,' Thrall comments. As for the old SSL... 'We still have it and we have plans for it, so it's not retired yet. I think we're going to make a small demo-preproduction place with it. A small "creativity" place.

'Studio D will be the next room that we'll tackle. We haven't decided what equipment we want in there yet, but we're planning a major reconstruction for that room. It was originally designed to be a mix-to-picture facility, and we really need it to be a high-end mix room, so we're going to look at it that way. So yes, a major redesign acoustically. I don't know who's going to do that yet but we're talking to a few people.'

Audio coding first

Italy: The AES will host the world's first conference on professional audio coding between in Florence, Italy on September 2nd–5th. The 3-day event will take place in an old tocsin villa on the outskirts of Florence and will focus on all aspects of high quality audio coding. Topics will range from high-definition coding to low bit-rate coding like MPEG4 and include 31 papers and 3 introductory tutorials

Chairmen are Marina Bosi (AES President) and Karlheinz Brandenburg (Fraunhofer Institute) supported by Juergen Herre (Papers chair) and Giorgio Parladori.

Registration for the AES 17th International Conference High-Quality Audio Coding via the Fraunhofer Institute. Contact Mrs Michaela Beasley. Tel: +49 9131 776 306, Fax: +49 9131 776 398 or e-mail: 17th_confinfo@aes.org. The program and online registration are available at the AES homepage. Net: www.aes.org.

Aphex and Telex patents

US: Patent Number 5,898,395 discloses Aphex' method of analysing the duty cycle of the most significant bit of an A–D convertor to determine if the data stream is centred (there is no DC offset). If it is not centred, a correction signal is then fed into the input of the convertor to make the correction.

DC offset in the output of a convertor is conventionally handled by a high-pass filter in the digital domain. When a signal whose peak is close to maximum input is fed into the convertor, however, the DC offset may cause overload in the conversion process. That means that while the high-pass filter will remove the DC, it will not take away the overload distortion.

▲ UK: Producer Nellee Hooper has insra led Genelec 1036 main monitors (seen above with engineer Pete Lewis) in his West London HQ, The Studio. The choice was determined in part by the bass requirements of Hooper's dance-based production work. Recently refitted to include an SSL console, the studio has a 5.1 surround theatre above it.

▲ UK: London's Royal Festival Hall recently hosted a concert by celtic band Iona, London's All Souls' Orchestra and a Yamaha-Meyer SR rig. Also the occasion of a live recording, a 56-channel Mackie desk, six Tascam DA-88s and B&W monitoring were under the command of engineer Nigel Palmer and supervisor Matt Parkin. The event employed some 70 mics including B&Ks with Millennium Bug mounts, Neumanns, Schoeps and AKGs. FOH were a Yamaha PM3500 FOH desk, Soundcraft SM20 monitor desk and MSL-4 and 650P cabinets, all supplied by B&H Sound.

The design development came when Aphex engineers Donn Werrbach and Gary Liden were designing an A–D convertor for use in Aphex' FM Pro Model 2020 analogue broadcast processor.

Telex' Patent 5,883,804, meanwhile, covers modular processing for digital system architecture. Handling digital audio and control infrastructure for managing a modular family of distinct but interoperable audio function modules.

ATI group buys API

US: The ATI Group has completed an asset purchase of Virginia-based API Audio Products.

Well known for its studio and broadcast mixing consoles as well as a strong line of external processing devices API developed all-discrete audio equipment throughout its 30-year history, notably the Legacy recording console, the Legacy Plus, and a range of console modules that can also be rackmounted. API also developed the industry standard 2520 op-amp.

The ATI Group is also the parent company of Audio Toys Incorporated and Uptown Automation. Audio Toys manufactures the Paragon mixing console and associated rackmount gear. Uptown Automation manufactures and installs moving fader and mute automation for analogue mixing consoles.

Magazine launch at IB0C

UK: *Content Creation Europe* (CCE) is a new magazine for the rapidly growing content creation industry, launching at IBC99. It will serve and represent a community of professionals who generate video -based content for a wide range of media—from broadcast television, film and video to games, the Web and DVD.

The sort of activities undertaken to achieve the above include editing, special effects, animation, 3D, authoring and design. The launch team at Miller

Freeman has recognised that there's an increasing degree of crossover between the computer-based professional tools employed to accomplish these complex tasks for a variety of distribution media. Their aim in launching CCE is to provide sharp and incisive coverage of those tools and technologies, and thereby appeal to a broad range of practitioners. If a professional product is used to create content anywhere in Europe, it belongs in the pages of *CCE*; everything from workstations, acquisition devices and codecs to nonlinear editing systems, 3D software and data networks. Through the agencies of news, reviews, profiles and roundups, the new magazine will relay crucial information to the potential purchasers and users of these products—and identify new business directions for this dynamic media sector.

Content Creation Europe will be published by Miller Freeman UK Ltd, whose established European sister titles include *One to One, Pro Sound News Europe, Studio Sound* and *TVB Europe*. Further sister magazines published by Miller Freeman PSN Inc. in the United States include *EQ, Government Video, Pro Sound News USA, Television Broadcast* and *Videography*. Editorial: Nicola Godwin (ngodwin @unmf .com). Advertising: John Hancock (jhancock@unmf.com).   **>**

▲ US: PreSonus Audio's ACP88 was an integral part of the annual 5-day Les Festival International de Louisiane where the 8-channel compressor-limiter-gate was used for processing the live broadcast of the festival. This festival is an annual visual and performing arts celebration of the cultural heritage of southern Louisiana—fusion of French, African, Caribbean and Hispanic influences. Over 350 bands and artists participated in the event. World Cafe, a 2-hour programme produced by Public Radio International (PRI) broadcast parts of the event. Worldwide access to the programme was provided by KRVS radio station whose signal originates in Lafayette, LA —the home of the festival. Presonus. Tel: +1 225 344 7887.

Studio Sound publishes World Report

International: Recognising the need for comprehensive, up-to-date market data, *Studio Sound* has teamed up with market research experts MTI (Market Tracking International) to produce two volumes of unique, essential industry intelligence.

The 350 page *Pro Audio World Report* (Vol. I) provides purchasers and manufacturers with an invaluable insight into the workings of this broad, dynamic market. It dissects the major issues affecting the international industry, analysing the impact of economic conditions and new technologies on a wide range of end-user markets across key countries. The 150-page *Pro Audio Facility Survey* (Vol. II), reveals the vital findings of *Studio Sound*'s 1998 survey of 750 professional recording facilities around the globe.

Studio Sound's executive editor, Zenon Schoepe, believes professionals in all aspects of the business have been crying out for this sort of research. 'The facts and figures contained in these volumes have been a long

time coming for an international industry affected by so many external factors. Now, for the first time, trends can be mapped and crucial decisions made on the back of solid, reliable information.'

The research indicates some promising forecasts for the industry:

'Studios are generally confident about the industry's future prospects... 61% of respondents expect their turnover to increase in the next 2 years.'

'Advances in technology will enable facilities to enter the advertising market... in 1997, total media spending was estimated at around $289 million...'

The in-depth world report, helpfully broken down into sections and subsections and including tables, graphs, company and country profiles, is the only comprehensive market data resource available to the global pro audio industry.

Studio Sound will publish more details of the report in the August 1999 issue.
Enquiries to Lianne Davey.
Tel: +44 171 940 8598.
Email: lkdavey@unmf.com

8

July

8–10
11th PALA 99
Singapore International
Convention and Exhibition
Centre (SICEC),
Contact: Ann Tan,
IIR Exhibitions.
Tel: +65 227 0688.
Email: ann@iirx.com.sg

26–28
**Conference:
Advanced A–D and
D–A Conversion
Techniques and
their Applications**
University of Strathclyde,
Glasgow, UK,
Contact: ADDA99 Secretariat.
Tel: +44 171 344 5472.
Email: adda99@iee.org.uk
Net: www.iee.org.uk/Conf/

30–1 August
ABC 99
Rajah Muthiah and Rani
Meyyamai Halls, Egmore,
Chennia (Madras), India.
Contact: Exicom
Tel: +91 22 641 2519.
Fax: +91 22 641 2522
Email: exicom@bom.2.
vsnl.net.in
Net: www.exicomindia.com

September

2–5
**AES UK Conference:
High Quality
Audio Coding**
Florence, Italy.
Contact: AES.
Tel: +44 1628 663725.
Fax: +44 1628 667002.
Email: uk@aes.org

5–8
Plasa 1999
Earls Court, London, UK.
Contact: P&O events.
Tel: +44 171 370 8228.
Email: sophie.matthews
@eco.co.uk
Net: www.plasa.org

10–14
IBC 99
Bolkesjø Mountain Hotel,

Amsterdam, Netherlands.
Contact: Gina Christison
Tel: +44 171 240 3839.
Email: show@ibc.org.uk
Net: www.ibc.org.uk/ibc/

23–26
**Nordic Sound
Symposium**
Bolkesjø Mountain Hotel,
Bolkesjø, Norway.
Contact: Richard Andersen
Tel: +47 67 54 14 83.
Email: riander@online.no

24–27
107th AES Convention
Jacob K Javits Convention
Centre, New York, USA.
Contact: Chris Plunkett, AES.
Tel: +1 212 661 8528.
Email: 107th_exhibits@aes.org

October

8–17
Telecom 99
Palexpo, Geneva, Switzerland.
Tel: +41 22 730 5969.

15–17
MusicBiz 2005
Ex'pression Centre for New
Media, Emeryville, California, US.
Contact: Keith Hatschek
Tel: +1 415 227 0894.
Email: info@hatschek.com.

21–23
Broadcast India 99
World Trade Centre, Mumbai
(Bombay), India.
Contact: Kavita Meer, Saicom.
Tel: +91 22 215 2721.
Fax: +91 22 215 1269.
Email: saicom@
bom2.vsnl.net.in.
Net: www.saicom.com
/broadcastindia

November

2–3
**24th Sound
Broadcasting
Equipment Show**
NEC, Birmingham
Contact: Point Promotions.
Tel: +44 1398 323 700.
Email: info@pointproms.co.uk
Net: www.sbes.com

19–22
**SMPTE Conference
and exhibition**
New York Marriott Marquis,
New York, US.
Contact: Bryan Nella.
Tel: +1 914 761 1100.
Net: www.smpte.org

22–24
Messe Frankfurt
Trade exhibition and conven-
tion for audiovisual system
installation
CMF, Ludwig-Erhard-Anlege
160327 Frankfurt
Contact: Metin Ergül
Tel: +49 69 7575 6130.
Email: metin.ergul@
messefrankfurt.com

December

8–10
Convergence India 99
Pragati Maidan, New Delhi, India.
Contact: Exhibitions India.
Tel: +91 11 463 8680.
Fax: +91 11 462 3320.
Email: exhibitionsindia@vsnl.com
Net: www.exhibitionsindia.com

June 2000

6–9
**Broadcast Asia 2000,
Cablesat 2000 and
Professional Audio
Technology 2000**
Suntec Centre, Singapore.
Contact: Singapore
Exhibition Services.
Tel: +65 338 4747.
Fax: +65 339 5651.
Email: info@sesmontnet.com
Net:www.sesmontnet.com
UK contact: Overseas
Exhibition Services.
Tel: +44 171 862 2080.
Fax: +44 862 2088.
E-mail: singex@montnet.com
Net: www.montnet.com

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BUSINESS PRESS

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Total average
international circulation:
21,648
Jul-Dec 1998
BPA audited

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A United News & Media publication

Class distinction

I WAS QUITE HAPPY with Mr Watkinson's explanation of class-D amplifiers (*Studio Sound*, April 1999) in his excellent audio basics series. The operation principle was nicely explained with no traditional prejudice in sight. The subsequent installment was less satisfying, however, when dealing with the proposed class-D/class-A hybrid.

Such a contraption is quite unnecessary, if only one were aware of the current state of the art of class-D amplification. Some examples include a number of class-D amplifiers such as those designed by our laboratory (Philips ITCL, Belgium) that are putting out hundreds of watts at distortion figures of 0.003% and less, with sound quality good enough even for 'tubeophiles' to consider an upgrade. At least three parties (including myself) have working all-digital class-Ds (operating without error correction of any kind) delivering better than 0.03% distortion figures. ITCL's upcoming G4A amplifier has expected distortion performance of <0.001% and output impedance <0.01Ω at all audio frequencies.

I'd like to see anyone do that with even the classiest of class-A amplifier technologies.
Bruno Putzeys, ITCL, Belgium

Ribbon 1

I ENJOYED YOUR recent article on ribbon mics (*Studio Sound*, May 1999). Regarding the Royer units and the upcoming stereo version, it has in fact been around for several years under its original manufacturer's name: the Speiden SF-12. I have one here, serial #127. On their web-site (www.royer-labs.com) they say:

'After testing an R-121 production prototype, David's long-time friend Bob Speiden offered his highly regarded Speiden SF-12 stereo ribbon microphone to the company for consideration—it is now Royer's second production ribbon microphone.'

It's an excellent unit and quite durable, although mine seemed to go a bit funny after recording a local percussionist in a concerto for percussion and wind band, and I had to send it in for servicing. I find it's really useful for spot miking piano, especially in m-s mode, and a friend who is the head engineer for the CBC here in Vancouver used his as a spot mic on Cecilia Bartoli for a broadcast a few years ago. I hope Royer send you one in for evaluation!
Stuart Tarbuck, Vancouver, British Columbia, Canada.

Ribbon 2

MAY I ADD to Dave Foister's excellent article extolling the virtues of the Ribbon Microphone (*Studio Sound*, May 1999)?

I've always heard such microphones referred to by professionals for their warmth of sound and presence—the latter without recourse to frequency shaping. I've used beyerdynamic M160s for

ACTIVATE

over 30 years, and find that by comparison with today's large-diaphragm condensers the sound is 'solid' and 'immediate' without the 'separated' and 'transparent' features associated with loud mixing. Not 'boomy' or 'muddy', but clear and outstanding if used correctly, particularly on mid-frequency textures.

An example to emphasise this: A record issued in the mid-seventies in the UK on Epic entitled *Picnic at Hanging Rock* by Pan Pipe virtuoso Gheorghe Zamfir comprised tracks from two sessions—a set done in 1970 recorded live in a Swiss church using (if I'm not mistaken from a photo I've seen of the session) a couple of pairs of M160s, and secondly a studio set done in 1971 using condensers. The 1970 tracks sound natural and 'woody' from an instrumental point of view, whereas the 1971 tracks sound metallic, thin and processed. The 1970 tracks are much easier on the ear, and the music comes over with

greater meaning. The difference is immense; well worth an investigative listen.

I've always understood that the reason for beyerdynamic's dual-ribbon approach was not for producing a hypercardoid response (I have a number of single-ribbon M260s which also exhibit the same response) but for producing a 'linear' amplitudinal response, a point the sales blurb used to refer to. My conversations with the beyer-dynamic sales rep from many years ago suggested that by putting one ribbon on the leading edge of the magnet and the other on the trailing edge caused any logarithmic artefacts of the front ribbon to be cancelled out by the anti-logarithmic artefacts of the rear one, and vice versa.

Does this not suggest that every other microphone is 'nonlinear'? Correct me if I'm wrong, but do not single element microphones have an amplitudinal response which is somewhat proportional to the square of the distance

between the diaphragm and the back-plate/magnet? That being so, have we spent the last 50 or so years conditioning ourselves to a nonlinear sound?

Dave mentioned source impedance. Again, my understanding from the beyerdynamic rep of the time suggested that the ribbon was effectively a 'single-turn' transformer primary, with the secondary providing the usual 600Ω output. Agreed, the output is very low compared for example with that from Sennheiser's MKH 405 and 415 series using the Varactor principle, but did not an American company many years ago (Cambridge, Massachusetts?) produce a transformerless ribbon with inboard electronics which overcame the sensitivity problem?

A good article, Dave, and thanks for bringing to the fore a genre which for those of us unaffected by technically sounding music was never out of vogue.
**Christopher Cathles,
CAMM Music, UK**

RUBY REWARDS

STUDIO SOUND'S 40TH ANNIVERSARY party continues with Allen & Heath's GS3000 mixing console taking centre stage. As you can see from the accompanying listing of equipment, many of pro-audio's top manufacturers have conspired to make our celebration one to remember by building a one-off custom model. In this third installment, it's a 24-channel A&H GS3000 console. The GS3000 (*Studio Sound*, July 1998) is an analogue 8-bus in-line console with twin-fader, dual path inputs, designed with project and commercial studios in mind. It features two valve preamps, which can be patched to individual channel inserts, groups or the LR path. The preamps incorporate A&H SVT (Symmetrical Valve Technology), allowing them to be used in balanced mode for regular inputs or in single-ended 'guitar' mode, which drives the valve to give the type of harmonic distortion much sought after by guitarists. Each mic-line input has 4-band EQ with fully parametric mids, that can also be split between the monitor and channel paths, and the console features MIDI mute automation and MMC.

ALL YOU HAVE to do to secure a unique Ruby GS3000 is to answer the questions below and clear your conscience of anything that might displease Lady Luck.

THE QUESTIONS

Q1 What is the name of A&H's first digital mixer?

Q2 How many valves are used in a GS3000?

Q3 What colour are the GS3000's pan pot caps?

CLOSING DATE FRIDAY 1 OCTOBER 1999

TO ENTER, you can either email your answers to ruby.competition@unmf.com, fax them (to +44 171 407 7102) or send them on a postcard to Ruby Competition, *Studio Sound*, Miller Freeman Entertainment, 8 Montague Close, London SE1 9UR, UK.

As long as you are a registered *Studio Sound* reader, you may enter any number of installments of the competition as long as you do so separately (multiple entries will be collected and used as fuel for Studio Sound's summer barbecue season), and include your Unique Reader Identification Number.

...include your Unique Reader Identification Number.

The Unique Reader Identification Number is the 9-digit number located in the middle of the top row of your *Studio Sound* address label.

REWARDING RUBY PRIZES

AKG 535 stage condenser, C1000, and C3000 microphones

A&H 32-channel GS3000 console

CEDAR Series-X DHX Dehisser

Drawmer DS201 dual gate

EMO E520 Single DI box; E445 cable tester; E325 3-way mic splitter

Genelec 1029 monitors

Joemeek VC1 compressor

KT graphic DN-360 graphic

Marantz CDR640 CD recorder

TL Audio C1 Classic compressor

Purple Audio MC76 compressor

Ongoing thanks are due to all those who have so readily contributed equipment, time and advice in the preparation of this competition.

TOA IX3000-IX5000

Breaking technical and geographical boundaries, TOA's IX-series digital consoles are making a serious play for the international market. **Rob James** sits in on the latest iterations

THE NAME OF TOA will be well known to people working in the spheres of installation and live sound, but may be almost unknown to others. This is not the case in TOA's native Japan where it began operations in 1934 and has long been a major player in the market. It will also come as a surprise to many that TOA installed its first digital console, an IX9000, not in Japan, but in Europe in 1990. This console was born out of a 2-year collaboration with the Vienna State Opera. A few months later TOA began another joint venture with the Japanese broadcaster NHK to produce a fully automated broadcast console. The result was the IX1100.

Outside Japan, notable installations include four desks at ORF the Austrian Broadcaster.

Anyone with any experience of the development of a digital console will be aware of the steep learning curve, the anguish involved in getting the thing to fly, and the level of support necessary. Digital console manufacturers have targeted music recording and sound for picture, at least in part because 'teething troubles' are more tolerable in these areas. TOA is the exception in aiming its first digital consoles at the fault intolerant theatre and broadcast arenas. With this experience, the company is seeking to broaden the range of applications and to widen its market. The first fruits of this initiative are the IX3000 and IX5000 series consoles.

Both consoles use the same core hardware. Elements of the control surface and software are tailored to suit the specific application and, correspondingly, versions are available to suit broadcast, theatre, post and music applications. The unit under review was built for demonstration purposes with elements of each type. As such it fulfills its purpose of showing the versatile modular architecture. The software on this particular console is best suited to theatre work so I intend to focus on this area.

TOA has elected to follow a 2-unit approach, keeping all the audio processing and I-O in 19-inch racks separate from the control surface. This is connected to the rack by a single, lightweight, Hybrid Bus (H-Bus) cable. The H-Bus is able to network multiple consoles, racks, remote head amplifiers and GPI-MIDI interfaces. A TDM (Time Division Multiplex) bus is employed between the I-O and DSP racks. There are many possible I-O configurations with a maximum of 256 inputs and outputs. Available options are AES-EBU or SDIF-2 digital with or without sample-rate convertors, 20-bit sigma-delta 64x oversampling A–Ds and 20-bit sign magnitude 8x oversampling D–As. These come in multiples of 16. I-O may be patched as required from the control surface. The overall limitation on maximum configuration is the number of signal channels—currently set at 160.

The console can be factory configured for a maximum of 60 buses, for which all channels have full processing at all times with the exception of delay which is optional. The inherent processing delay is fixed at about 1.3ms including conversion. However, if buses are reassigned (for example, groups to main outputs), the delay is doubled because you are routeing through two layers of processing. Operating level is factory defined and may be at any of the usual standards—the demo console was set to the EBU level of -18dBFS. Internal sampling rate is fixed at 48kHz and the console is intended to be synchronised to external word clock, although it will generate its own if necessary. The control model is an interesting mixture of assignability and layering with some unusual and clever extras. The control-surface construction has a satisfying heft. The modules slot into the frame in three planes. The faders are on the level behind the big armrest. Quick access keys and four rotary assignable parameter entry knobs are sloped and the pan-trim control, bus assignment indicators and meters are in the raked upstand. The angles are well chosen, at any rate they seem comfortable to me with a good compromise between keeping things within reach and keeping a reasonably low profile. The channel strips come in eight wide modules. System control uses a mixture of dedicated and assignable keys

and an LCD touchscreen. In the past I have never been a fan of the touchscreen, but here it is well integrated and feels natural in use. This sensation is helped by an audible relay click when anything is selected on the screen. A nice touch.

A single channel strip starts in the upstand with two red clip indicators, LED bar-graph level meters and gain reduction meters, gate indicator, LED bus assignment indicators—the exact arrangement depending on the factory configuration—a horizontal LED bar pan indicator, red alphanumeric function indicator and PAN-TRIM knob with Select keys. All knobs are touch-sensitive and need a positive grip to activate them. I found this disconcerting at first since I am used to flicking knobs with a finger, but I can see the wisdom in this approach on a live console since it protects against inadvertent adjustment. Each knob has a concentric arc of 19 indicator LEDs. The metering point is user-selectable between pre-EQ and post-fader positions. The sloping panel below the upstand is divided into five blocks. Four of these are identical and each consist of knob, concentric indicator, ON-OFF key and yellow alphanumeric annunciator. The function of each of these blocks changes according to the nine access or global SELECT ALL keys at the bottom. Normally, these knobs control and display parameters on a strip-by-strip basis, but it is also possible to view and adjust all parameters associated with an individual channel by using all the knobs and indicators on the eight strips in a block to display and control a single channel. The fader section starts with a big, square ACCESS key, a STEREO-STAY indicator, green alphanumeric

6-character channel name display, PFL and AFL keys and a red MUTE key that lights when the channel is cut. The motor faders are 100mm ALPS types with adjacent Write-Safe-Isolate indicators, position indicators and Relative-Replace and Match mode indicators with a recessed MODE key to locally switch between them. A single strip can control a mono, stereo, LCRS or 5.1 source.

The control section is usually located in the centre of the console, but can be specified anywhere in the frame if the application requires it—such as when the monitoring sweet spot is not in the centre of the console.

The touchscreen in the upstand is a bright 640 x 480 TFT colour device and is used to set up the console and display information while mixing. For example, if EQ is adjusted on an individual channel the screen displays a graph of the combined effect of all the frequency bands together with numeric data for each parameter. Routeing maps allow the operator to look forward at all the destinations a source is routed to or backwards from a destination at all the sources feeding it. Two blocks of four keys flank the assignable parameter entry section; the left-hand block controls display windows on the touch screen with keys for INPUT, AUX, GROUP and MAIN outputs and VCA-style fader ganging.

The right-hand block has system functions. The RESUME key reloads all the console settings to the same state prior to switch off. PRESET stores up to 50 sets of console settings for recall. DISK allows access to the diskette drives for further preset and parameter storage. CLOSE closes windows on the screen and restores the channel display.

To enable a modest number of strips

to control a much larger number of channels a paging arrangement is employed. This operates on the block of eight strips. Eight FADER PAGE keys allow near-instant switching between user-defined pages of fader assignments. Individual faders may be locked on the surface independently of the FADER PAGE key selections.

TOA uses the Motorola 56k series of DSP chips along with custom silicon and Motorola CPUs. As an early entrant on the digital console scene the company has proved its ability not only to develop a viable console but to maintain the currency of the technology through several generations of components. Each channel strip can have any or all of the following processing blocks. LP and HP filters at up to 24dB per octave in 6dB per octave increments. 4-band parametric EQ in sixth octave steps with ±18dB of boost or cut and Q variable between 0.1 and 5.0. The maximum Q does not provide a particularly narrow band, but is well chosen for sweetening. If steeper filters are required bands may be stacked on top of each other to increase the depth, and, if demand exists, the Q may be increased. A compressor-limiter with soft-knee characteristics, Ratio, Threshold Attack and Release controls may be employed or gate-expander with the same controls plus hysteresis which sets the difference between the attack and release thresholds. Alternatively a combined compressor-gate may be used. External devices may be inserted pre-fader or pre-EQ and there is a variety of possible AUX configurations depending on specification. Standard configuration is from 12 to 16 auxes; N-M or mix-minus operation particularly applicable to >

◄ broadcast work is also catered for.

Output busing varies according to the specification of the individual console. One 'standard' configuration is two independent stereo mains and eight groups which can be reassigned to the mains, but note the earlier remark concerning latency. The rest of the possible control surface configurations vary according to the intended purpose of the console—with a several options available for monitoring control, talkback and oscillator functions. Theatre desks may be specified with eight buses set up to feed multiple speaker destinations with dynamic automation of panning. Motorised joystick panners and full 5.1 panning and downmixing for post use are under development.

The automation software also varies according to application. For theatre use a combination of static and dynamic snapshots give up to 200 snapshots that can be stored in local memory. Snapshots can be named and the order varied for performances or rehearsals. Dynamic shots allow variation of parameters over time to be initiated at a key press or time-code trigger. It is possible to preview shots on the console surface without affecting the audio. It is evident that this automation has been developed over considerable time with a great deal of feedback from real-world theatre operation. I was unable to test the more conventional dynamic automation to time code which would be used for music or post applications, but to judge by the performance and features of the dynamic shot automation it should be fine.

In the short time I was able to use the console I was impressed by the thought and development that has gone into it. Any successful console in these target areas is aimed at making life simpler and safer for the operator. Comfort factors include redundant power supplies and hot swapping of modules. Automatic substitution of defective modules is under development. Other safety features include Resume which will restart the console with exactly the same parameters and the ability to continue passing audio while rebooting the surface. In the time I had with the demo unit I can only hope to skate the surface of what is possible. Suffice to say I soon felt at ease with the thinking behind the TOA console. The touchscreen displays the information needed when you need it, and the surface controls are reminiscent of a top end analogue desk. The general layout, and colour, and size of keys lend themselves to the instinctive operation theatre and broadcast work demand. The control surface construction feels robust, and may tolerate the abuse that years of use generate. Thanks to the assignable architecture the control surface can be kept compact. This is valuable in theatre work where a console occupying one extra seat can cost thousands a year in lost revenue. The network architecture also allows for an additional compact front-of-house console for rehearsal and programming in theatres where the control booth cannot be used. Once programmed the show can be run from the main console. If this console is anything to go by we shall be hearing a great deal more from TOA in the future. ■

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Drawmer DC2476

Taking advanced digital dynamics processing and adding a convincing valve emulation stage, Drawmer's entry to the digital boys' club is complete. **Dave Foister** signs them in

DRAWMER, ACKNOWLEDGED master of the analogue dynamic processor, surprised us all a couple of years ago by introducing digital elements. The combination of valves and A–D convertors in the 1960 was, while not unique, a decided departure for the company, but its latest range goes even further by operating entirely in the digital domain.

The unit offered to me for review is the DC2476 Digital Mastering Processor, clearly aimed at the same market as tc's Finalizer. It comprises a collection of processes relevant to mastering, from EQ to dynamics, but does not set out to be a Finalizer clone; its approach is quirkily different and it has some strong ideas of its own.

Drawmer's front panels have traditionally been the epitome of restrained blackness. Every one of the familiar analogue processors has commendably put function before style, making them easy to get round and fast to set up. The new range could be thought to go to the other extreme, with lurid

main parameter area, and its current state is shown by a black arrow in the corner. Whichever is selected, there is a four-direction backlit rocker-button beside the display for moving around it, and a big dome-shaped knob with a built-in push switch that adjusts the various parameters. The rocker takes a bit of getting used to, as it's not difficult to move it sideways when you mean down, and to think you're moving around the parameters when in fact you're on the menu bar. But this is easily mastered, and then becomes quite intuitive.

The palette of treatments this lot gives access to is very broad and comprehensive, with heavy emphasis on analogue-style processing. A full set of dynamic processors is included, with expanders, compressors and limiters all operating on three definable frequency bands. There is also a full-band compressor attached to the dynamic EQ section. There is 5-band EQ modelled on analogue circuitry, and the *pièce de resistance* is a valve emulation stage.

out uses the bootstrap approach, where increasing compression keeps peaks in the same place while pulling up the quieter elements. This avoids the need for gain make-up either manually or automatically. The only adjustment for the limiters is the release time, as the threshold is set at digital full scale.

Note that the valve section follows this, as does the output page where the relative gains of the three bands can still be adjusted, but the overall gain management system avoids the possibility of these undoing the work of the limiter. Alongside it for some reason is 3-band control of the stereo width, an interesting and effective option that allows the top end to be opened out while the bottom retains its tight centre or can even be narrowed.

The two EQ sections complement each other well. The dynamic EQ is deceptively simple, consisting of a single parametric band with variable dynamic gain up or down. The filter section can be monitored for fine tuning, and then the whole thing can

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colours, the now-obligatory curvy bits, and most things controlled from a screen and a few buttons with just one knob. It is quite a relief then to find that it is still easy to navigate, and that Drawmer's accessible style has been transferred to the medium quite well. Although I belong to the minority who think that manuals are there to be read rather than lost, it is always good to be able to get started before reaching for the guide book.

On the 2476 there is one key button next to the screen without which you will get nowhere; it switches control between a menu line at the top of the screen and the

Very comprehensive control over all these processes is provided, aided by a few automatic variables. Time constants for the dynamic processors are 'programme adaptive' and the entire chain is subject to a gain management system that monitors the signal level at several critical points and ensures that no internal overloads occur.

Nice touches abound, starting at the input stage where the screen shows the incoming signal level as a kind of sweeping oscilloscope display. This kind of graphic approach applies to several sections, most notably the valve stage, where three valves appear on the screen, a bar within each showing how much effect has been applied.

More importantly, the processes themselves are all very strong. The dynamics shine, as one would hope on a Drawmer design, with easy control over how gentle or aggressive the treatment will be, and the clear mastering benefits of multiband operation. Of course all the bands can be adjusted together, bringing the advantages of avoiding interaction between areas of the spectrum, but also each band can be adjusted separately, allowing complex tonal shaping at almost every stage of the chain. The effect of using a lower threshold for the upper and lower bands than for the middle is something familiar to mastering people but few others, and can beat any EQ for sheer enhanced loudness. Compression through-

do a good job of enhancing a particular area without bringing up all the low-level signals, or dealing with pops or esses in a simple and effective way. Attached to the dynamic EQ is a simple full-band compressor that operates completely separately from the big 3-band one.

The main EQ has five fully parametric bands with switchable shelf characteristics for the outer two, and a clear graphic display of the resulting curve. The shelf filters have a variable peak characteristic near the turnover frequency that increases the flexibility even more. Sonically it is smooth and easy, with enough adjustment to make it brutal if necessary.

Finally the valve section offers colours all of its own, with individually variable drive levels for the three frequency bands. To have the thickness, warmth, brightness and even distortion of the virtual valves under such detailed control allows a wealth of interesting possibilities, and the simulation of the real effect is very convincing and great fun.

A small yet effective selection of factory presets shows what the 2476 can do. Drawmer cheerfully admits that some of them are a little over the top, preferring to impress with power rather than subtlety, safe in the knowledge that any excesses can be reduced to taste very easily. There are several here—Beef it Up is a good one—that will throw you back in your chair if you're not ready for them, **>**

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SONIC SOLUTIONS

> With all this on offer the 2476 constitutes a toolbox for mastering, with obvious elements combined with some surprising ones. Many of the parts have been seen before so the overall concept is not new, but the presence of the valve stage and the general flavour make the 2476 distinctive

< with a good selection of EQ shapes, compression characters and valve effects. Some are more subtle, and there's a neutral patch for starting your own from scratch. The various processing blocks can be loaded individually from the presets if required, allowing a mix-and-match approach.

There is something here for any application, from clean tweaking to in-your-face maximum loudness. In case you can't hear it, there is plenty of metering all over the front to show the activity of the various dynamic stages, and some patches make it light up like a Christmas tree. I noted that with some of the louder settings it was possible to push it hard enough to produce a very occasional digital over on the output; as it was a downstream meter rather than my ears that alerted me to the fact, you can make your choice as to whether this would bother you enough to treat it with a bit more caution.

Several means of keeping track are provided, including a compare function and the facility to solo or bypass individual blocks as well as an overall bypass. Currently the output block gets saved with the word length and dither settings, although this may change. I found it awkward to have to change the word length every time I selected a new factory patch, although the workaround is to use the block load function and simply not load the output block—simple.

Software upgrades can be installed by the user via a MIDI adaptor box, having been downloaded from Drawmer's net-site or acquired on diskette. The MIDI ports allow the usual comprehensive control, and contribute to a well-filled rear panel with high quality SPDIF phonos and BNCs for word clock in and out. Conversion both ends deals in the increasingly standard 24-bit 96kHz format as well as almost anything else you can think of, with a choice of output word lengths and dither types.

With all this on offer the 2476 constitutes a comprehensive toolbox for adventurous mastering, with all the obvious elements combined with some surprising ones. Many of the component parts have been seen before elsewhere so the overall concept is not new, but the presence of the valve stage and the general feel and flavour make the 2476 quite distinctive. I found it lent itself particularly to big fat sounds, which just seemed to fall effortlessly out of it, but it is capable of subtlety and gentle correction and enhancement as well, all with superb quality. I had a great deal of enjoyment from it and would warmly recommend trying it. ■

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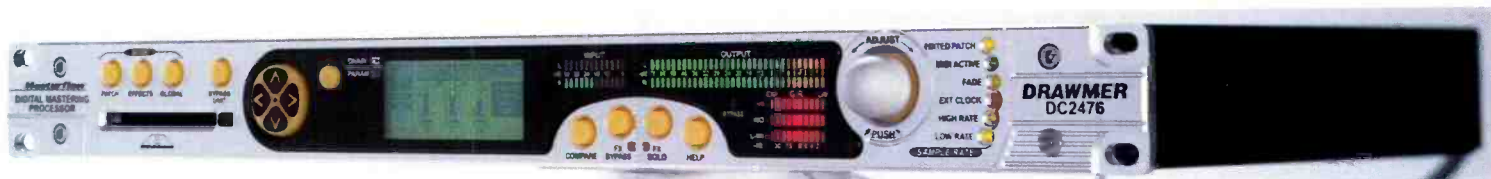
Merging Technologies Mykerinos

Following the continued interdependence of audio software and computer hardware, Merging Technologies' has adopted the Mykerinos PCI board. **Terry Nelson** plugs in and checks out

THE *STUDIO SOUND* READER will not have failed to notice the continuing emergence of Merging Technologies in terms of products and software. The latter is exemplified by the Pyramix Virtual Studio, while the former includes audio boards and more recently, a line of A–D/D–A convertors.

In terms of audio cards, the Kefren first provided 8-channel I-O and was the basis of the Pyramix Virtual Studio. The next hardware development for Pyramix was the Kheops 16-track system last year, but it was already evident that things were not going to stop there. The latest offering from Merging Technologies is the Mykerinos PCI audio board, which finds applications for both the forthcoming Pyramix v3 and OEM functions.

Merging Technologies claim that Mykerinos is the most powerful single-processor PCI audio board available, and a closer look at the specification seems to bear this out. The board employs a single VLIW (Very Long Instruction Word) processor that replaces a complete DSP Farm—or collection of processor chips—and provides what is to all intents and purposes, real-time processing by being connected directly to the PCI bus. The chip features a sustained 240MFlops of processing power with a peak of 480MFlops, thus making it extremely efficient. The processor also provides a very high memory bandwidth of 480Mbytes/s which makes it ideal for space-hungry algorithms such as high-quality reverberation, flexible surround panning and room simulation.

In terms of track density, the new board features enough I-O capability to deal with most demanding situations. In terms of numbers, the IT can handle over 128 tracks at up to 32 bits for playback and recording, or, in a more real-world situation, 64 tracks of simultaneous record and playback. It is interesting to note that the system now becomes governed by the capacity of the PC motherboard PCI chip-set rather than by the audio board. For example, two hard disks in parallel are recommended for 64 track I-O from a single Mykerinos card.

The I-O capacity also opens up a new world in virtual mixers, and a single card can be used as a real-time crosspoint matrix (or switcher) in up to a 64 x 64 configuration. Additionally, each crosspoint—or matrix node—features control of level, phase and delay. This paves the way for large matrix systems

having control of the three most important parameters, either for distribution or sound-reinforcement applications.

The logical progression from a matrix is to a consoles, and Mykerinos enables large virtual mixing consoles to be configured. A single board allows a 24 x 16 matrix mixer featuring individual gain and delay at the 384 matrix nodes—or bus crosspoints. In such a configuration, the mixer's low DSP load leaves plenty of space for audio effects processing.

Flexibility is the watchword of this

PCI board and the range of effects available includes 4-band parametric EQ for up to 64 channels, 10-band graphic EQ for up to 32 channels, dynamics processor for up to 16 channels and real-time reverb for up to 8 channels. The last effect is likely to be of particular significance now that the audio world has rediscovered surround-sound, as this is now a requirement for multichannel reverb units capable of creating convincing ambiences that are literally all around you. It will be interesting to see who will be first to take up the challenge.

If with all the power available here you're tempted to think that this is a case of 'never mind the quality, feel the width', I have to tell you that quality is also of the highest order. For those of

you into numbers, Mykerinos features 32-bit floating-point processing which provides extremely accurate internal calculations and a maximum headroom of +700dB on the high end and a total precision of -700dB in the low end. Quite where you would use all of this is another matter, but it is there.

In the mundane world of I-O, this provides 144dB of dynamic range for 24-bit I-O and up to 192kHz sampling is available for applications such as high resolution DVD audio premastering.

Of the system's low processing time.

Merging Technologies claim that Mykerinos is the most powerful single-processor PCI audio board available, and a closer look at the specification seems to bear this out

Merging Technologies is quick to point out the advantages of what it calls 'Low Latency, Live In to Live Out'. The actual processing time from input to output is under 4ms at 48kHz and this makes the board perfectly usable for real-time audio events such as live recording, dubbing, mixing and even sound reinforcement. Another advantage is that the processing time is a fixed delay that is independent of any real-time effects that may be inserted on an input or output channel. This facility is maximised further by a very fast response time between mouse, keyboard or other external controller commands and the Virtual Studio response on transport commands and parameter changes (such as gain).

Delving more deeply in Mykerinos, >

< a distinct advantage can be found in the form of very flexible I-O formats —and all in a single slot module. The board uses a modular daughter card concept that allows it to be configured as required in the space of a single slot, whether it be 16 channels I-O of 2+2 ADAT optical, 8 channels I-O on mixed analogue + AES-EBU, 56 channels I-O of MADI or combined audio and video. A later release will be 64 channels of IEEE 1394 (or Firewire). The board can also be used without a daughter I-O in order to provide expanded DSP power.

Mykerinos features an HDTDM (High-Definition Time Domain Multiplex) bus and this allow up to 8 boards to be connected either in parallel or as a daisy chain. This provides up to 256 bidirectional audio channels (or 128 channels in each direction) and points the way to large systems. The high definition floating point 32-bit bus means that internal clipping or noise and distortion build-up due to truncation is all but eliminated.

Audio today is never far from video and Mykerinos features flexible AV sync possibilities that Merging Technologies claim makes it 'HDTV ready'. The onboard sync connector contains built-in video sync input-output, VITC reader-generator, LTC reader-generator and time code insertion in the video window. Audio monitoring is provided via a two channel 24-bit/96kHz output. All

those working regularly—and not so regularly—with digital audio will know the importance of sync and lock. The Mykerinos board features an ultra-low jitter clock for sampling rates from 32kHz to 192kHz that can be run in internal (master) mode or locked (slave) to external video (auto detection of PAL and NTSC frame rates) or external wordclock. Lock to external time code is sample accurate.

Merging Technologies tend to make its products standards compliant and the new board is fully compliant with PCI v2.1, and will support both 3.3V and 5V operation. Big and little Endian are also supported. Over on the software side, Mykerinos is compliant with Windows 2000 (when it sees the light of day), Windows NT4 and Windows 95/98.

As it stands, the Mykerinos motherboard is delivered with 4Mb 120MHz SDRAM and free Pyramix editing software (a simplified version of the full Pyramix system). I-O daughter cards are supplied according to requirements, and, as we have noted previously, the board still occupies only one PCI slot when a daughter card is fitted.

Though the hardware basis of the forthcoming Pyramix v3 Virtual Studio, Merging Technologies has its eye firmly

on the OEM possibilities for the board, calling it the VS3 OEM Audio Engine (interested parties, please take note).

Other potential OEM solutions offered by the board include: Multiple 'Virtual Studios' run simultaneously and independently from one board, with each studio configured with it's own I-O routeing and mixer setup. The only limits are those of the total amount of I-O available and total hard disk tracks. This means that operations such as independent playback-record-mixing-CD mastering in different studios can be accomplished at the same time with one card—with the obvious benefits in time-saving. Film-type operation is also possible where up to eight 8-track digital recorder-dubbers may run together in an independent manner. And all of this with real-time processing.

The Mykerinos PCI board is undoubtedly a big step forward in digital processing cards and the fact that it is also deliberately aimed at OEM use means that the user should not be locked into one person's system, but rather on a common platform. It will be interesting to see what synergy will come out of this with other manufacturers (a combined AV non-linear workstation, for instance). In the meantime, do yourself a favour and check it out for yourself. ■

Quantec Yardstick 2402

Matching old-fashioned quality with unapologetically modern design, Quantec has delivered its Yardstick reverb processor. **Dave Foister** blesses the marriage of Art and Science

ONCE UPON A TIME the Quantec Room Simulator was one of the elite. At a time when good digital reverb cost serious money, the QRS was one of the best, in very select company and very highly regarded. Unlike AMS and Lexicon, with whom Quantec shared such elevated status, the company put all its eggs in the QRS basket with the result that it eventually got sidelined, and, although it still gets used by those who know what it can do, many engineers new to the business have never heard of what was once a highly sought-after processor. I remember looking at it for *Studio Sound* when the CARL (anagram of LARC—geddit?) remote control was introduced, and, although it had a very idiosyncratic way of doing things it was obviously offering some of the finest reverb to be had.

Now Quantec is back, hoping to prove that not all classics have valves. The outlandish new Yardstick 2402 sets out to recreate the original eighties algorithms using nineties DSP, maintaining that the QRS design was so advanced that off-the-shelf chips have only just acquired the power to recreate them. Originally the QRS used dedicated hardware; now the Motorola 56009-80MHz processor is pressed into service for the job, although even now some streamlining of the original algorithm has been necessary to make it possible. At the same time, the price has followed the trend normally expected of digital technology—it has dropped, making the Yardstick far more affordable than it was originally.

Most recreations of favourite equipment go to great lengths to physically

is hard to believe there can be any electronics in the slim box at all, never mind a full-blown Quantec Room Simulator —complete with mains power supply on board (no wall-warts for Quantec). Sadly this aspect is hidden as soon as the Yardstick is in a rack, but the appearance remains most striking.

Compared with some modern processors the Yardstick's display arrangements at first appear rather rudimentary, although it soon becomes clear that they are perfectly adequate for the job and quite elegantly arranged. There is a small LCD screen for navigating and editing the presets, and just enough LEDs to keep you informed about what is going on. There are 3-LED meters for input and output, and a row of LEDs showing the status of the incoming digital signal. That is all there is: the Yardstick has no analogue inputs or outputs, and, in fact, only has AES-EBU connectors. This means the back panel is as thinly populated as the front. Alongside the XLRs are three MIDI connectors and a 9-pin for data transfer to a PC—and the IEC mains input that leaves you wondering what on earth it can be feeding.

It would hardly be possible to have fewer controls than the Yardstick possesses. There is a big knob, strangely called the Pulsar, made of blue aluminium to match the panel and protruding conically, and a pair of bright orange push buttons—and that is it. The surprise is that the Yardstick still manages to be one of the friendliest and fastest processors I have used. Everything is to hand without the need for special tools or utilities menus, and nothing is more than a few twists of the

Quantec's philosophy on this is clearly explained in the manual. The most obvious difference between the QRS and other reverbs is that the QRS has only one First Reflection parameter where others use a multi-tap delay to generate a whole sequence of reflections. Quantec points out that a real room is a single signal processor, and generates early reflections, and the smoother decay by the same means; the feeling is that it is wrong to separate the two aspects. We all know that the early reflections play an important part in defining the character of a reverb, but Quantec maintains that this initial part should not be extracted from the rest of the effect. From this it follows that the First Reflection is not part of the reverberation itself but an extra

'Where's the rest of it?' is the comment passed by those who have seen it, as it is hard to believe there can be any electronics in the slim box at all, never mind a full-blown Quantec Room Simulator complete with mains power supply on board

replicate them, hunting down original bakelite knobs and toggle switches, matching colours, and copying every aspect of the layout so that only the manufacturer's name gives away the fact that it is a replica. The Yardstick makes no attempt whatsoever to do this. Instead it goes to the opposite extreme, presenting itself as perhaps the most bizarre design I have ever seen. The 1U-high panel is a vivid electric blue with the absolute minimum of controls, but the most striking aspect is its extraordinary leanness behind the panel. 'Where's the rest of it?' is the comment passed by those who have seen it, as it

Pulsar away. In part, this ease of use can be attributed to the basic concept of the QRS algorithms themselves, where the user has fewer parameters to play with, and the processor itself does most of the work. I find it refreshing to be able to get the sound the way I want it to be so easily, without having to wade through screen after screen of tiny adjustments. I would be interested to know how often they get used anyway; on a box with 300 presets, who really does much more than find one that's close and tweak the obvious things like reverb time, brightness and predelay? Answers on a postcard.

effect to add a distinct echo. It has its own level and delay parameters, and is only used in two of the factory presets—Backyard and Railroad Station —both of which show its specific use to create a slapback effect. This is an example of how reading the manual helps. If you do not realise that the real early reflections are incorporated into the main reverb algorithm it can be puzzling to find what looks like the early reflections switched off in all the Hall, Church and Plate presets, yet turning the early reflections on can mess up the naturalness of these effects completely.   >
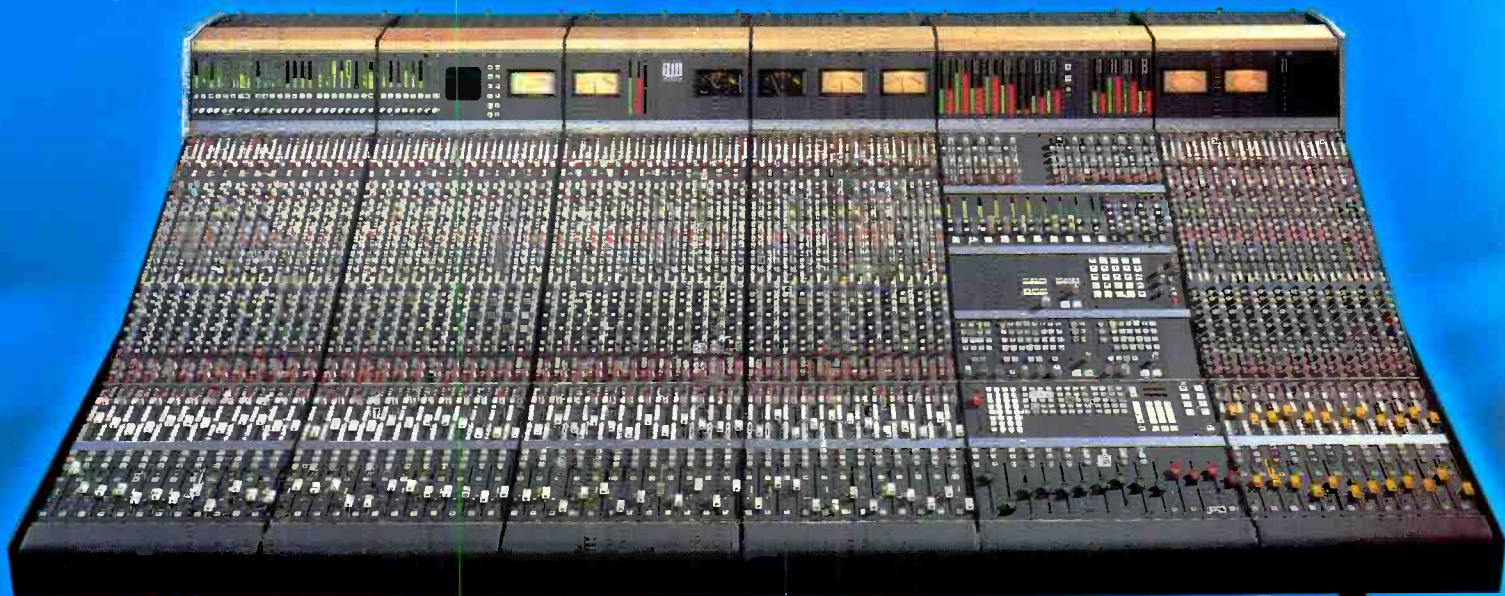
Q2

sometimes analogue is
the right solution

1998 saw Calrec introduce two new digital consoles, emphasising
Calrec's commitment to a digital future. However Calrec recognise that
for many applications an analogue console is the right solution.
That's why Calrec still supply a full range of high performance
analogue production and on-air consoles to broadcasters world-wide.

THE CALREC ANALOGUE RANGE
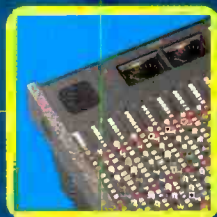
C2   MINIMIXER   S2

Believe everything you hear

CALREC

> Given a density parameter variable in increments of 1%, who can hear the difference between successive settings? Much better to have fewer options with a real difference between them—as long as the adjustment does not become too coarse

< This integration of the whole reverb envelope into one effect makes editing very simple indeed. Each preset type has only a handful of parameters for the reverb itself: main reverb time (up to 100s or Freeze), reverb delay, density, bandwidth, and high and low times as a factor of the main time. These last elements are also important to Quantec, following a similar line to the early reflections argument in that the room is seen as a single processor that deals with the entire spectrum continuously, not divided into bands. That is not to say that all frequencies are treated equally; the QRS algorithm treats the boundaries of a room and the boundaries of the diffusors inside the room as resonators at low frequencies or absorbers at high frequencies, with a smooth transition between the response characteristics across the spectrum.

Many of the parameters are adjustable in surprisingly big increments, which, again, I find quite logical. Given a density parameter variable in increments of 1%, who can hear the difference between successive settings? Much better to have fewer options with a real difference between them —as long as the adjustment does not become too coarse. An example is the global Correlation setting, controlling the inter-relation of the reverb on the left and right channels. 0% is intended for headphone listening, creating maximum out-of-head localisation, while 50% is for conventional loudspeaker stereo. 100% is mono.

Navigating this system is as easy as it could be. The Pulsar scrolls through the presets (there are only 17!) either autoloading them or waiting for the DOWN button to be pressed to select one. The chosen preset can be edited by hitting the DOWN button to get at the parameters, scrolling through them with the knob, hitting the DOWN button again to edit the one you want and hitting the UP button when you have adjusted it. The system setup stuff is accessed by scrolling down below preset 1 and editing in the same way, with options like hard-soft bypass, noise shaping and LCD meter decay. This approach is fast and simple, the sort of system that makes you want to play with the presets rather than giving up in exasperation when the reverb time is buried six menus deep. There are user-memories available too.

The sounds it produces are stunning. If you want fancy multiple effects and choruses and all the rest, forget it; if you want real reverb, smooth, natural and convincing, you won't find much better than this. It is good to see the Taj Mahal preset carried over from the original, along with St Peter's, and the other halls are all persuasive. Add the specials like the swimming bath, the theatre and the muffled cinema and you see the potential for simulating real spaces with complete conviction, with an astonishing ease of control. The Quantec Room Simulator was always special; now its Yardstick reincarnation stands to become equally highly regarded, and is certain to get noticed in every respect. ■

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Otari high speed archiving

Otari's new archiving system tackles the problem of transferring analogue archives to digital formats with a means of reducing the vast amounts of time required. Various high speed reel to reel and/or cassette playback transports are available, transferring up to four audio channels simultaneously to an ADC unit that incorporates level and machine operation controls. The resulting digital audio is temporarily stored on a removable medium (hard disk, MO or DVD-RAM are available) for later downloading to external mass storage systems, or alternatively it can be transferred directly via SCSI. The system requires no specialist operating skills, and as with other digital archiving schemes it expects that defects in the sources will be dealt with later in the digital domain when the material is required. Also new from Otari is a CD to cassette duplication system that runs at 8x or 16x speed and can either incorporate two CD drives for simultaneous copying of the two cassette sides or automatically break the copying across the sides from a single source CD.

Otari, Europe. Tel: +49 2159 508 613.

EV radios and specials

Electro-Voice's new offerings include three wireless systems and two specialised wired microphones. The ENG 618 is an integrated boom-pole/shotgun system with a back-electret microphone permanently attached to a K-Tek telescopic graphite pole. The base of the pole incorporates batteries for power, LF roll-off, output connector and even a headphone amplifier. At the other end of the size scale is the RE90L ultra-miniature lavalier condenser, again using a back-electret capsule to give smooth omni response up to 18kHz. Top of the wireless systems is the MS3000 UHF diversity system, with the MR3000 receiver handling a choice of hand-held and body pack transmitters and a wide range of microphone capsules. A similar selection is available for the more affordable R200 system, also UHF diversity but operating in a narrower range. Lastly the R100 system is a new VHF diversity package with similar options. All use E-V's Secure-Phase circuitry that uses the signal from both antennae at all times rather than simply switching between them.

EVI/Telex, Germany. Tel: +49 9421 706 317. >

Alesis MasterLink ML9600

The push to bring 24-96 to the masses has started with an unusual concoction of hard-disk recorder, editor and CD burner. **Zenon Schoepe** previews the Alesis solution

ALESIS HAS FINALLY MOVED into hard-disk recording and editing, but with a typically unusual twist. The MasterLink ML-9600 is described as a mixdown and mastering system that combines hard-disk recording and editing, digital signal processing and CD creation in a single stand-alone unit, but has the distinction of being able to burn stereo 24-bit, 96kHz audio on to standard CD-Rs, in a CD24 proprietary format, in addition to traditional Red Book format. The Alesis CD24 employs the ISO 9660 disc format and AIFF audio files on standard CD blanks that can be played back on 9600 machines and accessed by and archived to by DAWs.

Alesis clearly sees the machine as a replacement for DAT and as a cost-effective way into stereo high resolution recording, storage and archival. It is expected to ship in the third quarter of this year for $1699 (US).

MasterLink has a 3.2Gb internal hard drive and a 4x CD-R drive. The system includes an internal sample-rate convertor allowing high-resolution files on the hard disk to be used to create Red Book CDs. Editing is accomplished using front-panel controls, that allows audition use of edit start and end points.

Editing features allow the user to reorder tracks, adjust gain, build playlists, trim heads and tails of programmes, set start times, preserve best takes and delete tracks before committing the material to CD. Join and Split permit material from multiple mixes to be resequenced.

Front-panel controls include standard CD player transport with scan and skip, track controls (new track, delete track, track information), cursor controls, CD resolution, input source (analogue, digital, optical), word length, sample rate and Create CD. The 2U-high rackmount has a wireless remote and front-panel headphones circuit.

Onboard DSP includes equalisation, compression, normalisation and peak limiting. Analogue and digital I-Os are provided on XLRs and phonos. Convertors are 24-bit and at maximum resolution the 3.2Gb drive stores 95 minutes of audio; although the machine can be configured for combinations of 44.1kHz, 48kHz, 88.2kHz. and 96kHz sample rates and 16-bit, 20-bit or 24-bit word lengths. Firmware is updated through the internal CD drive. Points to note include the fact that with regard to DVD capability Alesis says the MasterLink is 'currently' a 2-channel format and that the machine's architecture is such that proprietary plug-ins could be developed for it, not directX or VST compatible. The machine does not have word clock connection, but locks to incoming AES-EBU. A CD24 format disc can hold around 24 minutes of 24-96 stereo.

The move is an interesting one as it does not replace the company's 20-bit ADAT Type II multitrack format, but rather serves as an accompanying system for mastering to. Most significantly Alesis has jumped aboard the 24-96 bandwagon at a price point that will make this technology widely accessible and has simultaneously answered requirements for CD-R burning and a logical DAT replacement. Quite clever really, and you could be forgiven for suspecting that more along the same sort of lines is likely. ■

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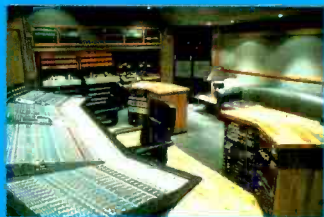
Sony MDS-E11

Taking a consumer format and adapting it for professional use is not a new endeavour. **Zenon Schoepe** assesses Sony's latest MD

MINIDISC has been embraced by broadcast with the sort of gusto that could not have been imagined when this logical replacement for the humble compact cassette was foisted on to the suspecting consumer. It now enjoys a remarkable level of acceptance in sound acquisition and in radio playback duties that has offset MD's use of compression against the sheer convenience and smartness of the format. What seems to have mattered most is that you can record on it repeatedly, edit of a fashion on the machine, access is fast and running costs are low.

Of course the big achievement of Sony's MDS-E11 MD recorder is that the device only occupies a unit of rack space. If its been designed well to retain all the functionality then this has to be an advantage and aside from the loading of discs you can operate all the top layer playback and recording functions from a supplied infrared remote control. However, you will still need to have the machine close enough to be able to see the front panel display clearly.

In terms of connections, most of the bases are covered with phono and balanced XLR I-Os

The arrangement of the multiple key options normally associated with MD can be a issue. Sony has been clever in giving the box a simple look but with deep levels of control using a small LCD

with paralleled simultaneous outputs and switchable -10/+4 on the latter. Digital I-O is provided via coax (complete with automatic SRC) and the cluster of connectors on the rear panel labelled 'remote' is of particular interest.

A mini jack socket permits wired control from a suitable remote (which defeats the infrared unit) while three additional stereo mini jack sockets permit multiple MDS-E11 to be daisy-chained together in two configurations for continuous sequential play and recording duties. Furthermore, 9-pin D-sub connectors allow parallel and RS-232C operation for complete integration into a radio studios scheme. Despite this and the XLR I-O pro stance the MDS-E11 supports SCMS.

With such a small front panel the arrangement of the sort of multifarious key options normally associated with MD can be a issue. In fact Sony has been clever here and given

the box a rather rudimentary appearance with deeper levels of control relying on a small LCD. There are a clutch of transport controls that take in forward and rewind cue, pause, stop play and record in perfectly acceptable proportions. There is also a ganged record level pot working with an input selector and a headphones circuit with level.

And that's nearly it apart from a dial marked AMS (Automatic Music Sensor in very hi-fi parlance) which can be turned to access tracks directly or, more importantly as track access is far more convenient and varied from the remote, pressed to select functions and parameters in editing.

Two small buttons to either side of this dial marked NO/EDIT and YES, as you can imagine, get you in, around and out of the menus. As we are into the software stage it is worth outlining what this unit can do. In terms of editing you can erase a single track, all tracks and part of a track (by employing divide), divide, combine, move and, a new one for me, name the MD and tracks. All these routines are entered on the NO/EDIT button and involve spooling through menus with the AMS dial, and progressing through to the point where you hit YES to complete and exit, or NO to abort. Quite straightforward, and thankfully there is a consistency to the logic of how things happen.

Crucial to all editing activity is the entry of track numbers which can be marked manually with a RECORD button press while in record, automatically with digital sources containing existing track numbers; automatically when levels drop below a threshold and then exceeds it again; or after recording has been completed by using Divide.

A particularly useful feature is the grandly titled Time Machine Recording mode. The machine captures the last 6s of audio when in record pause mode and these 6s can be transferred to disc along with the commencement of recording with a push of the AMS button. The machine will also auto insert 3s gaps between tracks, will auto pause after track playback and, usefully, when it sees 30s of silence from a digital input in recording will intelligently replace it with 3s of silence and park itself in record pause. The first time it does this it surprises you, but after that you begin to admire it.

The remote also permits control of a Sony CD player and instigates Synchro-Recording and it is worth mentioning that Sony has a companion 1U-high rackmount CD player planned for arrival soon. You can also run the MDS-E11 with a timer for recording and playback, and you'd expect track repeat, >

NEW TECHNOLOGIES

Tascam Dubber

Tascam's MMR-8 and MMP-16 dubber range was shown at NAB with new developments designed to broaden its appeal and its integration with third party systems. On show was Rorke Data's Galaxy 5 RAID array storage, which the Tascam units were sharing with a Pro Tools system, Tascam connected via SCSI and Pro Tools remotely via Fibre Channel. Tascam intends that this kind of centralised shared storage across platforms will increase work flow by removing the need to physically move disk drives. Also new is software Version 3.2, adding support for Sonic Solutions format audio. This allows the Tascams to play 16 and 24-bit audio projects from Sonic Studio and convert them to WaveFrame, Pro Tools and OMF formats.

Tascam, US. Tel: +1 213 726 030.

Calrec Lip Stick

Calrec has launched a novel system for dealing with the problem of mistiming between audio and video signals through complex signal paths. Lip Stick is claimed to remove the need for subjective operator estimation of the mismatch between picture and sound, instead using an objective

method of delay measurement. Details are not divulged, but it uses a transmitter and receiver pair to measure the delays incurred through any video path including MPEG links, and even when different paths are used for audio and video. Accuracy is quoted as ±2ms, and the whole system is dual-standard NTSC/PAL.

Calrec, UK. Tel: +44 1422 842159.

ATC for the Millennium

ATC's new SCM70 SL is a new design heralded as the most radical speaker in the company's history as its outward design breaks with tradition in both shape and materials. The cabinet is made from a combination of aluminium, MDF and Corian and incorporates gentle curves where straight lines might be expected, although ATC stresses that the shape is driven by sonic demands rather than aesthetics. As is usual for ATC designs it is self-powered, driven by a new triamplifier with 2nd order filters using discrete circuitry. Sophisticated features include isolated secondary power supplies, programme sensing, auto power off and infra red control. All the drivers are new, and comprise a 234mm Super Linear bass driver, a Mk2 midrange dome with improved waveguide, and a 25mm tweeter reaching beyond 20kHz.

ATC, UK. Tel: +44 1285 760561.

program play and shuffle play and they are all here along with repeat modes.

I thoroughly enjoyed using this machine, it has a robustness that in contrast to the accentuation of compact size and seeming fragility of the media and machines that is so apparent when using portable models. I want one in my car.

The remote control possibilities and options are admirable and the front panel's simplicity belies its power. However, ultimate operational convenience is grasped by a curious combination of the means in accessing the functionality—front panel, supplied remote control and additional remote. Many will prefer the direct track number key access from the infrared remote while the convenience of a custom parallel configuration will be best for an install, but you will always need to edit from the front panel which is probably not a bad thing. There are enough

control permutations to qualify the MDS-E11 as very flexible.

It is a splendid experience to encounter an MD machine like this that is able to deliver the full features of the format. If anything this extended functionality explains why the format has been so well received by pro users as there is so much you can do with the data as it goes in and after it is on the disc. I could almost admit it is wasted on a consumer. Anyone who has used consumer portables as anything more than a convenient acquisition medium will know the pain of accessing the cool tricks of MD. The MDS-E11 makes the whole business delightfully simple by comparison. In a way it is a shame that most people's first encounter with MD has not been through a machine such as this. If it had been so Pro perception of the format would, I believe, have been even favourably advanced by now. ■

Roland VM31000 Pro

From the digital studio to the digital mixer, Roland is playing a clever hand in the digital studio card school. **Rob James** sits in and cleans up

ROLAND'S FIRST offerings to the digital mixer market are here. The VM-3100 and its 'Pro' version (under review) will be familiar to anyone with experience of Roland's V-series workstations —although there are differences that can be endearing or infuriating depending on your viewpoint.

The compact package offers 20 channels each with 3-band EQ, two internal effects processors, a pair of compressors and a number of connection options. There are some limitations and compromises, but these are not exacting. Inputs 1 & 2 have a choice of XLR or balanced ¼-inch jacks with phantom powering. Inputs 3-8 are all on balanced ¼-inch jacks with an extra guitar jack switchable with Input 4. Analogue stereo sources are best connected to Inputs 9–12 via phonos. SPDIF in coax and optical flavours take care of the single stereo digital input. This is normally patched to Channels 11 & 12, but this can be altered. Analogue outputs are Master LR, Aux 1&2 and Monitor (headphones) on

jacks with LR Bus outs on phonos. Both the coax and optical SPDIF connectors output the same signal as the analogue master. In the absence of a separate wordclock input, sync is taken from the digital input. An external 2-track digital recorder is obliged to sync to the VM3100 output. If the digital output of the recorder is connected to the digital input of the mixer without an external sample-rate convertor the system ends up chasing its own tail in silence. The Pro version is also equipped with an RDMB II multichannel digital I-O connector that accounts for Inputs 13-20. When used with the optional extra DIF-AT interface box this enables connection to ADAT or Tascam 8-track machines.

I used the VM3100 with a stereo digital source and added guitar and mic, and had a lot of fun once I began to get a grip on some of the arcane logic. Roland is becoming infamous for its manuals and this one is no exception. It appears thorough until you actually try to do something. Then you find yourself jumping backwards and forwards like a jack rabbit. The short-throw channel faders are layered using Audio Channel and MIDI Channel keys. Thus the faders can control 16 channels of external MIDI control change parameters. In the absence of motor faders the layering could be confusing but since the display shows the fader positions it could be worse. The buttons feel positive and most have internal lights. The small display is clear and makes much use of soft keys. Scene memories are used for onboard automation: 32 memories in eight banks of four store a vast array of parameters. Transitions can be instant, do nothing until the controls are nulled, or, for level changes only, crossfaded over a predetermined time.

Roland has also employed its 'EZ' routing trick here to some effect with 16 preset and 16 user-memories. Monitor select allows comparison between desk output and digital input. Compressors and EQ also get 16 preset and 16 user-memories.

The 3-band EQ uses dedicated knobs for boost and cut. Mid has variable Q. All the rotary controls apart from the parameter wheel look like the usual analogue controls in that they have end stops. The control does not actually do anything until the knob arrives at the previously set value then it behaves normally. All the EQ settings may be controlled from the parameter wheel, if you prefer. The effects are up to Roland's usual standard—100 preset and 100 user-patches are provided and they will be used with the number of possibilities on offer. The VM3100 Pro also has Roland's speaker modelling algorithm available on FX 2. This was first intended for use with Roland's own DS-90 digital powered speakers. I was amused by the rather coy descriptions of the some of the speakers modelled—'Wh.CO1NE—sealed enclosure 2-way speakers known for their white woofers and widely used in recording studios'.

The VM 3100 is well equipped as a MIDI slave and also for use with a sequencer, but sadly it does not output MIDI timing messages.

Roland suggests a number of uses for the VM3100 Pro. I think it is best suited to demos or maybe as a keyboard submixer, I also see it becoming popular as a 'sidecar' simply for its effects. I do have a couple of reservations. Internal processing is 24-bit so it is easy to run out of headroom when using steep EQ. However the more serious concern is the steepness of the learning curve. Usually once I have found a way to do something the knowledge sticks. In many instances this was not the case with the VM3100. I feel Roland could find more intuitive ways of controlling so much power from so small an interface. For all that I cannot help liking the thing. The effects are a pleasant change from the rest of the herd, the EQ is typically digital, it feels well built and I look forward

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MTR double splits

MTR, well-known for handy problem-solving boxes, has introduced a doubled-up version of its established PS-4 passive splitter. The PS-8 has eight outputs on stereo jacks, mounted on in internal PCB. Each output has overload protection and a link on the board can be moved to convert the unit to mono operation. The obvious application is feeding multiple headphones from a single source.
MTR, UK. Tel: +44 1923 234050.

SCORE v4

San Andreas Press SCORE, the music notation program for PCs, is now in version 4 which provides a completely new menu system. The intention is to bypass the necessity to resort to the manual or memorise commands to offer far easier access to the power of the program's functionality. SCORE is said to be the standard for music engraving software, capable of notating anything from rock to the toughest avant garde classical.
New Notations, UK. Tel: +44 181 875 0429.

QSC ups the power

The latest addition to QSC's PLX Series of power amplifiers is the most powerful yet, capable of delivering 700 Watts per channel at 8Ω. The PLX3402 is still only 2U and 9.5kg,

and offers very low hum, noise and distortion, proportional-response clip limiters and user selectable low-frequency filters.
QSC, US. Tel: +1 714 754 6175.

CobraNet goes 24-bit

Peak Audio's CobraNet Ethernet-based audio networking technology now has full support for 16, 20 and 24-bit operation. This applies to all CobraNet products now on the market, which are all capable of transmitting audio up to the resolution of the convertors used. In addition, all CobraNet compatible products can receive audio up to 24 bits regardless of the convertors used, giving full compatibility across all the products from the various manufacturers licensing the technology. Currently these include QSC, Peavey, Crown, Level Control Systems, Rane, EAW and Crest, and all their CobraNet products are software upgradable, ensuring continued compatibility in the future. Developments on the way include support for Gigabit and Switched Ethernet, ATM, and SNMP management.
Peak Audio, US. Tel: +1 303 449 9337.

AKM codec

New from Asahi Kasei Microsystems is the AK4524, a high performance 2-channel 24-bit codec for use in 96kHz recording systems. The ADC uses 64x oversampling with a dynamic range of 101dB, and the A–D >

CEDAR BRX+

Latest in the line of stand-alone CEDAR processors,
the BRX+ is intended to banish buzz in all its incarnations.
Dave Foister chews on the bricks

I RECENTLY RECORDED an orchestral concert in a church and was plagued by a subtle unacceptable hum (and it wasn't the verger). Of course I checked all the connections, and while I was on my knees inspecting the only join in my multicore I noticed that the stone I was kneeling on was warm. Surprise, surprise, the one vulnerable point in the cabling was sitting on top of a very efficient hum generator otherwise known as the under-floor heating. I moved the join a foot or two away and the problem disappeared. It was a useful experience to file away for the future, but what if I had not found the source of that hum? I would have had to filter it out somehow, almost certainly at the expense of some of the musical content.

It was handy, then, to have some of the rehearsal on tape when the latest CEDAR Series-X box showed up for review. CEDAR restoration processes seem to move freely between the computer-based systems and the dedicated hardware units, and the DeBuzz algorithm is the latest to migrate to its own 1U-high habitat. Within Series-X the DeHisser is the DHX, and the DeClicker the DCX, but applying this logic to the DeBuzzer would have landed them in court so the hardware buzz remover is called the BRX+.

The original DeBuzz process for CEDAR for Windows was reviewed here a while back, and a glance at its implementation makes it unsurprising that the Series-X hardware version has a rather more elaborate front panel than the first flush of Series-X units. Those first three units shared a common board and differed only in the number of pots sticking through the front panel. The latest entries need more than this, and so the BRX has a numeric display and a rotary encoder with built-in push-button alongside the expected couple of pots and three small buttons. Even so the simplicity of operation belies the complexity of what is going on inside in the best traditions of CEDAR.

The idea here is that the BRX will, with your help, identify unwanted buzzes of various types and frequencies—mains hum, thyristor noise, camera noise and their kind—and remove them using methods altogether more subtle than the kind of notch and comb filtering approaches normally employed. As with all CEDAR's treatments, the concept of the user helping the algorithm, rather than setting up the parameters on a dumb process, is important. Here the user has to identify the general area of the buzz's fundamental, and the

process does the rest.

The rotary encoder is used to select a fundamental frequency for the buzz, and this is the most important parameter to get right. For instance, if you choose 25Hz for a 50Hz-based hum, the BRX will do unnecessary things at 75Hz, 125Hz, and so on, and likewise selecting 100Hz will miss harmonics at 150Hz and 250Hz. To help, a set of presets is available; pushing the knob and turning it steps through a selection of likely candidates for both sides of the Atlantic. The chosen value can be adjusted manually if necessary to compensate for speed errors in the recording (or even in the mains).

Manual adjustment has coarse and fine controls, toggled by pushing the encoder and indicated with LEDs. Around 50Hz the fine adjustment is in increments of 0.01Hz, and assistance with getting it right is offered by red up-down arrows that light to show that the BRX has detected something near to the currently set frequency, and which way to go to lock on to it. Once the basic value has been set, the BRX is able to track variations in frequency in a window of 1Hz at 50Hz to allow for fluctuations in speed.

The next step is also typical CEDAR—crank the other adjustments up to full so that they make nasty noises and then back them off until they're right. In this case the two adjustments are Buzz Power and Attenuation, and even hopelessly over-adjusted they do not do the kind of damage to the signal you might expect. First the Buzz Power is backed off until the Buzz is just about to reappear, then the Attenuation is set to the required amount. Getting it right is easy, and far less critical than some CEDAR processes. My heater hum disappeared completely without affecting the low end of the musical signal at all, and CEDAR's demo recording, a conversation with a generous helping of one-legged style buzz, was equally impressively cleaned up, despite having a low level of wanted signal and a fair bit of hiss. I also faked a setup with one open-circuit microphone channel mixed with a clean signal, and again the problem disappeared without affecting the clean signal at all.

The BRX+ is actually much simpler to use than its software precursor. Once again it does a specialised job with an effectiveness and integrity that perhaps nothing else can match, and as such should become standard equipment in any facility that ever has to deal with this kind of problem seriously. ∎

< has 128x oversampling and a 24-bit x8 digital filter. The D–A uses AKM's newly developed Advanced Multi Bit architecture that achieves low outband noise and high jitter tolerance by use of switched capacitor filter techniques. Applications for the device include MiniDisc, digital VTR systems, musical instruments and professional mixers.
AKM, UK. Tel: +44 1923 226 988.

Tannoy Reveals

Claiming to be the most affordable self powered professional monitor, the Reveal Active from Tannoy follows the success of the passive Reveals. The blue cabinet with 40mm

curved baffle contains a long throw 6.5-inch bass driver and a 1-inch soft dome tweeter, both magnetically shielded. Each is driven by its own 50W amplifier delivering continuous SPLs up to 114dB with a frequency response from 65Hz to 20kHz.
Tannoy, UK. Tel: +44 1236 420199.

Discmatic CD duplicator

ONYX from Discmatic is a tabletop multidrive CD duplicator with new autoloading features that improve CD handling. It supports up to four 8x drives with a capacity of 100 discs, and its autoloading mechanism picks up each disc by its rim, lifting it off the spindle and loading it into any of the recorders. All contact with recording and printing surfaces is avoided. Further streamlining is provided by a batch copying feature that allows several masters and blanks to be stacked and automatically copied in sequence.
Discmatic, US. Tel: +1 516 864 9700.

Switchcraft plugs upgrade

Switchcraft has introduced two new ranges of chassis XLR connectors. The E Series have quick release inserts to facilitate part-assembly manufacturing processes and have various improvements including reshaped ground springs, a redesigned latch, and subtle changes to the body functionality and appearance. Complementing these are the EMI/RFI shielded F Series, incorporating a ferrite material to provide excellent shielding and filtering characteristics. This is intended to reduce material and labour costs and free up board space by eliminating a number of discrete components normally associated with EMI/RFI protection.
Switchcraft, US. Tel: +1 773 792 2700.

Westlake BBSM-5

Studio Sound's 'bench test' loudspeaker reviews continue with the BBSM-5. **Keith Holland** reports

THE WESTLAKE BBSM-5 is a 2-way passive loudspeaker system comprising dual bass-mid drivers of nominally 125mm diameter arranged horizontally. High frequencies are handled by a 28mm soft-dome tweeter mounted between, and slightly above, the bass drivers. The cabinet has external dimensions of 455mm wide by 255mm high by 300mm deep, and has a bass reflex port below the tweeter. The back panel of the loudspeaker has four screw terminals which allow the option of biwiring. Westlake supplied an unusual accessory with the review loudspeaker that is claimed to improve subjective performance. The add-on consists of a block of open-cell plastic foam that fits around the front of the cabinet, presumably to control edge diffraction effects. The main review was conducted without the foam 'muff', but the on-axis frequency response was remeasured with the foam in place. No power handling specifications were supplied with the loudspeaker, so no comment can be made about maximum levels and so on.

The on-axis frequency response of the BBSM-5 is charted in Fig.1. The response is maintained between ±4dB limits between 40Hz and 20kHz, which is an acceptable result, and the average sensitivity is about 90dB SPL for 2.83V input at 1m distance. There is a distinct 'hump' in the low-frequency response at 50Hz prior to a classic 4th-order roll-off giving -10dB down at 35Hz. The harmonic distortion performance is truly remarkable for a loudspeaker of this size. The levels of harmonic distortion generated by this loudspeaker are among the lowest recorded in this series, and rival those produced by much larger systems. Second harmonic distortion is seen to peak to -27dB (4.5%) at 30Hz and -45dB (0.5%) at around 100Hz, but between these peaks the distortion is below -60dB (0.1%) at 50Hz. The reason for this drop in distortion at 50Hz is probably due to port tuning. The 'hump' in response at 50Hz suggests that the port is responsible for most of the output at this frequency. At low levels at least, the motion of the air within a port tends to be more linear than that of a drive-unit. Horizontal and vertical off-axis response are shown in Fig.5 and Fig.6, respectively. Because of the horizontal spacing of the bass-mid drivers, there is a severe interference notch in the horizontal off-axis response at 1200Hz. The vertical directivity shows no such problems however, due to the low-mounted tweeter. High-frequency off-axis behaviour is well controlled with little evidence of side-lobing. The time domain performance of the BBSM-5 can be seen in Figs 2, 3, 4 and 7, which show the acoustic centre, step response, power cepstrum and waterfall plots, respectively. The movement of the acoustic centre to just over 2m behind the loudspeaker at low frequencies is typical for a ported loudspeaker without a high-pass protection filter. The step response shows that the high frequencies radiate some 0.5ms before the mid frequencies, indicating a crossover/time alignment problem. This delay is also evident from close inspection of the acoustic centre plot at about 800Hz or so (it is not known, however, how audible these forms of transient distortion are, but it is likely that, all else being equal, a loudspeaker with accurate transient response will out-perform one with problems). The power cepstrum shows evidence of echoes at about 0.2ms and 0.5ms which may be responsible for the slightly irregular frequency response in the mid-range of frequencies. The waterfall plot shows that the hump in response at 50Hz, followed by a 4th-order roll-off gives rise to **>**
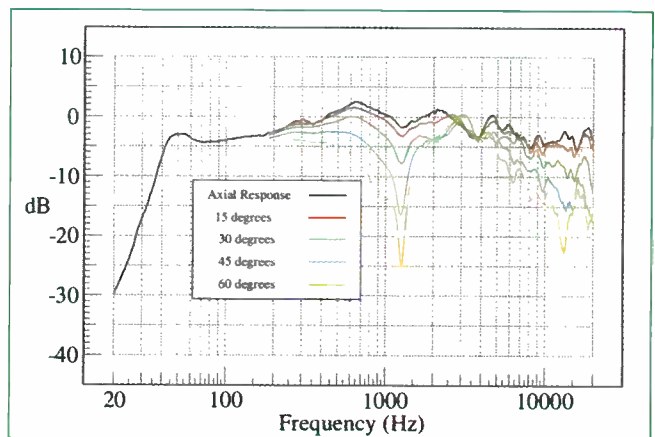

Fig.5: Horizontal directivity

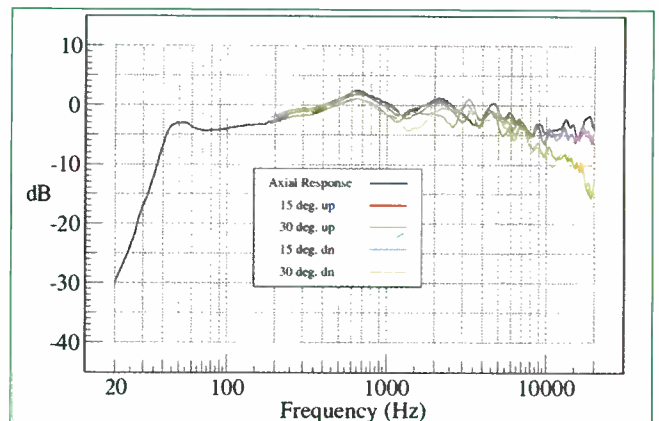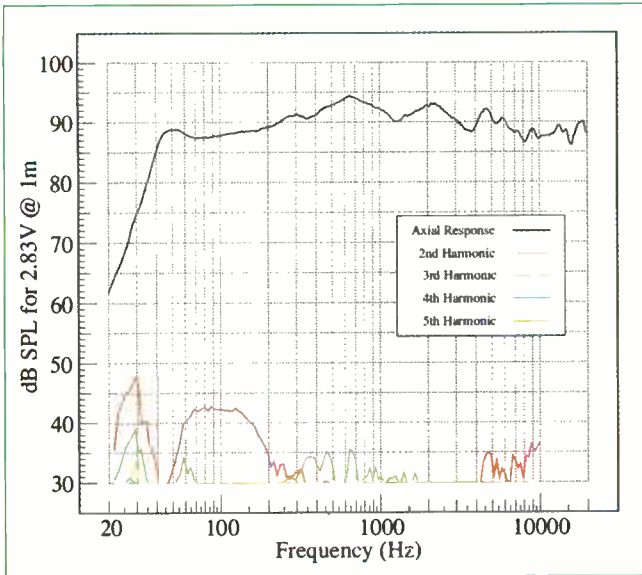**Fig.6: Vertical directivity**

Fig. 1: On-axis response and distortion

Legend:
- Axial Response
- 2nd Harmonic
- 3rd Harmonic
- 4th Harmonic
- 5th Harmonic
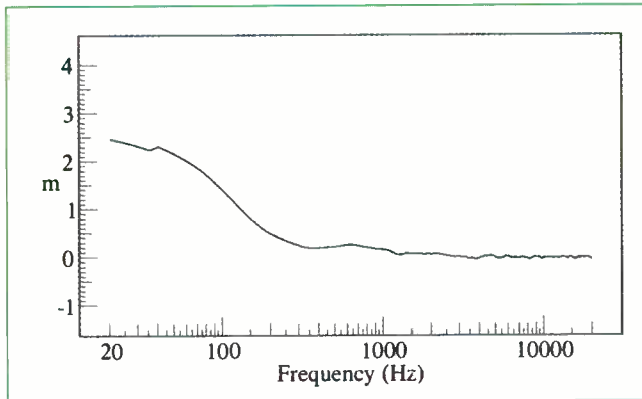
Fig 2: Acoustic centre
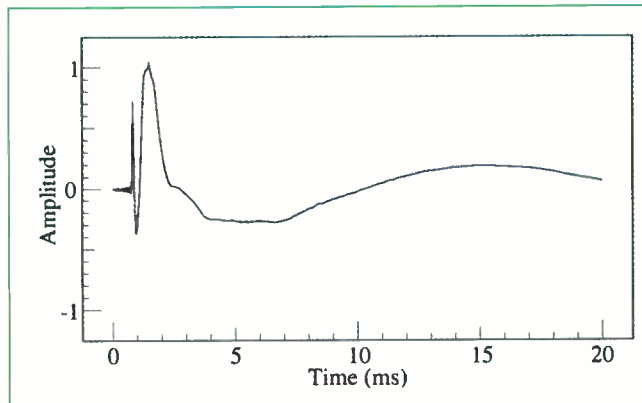
Fig.3: Step response

Fig.4: Power cepstrum

Fig.7: Waterfall chart

Fig.8: Effect of muff on on-axis response

but the time-domain performance is fairly good. Directivity performance shows the results of a different design compromise than many other loudspeakers in terms of drive-unit layout. With non-concentric designs, some compromise in directivity is almost inevitable. Finally, Fig.8 charts the effect of the foam muff accessory on the on-axis frequency response. The muff is seen to partially fill-in the drop in response below 500Hz, which demonstrates that the rising response up to 500Hz and the peak at 650Hz is due to classic rectangular cabinet edge diffraction effects; the muff reduces the effect of the edge by replacing it with a 'soft' baffle. If this loudspeaker were flush wall-mounted, the edge effect would disappear. ■

< ringing at 50Hz. However the mid-frequency and high-frequency decay is very well controlled. The Westlake BBSM-5 is a good performer overall. It has some exceptional qualities, especially the very low harmonic distortion. It is let down slightly by an uneven on-axis response that just fails to meet the ±3dB limits,

Symphony in Surround

A Swiss orchestral session provided the ideal opportunity to field-test the SPL-Brauner surround recording system. Things were ticking along as **Dave Foister** found out

THE FULL SCOPE of what 5.1 surround can do is only now becoming apparent, along with the new methods we need to develop in order to exploit it. Faced at last with the reality of surround systems in the home, the race is on to make music work on the five speakers of a home cinema setup.

This has given Wolfgang Neumann of SPL the chance to realise an idea going back 20 years: a dedicated surround microphone array optimised for natural pickup of the acoustic event it is placed in. The idea became hardware with the involvement of Dirk Brauner, the microphone specialist whose top-flight models have caused such a stir, and the result is a striking piece of kit that looks like a space station or a chandelier depending on your mind-set. Brauner provides the microphone array, SPL the control electronics and the whole system is known as Atmos 5.1. Early showings aroused a high degree of interest, one of the keenest observers being Thore Brinkmann of Swiss recording outfit Sound Arts. So this is how I came to be sitting in one of two temporary control rooms at the Kultur-und-Kongresszentrum in Lucerne, Switzerland, as Sound Arts conducted their boldest experiment yet with surround recording of an orchestral concert.

But first, back to the system. The Brauner ASM 5 array will be familiar from photographs; less familiar, perhaps, is the SPL control console that interfaces it to the outside world. This is a 5U-high box, well laid out with sensible controls and functions that are readily apparent once you know the system. Although SPL is known for extremes of technology from valve to all-digital, the Atmos electronics are solid-state analogue. The microphone array is connected to its controller via

a heavy multicore that carries signals, power and control voltages.

The microphones are side-firing capsule assemblies mounted on the ends of five arms that place them in positions corresponding to the conventional 5-speaker playback layout. There are three fairly closely spaced at the front for left, centre and right, and two further away at the back for left and right surround. But this is not a 'take it or leave it' system that makes you do things in one particular way; a surprising amount of fine tuning and adjustment is available to adapt the device to a specific task. It might be thought that given a standard playback speaker layout, a fixed microphone rig specifically set up for that layout would do the job best, but the Atmos system aims to allow for a wide range of acoustic conditions, microphone placements dictated by circumstance rather than choice, and different arrangements of the source material—in this case the orchestra.

Each capsule can be swivelled from side to side (though not up and down) on its mount so it does not have to remain in line with its arm, and a great deal of further flexibility is built in to the control unit. For a start, each capsule can have its polar pattern adjusted independently. This is an infinitely variable adjustment, not switched, all the way from omni to figure-of-eight via cardioid in the middle. The centre capsule could be cardioid, the left and right wide cardioid, and the surrounds omni, if that is what suited the circumstances. From here the signals pass through five preamp modules, each fitted with the full complement of necessary functions—input gain, pad, phase reverse, highpass filters and output fader (actually a rotary pot). An illuminated vu meter at the top of each channel shows output level. There is a neat trick fitted to allow

for the problem of adjusting all the gains to achieve correct modulation without disturbing the balance of the channel outputs: the input gain pots are motorised and channel 5's pot can be set to be the master, the other four following its changes.

Under many circumstances (and the Swiss concert was one of them) the preamplified microphone signals go directly to the five corresponding surround output buses for recording on a suitable multitrack. Further flexibility is added by a full-blown surround panning module on each channel. Any microphone can be routed to any bus, and panned LCR with divergence control, and between front and surround. I was puzzled as to why this would ever be useful, until it was suggested that an over-reverberant room could be compensated for by panning the left and right signals a little in from the centre, and conversely exaggerated effects could be created if desired. These facilities are augmented by controls for the stereo spread of the front left-right and surround left-right pairs, each having an associated correlation meter.

Metering is also provided for the bus outputs, and the sharp-eyed will notice that there are in fact six. These allow for a sixth channel derived from a selection of the existing five, the obvious use for this being a low frequency .1 channel. With this in mind the relevant output is fitted with a low-pass filter set at 130Hz, although this is switchable to allow other filter characteristics to be added externally.

Below this on the panel is a small set of controls for a 2-channel external input to be added to the surround mix. It can be sent to the front or back pairs or summed to the centre bus, or any combination, and has a single gain control.

It does not stop there. Each channel

has a separate microphone input to allow use with arrays of other microphones, and two insert points to allow the use of EQ and dynamic processing. As Wolfgang Neumann noted, there is a growing need for ganged 5-channel signal processing since it would be a nightmare to EQ this lot unless it were simultaneously adjustable.

Sound Arts is a 3-man team of Detmold Tonmeisters, Clement Spiess (unavailable for this job), Koichiro Hattori and Thore Brinkmann. Their presence near Lucerne led to an early involvement in the new concert hall and the recording of a series of concerts given by the Lucerne Symphony Orchestra under Jonathan Nott. Nott's enthusiasm for surround coupled with the excellent variable acoustics of the hall prompted the idea of a series of recordings to establish a preferred way of capturing the performances in 5.1 surround, and after some trials with various *ad hoc* arrays this was to be the first with a dedicated surround system. Extra spice was added by the need to provide a conventional stereo recording for Swiss radio, and this necessitated two control rooms. Hattori was to handle the stereo mix while Brinkmann dealt with the Atmos system, with Wolfgang Neumann and Andre Indefurth of SPL on hand to assist.

The Atmos controller sat in one room, its outputs fed to an 02R in the stereo control room. Hattori created his own stereo mix from around 20 overhead and spot microphones, and the spots were mixed with the Atmos feeds on to a Genex 8000 while the stereo mix ran separately to another Genex and DAT. The 02R allowed all the spots to be independently panned to a stereo bus and to a 5-channel surround group, although the balance between them had to remain the same on both. The surround mix was returned to the surround room for monitoring, and meanwhile the raw Atmos outputs were recorded on a DA-88 for comparison with the full mix.

The surround monitoring was done on KS Digital ADM-2 active monitors, with features that made them in many ways ideal for the job. These speakers have analogue and digital inputs, with the crossovers and phase linearity processing implemented digitally. They automatically divide themselves into stereo pairs when fed with a looped AES-EBU input, and have global remote control of volume and input selection. For this session the digital signals from the full mix were fed to one input and the analogue output of the Atmos controller to the other, allowing simple switching between them with accurate overall volume control. The Atmos control unit was routed to the DA-88 and back again via its insert points, allowing tapes of rehearsals to be played back for checking and comparison. This also meant that the Atmos's monitor level

control could be used to match its volume with that from the complete mix when the monitors were switched over.

Lucerne's new concert hall goes to elaborate lengths to provide adjustable acoustics. Its interior feels intimate for a 1,900-seater, but there is a further 8,000m³ of space behind the walls to act as reverberation chambers. Hydraulic doors can be opened and closed to allow these chambers to become part of the acoustic space, and the chambers also have heavy curtains that can be drawn across their walls to control their effect. The setting for the concert being recorded gave a delightful bloom to the orchestra with a consistently pleasant sound throughout the various seating areas.

The concert programme was adventurous, ranging from a Beethoven symphony to Adams' *A Short Ride in a Fast Machine*, beginning with a bizarre piece by Ligeti that could have been designed as a surround demo.

The piece requires 100 metronomes to be set up all round the hall, set to different speeds, and started ticking while the audience is coming in. They are then left to run down, the concert starting properly when the last metronome (the slowest) stops. This meant that the audience sat listening for around a 15 minutes surrounded by ticking metronomes with the orchestra on stage about to play.

The results in the control room were remarkable. The sensation of being immersed in the sound of metronomes was utterly convincing, and as the faster ones began to drop out it became possible to pinpoint individual metronomes all around the listening position. But perhaps more important is the way the system presented the orchestral works.

Many will agree that a good surround recording of an event like this will not attempt to place the listener in the middle of the orchestra, but in the best seat in the house, recreating the ambience of the hall in the listening room. It was immediately obvious in rehearsals that the Atmos system was doing just that. Newcomers to surround often expect dramatic effects from the rear speakers, but this kind of recording will rarely deliver them; indeed, it is often hard to tell just what the surround channels are contributing until they are turned off. The immediate collapse of the image into a flat line across the front is almost as dramatic as any deliberate effect would be. More than that, the presence of the rear channels seemed to enhance the depth beyond the front speakers, adding to the illusion in an unexpected way.

The issue with any surround system is the size of the sweet spot. Some are happy to be strapped into a chair in the centre of the speakers, while others prefer more latitude as to where reasonable results can be heard, and this is obviously hard to deliver. However it is not impossible, and the Atmos

system shows this well; there may be an optimum position to hear the recording from, as there might be a best seat in the hall itself, but the reproduced soundstage works well from anywhere within the loudspeakers and even outside them. Another test of a surround system is whether the individual loudspeakers draw the ear, and the combination in use here passed that one too. With everything running properly, the loudspeakers disappeared, leaving a seamless sound stage enveloping the listener.

The ability to add panned spot mics into the Atmos signal without special equipment, and without compromising the surround image too much, is a great benefit of the system. The mixes with and without the spots both had merits; the spots inevitably added touches of detail, but the Atmos alone won for me in terms of the realism of the surround and the sense of space and depth. The Brauner microphones are superb, helping to deliver a memorable impression of the actual acoustic experience.

The conductor, Jonathan Nott, was also impressed, sharing the view that while both mixes were convincing and each scored in some way over the other, the Atmos system as it stands did a good job of delivering a faithful rendition of what he was hearing in the hall. Sound Arts plan to try further ideas in the magnificent Lucerne hall, but the Atmos package certainly proved itself on this occasion. As a dedicated surround microphone system it is not alone, (it has, perhaps, only one serious competitor) and its simplicity and flexibility make it very attractive to those who want to exploit the growing surround market for serious musical recording. ■

From Muscle Shoals to Nashville to Memphis, Norbert Putnam seemed fated to help shape an alternative to conventional country music. **Dan Daley** plays the name game

Norbert Putman
Muscle Bound

THEY NAMED a street after Chet Atkins and a park after the late Owen Bradley, off Nashville's Music Row. These two producers are regarded as establishing the Nashville Sound as it is known today. In time, Norbert Putnam may receive a similar accolade as he has owned and traded about the same amount of real estate around the Row as both Atkins and Bradley, and owned more studios than the two of them put together. And he was the origin of that other Nashville sound, the one that's got little to do with steel guitars and fiddles, the one Nashville wishes it could hear more often.

While Nashville was focused on country music through the sixties, seventies and eighties, Norbert Putnam was producing much, if not most, of the non-country music that passed through the city during that time. His productions included records that established such artists as Jimmy Buffett, Joan Baez, Dan Fogelberg, Kris Kristofferson, Brewer & Shipley, Pousette Dart Band, Donovan, John Hiatt, JJ Cale, the Flying Burrito Brothers and the New Riders of the Purple Sage. If it wasn't country and it wanted to come to Nashville, it usually wound up on one of Putnam's doorsteps—the ones leading to the four seminal recording studios that he was involved in, including Quadraphonic, The Bennett House, Georgetown Masters and Digital Recorders.

Norbert Putnam was part of the Muscle Shoals crew brought to Nashville by Elvis Presley producer Felton Jarvis in 1965, because TCA Records artist Presley used his label's studios there, and Jarvis wanted musicians with more of an edge than the current Nashville A-Team that included drummer Buddy Harmon, bassist Bob Moore, and guitarist Grady Martin. Along with Putnam came keyboard player David Briggs, drummer Jerry Carrigan, and saxophonist Billy Sherrill, a crew that would become Nashville's second generation of session players. Putnam, a bassist, became a musician simply because, he says, 'There was nothing else to do in [Muscle Shoals] back then. If you didn't play music there was nothing to do. It was such a small town that if you stole a car they'd know who stole it.'

Muscle Shoals is a small town with a big history, tucked in the hill-and-lake country in the north-west corner of Alabama about three hours from Nashville. The place was a momentary blip
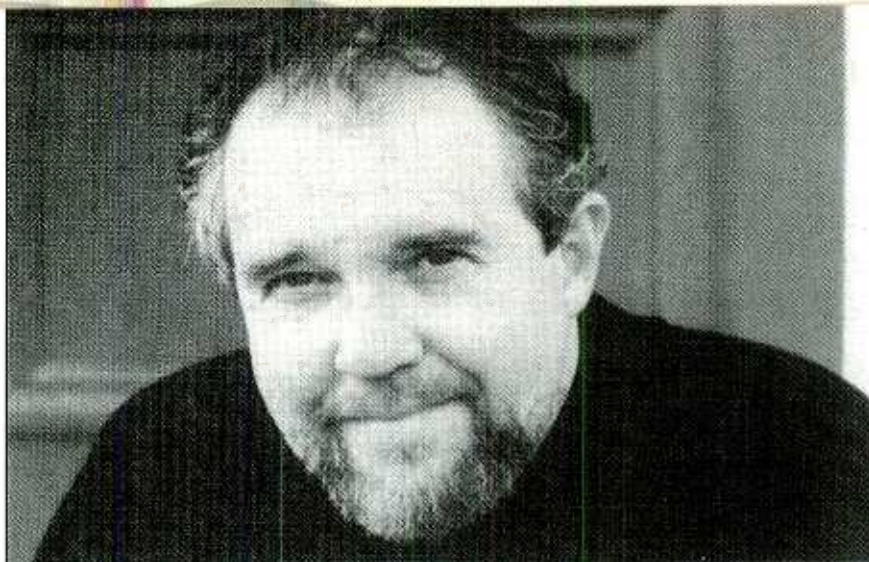
on the peripatetic journey of R&B music on its way to Detroit and elsewhere, leaving behind a small yet talented collection of white southern musicians and writers on whom R&B's voodoo had a lasting effect. The first hit black artist the town produced was the late Arthur Alexander and his recording of 'You Better Move On'. Alexander was the bellhop at the Muscle Shoals Hotel. The manager of the local movie theatre, Tom Stafford, then a gangly kid, would offer the musicians free passes to the movies in exchange for playing on his demos, recorded in the dilapidated offices above his father's corner drugstore, playing to an old Roberts mono tape recorder with only enough microphones to record half the drum kit and a Heathkit recording console with no volume fader—the fade-outs were accomplished by turning the console off and letting the tubes die out.

'We'd hang out with Tom, go to the movies, then go upstairs over the drugstore,' recalls Putnam. 'Occasionally he'd go down a fill a codeine cough syrup prescription and share it with us. Arthur Alexander was coming over and bringing songs to Tom. Tom got Arthur together with Rick Hall [a future music producer and publishing magnate who would go on to produce Alexander and other country, pop and R&B artists], who was the bass player in a band called the Fairlanes. Billy Sherrill [who went on to produce Tammy Wynette and run Columbia Records in Nashville] was the sax player in that band. Rick called up Briggs, Carrigan and I. We did two songs and got in his car and drove to Nashville and played the tapes for Chet [Atkins] and Owen [Bradley] who said they couldn't do anything with it because it

wasn't country. So Rick found Noel Ball, the leading DJ in Nashville. He had a sock hop on a local television channel on Saturday. He was the Dick Clark of Nashville. He also worked for a record company.' The record made the Top 10 two months later.

'That launched Muscle Shoals as a recording centre,' says Putnam. 'It wasn't as soulful as Memphis. We were listening to Burt Bacharach and Bobby Blue Bland. We had experience playing black music, but it didn't come out as black as Memphis. The profits from that record funded Rick Hall's Fame Studio. It also attracted Bill Lowery out of Atlanta with his R&B acts. He brought Joe South, Billy Joe Royal, Ray Stevens, Jerry Reed and Tommy Roe and his bubble gum hits. It was black music but with a white rhythm section. Felton Jarvis would come down to do Tommy Roe for Bill Lowery. Once we got there, we ended up playing for [Monument Records founder and producer] Fred Foster on sessions for Presley and Roy Orbison.'

Through the sixties, the Muscle Shoals musician club grew, mainly based in producer Rick Hall's Fame Recording Studios (actually in nearby Florence, Alabama; the name Fame was an acronym for Florence, Alabama Music Enterprises), and together they played on over 500 hit records of all genres. The Muscle Shoals players who moved to Nashville were soon playing demos for producers Wesley Rose and Jerry Bradley at RCA. 'From those demos came all the master sessions. The producers hired us based on those demos,' says Putnam. 'By 1970 I was doing about 600 record dates a year.' Those sessions included Bobby Goldsboro's 'Honey', Tony Joe White's >

< 'Polk Salad Annie', JJ Cale's 'Crazy Mama' and several Presley hits, as well as the Monument artists Orbison and Boots Randolph.

The Nashville Putnam came to in 1965 had few studios; he recalls only RCA's two rooms, Studios A (now Javalina Studios) and B (now a museum run by the Country Music Foundation), and the Quonset Hut, built by Owen Bradley and later sold to CBS Records, after which Bradley built his famous Barn studio, where he recorded classic records with Loretta Lynn and others. Fred Foster had Monument Studios, part of his record label, where he produced records for Roy Orbison.

Like his cohorts in Nashville's factory-like studio trenches, Putnam was approaching burn-out from all those sessions. But when Bob Dylan came to the city to record his influential *Nashville Skyline* album in 1969, it put Music City onto the larger map of the new mainstream pop music. Dylan changed folk music forever when he went electric at the 1965 Newport Folk Festival; following his lead, bands like the Byrds were suddenly looking to Nashville as a new resource for music away from the already jaded environs of LA.

And it was a veteran folkie and a brilliant and dissolute songwriter—each of whom had revolutionised their own milieu, that changed Putnam's career course from musician to producer. Putnam had played on several Joan Baez albums in Nashville, produced by her long-time recording mentor Maynard Solomon, who also owned her record label, Vanguard Records. But in 1970, Baez had become enamoured of composer Kris Kristofferson, who had stood Nashville's staid songwriting community on its ear only a few years earlier with songs like 'Me And Bobby McGhee', which blues-rocker Janis Joplin had turned into a radio anthem, and the brooding 'Sunday Morning Coming Down'. She chose Kristofferson to produce the record.

Meanwhile, Putnam and David Briggs had moved forward with an idea for a recording studio in Nashville that would let them indulge their fascination with technology, do their demos as would-be producers and perhaps make a few bucks in the process. Engineer Elliot Mazer, who worked on Linda Ronstadt's *Silk Purse* recording in Nashville, upped the ante by telling Putnam & Briggs he would bring projects to a leading-edge facility there.

They fitted Quadraphonic Studios —still located on Grand Street off Music Row—with a Quad-8 class-A, all-discrete desk with Jensen transformers and high-low shelving EQs and a 16-track Ampex MM1100 2-inch deck. Putnam had been talking with CBS Laboratories engineers in Connecticut about their work with the nascent quadraphonic format, and sensing a trend, he chose

Quadraphonic as the studio's name.

'We stuck a pair of JBL 4310 speakers in the back of the room and bang, we had quad,' laughs Putnam. The studio had quickly become a hangout for those disenfranchised by the Nashville oligarchy and for the swelling influx of young musicians and writers who were coming to Nashville in Dylan's wake. So it wasn't surprising when Kristofferson agreed to use Quad to make what would be Baez' *Blessed Are* album. It also wasn't surprising, at least in retrospect to Putnam, that Kristofferson, who was painfully shy in the studio, intimidated by the technology and the musicianship and who needed a belt to work up the nerve to bring his gravelly voice in front of a microphone even for demos, came to the first session tipsy and pleaded with Putnam, the bassist and arranger, to take over production chores, as well. The sessions went quickly, including the recording of a cover of The Band's 'The Night They Drove Old Dixie Down' which was Baez' first and only hit single.

This event caught the notice of Clive Davis, then president of Columbia Records in New York. He flew Putnam to Manhattan and feted him for making a record with a folkie and selling 1.5 million copies. 'He called me a genius,' says Putnam, still bemused by the meeting nearly 30 years later. 'I had never been called that before. So I said, "Mr Davis, it's my only record as a producer. I might have gotten lucky". He said you must know something, because she's made 10 records and none of them ever sold more than 100,000. Then he looks at me and says, like this was going to be the greatest honour a man could bestow upon me, "I want you to produce all the folkie artists on CBS Records". I thought to myself, I have died and gone to hell. This would be like conducting an orchestra made up of banjos. I had grown up in Alabama and I liked soul music. Clive said he had just signed Gamble & Huff out of Philadelphia to take care of that for him. Then Clive grabs a tape box and slides it across his desk to me. It was a demo of a guy named Dan Fogelberg.'

Putnam produced Fogelberg's debut record, *Home Free*, released in 1973. The record, however, didn't live up to Davis' expectations and it wasn't promoted well. But there was hope. 'It was beginning to look like his career was over before it had started, but Dan called me from the road one day that year,' says Putnam. 'He was playing in Jackson, Mississippi, and he was screaming into the phone telling me that 2,000 people had turned out to see him. Seems a local radio station had played the record and it was getting great response. So a while later, his manager Irving Azoff went to see Clive and asked Columbia to release him. Clive >

Gary Rydstrom, effects mixer

< when we recorded the score, with large portions of the film still being in temporary form due to visuals; primarily optical effects not yet finalised and computer effects not yet manufactured. Given the structure of some sequences in the picture we were relying on those final form opticals, and so what we did was score to a picture that was an approximation and then the music editor, Ken Wannberg, was assigned the task of aligning the existing music to the new picture. What you hear in the picture is a manufactured track from material that was meant to be there, but in adjusted synchronisation with the new picture, whereas what you hear on the CD is more of a compilation of the original material that was recorded for those sequences and in some cases cut down due to time constraints.'

In terms of the Foley, there was a similarly rational approach. After all, having worked on the project for so long and ensured that there was a comprehensive temp mix, Ben Burtt had a fairly good idea as to where the music would be and what could possibly be heard over it.

'We would say things like "Don't worry about this. You'll never hear the footsteps",' Tom Bellfort recalls. 'However, Foley also contributed to some wonderful effects, and so they did a really terrific job.'

'We did everything that moved,' asserts Dennie Thorpe, who, along with Jana Vance, was one of the Foley artists on the film. 'You see, even if we're only going to use Foley in conjunction with sound effects or production effects, we

Tom Bellfort, co-supervising sound editor

have to be there when all of the production sound gets stripped out for the foreign versions.'

Thorpe and Vance divided the main characters between them; the former taking care of footsteps and so forth for Anakin Skywalker, Queen Amidala and Obi-Wan Kenobi, while the latter did the same for Jar Jar Binks, Qui-Gon Jinn and Darth Maul.

'With whatever we do, we always want to convey that there's a reason for us to make a specific sound,' says Thorpe. 'Obviously the actors are telling the story and there are certainly production effects and other sounds cutting in and out, but we're all working together to create a sort of translation of what you're seeing into sound. For our part we had to deal with plenty of wet, gushy sounds, metallic sounds, zingy sounds.

'For instance, there's kind of a cute scene between Jar Jar and another character named Sebulba that takes place in a marketplace—Jar Jar sees all of this food, he gets hungry and wants to take something, and in sort of slapstick fashion there's this iron grille that the food is hanging off. There's very little happening except for the characters' voices and what they're doing, and so we were happy to be assigned the squeaks of the iron grille, as well as other effects such as Jar Jar's tongue zinging out to grab a piece of food, the food then flying through the air and splashing Sebulba as chaos ensues. To that end I was crunching food, mashing my hand inside a cut orange for the sloshing sounds, and flicking a metal Slinky inside a box for some of the more zingy sounds. Added to that were the footsteps and all of the other background stuff that we put in, helping to make the whole scene come to life.

'By that time we'd already gone out shopping and acquired all of our props, because on this film we weren't completely set up in the studio for everything. We looked at the film with Ben and with our sound editors and Foley editors, and then Jana and I went out shopping to various junk stores and recycling places for all of the materials that we wanted. Then, in terms of the recording, our top mic was a Neumann U87, and we used that in conjunction with a PZM and a room mic, and Tony Eckert mixed those three so that we could get perspective on our sound. That's something we really like to do in Foley. We don't want it to sound like it's in your face all of the time, and so as the characters walk through the hangars and the palaces and the courtyards their footsteps sound different on

every surface.

'That worked very well for us, a large part of the solution being the EQ that Tony came up with. Then, when the characters were shot up or blown apart, we'd use various old vacuum cleaners, jacks, drills and heavy pieces of iron, so that each head and arm and leg and hand and maybe torso had a sound of its own. Added to that were the sounds of the characters touching themselves or each other, and being that a lot of them are not people we would use pineapples or coconuts or even cantalopes for their skin surfaces.'

Dennie Thorpe also did Foley work on *Return of the Jedi* back in 1983, a film in which C-3PO had a somewhat more complete appearance than the skin-less, work-in-progress look that he boasts in *The Phantom Menace*.

'He had a whole different set of sounds in Jedi,' says Thorpe. 'For this movie we just ended up using a large stainless steel salad bowl with wires, whereas in *Jedi*—where you couldn't see inside of him—we manipulated ice trays along with some other aluminium,
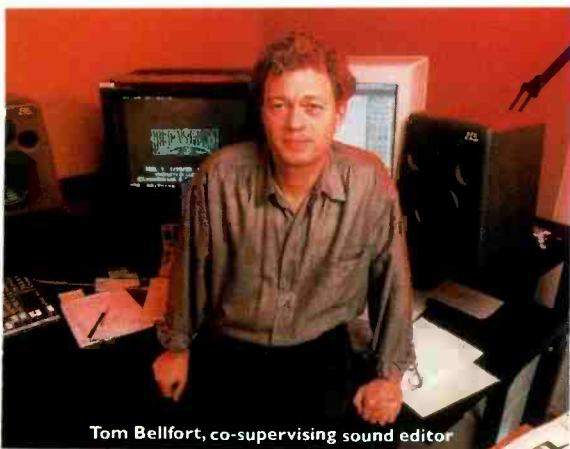

metallic sounds. There again, in both films his feet are complete, and so for those I used these really old-fashioned, 18-inch-long kid's skis that sound sort of like flippers when you put them on and walk on the various surfaces.

'Foley can be a bit mysterious when you're sitting up in the control room because you don't know what we're using, but then when people come down onto the Foley stage and see us work it appears as if all we're doing is playing all day. They really like that, while at our end we're sometimes wondering if our stuff is going to cut through. However, the sound effects department always leaves room for Foley—we've worked with those people for quite a while now and they know what kind of things they can leave to us.'

'We knew right from the beginning what effects could do and what Foley could do,' adds Tom Bellfort, 'and so the tracks themselves were not as typically dense as on other movies. You know, what happens frequently is that everybody duplicates everybody else's work and then you throw it out, but that rarely happened on this project.'

Another departure from tradition was the uniform use of Pro Tools at >

Matt Wood, assistant sound editor

< Skywalker. Normally the dialogue people work on Studio Frames while the effects department uses Pro Tools, but this time around it seemed logical for everyone to be on the same platform while elsewhere Avids came into play.

'That meant I was able to transfer files back and forth very easily up to the main house over the network here at the Ranch,' says Matt Wood, who assisted both Tom Bellfort and Ben Burtt in terms of the sound editing. 'George would work on an effects scene for something like the space battle, and we'd take those tracks and we could cut something together quite quickly. We'd even do it Dolby encoded with surround information and send it back over the net, so that they could play it off their Avid right into a theatre up there at the main house and get a good idea as to where we were going.

'In terms of the ADR recording I was really interested in doing something different. The new Apple Mac Powerbook G3 had come out, and I thought it would be a good test to see if I could take some of the stuff that I had been doing in the studio with the Pro Tools and try to apply it to a portable-type system. I therefore built a portable ADR recorder; it's got a preamp, an A–D, a mixer and video distribution with streamers, and I had all of the information that Tom spotted in the computer, so I was able to effectively be a recordist. I could go into a quiet room anywhere and record, and when we worked at places like Mag Masters in London and the studio in the Bahamas I had a complete backup in case it failed. With that setup, the second we had completed some ADR I could take that information, back it up to CD and I could also send it back home over the Internet so that they could be cutting it immedi-

The Foley-stage team rattle their wares

ately. Towards the end, when we were still looping in the middle of print mastering, that was a necessity.'

In addition to helping assemble the music editing system, Matt Wood was in fact also responsible for recording some of the raw sound effects for the film. These largely consisted of those used for the pod race, entailing capturing the sound of cars going around a race track, while the overwhelming majority of both new and classic effects came from Ben Burtt's vast library of sounds.

'A lot of it is on ¼-inch tape,' says Wood, 'so we actually went straight from that format through an Apogee A–D convertor and into our system, enabling us to get the purest sound out of the source tapes. We ended up with 1,500 newly recorded sounds for the film, and, unlike most sound designers who will just give you a list of raw sounds, Ben cut them into the film as we went along and was able to distribute cut sessions to the editors. From an editing standpoint that was a great template.

'The pod race is one of the main scenes in the movie and Ben worked really hard on it for a long period of time. It's not music-driven, it's all visual effects, and George wanted a different sound to represent each pod. We had to get quite creative there and editorially it features a lot of quick cutting, so it was hard to really get a sense of the slow-medium-fast approach in a shot that lasts two seconds. It was challenging, but throughout the project we had a good idea of what was coming in terms of visual effects. Ben also worked in the animatics department and they put together a virtual storyboard—a low-res video representation of what ILM was going to produce for each shot —and it was great to have that temp version to work with early on, transfer our efforts to his Avid so that he could listen to them in a theatre, and then get his feedback.'

Among the aforementioned 1,500 newly recorded sounds were recognisable oldies such as the light sabres and lasers, which often amounted to the original elements being tweaked to meet with present-day demands—sample the new double-edged light sabre.

'We used a lot of tools for processing,' says Wood. 'One of them was the Symbolic Sound Kyma, which is basically a giant DSP device that has the software control to do anything. In addition to that Ben also has a Synclavier which he used a lot, and then there was a vast array of plug-ins on Pro Tools.'

At the same time, the comprehensive use of Pro Tools at the Ranch also

proved to be advantageous when, at the last minute, Tom Bellfort was able to help cut some of the Foley and dialogue without even a hitch.

'It was really smooth sailing all of the way,' he says. 'George knew exactly what he wanted. He would sit through the mix and give us his notes, go to his editing suite and do some of the fine cutting, and then we would have playback. It couldn't have been more straightforward. The lack of stress was exceptional, and I cannot think of too many films that I've worked on that have been like this. It was a really sweet job and there are few of those around.'

After each big effects session on the Pro Tools the results would be divided into premixes for Gary Rydstrom, who would then solo up to 24 outputs per premix on the Neve Capricorn and go to an MMR-8 24-bit digital drive.

'The main challenge on this project was to live up to what people heard before on these movies as well as to their expectations for the new movie,' he says. 'That's why Ben Burtt's contribution was indispensable. He knows the sound universe of these films well. He knows how we should deal with the laser swords and space ships, and so on, and he really kept an eye on how we did things so that they were consistent with what was done before. I would then try to go just a little wilder, making the laser sword fight one step up from the previous movies by spinning the sound into the surround and over the audience's head, and doing all sorts of fancy tricks that would place viewers more within the action.'

The result is an interesting blend of the cosily recognisable with the subtly new, highlighting the sonic advances that have taken place since the Jedi last appeared on-screen back in 1983.

'Just having digital editing and digital mixing is so revolutionary,' says Rydstrom. 'Finally we can control where things are panned and how we blend them together. It's quite striking. Our use of spacial playback in the theatres is much easier and much better these days, and in that vein we had one more surround channel that we could use for this film. I had three surround channels and three front channels, and it was really nice to take advantage of that.' ■

The Mix So Far

With some 60 years experience in the console business between them, Sony's **John East** and **Antony David** chart the development of the mixing desk

I N THE EARLY DAYS of recording, a microphone amplifier would be connected to a gain control and the output fed directly to a cutting lathe. The only other control on the console was the switch for the red light to cue the musicians. Some might say that it has been downhill all the way since then.

If so, the rot started when equalising amplifiers were introduced to straighten out the nonlinearities in microphone frequency response, the object being to produce as flat a response as possible. Only men in white coats were allowed to use the ±3dB available to make the sound more true to the original. Things got worse when multitrack technology allowed music to be recorded sequentially. Soloists got a second, third or fourth chance to get it right or—when digital tape machines arrived—to be dropped in a demi-semi-quaver before that tricky line in the second half of the chorus, and later—with pitch shifters— just to get the words in the right order, and later still just to utter to right syllables in any order into a sampler.

Initially, mixing consoles were designed by the organisations that used them. This continued for longer in some industries than others—National Radio stations, for example, continued to build their own until recently. The BBC started with Type A consoles in the late thirties; these had discrete mic amps and standard configurations of modules in an external rack with a one piece control panel and were used for recording talk shows and drama. These were replaced in the fifties by the Type B, still using valves, still modular but introducing features such as SLS (Studio Loudspeaker) muting with mic open + channel fader up + group fader up +main fader up and clean feeds (n - 1 outputs for cueing external contributors). It was not until the seventies that the BBC started buying consoles from manufacturers based on the GP specification. Neve, Audix and Calrec built a series of consoles based around these specifications for different applications. Then in 1979 the first, albeit modified, in-line console, an SSL 4000E was installed in BBC Manchester Studio 7 then rolled out into all major studios and trucks. This was probably the first time an up-is-louder fader was introduced and it caused a lot of consternation. To this day some consoles at the BBC still operate in the traditional down-is-louder mode.

TV broadcasters embraced multitrack recording for production and postproduction. This enabled them to improve soundtracks after the event and reduced pressure on the expensive shooting phase of production. Mixers were adapted from music multitrack consoles and TV production consoles had multitrack routeing and monitoring added.

The film community also developed a methodology that differed from music production. With relatively large budgets and a seemingly endless supply of mag dubbers, film dubbing studios were working with larger numbers of sources from an earlier date but had different routing requirements. The split of sources into dialogue, music and effects gave rise to a different mixing approach which in Hollywood culminated with the 3-man monster—effectively three consoles connected more physically than electronically and designed to rationalise sources by way of premixes. Uniquely, film studios worked in reverse as well as forwards (a function of the way film transports were used for audio). Companies such as Quad Eight-Westrex and later Harrison dominated.

Film sound was originally mono but the arrival in the mid-seventies of Dolby Stereo necessitated the expansion to four buses of the console outputs. Mixes were finalised in all the different release formats and required not just the complete mix but also submixes of Music & FX (M&Es) and Dialogue so that other language versions could be readily prepared without the need to remix the entire soundtrack. This introduced further complexity into the monitoring system, as mixing had to be done in the context of existing premixes. To handle this, large integrated monitor matrices were introduced which benefited later when resettability was added, pioneered in this instance by Harrison.

Film formats developed further with the arrival more recently of Dolby SR.D (5.1), DTS (5.1) and Sony's SDDS (7.1) which incorporated variously a subbass channel, stereo surround and Inner Left, Inner Right. These formats further complicated the design of consoles which required great flexibility in signal routing to enable the engineers to create mixes in all the release formats.

TV stations started to transmit Dolby Stereo encoded mixes, which are broadly compatible with conventional 2-channel stereo. As consumers started to buy decoders for home use, TV production companies, waking up to the opportunity, made their more prestigious productions in Dolby Stereo and more recently in 5.1, adopted as the standard for DTV. As a consequence TV production consoles now increasingly feature surround sound capability.

The Music Recording Industry dabbled with surround sound in the early seventies with the Quad, QS and SQ —all 4-channel formats designed to be encoded onto vinyl records. Although none were ultimately successful, console designers left in the rear buses which were put to a variety of uses. Unfortunately panning circuits for these LF, RF, LB, RB formats do not readily adapt to the LCRS of Dolby Stereo which meant that multitrack music consoles from the seventies and eighties were not suitable for film-type work. This, and the differences in routing requirements, kept the designs of consoles for film and music recording quite distinct. More recently, designs are becoming more similar mirroring the convergence of the industries, with recording studios doing more sound for picture work, and movies increasingly dependent on TV and VHS distribution.

Consoles have been reputed for a variety of attributes—ergonomics, automation, technical performance, reliability, price-features ratio and pedigree but perhaps the most important and least quantifiable is sound quality. The quality of the entire signal chain involves a plethora of interrelated issues: mic amp performance, headroom, crosstalk, phase linearity, slew rates, intermodulation, distortion, output symmetry, frequency response and so on but one element of the console

often dominated the sound quality issue —the equaliser. Nowadays the unique sounds of different EQs are reasonably well understood but names like API, Focusrite and Neve and the sonic magic they stood for, had a big influence on perceptions of the consoles with which they were associated. Dynamics processors also played a role in defining the sound of a console—SSL's noise gates and Quad Compressor for example.

Digital signal processing had a profound impact on mixing consoles. First came delay lines and digital reverbs as external devices, then large-scale digital consoles - not without false starts. Most public perhaps was Neve's DSP built in collaboration with the BBC (1981-1983), from whence stemmed the immortal words to the impatient would-be recipients 'It's just a matter of a few lines of code'. A spin-off from a US military project saw AT&T's powerful DISQ, adapted as the processor connected to existing control surfaces which had recall interfaces. While neither of these projects was commercially successful, both showed that it could be done. Prior to both these, EMI had developed an in-house digital console by 1980. Eventually digital consoles came of age with AMS Neve's Capricorn and later Sony's OXF-R3 'Oxford' console.

By the end of the eighties a number of workstations based on proprietary technology appeared like AMS' AudioFile and DAR's SoundStation. Small digital consoles were designed to work with these (AMS Logic 1) but the rules were changed forever when Yamaha entered the market with 02D and later 03R. Priced at or below their analogue equivalents, these consoles brought digital technology into the mainstream.

For postproduction came integrated console workstations such as Lexicon's Opus in 1988 and SSL's Scenaria in 1992, which included a digital video scratchtrack. Integration of the mixing console with an external routing matrix has been another recent introduction. Stagetec's Nexus router, for example, works on a TDM (Time Division Multiplex) bus and can interface with the Cantus console. Systems like this facilitate network operation—another trend for the future.

When we look back from 2009 what changes are we likely to see? The postproduction workstation model suggests that full integration of storage and mixing functions will be extended to music recording consoles. We can expect more platforms (including perhaps Windows 2000) which can support applications from a variety of suppliers and more screen driven user interfaces where the challenge will be to get the connection between the necessary hard controls and the screen more intuitive and direct. The move towards higher bandwidths will settle at some level but another parameter will emerge as being

a key issue in sound quality. In short, the strange mix of technology, music and black art which has characterised the last 50 years looks set to continue in the same vein.

In the beginning everything was mono. Things were simple and mixing consoles had a small number of inputs and a single output. Consoles were physically small, so it was easy to reach any control. Stereo brought the requirement for twin outputs and later a pan control (typically a multiposition switch) to enable allocation to the left and right outputs. It was not long before the stereo recorders were adapted to record individual elements, not necessarily at the same time, on the two tracks and multitrack tape recording was born.

As the number of tracks grew, consoles got bigger to accommodate the increasing number of inputs and monitor channels. When 16-track recorders were introduced in the late-sixties console control surfaces moved towards having an input section on one side and a tape monitor section on the other.

As 24-track machines became available, this configuration became more unwieldy. Consoles were getting wider than could easily be reached from a central position and the distance between the input channel and monitor channel was a major inconvenience. Enter the in-line concept pioneered by MCI and Harrison in the mid-seventies, where monitor path and input channel sharing the same number were located in the same channel strip and could share resources such as filters or aux sends. This relatively complex idea took a bit of getting used to but reduced the width of the console to manageable proportions, made recording, especially overdubbing, much easier and began to offer some new and creatively useful possibilities. As a consequence channel strips became longer and knob-switch density increased.

Then along came SSL and refined the idea incorporating a dynamics section in every channel and putting a TV monitor for the automation system in the centre of the console. A corollary of this was that master sections grew larger to accommodate the display and keyboard. After two early iterations, the A-series and B-series, SSL's 4000 E-series became the new industry standard. 'No one will ever buy a mixing console with a TV in it', one famous producer was heard to say at the AES show in 1979—we don't know what happened to him.

During the eighties the size of consoles continued to increase as two 24-tracks were synchronised and 32- then 48-track recorders were introduced. Consoles purchased by high-end studios during this decade increased by an average 3-4 channel strips per year. By the early-nineties in-line consoles with 80 channels or more—and getting on for >
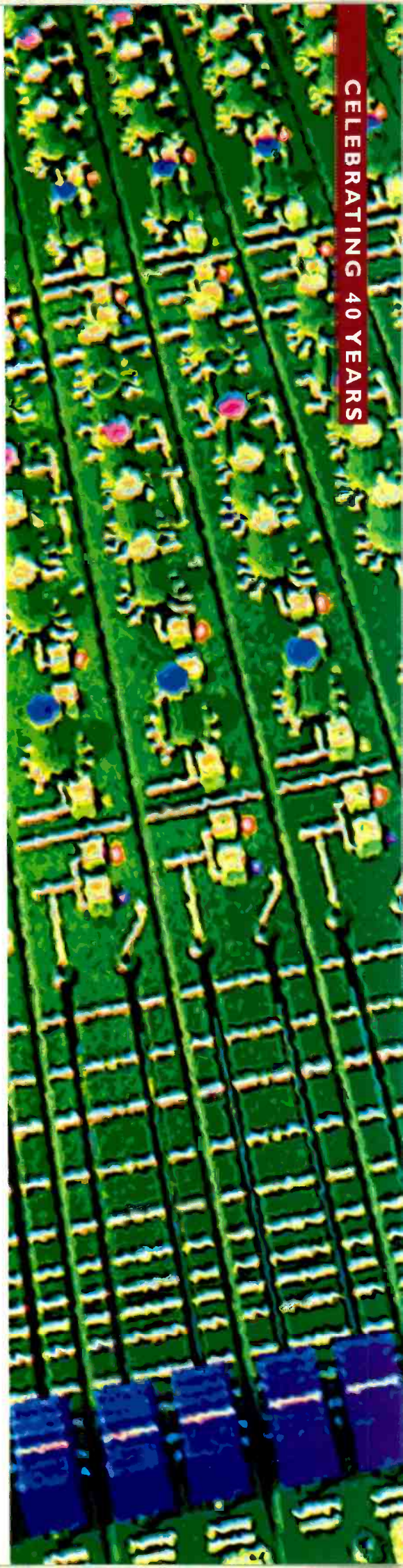
< 5m wide—were not uncommon.

The widespread use of in-line design and the not dissimilar Neve V Series consoles created a de facto standard which spawned a generation of freelance recording engineers who could readily move between studios. In the case of SSL studios this was enhanced by the ability to transport mix and setup data on floppy disc. However, size had once again become an issue, partly of ergonomics, partly of cost and digital control offered a way to reduce both with the added benefit of resettability. Early pioneers of the analogue assignable console include Trident (Di-An) and Harrison (Series 10) but the Euphonix models, starting with the Crescendo, were perhaps the first to sell in any meaningful quantity and really succeeded in shrinking the control surface.

An obstacle to the acceptance of assignable control surfaces, particularly for those not very computer literate, has been the ease (real or perceived) with which engineers can use such systems. Interestingly, computer-based systems which have received good acceptance in the lower end of the market and have a serial control system (the mouse) are now adding increasingly comprehensive control surfaces. At the high end, control surfaces such as Sony's Oxford have balanced assignability with a short learning curve.

EARLY MIXING CONSOLES were built with valves: apart from the obvious difficulties with heat, power, reliability and size, engineers and producers used to bemoan the 'roundness' and 'softness' of the sound they produced and wished for something with a bit more edge. They got it—the transistor, which gave designers a lot more scope for designing processors and made practical devices like noise gates and parametric EQs which were very difficult to implement with valves.

The 5534 integrated circuit which first appeared in the mid seventies was deemed sufficient both to make a mic amp and drive a 600Ω load; because of this it became a universal building block, reducing cost and perhaps most importantly, size. It was widely adopted and enabled more complex signal paths to be constructed within a channel strip creating more flexibility and creative possibility.

At around the same time VCAs were incorporated into designs as the controllable variable gain element of automation systems. Unfortunately early VCAs were a serious weak point in the signal chain because of their noise and distortion performance. This led to the introduction of motorised moving faders which, despite the mechanical challenges, put the audio through a conventional conductive plastic track.
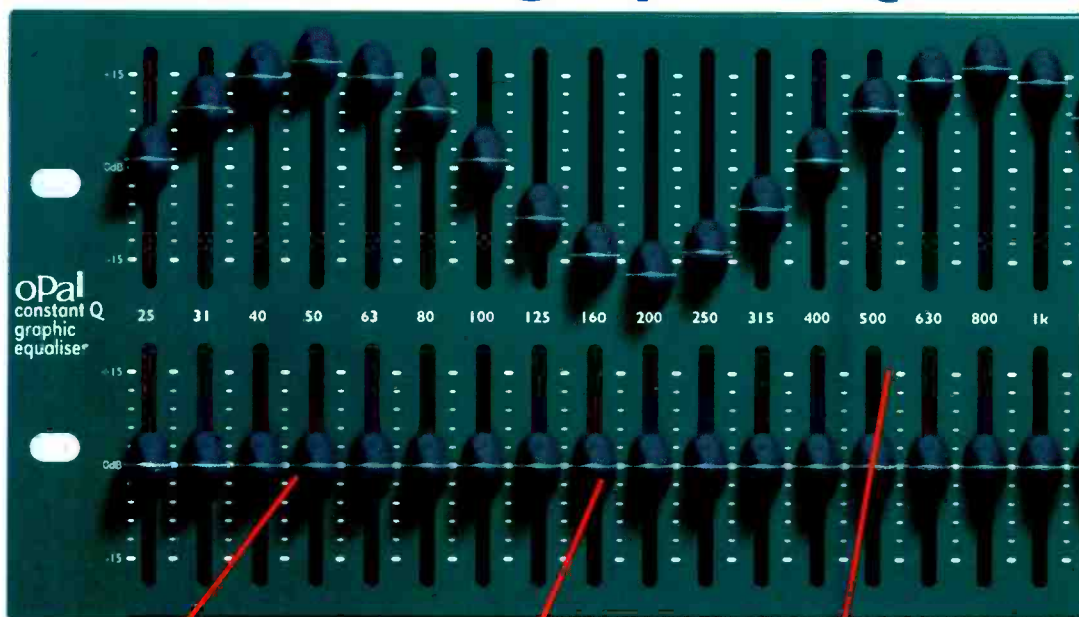
While developments in analogue technology have plateaued, developments in digital signal processing are accelerating with annual quantum improvements in processor speed, storage capacity and transmission bandwidth. This suggests that the cost-performance ratio for digital systems will continue to improve.

Improvements in A–D/D–A performance have now made digital systems capable of out-performing complex analogue chains. Once safely in the digital domain, properly implemented digital processors can preserve sound quality whereas in analogue systems crosstalk, noise and distortion are a constant concern. The enormous flexibility of digital mixing consoles and the potential to keep improving them with software updates clearly indicates where the future lies.

The challenge now focuses on software development: high level tools which enable skilled audio designers to accomplish their craft in the digital domain will be valuable in producing innovative devices and processes. The integration of applications from many sources will present challenges which may largely be solved by Apple, Microsoft and their successors.

Analogue electronics and the 'valve sound' in particular have provided a wealth of wonderful sounding devices whose value will only increase as time

goes on. Sampling and emulation techniques will ultimately be able to accurately reproduce these sounds but the originals will undoubtedly become collectors' items.

THE MULTITRACK mixing session of the early seventies typically involved the engineer refining and rehearsing the mix, often in sections, until he ran out of hands and asked the assistant for help, then the producer, then the bass player and so on (excluding the vocalist for obvious reasons). The mix was finished with a series of edits on the ½-inch (or ¼-inch) as the better elements from various attempts were combined.

The first VCA automation systems from companies like Automated Processes (API) appeared around 1973 and recorded multiplexed fader and switch automation data onto a spare track of the multitrack. Updates required the use of another track—the levels of undo being defined by how many tracks were available. Delays inherent in this process meant that after even two or three passes cuts were already becoming noticeably late.

Enter linear time code which migrated from videotape and allowed a solid relationship to be created between synchronising devices, including automation systems, and the audio on the tape by striping code >

< on one track of the tape usually with a blank guard track to prevent crosstalk. With this reference, data could be stored in the computer and updated ad infinitum. Time code also made the synchronisation of multitracks a possibility which contributed to the use of greater number of tracks and the consequent need for more channels in the console.

The moving fader which came a little later (Necam and GML) overcame the concerns for sound quality caused by the noise and distortion performance of VCAs and started a long-running debate on the relative advantages of one approach against the other. VCA systems offered trim updates which refined existing fader moves while moving fader systems provided the reassurance of a visual indication of the 'real' position of the fader. Hybrid systems like SSL's Ultimation which used both moving faders and VCAs were able to provide the best of both worlds but showed that moving faders had eventually won the argument.

Integrated machine control was a valuable by-product of automation and was extended to include track arming and offset storage largely eliminating the need to have the multitrack remote next to the console and making the engineer's life easier.

Automation of other variable controls such as aux sends and EQ was also added by some but the cost and complexity prohibited widespread use until the advent of assignable control surfaces. A popular solution to this was Total Recall, SSL's system for storing switch and knob position for subsequent display and manual matching via a colour screen. This enabled large consoles to be reset in a few tens of minutes and encouraged more efficient use of studio time and portability of projects to similarly equipped studios.

Next came digitally-controlled systems, both analogue and digital. Assignable controls which are not motorised required another method of displaying position or status: with fewer controls to accommodate, more real estate could be devoted to such displays and it became practicable to automate every useful function of the console including routing. This has opened up a new set of creative possibilities.

The Holy Grail of automation systems is complete dynamic automation of all useful controls and reset of all the interconnections and outboard gear involved in a session so that previous conditions can be exactly recreated. MIDI interfacing offers a way to achieve part of the external objective but the continuing use of venerable analogue devices means that this utopia will not be created. Perhaps this is a good thing—one moment can never be repeated. ∎

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< protocol, which it calls EASI (Enhanced Audio Streaming Interface). On a page on the Emagic web-site headed 'The true motivation for Emagic's engineers to develop EASI', Markus Fritze, chief developer for new technology integration at Emagic, does not pull any punches: 'Last year we thought that ASIO would be the natural choice to support new audio hardware, mainly because there were a number of drivers available. While we were implementing our ASIO host I soon ran into problems: the documentation is poor and the host side is almost undocumented. We had to do lots of trial and error experiments to get the ASIO drivers to work. In my opinion the architecture of this interface is too simple to be able to support the newest audio hardware with input monitoring, hardware mixers, etc. I think that Emagic—as an innovative company—can't afford

to rely on this proprietary interface and just wait and hope for improvements from Steinberg which again may or may not be documented.'

Felix Bertram, group executive for hardware and engines, adds: 'We do not have a not-invented-here mentality at Emagic, we are just striving for a better solution. We want to share the EASI technology with the music industry to establish EASI as an open, alive and widely accepted cross-platform standard.'

To this end, Emagic is making EASI public (no Non Disclosure Agreements here), and has placed a preliminary EASI description up on its web-site as a PDF file 'to underline our honest intention in creating a technical solution that will be beneficial to everyone', with the additional comment 'No politics, no hidden agendas, just straightfoward technical improvement.' Indeed, it seems that the company can not be faulted on either its honesty nor its honest intentions, but this is a story that has yet to unfold. Ultimately, what matters to users is that they get a system that works robustly and, hopefully, flexibly, but also that they do not get thrown into a dark age of format incompatibility and confusion. Today, Emagic's reach extends world wide. The company, that founded the Grass Valley, California-based Emagic Inc in 1994 to better serve the Americas and Pacific Rim territories, now sells into 44 countries, ranging from Iceland to

New Zealand, Hong Kong to Saudi Arabia, via its international distribution network, and employs more than 60 people. In recognition of the importance of the constant work to improve existing technologies and develop new, superior ones, Emagic places R&D at the centre of its business activities, with a third of its employees being engineers. The ASIO-EASI saga described earlier illustrates this orientation in action. Indeed, the pursuit of technical excellence comes across as being at the core of Emagic's philosophy. It is a pursuit that has stood the company in good stead as the power, functionality and flexibility of computer-based sequencing software has increased phenomenally over the past decade. What Emagic do next will be interesting to see—in particular what they do with the BeOS platform's multimedia, multiprocessing power—but the same attention to technical excellence will be at the heart and soul of it. ∎

Spreading the News

The changing requirements of British independent news broadcasting has seen ITN busy re-equipping its established HQ. **Kevin Hilton** seeks out the truth

TELEVISION NEWS can no longer exist in a rarefied atmosphere —it is programming much like anything else and so has to attract and keep viewers. Ratings are all and news providers have known this for a long time. As much as the likes of Walter Cronkite and Richard Dimbleby are held as examples of a different style and a different age, their reassuring presence still brought in the viewers. Times have changed; it is now brasher and more infotainment-based. Which is why American newscasters have become notorious for the bubble-headed bleach -blonde approach and UK television this year has seen a major shake-up of BBC and commercial news coverage.

The news organisations of both countries have shuffled around their presenters and reporters and introduced new graphics. In an even bolder move, the mainstream commercial network, ITV, has replaced the flagship *News at Ten* programme with a 6.30 bulletin and an update at 11, leaving the rest of the evening clear for uninterrupted movies and dramas. The news programmes for Channel 4, the cultural, minority interest commercial station, and Channel 5, the most down-market commercial service, have also been repackaged, a tough job given that one organisation,

Independent Television News (ITN), provides programming for C4 and C5, as well as ITV. As much new broadcast technology has initially been aimed at the news market, digital is a big part of the changes, particularly at C4, but this is understandable as the whole of ITN is in the midst of a major upgrade at its headquarters beneath the glass atrium in Grays Inn Road, which runs between the London districts of Holborn and Kings Cross.

ITN was founded in 1955 by the various ITV regional companies, operating as a wholly owned subsidiary with a remit to provide national and international news. At the time, BBC News shunned fancy visuals on the grounds that it would detract from the integrity of the coverage; ITN was given the brief to 'combine the verve of Fleet Street with the authority of the BBC'. It did this by importing many of the presentation stylings of American TV—most famously the 'double-header' approach, where two presenters anchor the programme—and picking up on the growing use of modern technology, fighting the BBC to be first with satellite links. ITN also succumbed to the lure of personality, hiring such flamboyant figures as Robin Day (a notoriously aggressive, bow-tie-sporting political interviewer,

who later became a BBC mainstay) and the idiosyncratic, toupee-wearing Reginald Bosanquet, who became an unlikely sex symbol of the 1970s.

Channel 4 has used ITN for its news coverage since it went on air in November 1982. The UK's fourth terrestrial service (behind BBC1, ITV and BBC2) had been the cause of much debate before it was finally brought to air. Channel 4's remit was clear: to serve minority interests and encourage innovation. This it has done for broadcasting and the cinema, becoming a cofounder of many successful British and Irish movies through its Film on Four seasons (which it has now developed into a new digital channel, FilmFour). Alongside eclectic domestic fare, C4 has built up a reputation for importing much of the best US TV, such as *ER*, *Friends* and *Cheers*. It has also been fearless with regard to sex, violence and bad language, angering the moral majority of Middle England to the extent that a now ex-chief executive of the channel was dubbed the 'pornographer-in-chief'.

To fit into this mix, ITN tailored its coverage, producing bulletins that give analysis and in-depth coverage of major (and sometimes not so major) stories. Until this year, *Channel 4 News*, which consists of short bulletins during the **>**

< day and a half-hour early evening programme Monday to Saturday, had to share a studio with *News at Ten*. With the contract to provide news for C4 up for renewal at the end of last year, ITN agreed to build a dedicated, integrated digital facility, with a new control room and studio, that went on air during January this year.

Having its own purpose-built studio is a key part of the new image of *C4 News*. The studio, which was previously the ITN kitchen, has full-length windows that look onto the atrium in the middle and into the newsroom to the viewer's left, the technical areas to the right. In keeping with the modern approach to current affairs, there is no intrusive desk for the presenter to hide behind; the main anchor, Jon Snow (who is as well known for his selection of vibrantly coloured ties as he is for his dogged interviewing style), usually stands in front of a video wall to introduce the main stories. Copresenters sit alongside a designer coffee table bearing a laptop computer. Sneakily, the studio is made to look much bigger than it is through the clever use of camera angles and deep perspective.

All this has modernised and sharpened up the programmes without detracting from the serious nature of the approach. Martin Swain, ITN's production standards manager, who selected the technical equipment and worked with installation manager Kevin Morrison to oversee the project, says it was time for an update because the previous look was 'getting long in the tooth'. The decision to build a new studio means that C4 News now has full access to a facility; previously there were occasions when it could not always get the time it needed.

'The studio has been built using our extensive knowledge of live television news,' Swain says. 'Our aim is to produce bespoke, polished programmes using relatively few operators. But this depends on the kind of programme —if it is a 2-minute summary, then obviously we do not need a large crew. This is why the studio had to be as flexible as possible, so that it could be operated by fewer people if necessary.'

This flexibility extends to having broadly similar, if not the same, equipment in all the complex' studios and control areas, making it easier to bring in freelance crew. 'Technology has moved together more,' says Swain, 'and what we've tried to do is build a core, centralised ITN. Stories come into that central system and are then out-putted along various "arms" to the different departments, where they are presented in the character of the specific programme. C4 quite liked what we did for C5 [a fast-paced bulletin with a presenter perched on a desk or walking around the studio, while journalists work in the background], but wanted a different version. At one time we just used to produce one product, but now we have to be flexible and supply what the customer wants.'

The first criterion that had to be satisfied was to establish a digital infrastructure for video and audio. All new TV facilities are being built with digital video capability, but it is not so commonplace for audio; many studios are sticking with analogue for live production work, arguing that digital mixing consoles are too expensive, unreliable, do not have the necessary headroom and cannot be used in the same way as the old technology. Swain agrees that television is still on the cusp of analogue and digital sound, which led to serious consideration.

'Initially we couldn't decide whether to stay analogue or go digital,' he explains. 'There is no sound desk that can handle serial digital video, which would have meant we would have had to strip out the audio. In weighing up analogue and digital, we looked at the costs and flexibility of the two. We decided that it would be nice to go digital, but that there was a cost premium, so we then weighed up the costs and the features.'

The leading digital consoles were evaluated using these criteria. As ITN had already installed Quantel Clipbox video-editing servers as part of its inte- >

Choosing the right audio Codec.

The Dialog4 MusicTAXI range is one of the most comprehensive codec packages on the market today. It contains all the standard ISO/MPEG audio coding algorithms in common use today such as Layer 2 and Layer 3, as well as CCITT G.722 for high grade voice bandwidth connections, and G.711 so it can talk to a plain old analogue telephone line, too. Connectivity features include upto three ISDN terminal adapters and X.21 port, for operation up to 384kbps. Dialing is quick and easy using the 96 entry directory.

The range of network protocols included means that it can be taken to virtually any part of the world. In the studio the audio i/o can be analogue or digital (AES/EBU & S/PDIF interfaces are both provided). The aux data channel enables embedded control data to be sent alongside the audio, and the unit can be controlled remotely from a PC or the external Remote Panel if desired. Most importantly automatic sensing of the codec at the other end of the call means that it sets itself up to communicate with the most commonly used systems in use today, i.e. Telos Zephyr, CDQPRIMA, Glensound and

others without complicated manual programming. Operationally the buttons are large and straightforward to use, while the illuminated LCD display gives a clear indication of what is going on at all times. No noisy internal cooling fan to worry about in quiet studio conditions. The Remote Panel can control a MusicTAXI from over 500m away via the RS422 interface. The online menu indicates online time, send-level, receive-level, adjusted headroom, Rx

and Tx audio configuration, SYNC flag of MusicTAXI at the other end.
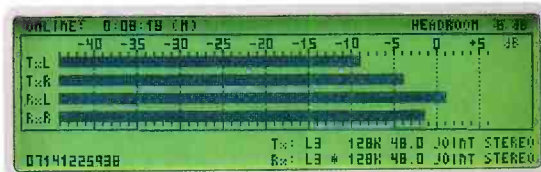
Tapeless recording and transmission on the spot is the answer to the enhanced requirements of correspondents. The CTAXI is the solution and is set to become the standard for mobile recording and transmission, because it satisfies the users demand: stereo recording, editing, file-transmission to computers, realtime-transmission to all well known codecs. The CTAXI is, of course, child's play to operate. You can use it as telephone, walkman, audio recorder, mobile editing station, transmission device. The size is as small as today's cutting

edge technology allows: 58 x 239 x 150 mm, the weight is 1150 g including 2 x Li-ION batteries. The charger is inbuilt and allows uninterrupted operation. PCMCIA flash cards or hard drives can be used for stereo recording. BWF format is supported.

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< grated newsroom system, Swain and his colleagues looked at another Carlton Group member, SSL. 'We have had a long relationship with Carlton,' Swain says, 'but SSL couldn't offer a desk with the capability we wanted at a price we were prepared to pay.' ITN turned to a company that, although established in recording and postproduction, is still behind its competitors in terms of television. 'AMS Neve were kind enough to offer us a good price to get a foothold in the TV market,' Swain adds.

AMS Neve designed the Libra Live as a specific TV digital production console. For C4 News, ITN wanted 36 live operational channels, with enough flexibility to quickly change the desk's functions. 'This is important because we can be doing a live programme one minutes and a studio chat show the next,' says Swain. 'The live aspect was something we looked at very hard because it is our bread and butter—we need to be able to make changes when we want. As part of the evaluation, a panel of four sound engineers looked at the console and had in-depth discussions to ensure that we had made the right decision.'

While saying that the Libra Live offers flexibility, Swain acknowledges that assignability does bring some problems. 'An analogue desk has a knob for every function, the SSL has a rotary encoder, but that puts the price up, so

we decided that we could live with the small restriction of assignability. A permanent channel strip would be nice, but we have two banks of 36-input faders. It makes sense once you get your head around the fact that the desk is capable of doing more.'

ITN had to address the perceived problems of headroom and unreliability in the early stages of the installation, but Swain comments these are no longer issues. 'As long as you're careful about the interfacing, there should be no headroom trouble,' he says. 'Once everything is in the desk, there is enough capacity. The only problems really come with the A–Ds and D–As.' The installation of the control area, a narrow room with a sliding glass door, began in November 1998, with trial runs in December. Swain admits to early problems: 'The desk has never taken us off air, but we have had one or two scares, particularly with software bugs. We specified one or two changes, including a requirement for back-stop prelisten. A major worry was when a software revision introduced a new set of bugs into the system, so we kind of went back one step to cure it. But Neve has been quite attentive in dealing with these situations.'

One of the staff operators admits that the control surface of the Libra is different, but that it is only a case of controls being in different places. 'Assign-

ability is very flexible,' he says, 'but it is a slightly different way of working.' Swain adds, 'When you're on air, you don't think about whether something is analogue or digital, you just instinctively reach for a knob. The technology shouldn't get in the way.'

The control area also features a 360 Systems Instant Replay digital audio playback system and the accompanying Short/Cut editor. While the majority of on-air material will be either live feeds or preprepared packages, there is always last-minute footage that has to go straight out, without postproduction. Some of this can be from overseas broadcasters with foreign language commentary. In such cases, the soundtrack is stripped off and suitable sound effects added as 'mute' visuals are disliked in TV.

Upstream of the Libra Live is a Pro-Bel matrix, whose output is shown on an in-house designed touchscreen system. This enables inputs and outputs to be routed quickly, a necessary function given the amount of live and recorded feeds that come into an on-air news studio. More centralised flexibility will come when the central video servers are implemented, giving the various control areas access to everything and the ability to manipulate it as is needed.

Which, we can hope, is the only kind of manipulation the news goes through. ■

My Lady's Pleasure

The installation of a console at New York's famed Electric Lady Studios signals changing times and changing plans.
Tim Goodyer enjoys a purple patch

WHEN THE TEAM came in to commission Electric Lady's new SSL 9000j console, it had already hosted sessions with Santana and Japanese band The Bees. Worse, the only window the studio could only find for the installation opened at 2am on Friday and closed at noon on Saturday, so the whole exercise could easily have ended in tears. As it turned out, however, as soon as the SSL staff were clear, Diana Ross moved in, followed by Bon Jovi, David Sanborn, Di Angelo, and Naughty by Nature.

The studio ran almost until the close of a session with Ron St Germaine just before Christmas and then the madness began. 'We made it by the skin of our teeth,' confesses studio manager Mary Campbell, happy to reconstruct this short passage in Electric Lady's eventful history with the aid of the studio diary.

'It was a big thing for SSL, too, because it's the first time since it opened that Electric Lady has been an all-SSL facility,' Campbell states with particular reference to the custom purple 9000j already installed in Studio B. 'In the early seventies it was all Neve,' she continues. 'There was an 8078 and two 8068s. First an 8068 went out and an SSL came in, then another 8086 was replaced with a Neve 8128 for a brief period of time. And then in 1988 the 8078 went out and the Focusrite came in—and stayed for ten years.'

The famed Focusrite had been a significant part of Studio A's character, its sound commanding a clientele in its own right, but mounting maintenance problems have seen it recast to support the new 9000.

'The console had its problems with busing and other things, but people were prepared to put up with all kinds of stuff because it was absolutely stunning. And that was part of the reason we kept it too,' Campbell agrees. 'So we had to decide whether we were going to sell it whole or keep it, and we came up with the idea that if we had a 9k with all the Focusrite mic preamps and EQs we'd have a one-of-kind room. Even if someone wanted one just like it, they never could have it. So we're in the process of chopping up the Focusrite and retrofitting it into the outboard racks.'

If the decision to replace the Focusrite was straightforward, the selection of a replacement console was less so—particularly as it involved changing the function of Studio A. With the Focusrite in situ, the room was basically used for tracking and overdubbing. Campbell wanted to make it more mix ready, and the clients and calls seemed to indicate a demand for another 9000j.

'People at the record companies were calling and saying "I need a S-S-L-nine-thou-sand." That was coming up a lot. You can read into it that they don't know what they're asking for, but they wanted to book a 9k. The size of the room was another factor; there is a lack of sizeable rooms with a 9k in New York.'

In further support of her case, Campbell quotes David Sanborn regular practice of recording and tracking at Avatar (formerly Power Station) and mixing at Electric Lady on the Focusrite. 'For this record he worked a lot in LA, then some at Sound on Sound, some at Avatar and then came down to mix in the A room on the 9k. He's an audiophile, he knows what he wants to hear and he was satisfied.

'With this console, 50% of the time will be top-notch mixing and 50% of the time will be tracking with the Focusrite modules, instead of having a room that's run at 60% of capacity throughout the month I wanted it to be 100% either way. There are two studios in New York that have 9ks and they haven't put them in yet—I don't know why. Mine went in in nine days—in and running and booked.'

On the matter of room rate, the newly equipped Studio A has fallen in line with the established rate for Studio B. 'The console itself is pretty pricey,' comments Campbell, 'so you have to set a certain rate in order to accommodate the price of the console and a profit margin. But it seems that it's not as hard a sell as you might expect. Sometimes a client will tell you they'd rather cut their record on an ADAT, but there seems to be a change of aspect where people want top-notch recording. Whether it's because their budgets allow it or whether their personalities want it or whether it's because it's technically correct, I couldn't tell you. There's a lot of rap and R&B artists with a few years success behind them who are booking out the giant room at the Hit Factory with a Pro Tools rig.

'Rates are staying real steady though, and there's enough business to go round. In Nashville, 9000 rooms are about half of what they are in New York even though the same amount of money goes into the control room; the same amount of money goes into the purchase on the desk. Sometimes we get the argument that "I could go to Nashville and get..." And sometimes they do—with airplane flights and accommodation, it's less than working in a New York City 9000 room—but that console really seems to be the ticket.'

If only the landlord's remodelling of the studio's frontage had met with equal enthusiasm... ■

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The **Element 78** series from Summit Audio, designed by Mr. Rupert Neve, presents to the user a unique combination of "Class A," discrete and solid state, plus transformer coupled designs. Digital implementation of storage and reset capabilities enables comparison of 25 memory settings, copying of settings between units and MIDI control.

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Quantum Jumps

Sharing its birthday with *Studio Sound*, Quantegy has been a driving force in the evolution of audio recording. **Carl A Snape** comes up to date on 40 years of tape manufacture

NOW KNOWN as Quantegy Inc, the history of Ampex tape manufacturing goes back 40 years. Ampex itself was founded in 1947 and its first product was the M-100 tape recorder, but it was not the need for analogue tape that initially made Ampex interested in manufacturing tape. It was video. During the early days, while developing its audio recorders, Ampex worked closely with 3M on tape development. In 1956 Ampex introduced its 2-inch video recorder—the VR-1000. In order to improve the performance of the video recording system the company needed to accelerate the tape development programme. The easiest way to do this was to have your own tape plant so Ampex established a small task force to look into the feasibility of building a coating plant near its headquarters in Redwood City, California.

Having found a site and worked out the details the task force presented its findings to the company. The plan was rejected, the company believing that in order to get on line quickly it would be better to buy an existing facility and in fact they had already identified a small plant in Alabama that looked quite suitable. It was a somewhat bitter blow to

those that had worked on the California project, they would soon leave and set up their own facility in California. The name of the fledgling company: Memorex.

The Alabama plant that Ampex bought was owned by John Herbert Orr, an American soldier in the Signal Corp who, at the end of the Second World War, had been involved in a serious traffic accident in Germany. According to Steve Smith, director of marketing at Quantegy, the story, as he understands it, is that Orr spent several months in hospital in Germany and apparently the nurse who looked after him had been a former BASF employee. During the period of his recovery they would talk about tape and tape manufacturing.

When he returned to his home town of Opelika in Alabama, Orr began to see the potential of magnetic tape and in 1952 opened his own tape plant. It was purchased by Ampex and became the Ampex Magnetic Tape Division in 1959. In 1990 the tape plant became a separate subsidiary and was renamed Ampex Recording Media Corp. In November 1995 Ampex sold the plant and it was reborn as Quantegy Inc.

'Unbeknownst to us,' explains Smith, 'one day after we had become known as Quantegy, 3M announced that they were leaving the magnetic tape business. Our initial entré to 3M was to get some capital equipment, plastic moulds, things like that for our moulding machines and also some automation equipment that they had.'

In fact in August 1996, Quantegy ended up buying all 3M's professional tape interests including its formulations and some of the equipment, but it had no use for the actual facilities.

'They are still actually doing some magnetic coating in Kansas and they still do some VHS tape which we purchase from them. All our VHS tape is made by 3M.'

During World War II the Opelika site had been a prisoner of war camp and one of the original buildings—the doctor's surgery—still remains in the centre of the plant, protected, as a building of historical interest. According to Ken Holsenback, staff engineer, Audio Media Products, today the site covers 30 acres and contains half a million square feet of production area and offices. The production areas are spread over several buildings, the main ones being Plant 1 which is used for coating audio products, and Plant 2 which is used for video. There is also a third coating area, that is used specifically for R&D work. Along with the Mix-Prep building, Plant 1 is the earliest part of the facility, and, although it does not have the high tech, automated look of Plant 2, first impressions can be deceptive.

At the heart of Quantegy's tape manufacturing philosophy is consistency. Unlike some other tape manufacturers who strive to produce a better tape everyday—and inevitably there will be good and bad days—one of the main driving forces at Opelika is to keep each batch as consistent as possible. That may not sound quite so glamourous, but it is what professional customers expect and it is every bit as demanding and exacting a challenge, more so in some ways, as trying to produce batch to batch improvements at every opportunity.

'In our analogue product line,' says Smith, 'our customers have said once they identify a sound, a characteristic,

runability—all the parameters analogue tape exhibits on a transport. Once they say that is something they will buy and use, they don't want us to change it. So our goal here, with all our products, is that once we have introduced a product, it doesn't change. 456 was introduced in 1974 and, except for one oxide change, because the oxide supplier stopped making the original oxide, except for that, it is virtually the same as it was 30 years ago. The opportunity was there to make it better, but what we have done is introduce new tapes. Ampex 499 was the upgrade to 456 and now GP9 is the upgrade to 499. We have been doing this about every 10 years.'

Analogue tape consists of three primary components: the base film, the magnetic coating and the back coating. The Mylar (polyester) base film is shipped to the plant on industry standard 660mm (28-inch) wide rolls. The coatings are manufactured on site from a variety of different ingredients depending on the application. The magnetic coating, for example, includes the magnetic oxide, binder, lubricants, whetting agent and solvent, and so on. For analogue tape there can be as many as 26 ingredients. Quantegy actually produces 57 different tape products using 16 basic formulae.

Contrary to popular belief few tape manufacturers actually produce their own magnetic particles. These are normally bought in from specialist suppliers.

The mixing of the ingredients is done in the Mix-Prep building. A critical part of the process is to ensure that the magnetic particles are evenly distributed throughout the binder. Another important consideration is to avoid breaking the magnetic particles as this generates non-magnetic particles, which reduces the sensitivity of the tape or creating clumps of particles, which increases distortion. Milling is the first stage of the process and this is done in large diameter drums containing either glass beads, steel balls or even stone pebbles. This can take up to 72 hours for an analogue tape. Video formulations tend to take less time as the magnetic particles are needle shaped and are therefore more easily damaged. The magnetic coating for video is also produced in individual batches whereas audio tape is treated differently.

Having the consistency of flour, the magnetic particles once mixed with the other ingredients, is stored as a slurry ready to be pumped to the coating area. For audio, typically 400 gallons of mix will be produced at a time and this 400-gallon mix is then added to a 1,500-gallon blend tank. This ensures that any variation, no matter how slight, is offset by the much larger quantity of mix already in the blend tank. This is important for consistency. It takes about a gallon of mix to produce a 2,500ft reel of 2-inch tape.

Studio Sound July 1999

In order to start the coating process the mix has to be pumped from the Mix-Prep area to the coating room (Plant 1 or Plant 2). It is at this stage that the actual catalyst (hardener) is added. One of the first jobs is to apply the back coat. This is done as part of a single continuous coating process in Plant 2, but a separate back coat line is used in Plant 1. Once the back coat has been applied to the base film the magnetic coating is put on the tape. Unusually Quantegy use a knife-coating process for all its analogue tapes.

'I don't think anyone else in the world, whether it's a small coating line in China making cassette duplicating tape or a brand new DVC video line—no one in the world uses a knife coating line,' says Smith. And the reason Quantegy uses a knife coater? 'We believe, sonically it makes the best sounding tape. It is the lowest technology, the oldest technology, but for the very thick coated analogue tapes it does it better than any other kind of coater. If you went to a coating >

Top right. 1: One of Quantegy's Audio Test and Evaluation laboratories, showing DASH and a selection of analogue open-reel recorders. 2: 'Jumbo' tape storage area. 3: The pebble media used in the ball mill. 4: An ariel view of the Quantegy factory in Opelika, Alabama, US. Top left: The horizontal ball mill.

< symposium and you said you used a knife coater, people would be laughing because it is 40-year old technology. We have tried making our analogue tape on reverse roll gravure and slot die coaters, but they don't sound as good.'

So why has everyone dispensed with knife coating? It's an unsophisticated, manual process. A knife edge sits across the base film. The height at which it sits determines the coating thickness. When the system is running a puddle of magnetic coating collects behind the blade and it is the operator's job to ensure the puddle size remains consistent. One explanation as to why this may produce a better sounding tape is the fact that the mix really does flow onto the backing film rather than beginning squeezed onto the film as in the case of the more modern coaters.

Once the mix has been applied, the magnetic domains in the coating need to be aligned. This is done by passing the through a strong magnetic field that aligns the North and South poles of the magnetic particles down the length of the jumbo tape. Once this has been done the coating is dried by running the tape through a series of large ovens that vary in temperature from 150°F to 100°F. The length of these drying lines, and the speed at which the freshly coated tape travels, varies according to the product (audio or video) being made. Oven lengths vary between 200ft–500ft and the tape travels between 200ft–800ft per minute.

What is remarkable is that throughout the drying process the tape does not touch anything, but is suspended on a cushion of air as it travels the length of the oven.

Once dried the surface of the magnetic coating is calendered. This is done by passing the tape between two rollers, one made from stainless steel and with
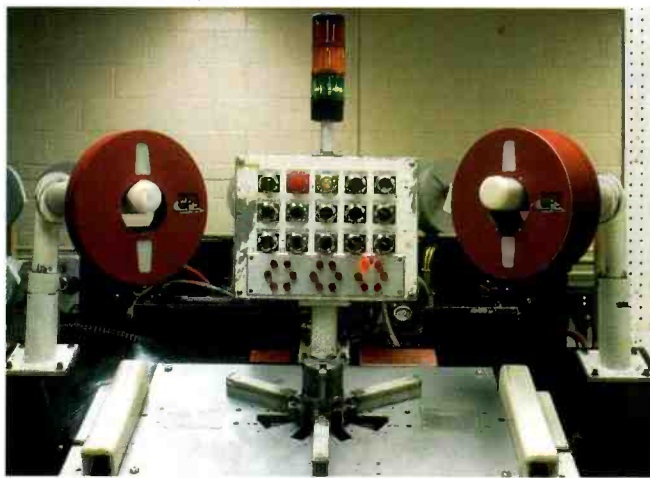
Tape reel assembly unit, showing flanges for the Quantegy GP9 analogue tape

a mirror like finish, and the other made of paper or plastic with a measured amount of give in it. This literally irons the surface of the tape forcing the binder that has risen to the surface during the drying process back into the magnetic coating and imparting the characteristic gloss like finish to the recording side of the tape.

Although I said earlier that the first stage of the coating process is to apply the back coat, 478 is an exception. 478's back coat is very rough (this is to allow the air to escape quickly during winding so as to give even packing) and the roughness of the backing can imprint itself on the calendering process so the back coat is applied after the tape has been calendered.

At this point the jumbo reels are now ready for slitting. This creates individual reels of tape (known as pancakes) in one or the other familiar tape widths (2-inch, 1-inch, and so on). After slitting the pancakes are burnished using silicon carbide wipes. This effectively removes any slitting debris and ensures the tape is physically clean. Each pancake is then tested, in some cases throughout its entire length. The appropriate test chart then put in an envelope and included with each reel of tape once the pancake has been wound onto

individual reels and cut to the appropriate length.

When it comes to professional products quality and reliable performance are clearly important issues and Quantegy has extensive test and measurement facilities to ensure the appropriate standard is maintained. The facilities include not only detailed chemical and physical analysis, but also listening tests and investigations at the submicron level with various atomic microscopes. In the Microscopy Lab, for example, it is possible to examine the smallest contaminants, things that could cause a dropout, and identify the nature and hopefully the source of the problem. Another area of interest is the surface roughness of the magnetic coating as this can affect head-to-tape contact and the frictional properties of the tape. In a lot of ways the Physical Lab is a torture chamber for tape, and, although Quantegy do not have a rack or thumbscrews they have the high-tech equivalents. It is here that production and R&D samples are heated, twisted, bent and pulled apart in order to discover the physical limitations of the base film and equally important the magnetic coating itself. Given that some videotapes have three separate coatings and in some cases have to negotiate a fairly tight and complex tape path makes these kind of tests so crucial. And if you are recording at the North Pole or in the Sahara then the conditions are going to be even more demanding. The ultimate test for tape, however, is probably the Galileo space probe. Way up there, in the most extreme of conditions, there's Quantegy tape happily recording what's going on at the edge of the galaxy.

Back in Opelika, a lot of the work done by the Chemistry Lab is concerned with quality control, both of incoming materials and during the >

< manufacturing process itself. The staff is also involved in research work as well. One of the key responsibilities is to ensure that the correct amount of catalyst is being added to the magnetic coating. With as many as 10 different mixes being use on a shift this has to be constantly checked. Another important criteria is the mix viscosity. This has to be carefully monitored and it is also critical that the mix ingredients are probably dispersed. All these things are constantly being checked.

'Dry' end of tape coating line showing coated 'jumbo' rolls of tape

But even when all the scientific testing has been done it is in the studio where the ultimate test lies. What happens when a reel of tape is put on a tape machine? In order to find out Quantegy has its own well-equipped audio testing facility with an impressive collection of recording machines. These include a Studer A800 and A80; a couple of Ampex ATR-100s; Nagra-D and IVS; Otari MTR-90 (Series 2 and Series 3); Ampex MM 1200; two Sony 3340 digital multitracks; a Panasonic SV3800 and a couple of Sony DAT machines; Tascam DA-88; Alesis ADAT recorder and finally a Sony BBU800/1610 CD mastering setup.

The first professional audio tape made by Ampex was simply known as Ampex A. This was replaced by 406 in 1968 which went on to become one of the best selling tapes of its time, however 406 did have some oxide shedding problems particularly with the larger widths (1-inch and 2-inch). The problem was addressed when Ampex introduced its 456 Grand Master formulation.

'456 was much more durable,' explains Smith, 'and it had this really "thumpy" bottom end that other tapes didn't offer. We believe this bottom end on 456, and 499 and GP9, is due to the knife coater.'

Like 3M's 996, Ampex' 499 Grand Master Gold could be pushed to high levels, but some engineers preferred 996. When Quantegy bought 3M some people wondered why the company did not continue with the tape.

'996 had a large following, especially in the UK, because it sounded different to 499. I could come into the factory and say, "Make this" and they can look at it and try to get close to it, but that's it. I can't come in and say I was talking to Gus Dudgeon or someone and he said the tape had a real airy quality, but the bottom end was fat and the high end was bright and twinkly. These guys are just going to look at me and say, "What, are you nuts? What does that mean?"'

In fact Quantegy did try to manufacture 996, but it is in the nature of manufacturing magnetic tape, that even with the original recipe books, and after spending a lot of money, and working on the project for something like 18 months, the essential sound of 996 still remained elusive.

'The outshoot of that was 3M had some good things in their recipes so we said why don't we see if we could get the best of both worlds. Use our manufacturing capabilities—knowledge of the industry—and the 3M recipes and see what we can get out of it. The result was GP9 which we introduced about six months ago.'

Although some tape companies constantly push new formats and high-tech developments Quantegy is much more choosy about what it manufacturers and for good reason. The company has experimented with metal evaporated techniques, for example, but the equipment has now been dismantled and sold off as Smith explains.

'Obviously Sony recognised metal evaporated as an opportunity. We see

the newer technology of dual coating as much more of an opportunity for some of the newer products such as DLT 7000, AIT or DVC Pro. Most of these are dual coated—ME hasn't really taken off.'

Smith also raises another important point. No longer part of Ampex, the company is no longer involved in initiating or promoting the introduction of new recording formats.

'Obviously Sony has excellent insight into that because they produce both tapes and machines. Fuji works very closely with Panasonic. I would say Maxell is in the same situation as us, and BASF is that way also. BASF is the world's largest coater of magnetic media. They make a huge amount of VHS and cassette duplicator tape. We don't have these product lines, so from a coating standpoint, among the professional companies, we are definitely the smallest. So when we decide to make a new format the format has to be successful.'

Quite a number of tape manufacturers, however, have got involved in making recordable optical media like CD-R, but Quantegy has made a deliberate decision not to go down this road.

'Optical is dye technology and that is quite different technology to what we currently have. We kinda pick our shots carefully and we are definitely not at the leading edge of technology innovation here. The markets has to get fairly large before we have an opportunity to get into them.'

Sooner or later, with the growing use of digital recording techniques the question has to be asked: How long will professional quality analogue tape be available? Will there come a point in time when the amount of tape being bought is just too uneconomical to produce?

'That won't be our decision; it will be the oxide manufacturer's. They are the ones that have the economy of scale. We can make one reel of tape a day. We could coat a jumbo a month and that probably wouldn't cost us a whole lot. And we are not looking to rip out Plant 1 and put two new metal evaporated coating lines in or anything like that. We have all the capability and capacity we need for video in our new plant. So the day after we make our last reel of analogue tape we will probably just put a big lock on the door and say, it's finished.'

For the time being at least, globally, there is still a big demand for oxide. Cassette duplication tape is still a huge user of analogue oxide, and so to is VHS, although this uses a slightly different type of magnetic particle and while studios may be using less analogue tape there is still a large demand for it in other areas of professional recording.

'The business was very large and con-

tinues to be quite large. Some of the bigger users of analogue tape include Third World radio. Ethiopian Radio, Indian Radio and Turkish Radio—these are all customers of ours and they use a lot of tape. So if a professional user says, "Gee, I used to use 200 reels of 2-inch a month and now I'm using a 100 or 75; your factory must be closed half the time". Well that's not true. Since 3M left the market we are making as much analogue tape as we ever have. But now the analogue market is declining. There is no question of that. So I can't say there is growth or anything, but our business hasn't declined that much and looking to the future I would say it is good for at least 10 years. We have just completed a 7-year plan and there is a lot of analogue business in that. So if you are using an analogue machine and you want analogue tape you can at least plan on 7 years.' ■

View of a tape slitter showing individual slit strands of tape.
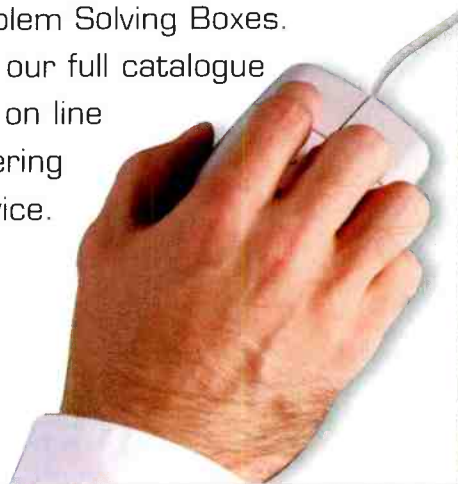
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US: Do nothing and die

The Internet is changing the face of the recording studio in ways never envisaged by recordists or netheads writes **Dan Daley**

IT WAS A TYPICAL David Bowie session, in New York—musicians and related personnel milling about, awaiting the Androgynous One's imminent arrival, catering nicely laid out on the lounge countertop, musicians tuning up. All in all, another quotidian day in the life of a recording studio. Except that this was not your run-of-the-mill session. The presence of a CNN video crew gave it away, as did the bearded, tie-dyed bunch manning a small flotilla of laptop computers in the edit suite. What was going on that day at Looking Glass Studios on lower Broadway in Manhattan was what the studio industry in the US —and increasingly, the rest of the world —is about to resemble.

David Bowie was about to make a netcast, in this case a recording session, but one monitored by an 8-lensed netcam situated in the rear of the control room, whose operation could be controlled by the millions of computer mice wielded by anyone who tapped into www.davidbowie.com that evening. It's still something of a novelty, as evidenced by the news crews and print journalists there to report the event. But the day is not far off when something like this won't be terribly newsworthy at all. It will be the day-to-day reality of a studio business on its way to a very different future.

What makes this transition so interesting is how smoothly it's taking place; only in retrospect will it appear revolutionary. The Wright Brothers' 1903 flight at Kitty Hawk was but a novelty at the time, and aeroplanes were still just that even as the guns of August 1914 began firing. But once a Frenchman realised that one could just as easily drop a hand grenade over the side of his scout plane and do some damage to the enemy, everyone was suddenly tripping over themselves to assemble entire air forces. The Internet has experienced the same effect; it's fun to go surfing along the Net, but once it was figured out how to make some real money at it, watch out. And that stampede is now hitting the studio business.

Though it is co-owned by Philip Glass, Looking Glass itself is a fairly conventional studio—two control rooms and a small digital editing suite. But Bowie didn't choose Looking Glass for its conventional aspects for the purposes of the netcast. It was the facility's extensive T1 and ISDN telephone lines, and a staff well-versed in Internet operations, that made it appropriate on this occasion. Those have been in use for some time there and at other studios for technologies like DolbyFax and other high-bandwidth audio telecommunications systems. In fact, the session was something of a sideshow —it was the payoff in a promotion to build the popularity of the artist's net-site, in which fans emailed in lyrics to which Bowie wrote music to those of the winner. Though the netcast was staged to make it appear that it was a complete recording session, the basic tracks had been previously recorded. Only the vocals would be recording during the event. In a sense, much of the technology, like the console and tape machines and monitors, that older studio choices were based upon, were irrelevant to this session. To put it bluntly, with one of the audio streams going out at 28.8kps with 22kHz of frequency response, two Dixie cups and a piece of string were almost a match for the SSL E-G console in Studio A.

That underscores the point that what will make studios economically successful and viable in the future will be entirely new technologies and areas of expertise. This is not solely Internet-driven, either. Not by a long shot. Rather, it is propelled by the changing economic environment in which studios exist. Pressured by artist-engineer-musician-based project studios and a constricted rate structure, conventional studios have to become unconventional in terms of the services they offer the entertainment industry. And in the process of learning to do that, they will reconfigure the industry as we know it today.

Europe: Redefining CD

Sony's SA-CD standard challenges DVD for the attention of the serious audio consumer, but can it be made to deliver, asks **Barry Fox**

AT THE END of May, Sony launched SA-CD in Japan as a $5,000 stereo-only system, with the promise of surround in the future. At the Technics annual European seminar—held in Salzburg, Austria at the same time as Sony's launch—Matsushita (Panasonic-Technics) confirmed that it will launch the rival and incompatible DVD-Audio system worldwide before the end of the year. Matsushita also unveiled a whole range of 'DVD-Audio ready' amplifiers and loudspeakers.

Audiophiles can already get 96kHz and 24-bit stereo from ordinary DVD and Dolby AC-3 or DTS multichannel surround. It is going to be a hard job selling a system that offers bitstream stereo (SA-CD) or 192kHz/24-bit surround (DVD-Audio).

Sony has only limited support from the software industry, mainly from its own music company, Sony Music. Technics claims wider support and Peter Olliff, ex-PolyGram, now with Universal, was there to promise titles on DVD-Audio. But the man from Warner, Olivier Goulon (MD Ancillary Product with Warner Music Europe), never turned up to fly his flag. Nor did he send anyone in his place. I'd like to be a fly on the wall when Warner's Mr DVD, Warren Lieberfarb hears about that.

The SA-CD system has been sold on the promise of hybrid discs that can be played either on the 600 million CD players already in use, or new SA-CD players. Philips and Sony have dubbed hybrids a 'defining attribute' of their system. But when Sony launched SA-CD in Japan in late May, Sony's own record company did not support the launch with hybrids.

The official line is that while Sony regrets this, Sony's music division is independent and can do what it likes. And as the player sells for $5,000 the audiophiles who buy it are unlikely to worry about CD playback.

David Walstra, General Manager of Sony Europe, is confident that when SA-CD launches in the US, there will be hybrids. He is also confident that they will play either on legacy CD players or new SA-CD hardware. But at the AES in Munich, doubts were already surfacing.

Set aside the issue of cost, and whether the record companies want to spend extra on hybrid pressing when so few people have SA-CD players, and the market is stifled by the format war with DVD-Audio. Set aside the issue of whether artists will ask for higher royalties if one disc carries two different versions of the same music and thus deters people from buying CD pressings now and SA-CDs later. Set aside the issue of whether the consumer will be interested in paying extra for a CD simply because they might one day want to play it on a SA-CD player. Consider just the technology of hybrids and how reliable it will be outside the labs and out in the real world.

The hybrid disc stores the two recordings at different depths in the surface. The lower layer, beneath 1.2mm of clear plastic, is a conventional CD recording to the so-called Red Book standard for CD. Another layer, at 0.6mm, conforms to the new Scarlet Book standard and is made of

> **Ted Abe head of Matsushita's Audio Technology Group:'Our tests suggest that 30% of legacy players will not be able to play hybrid discs'**

semi-reflective material like a 2-way mirror. This layer carries the Direct Stream Digital recording used for SA-CD.

The laser optics in a standard CD player or ROM drive routinely focus at 1.2mm, and should ignore the semi-reflective layer of a hybrid disc. A DVD-Video player can focus at either depth, but is not equipped to decode the SA-CD recording so it should ignore the semi-reflective layer, focus down to 1.2mm, and play the disc as a conventional music CD. But if the laser in

In Nashville, where studios have been under intense economic pressures for over two years, I've watched the successful ones base their survival on new services, such as the broadcast access service that Emerald Recording developed in which record labels can use the audio facilities with an interface that allows their artists to give real-time interviews on radio stations anywhere in the world. Starstruck Studios implemented something similar, and are in the process of expanding that to video for television broadcasts. These are services that personal studios can't offer, and for which dedicated broadcast facilities would charge exorbitant amounts of money for. But with the right balance of technologies, expertise and a certain amount of vision and willingness to adapt, they are services to which that audio studios are perfectly suited.

This is not the old paradigm of adding some cassette duplication decks in an unused closet. This is a fundamental realignment and redefinition of what a studio does as its core competency. It's a fine line between evolution and revolution. But both are going on under our feet at this very moment. The trick is going to be figuring out, on a case-by-case basis, what new technologies and endeavours will work the best for each studio and each market. Change is unsettling, but it's also exciting. There are, in a sense, no wrong answers to the question of whiter the studio of the future, save one: doing nothing.

a CD player or ROM drive detects the semi-reflective layer, either because the player is cheap or old, or new and designed to read erasable CD-RWs discs which have lower reflectivity, the player will reject the disc as unplayable. DVD players may not switch to the lower CD layer when they fail to decode the DSD signal.

Ted Abe head of Matsushita's Audio Technology Group, and his engineers have made hybrid discs and tested them on a wide range of real-world players. 'We have very serious concerns about backward compatibility of hybrid discs' says Abe. 'Our tests suggest that 30% of legacy players will not be able to play them. It's impossible to predict which ones will reject hybrids and it's impossible to do anything about it when they do fail. We stopped research on hybrid discs and said No Way.'

David Walstra assures 'We are 100% certain that hybrid discs will play on players if they conform to the Red Book CD standard. We have checked our own DVD players and they play hybrid discs. We can't speak for other manufacturers'.

Payl Reynolds, Philips' director of new business development, says the company is now starting pilot production of hybrid discs at Eindhoven in the Netherlands to 'perfect the technology'.

Ted Abe, warns that many real world players do not exactly match the Red Book standard, but play most CDs. Their owners will make a lot of noise if they buy discs that do not play.

Decaying orbit

Rupert Murdoch's satellite lovers may be enchanted by spin, but his digital lures for terrestrials seem ill-prepared writes **Kevin Hilton**

NOTHING LASTS FOREVER. The reality is that the really good times are only around for a relatively short period. I wonder if somebody has told Rupert Murdoch that? Probably not, because, in our money and power obsessed world, the majority of people would consider it unthinkable that a rich, powerful figure could suddenly start to look vulnerable and weak. But it can happen.

The media is no different to anything else and so has been pulled and shaped by trends, times and, more recently, technology. The all-powerful press barons, who dominated newspapers in the 1940s and 1950s, slowly gave way to a new breed of entrepreneur, most obviously embodied in Murdoch. While other media groups are exactly that—faceless amalgams of cash and expertise—Murdoch leads from the front. There may be chief executives running the various branches of his empire but, to the public at large, Murdoch is News International/Corp and the Fox Network. This was reinforced by his cameo on *The Simpsons*, playing up to his megalomaniac image.

Now 68 years old, Murdoch has begun to address the question of his successor, as he contemplates retirement and old age with his new and young wife. His children are circling conspicuously, looking at each other, trying to work out who is going to ultimately take charge of the empire. They could be disappointed. In April, Mark Booth moved from his post as chief executive of BSkyB to head up News International's fledgling new media division, epartners. It is understood that Booth did not want to stay with the satellite broadcaster and had been approached by a 'major Internet provider'. Murdoch did not want to lose him and so engineered the move. No replacement has been announced but it was made clear that it would be someone from beyond the existing company structure, a snub to Murdoch's daughter, Elizabeth.

News Corp is putting $300 million into epartners, which will count the Internet, interactive television and wireless communication within its remit. Murdoch made his first foray onto the Internet through the British tabloid The Sun and in June announced that BSkyB was joining the growing number of free Net service providers in the UK, the first broadcaster in the country to do so. This move, and the founding of epartners, is obviously part of the growing convergence between traditional media and new forms; arguably these moves were also prompted by Rupe's apparent paranoia over the growing influence of Bill Gates and Microsoft.

Murdoch showed himself to be master of new publishing technology during the 1980s when he moved *The Times* and *The Sun* out of Fleet Street and into new, computer-based premises in East London. With broadcasting, he has revealed himself to be less canny. In the early satellite TV wars, Murdoch adopted the existing PAL standard for Sky Television. This undoubtedly gave him the advantage over BSB, which was courageously attempting to use new technologies (MAC and the Squarial) for a new age of broadcasting.

The price was to be absorbed by its rival. And now there is digital. Sky Digital's launch was delayed and there are signs that all is not well. Murdoch has successfully used price-cutting in the newspaper business and is again applying tricks from one discipline to another: in May, Sky began giving away set-top boxes in an effort to attract more subscribers. Those signing up will pay only a £40 installation fee, but service charges have risen by as much as £2 per channel in some cases.

On a more worrying level, Sky insists that the decoders be connected to the subscriber's telephone line. Many people seem unaware of this and there is speculation as to what information Sky is gathering when its computers connect up with the set-top boxes. When London entertainment magazine *Time Out* voiced these fears, Tim Allan, Sky's director of corporate communications, repudiated any 'sinister' undertones, saying that only one call a month is made, purely for billing purposes. Perhaps it is only coincidence that *The X-Files* is one of Sky's biggest successes…

Immediately after the free decoder offer was announced, Sky's share price rose by nearly 5%. It fell back again the following day. ONdigital, the UK's digital terrestrial service, soon announced its own special deal, but this did not help its already falling stock. The real winner in this is set-top box manufacturer Pace, although there must be the realisation that the market will change dramatically when affordable integrated sets start to appear later this year.

Sky Digital says it is on course for its target of 1 million subscribers, while ONdigital recently announced that 110,000 people had signed on since its began in November. However, it could be that a purely digital version of television as we now know it is not the future. The US has indicated that the real advance is not more channels, but high definition; HD is not a priority in Europe but interactivity could replace it as the extra component that truly makes a difference. And, despite epartners, it could prove that Rupert Murdoch is a creature of the analogue age who gives way to yet another breed of media overlord.

> The US has indicated that the real advance is not more channels, but high definition

Compress to conquer

MP3 champions the Internet and threatens the copyright of the record companies.
Ulrike Godet discusses the file format set to conquer the Internet and broadcasting

MP3 HAS BECOME one of the search-words most used on the Internet. Furthermore, its existence is a cause of much anxiety to record business executives. What began as a technology simply intended to make economical use of the telephone network has become a major threat to the operation of big entertainment companies. MP3 is about to transform the way music is sold, and to create alternative opportunities for musicians making their debut on the Net.

MP3—a file format based on MPEG-3 coding—offers a reduction in the data transmission rate of audio files. With it, you can listen to live events via the Net or download a song to hard disk. With current transmission bandwidths and storage constraints, and without some form of data compression, audio transmission is slow and its storage requirements too greedy. One minute of 44.1kHz, 16-bit 'CD quality' audio requires 44,100 samples/s x 2 bytes/sample x 2 channels x 60 s/min—around 10Mb of storage space. Using a 28.8kHz modem, it will take about 50 minutes to download a single minute of music. Clearly this is not going to be attractive to music buyers.

What troubles record companies today, was once a major achievement for the engineers and researchers of the Moving Pictures Experts Group (MPEG). It required extensive research, testing, and endurance, involving engineers from Fraunhofer Institute along others from the Friedrich Alexander University in Erlangen working in the field of high-quality audio coding. First intended to transmit speech via the existing 'copper-based' telephone network, the work on AC3 gained pace with the introduction of ISDN. The aim became to transmit not only speech but also music via phone line or other channels, where you have to be economic with data rate.

The first real-time stereo encoding-decoding system was implemented in 1987 to deliver the highest practical quality (for lower bit-rates) to the first MPEG test. Based on collaborative work with AT&T, Thomson Multimedia, CNET, it was completed in 1990.

The decisive breakthrough for audio compression came about because of discoveries in psychoacoustics—the field of research describing the human auditory system. The key issue was to identify the frequencies that could be heard and the perception created when listening to beating sounds. The theory being that signals below the audibility threshold (those outside the perceptible frequency range) are superfluous and may be left out as irrelevant redundant data. Addressing the perception of sound by the human ear is the reason these techniques are termed perceptual audio coding.

The MPEG-family of audio coding schemes shrinks the original volume of data from a CD by factor of up to 12 while still preserving audio quality. Higher factors of reduction that are much better than reducing sampling rate or resolution of samples are also possible. Depending on the algorithms one will get different ratios. Keeping CD-quality the different 'Layers' gain different compression factors: Layer-1: 1:4, Layer-2: 1:8 up to 1:10 and Layer 3: 1:10 up to 1:12.

Based on the application of MPEG Layer-3 (MP3), the table shows several data rates and the resultant quality:

Data Rate	Sounds like	Channel
8kbps	Telephone	mono
32kbps	AM-quality	mono
64kbps	FM-quality	stereo
96kbps	nearly CD	stereo
128kbps	CD	stereo
256kbps	Studio	stereo

Source Fraunhofer-Institute for Integrated Circuits IIS http://www.iis.fhg.de

High quality audio-coding gave the market of audio-compressing impetus, and a lot of new players and companies appeared. Some used the MPEG algorithm, others used proprietary methods. There are several dominant audio codecs today within two areas of application: streaming or downloading audio files. These include WinPlay3 (Fraunhofer); Audioactive (Telos); RealPlayer (Real Networks); MediaPlayer (Microsoft); and Quicktime 4.0 (Apple). They all enable you to store, edit or arrange the audio data, but players for streaming work at lower bit-rates due to the narrow transmission capacity of the Net and the modem in use.

An evaluation test of the European Broadcasting Union, done at the Institute for Broadcasting Technology IRT (Institut für Rundfunktechnik) in Munich identifies a direct correlation between audio quality and data rate. Best results were delivered by one ISDN channel with 64kbps. The quality level of MPEG Layer-2 and MPEG Layer-3 (MP3) is comparable with good FM radio, no artefacts were only noticed with Layer-3 coded signals at that bit rate. If you want to get CD-quality, the encoder has to work at a bit rate of 128kbps (2 ISDN channels). The evaluation showed that the auditory impression was 'very annoying' when transmitted via a 14.4kbps analogue modem, but improved significantly at the data rate of an 28.8kbps modem. MP3 and Advanced Audio Coding AAC, were also devised by engineers of Fraunhofer IIS, these were the only ones that sounded better than AM radio at that transmission speed

The ongoing work of international standardisation at MPEG is leading to a better prospectus in the field of streaming audio. In particular, MPEG-4 audio provides excellent audio quality at extremely low bit-rates, for example with AAC. On the other hand the increasing capacity of the network promises better interaction of digital content with the transmission speed of the Net.

Today, broadcasting goes on-line. This new possibility has a many benefits, for producers and for the audience. It is a new medium for journalists and programme designers. Editing and production can work in different places and are not dependant on each other. News programmes can be distributed immediately at a lower cost compared to terrestrial, or cable and satellite broadcasting technology. That is mainly relevant for customised programme content or smaller radio stations. Information can be tailored to individual and specialist interests.

Transfer of audio data via the Net has pertinently gained importance in the last few years because of this emerging commercial potential. It is widely believed by experts that revenues of music and sound storage media distributed by the Net will dramatically increase while the turnover forecast of traditional music stores market will weaken. According to a survey of the GWU's WWW Surveying Team at the College of Computing at Georgia Institute of Technology, Atlanta, US, 50%–60% of intermediate and experienced Internet users are going on-line once a month or more >

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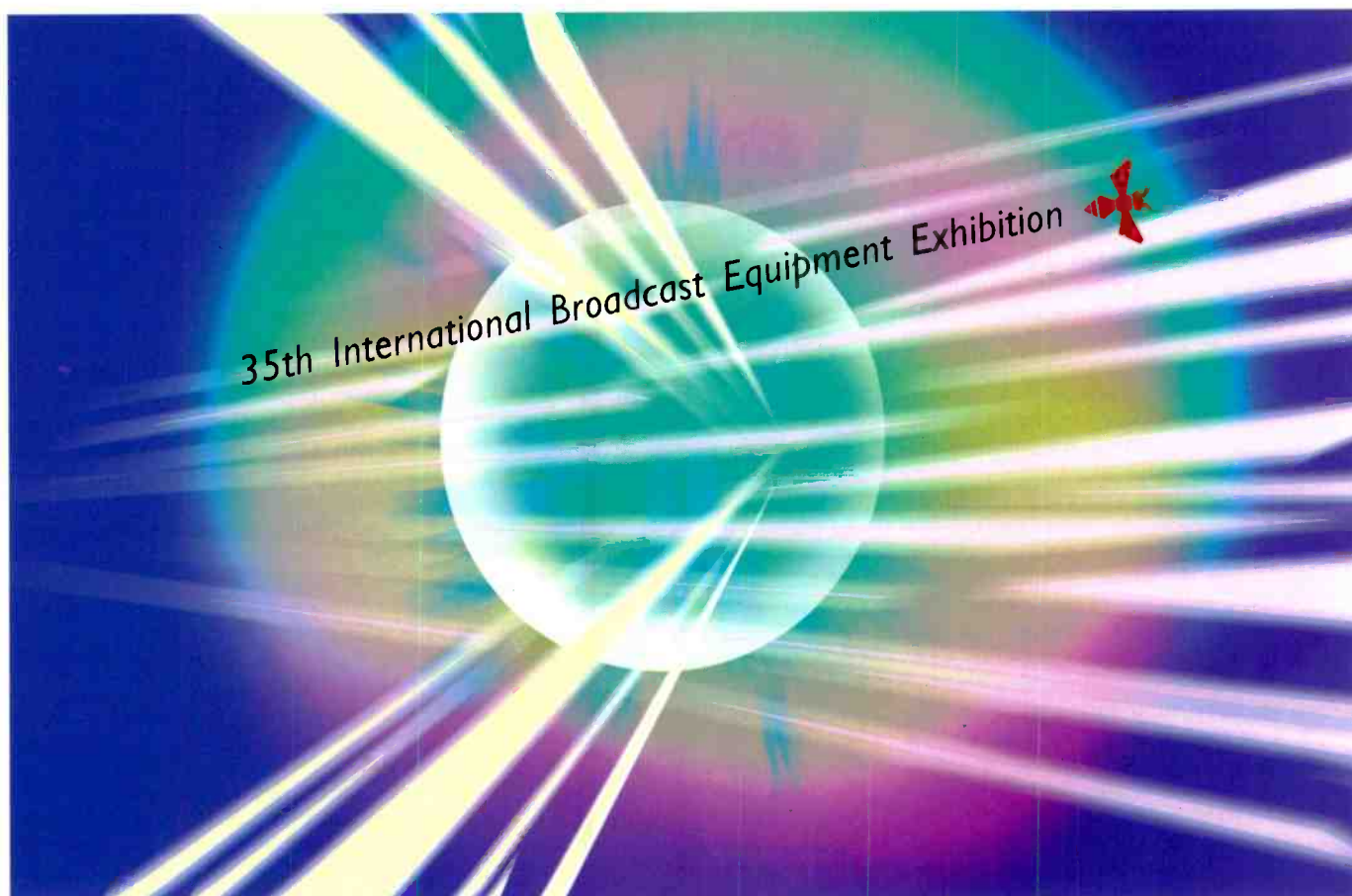
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< with an intent to buy, both in the United States and in Europe. They are aged under 26 years and even older ones are active. Music is among the top five products purchased on-line. Other leading products include software and hardware, followed by books. The growing number of Newbies on the Net is increasing all the time.

The biggest expenses for the record industries are in production—pressing the data on to the discs—marketing and distribution. These companies also invest huge sums of money to develop their stars and promote them. When an artist signs a completion contract with his label, more money is spent. Overhead costs make music expensive to distribute. New business models, such as Music-on-Demand, Net Shopping or Internet R@dio liven things up and befit the changing, youngster target group growing up in a purely digital world.

MP3 is the big thing here—the darling of the public. Compared to other audio codecs, MP3 promises viability and profit, it preserves sound quality and Net-users will quickly get their tunes.

Even so the record business is a tad jittery about the infringements of copyrights and royalties, but Michael Robertson of MP3.com looks into the future with less concern. From his net-site www.mp3.com 200,000 visitors download songs daily from more than 10,000 artist—and for free. There are no illicit copies. This business runs on his own concept: either download or buy CDs. MP3.com stimulates the sales and the artists keep their rights while getting 50% of the proceeds. In addition the company is building up a database, keeping track of the fans and where they live—useful for successful live-events. Robertson talked in an interview with the German lifestyle and computer magazine *Konr@d* about the advantages for artists and the business promoting them: 'Why shouldn't a record company invest $20,000 first (instead of $2million) and wait to see whether a junior musician will succeed on the Net? They

Karlheinz Brandenburg: 'People are smart at finding ways to trade music without being detected. What is needed is a convenient, competitive and legal way of electronic music distribution'

can ask who of their artists are successful and worth investing in.' The attractiveness of MP3 is no longer a matter for newcomers to the music scene. Since stars and divas have also made deals with MP3.com, it has got a little more mainstream. The most recent news is of the live recordings from Alanis Morisette and Tori Amos now available for free on the MP3 net-site.

A major contributor to MP3, Karlheinz Brandenburg, discussed the task of protecting intellectual property rights with journalists at the last AES Convention in Munich: 'We think that MP3 pirating cannot be overcome just by hunting down illegal copies of music on the Internet. Audio coding cannot be "uninvented" and people are smart at finding ways to trade music without being detected. What is needed is a convenient, competitive and legal way of electronic music distribution.'

The Universal Music Group, one of the biggest music companies began by reacting to the force of the market, as the American media announced at the beginning of May. Universal Music Group is probably setting trends with its recent decision to establish a piracy fighting digital distribution system. Testing installations have begun in co-operation with the InterTrust Technologies Corporation, based in Sunnyvale, California, that supports electronic commerce and digital rights management systems. Together they are setting new rules for on-line music purchase. The device is embedded in the audio file and

allows consumers to listen to the music twice for free, but they must then decide whether to buy it or not.

For more mobility and flexibility in using MP3 files you are not confined to your PC. New portable gadgets are about to shake up the market. They look like Walkmans and are called Rio, MPMan, MPLayer3 or Yepp. These devices read the music data directly from a stamp-size chip that can be recorded on over and over again, and there are no mechanical parts to reduce the listening quality. Joggers will now be able to enjoy their music in CD quality without vibrations causing tracks to skip.

These solid-state players either work with an integrated chip or flash-ROM-memory cards, or both, at a up to 64Mb of capacity. The music data—compressed MP3 files—are transferred by a PC onto the memory chip from personal archives or CDs via a serial interface. Selection and sequence of your personal favourite tunes can be varied at choice.

In professional audio there is the the Flashman, a solid-state recorder with no need for revolving tapes or disks, it uses CompactFlash Cards with a storage capacity of 96Mb and are on the market now . This means 1½ hours of high quality music in MP3 format. It is only a matter of time before the capacity of these carriers increases further. Flashman is also the result of collaborative work between the Corporate Computer Systems CCS Europe and Fraunhofer IIS. The Flashman will enable journalists to record all important broadcast formats such as MPEG Layer-2 and Layer-3 as well as using linear, uncompressed, recording. This direct access to the recorded audio will save valuable time. But data must not be loaded onto the PC or laptop in the way as DAT recorders—five minutes of music will not take five minutes to transfer.

It remains to be seen whether a multifunctional piece of equipment such as the Flashman will change the professionals market when introduced later this year. ■

Time code

Last month we looked at how the different frame rates used in time code arose. Here in part two **John Watkinson** continues with the details of how time code actually works

THE FORM that time-code signals and recordings take today was moulded by the need to record the signal on a conventional analogue audio track. Thus, although the information in the signal is fundamentally digital, it had to be carried in a waveform that was capable of passing through analogue audio circuitry, cabling and media. A further issue was the requirement to read time code at shuttle speed in either direction. This is necessary to facilitate editing or chase synchronisation. Almost any type of signal would do if operation at play speed was the only requirement, but running at variable speed and in reverse needs more thought.

The characteristics of analogue audio recorders and cabling make it most unlikely that any DC response exists. Analogue tape heads, balanced line transformers, coupling capacitors, and so on, all act as high-pass filters blocking DC. In audio this is actually a good idea since DC offsets cause thumps at edit points, and needless dissipation and offsets in speakers.

Fig.1a shows that a pure binary signalling system is of no use because different bit patterns result in different DC components. Fig.1b shows that pure binary does not work in a timing sense either. Adjacent bits can be identical, and when serialised there is no change to signify the occurrence of a new bit. There might be a way around this at constant speed, but at variable speed this becomes impossible.

The solution to these problems is the adoption of a modulation scheme known as a channel code. In general, channel codes are designed to get real data down real channels despite the shortcomings of those channels. The channel code used for time code is variously known as Biphase, Biphase Mark Code (BMC), Manchester Code (so called because it was raining when it was developed), or Frequency Modulation (FM).

Fig.2 shows how FM works. The information content of the FM waveform has nothing to do with the voltage. Instead information is carried in the times at which the voltage changes from one state to another. These changes are known as transitions. The existence of a transition is all that matters, the polarity is irrelevant. As a result, a time-code signal can be inverted or polarity reversed without causing a problem.

Each bit is allocated a transmission period known as a bit cell and there must always be a transition between adjacent cells. This maintains clock content in the signal irrespective of the data content. In the centre of the bit cell a further transition may be inserted if the data bit is a one, or omitted if it is a zero.

Fig.2 shows that the result of this simple coding rule is that the signal has a rich clock content, which is useful in variable speed applications. It is also DC-free. This should be clear because the waveform for a single one is half positive and half negative and the same is true for a pair of successive zeros. Fig.2b shows that the lowest frequency results on a string of 0s and the highest results from a string of 1s. The fundamentals of these frequencies are only one octave apart and so it should be a relatively easy matter to equalise such a narrow-band signal. An advantage of a DC-free signal is that after heavy low-pass filtering the waveform may be unrecognisable, but the position of the zero crossings on the time axis does not move, hence the use of these to convey the data.

Fig.3 shows what a time-code signal may look like after it has been replayed. The waveform here is called an eye pattern and is obtained by retriggering an oscilloscope and superimposing many traces. Note that there is amplitude variation between the 1s and the 0s because of frequency dependent loss, there is noise added which shifts the signal in the voltage domain and there is jitter, which moves the transitions along the time axis. The decoder has to reject all of these problems.

Fig.4 shows how the decoder works. The first stage is called a slicer. This compares the incoming signal with a threshold and recreates a binary signal with jitter. As the signal is DC free, after slicing the average duty cycle should be 50% so it is possible to modify the slicing voltage until this is the case. Adaptive slicers like this avoid drift.

The jittery binary signal then goes to a phase locked loop that regenerates a stable clock locked to the signal. The phase locked loop contains a phase comparator comparing the phase of jittery incoming transitions with transitions from the oscillator. The jitter results in noise on the phase error. By filtering the phase error, the control signal to the oscillator reflects only the average phase
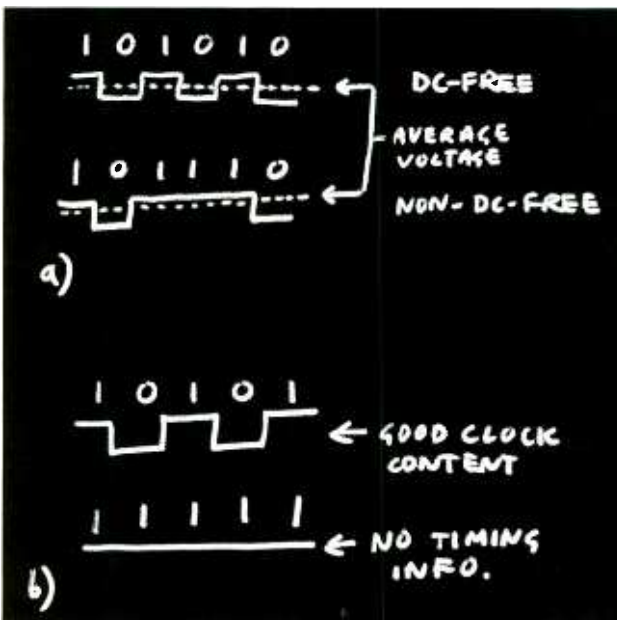

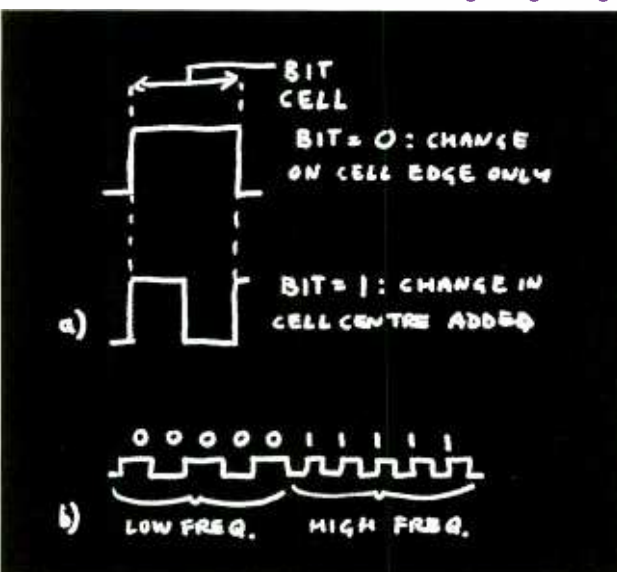**Fig.1: Raw data can not be used for recording or signalling**
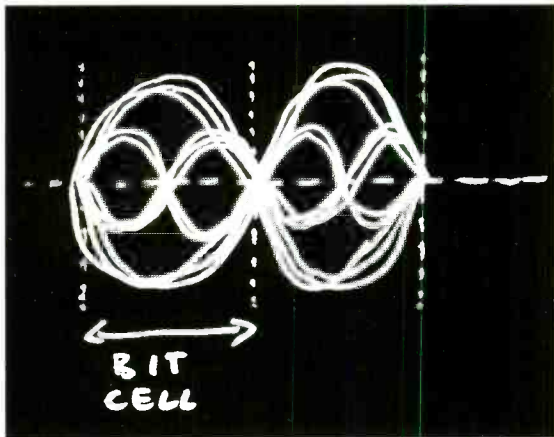

Fig.2: F M coding
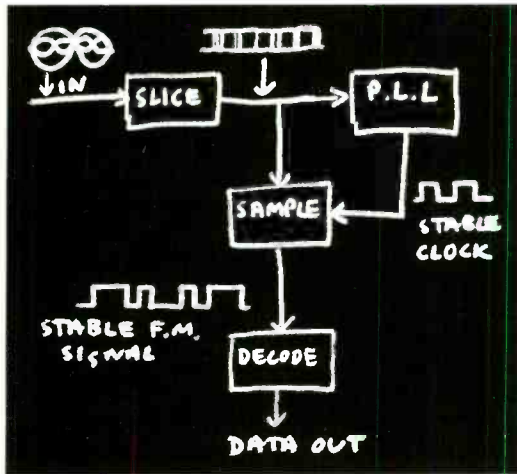
Fig.3: F M eye pattern

Fig.4: F M decoder rejects noise and jitter

and not the jitter. Fig.4 also shows that when a stable clock exists, it can be used to sample the jittery sliced signal. When successive samples are different, there must have been a transition. When they are the same, there was no transition. The exact position of the transition between the samples is irrelevant and so the jitter can be rejected. In this way the FM waveform can be 'cleaned up' and returned to a stable signal for decoding to binary. As all bits look the same it is important to synchronise the serial data message by beginning with a standardised bit pattern that the decoder can recognise. By detecting this sync pattern, the decoder can then count off the bits and recreate the original binary coded decimal numbers representing the timing information. The sync pattern used in time code is not only unique; it is also asymmetrical so that the decoder knows if the signal is coming from a machine running in reverse.

The time-code data only change once per frame period, but the decoder can obtain higher resolution than this. The structure of the time-code recording is, however, closely defined and the frame always contains the same number of bits. As a result the decoder can obtain higher time resolution by counting the bits through the frame.

Once a decoder has obtained the actual time data from a tape then further processing is needed to perform useful functions. One of the simplest functions is to position the recording at a predetermined code. This might be used in an editing application or in an autolocator. The required code is entered into the system, and the controller simply compares the desired code with the actual code to determine which way to go. This decision is used to shuttle the tape transport and as the tape winds, the difference between the two time codes will fall. As coincidence is approached, the shuttle speed may usefully be reduced to prevent an overshoot.

For synchronising two machines together, one of them has to be a master, which means that it runs at a constant reference speed, and the other has to be a slave, which means that it will change its speed to follow the master. In the case of an audio recorder and a video recorder, the VTR would need to be locked to station reference video timing, so it would make sense for the audio recorder to be the slave.

Assuming that a common time-code generator was used when the recordings were made, synchronisation is obtained by comparing the time codes replayed from the two recordings. Clearly this requires two decoder stages. The time code from the master becomes a reference and the time code from the slave is used as a feedback signal. The slave transport is speeded up or slowed down slightly in such a way that the slave time code has the same value as the master time code at all times.

In addition to the obvious requirement for a synchroniser, the slave machine must have some means to accept an external control input. In the simplest case this may be no more than an analogue input that can slightly speed up or slow down the capstan. Different machines have different values of capstan inertia and different sensitivities to the control signal and it will generally be necessary to set up the synchroniser for a particular model of slave recorder.

If this is not done the gain of the system may be too low or too high. The gain is defined as the degree of acceleration of the slave resulting from a given time-code difference. If too low, the accuracy of lock of the slave may be poor; if too high, the slave may hunt about the right speed, slowing down and speeding up alternately and doing nothing for the sound quality.

In more sophisticated systems there may be full control of the machine so that the master can be shuttled in either direction and the slave will continue to follow it. If one machine shuttles faster, it will have to wait for the other to catch up before play can recommence. ▩

To show, or not to show?

The trade show season now extends throughout the year setting the exhausting challenge of a perfect presentation every time. Why attend them all?

Dave Caulfield of E-mu /Ensoniq polishes the shop window

HERE'S THE CHALLENGE: To uproot your place of work, transport it several miles and rebuild it, paying special care and attention to decor (lots of plants are nice). If you are very lucky, you have two days to build it. Then you spend several days living in it, including a weekend (for which your staff thank you dearly) and then take it all back again. Oh yes, you may have to do this several times a year.

As you may have guessed, we are talking about exhibitions. Fairs, Shows, call them what you will. Exhibitions are a manufacturer's main opportunity to present their own 'shop window' to the buyer. Gather all the manufacturers under one roof and, bingo, you have a venue that every single buyer will want to visit (that is the theory anyway).

There is no particular season for exhibitions. Related shows (recording, broadcast, postpro and music equipment), take place in most months of the year. The ongoing challenge is always, 'Which ones shall we do?' (quickly followed by), 'Which ones can we afford?' The reason this topic has come up so much recently, is that there has been renewed interest and attention given to the promotion of quite a variety of shows, covering a wide range of interests.

The brutal truth is that not many companies can afford the budget to exhibit at all available shows (if I am wrong, let me know and I will start an exhibition company). Each company has to choose its promotional activities with care. The product line-up has to be considered to see where major opportunities lie, and the budget sliced up between; advertsing, PR, exhibitions, point of sale, and so on. Not every product fits every show, so the decision has to be taken—to partake or to resist?

E-mu/Ensoniq's products cover all of the above interests—so you can imagine that we have an unenviable task of deciding which shows we will have to pass over—never an easy decision, not least because it is generally considered desirable to have a presence at major exhibitions. Every exhibition stands or falls on the number (and calibre) of the visitors. The overall size of the show will determine how much the organisers will spend on show promotion, thereby attracting enough visitors to satisfy the exhibitors—a paradox, because if the potential exhibitors cannot be satisfied in advance that there will be enough visitors, they may not participate in the first place. The launch of each exhibition sees a testing of wills—the exhibition organiser presents the exhibition in the best possible light; 'nearly sold out', 'national TV coverage', while potential exhibitors jockey for the best positions and try to negotiate the price down.

So what makes an exhibition organiser want to do the job in the first place? First, you must understand that there are two distinct types of Organiser; the professional who does it for a living, and the Figurehead, who is usually a member of an industry association.

The former is motivated by profit—it is their business after all, while the latter is usually motivated by the desire to showcase the association and the members—and will often be supported by a professional organiser. There is nothing wrong with either, in principle, and the potential exhibitors can make their own decision, based on the given information.

I have to confess that I love exhibitions. It is great to be able to meet customers, observe competitors and contact friends. There is always a high level of excitement in the planning.

However, exhibitions have a lot to do with image and many things can go wrong between the planning stage and the show itself. Those of you who have seen *Spinal Tap* appreciate the pitfalls of trusting a designer to produce the finished article from a quick sketch. The weeks leading up to a show can be tiring, as all the details have to be brought together, while conducting the normal day-to-day business. A loss of attention and an important detail can be lost, until the day of the show of course, and *voilà*—instant *Spinal Tap*.

But what of the visitors? Why aren't there a larger number of visitors? Let us take what is normally accepted as a 'trade' show (one that you would not usually find the public attending). I am mystified why they do not attract a more people. It cannot be cost—I simply do not accept that someone whose livelihood depends on the industry can not justify a visit to an event where they can see most of the industry under one roof, plus the possibility of meeting many of their peers. Is it because it is felt that the suppliers are there only to sell? Well of course they are, but we all have to sell in some way to survive in business don't we?—even if we are selling ourselves.

A show can never exist for its own sake—not for long anyway. Despite the incredible technology at our disposal, I feel that our industry's greatest strengths are derived from direct communication. Yes, its the old 'one on one'. See you at the next show. ■

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All FR Series Power Amps have: • Variable frequency low-cut filters • Limiter with On/Off switch • Detented gain controls calibrated in dB and volts • Selectable stereo, mono & bridged amp modes • LED level display with Signal Present & Overload indicators • Balanced XLR & TRS inputs • XLR thru outputs • Superior T-Design cooling • 5-year transferrable Limited Warranty

M·800: 800 watts @ 4 ohms bridged • 400+400 watts @ 2 ohms stereo • Constant directivity horn EQ/Air EQ • 1/4" & 5-way binding post outputs

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Switchable "soft knee" limiter helps maximize SPL's without over-driving your PA cabinets (or the M·800).

Quick switch configuration for **stereo, mono** or **bridged** output.

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Built-in variable CD horn EQ. Constant directivity horns achieve wide dispersion at the price of reduced high frequency response. The M·800 lets you restore lost treble or just add some sizzle to your PA system's top end.
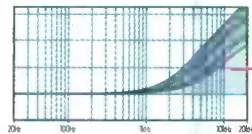
Generous front panel intake gets its cooling air from the room instead of from inside a hot road case.

Other cool stuff (pun intended) that our overworked art department can't figure out how to draw a red callout line to...

• Short T-Design cooling air paths for smoother thermal gradient and thus increased reliability.

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