Studio 4

VINTAGE KEYS
CLASSIC ANALOG KEYBOARDS

MASTER  EDIT  < CURSOR >
HOME/ENTER  — MIDI

DATA

Instr. Select

Instr. Param.

Device Param.

MIDI Mode

PRG Select

Tables/Maps

Pan/Glide
Name

MIDI Param.

Volume/Tune

Envelopes
LFO's

Volume
Filter

Oscillators
Waves

Mod-Macro

Env-Macro

Fast Access 2

Fast Access 1

Mode

Param./Value

Recall

Compare

Store

waldorf

Shift

O.K.

CLAVES  MARACAS

# MIDI

# STUDIO SOUND
### AND BROADCAST ENGINEERING

**On the Air — see page 48**

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www.americanradiohistory.com

# Mo' MIDI

There was a time when the use of MIDI was limited to the lower strata of the musical order. And when it first appeared (on the Sequential Circuits *Prophet 600* in 1983), this was its apparent destiny: to be an interface facilitating the interconnection of synthesisers, drum machines and sequencers. It was conceived by Dave Smith, then president of US company Sequential, as a solution to the compatibility wars being waged between manufacturers of these instruments . Would it not be in the commercial interests of the companies concerned, as well as the musical interests of their customers, to have an agreed standard allowing different manufacturers' equipment to be part of the same setup? He reasoned. But in adopting the MIDI standard, Smith and the newly-formed American and Japanese MIDI administrative bodies had created a monster.

Like Mary Shelley's monster, MIDI possessed a life of its own. Releasing details of the specification caused a completely new sector of the musical instrument industry to appear — independent programmers paired their technical knowledge with their musical needs and wrote the software the exercise dictated. It did not require the resources of a hardware-based industry and the resulting programs were potentially attractive to anyone owning MIDI equipment. Programmers became businessmen and MIDI software became big business for them. Software sequencers, along with samplers and drum machines gave a 'new breed' of musician a golden opportunity — and the studio business a kick in the groin.

'MIDI is a 4-letter word', 'MIDI: Musical Idiots Demand It', joked the pros. Yet MIDI has irrevocably changed the music industry, the way in which certain music is made and even the music itself: from sample-based dance music to experimental classical music. It's no joke.

The rising popularity of mix automation in professional installations was pre-empted by MIDI mix automation — admittedly it lacked the definition of pro systems sophisticated touches such as moving faders, but it did mean that many 'lesser' equipped studios were offering automating mixing before their professional brethren. In fact, more and more professional equipment now unashamedly incorporates MIDI. It began with the inclusion of MIDI sync facilities — the Trident *90* console, for example, generates MTC and Song Position Pointer allowing it to talk to a sequencer as if it were a tape transport. Korg's *SoundLink* direct-to-disk system actually incorporates an on-board 16-track MIDI sequencer as well as supporting MIDI Tempo Map generation and the ability to send Program Changes to other MIDI equipment. And the new Air Lyndhurst complex is incorporating MIDI tie lines amongst its host of 'professional' patching facilities from the ground up.

In certain studios, MIDI has offered a completely alternative approach to tapeless recording, since a sequencer provides a perfectly adequate method of managing a sampler-based audio recording system. Tim Simenon's Dijiland Studio (UK) employs Steinberg's *Cubase* software and a selection of Akai samplers to exactly this end. The setup also lends itself to live performance without the necessity of preparing samples of 'studio-generated' sounds or backing tapes where they might otherwise be necessary.

None of this is to say that MIDI is perfect — there is always someone trying to rock the boat with talk of 'MIDI 2' or 'Super MIDI' on the grounds that MIDI is a slow, serial protocol — but the present system is holding up well, and offers pro studio owners as many opportunities as it does owners of project studios. Additionally, MIDI reminds us of the importance of standardisation, something many users of digital audio interfaces wish had never been forgotten.

MIDI has now come of age. It has weathered the snobbery that might have kept it out of the pro studio and now most professional studios make use of it in some capacity — or their clients do. MIDI also proves that the leads do not always come from the top. It is easy to believe that the most expensive, most esoteric equipment represents progress, but music itself tells us that it ain't necessarily so. ■

**Tim Goodyer**

Cover: Equipment courtesy of The Synthesizer Centre, London. Photography: Nik Milner

# AMS AND NEVE.

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When **⊜Ⅲ⊜** and ∧ Neve were merged at the end of 1992, it was the talking point of the year.

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But instead, we'd like to let you know that in '93 AMS Neve plc has combined expertise and resources to push ahead in both traditional and new technologies, developing products that will always keep pro audio people talking.

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AMS Neve plc, Billington Road, Burnley, Lancashire, BB11 5ES, England. Telephone: 0282 457011. Fax: 0282 39542.

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# Abbey Road's Capricorn

Abbey Road Studios have become the first facility worldwide to offer the Neve *Capricorn* console. The 45-fader digital console has been installed in the specially refurbished Penthouse Studio where it replaces a seven-year-old Neve *8128*.

The installation itself was said to go 'very smoothly', and due to the console breaking down into easily manageable sections, there was no need to remove control room windows, doors, etc — 'No more difficult than getting a sofa into a house,' said Technical Manager Neil Aldridge.

'The potential of the *Capricorn* is enormous,' stated Abbey Road's Director Ken Townsend, 'it is hard to visualise all the possible applications of such a sophisticated piece of equipment, but we are confident that it will be a great success for recording, remixing and a host of other tasks.'

The console is configured with 96 signal paths and can be interfaced via MADI to the studios' Sony and Studer 48-track digital machines. In addition MADI has been used to link the *Capricorn* to Abbey Road's large orchestral Studio 1, providing 32 mic inputs. This link-up will allow direct comparisons between analogue and digital consoles, and will have a strong bearing on Abbey Road's decision to install the *Capricorn* in Studio 1 this year.

The first sessions in the newly equipped Penthouse have been remixes for an album of Jules Styne overtures taken from famous shows including *Funny Girl*. The sessions are being produced by John Yap and engineered by John Kurlander whose initial impressions of the console are very favourable.

'It's a brilliant design,' said Kurlander. 'Operationally its very easy to use and I find that I can do a lot of things in a fraction of the time it would take on an analogue desk. The sound is very natural without a trace of coloration — what you put in you get out.'

On the recent announcement of 'AMS-Neve plc', Abbey Road's Technical Manager Neil Aldridge confirmed: 'We have been given concrete assurances that the *Capricorn* will remain fully supported by the new company in the future.'
**Patrick Stapley**


**Penthouse pet — Abbey Road's *Capricorn***

# Morse: Ampex on the case

As the seventh, and final, TV series of the Beamish-sponsored *Inspector Morse* hits the ether in the UK, the soundtrack CD derived from the first series chalks up sales in excess of 500,000 and earns itself and its creators an Ampex Golden Reel award. While it is obviously beneath the powers of Morse to infer that the tie-in between the music and the award is the use of Ampex *Grand Master 456* tape to capture the music, it was beyond the powers of composer Barrington Pheloung to predict the popularity of his work. He was left in no doubt, however, when the final day's recording for the final series was concluded by the presentation of the award to Pheloung, Music Supervisor Graham Walker,


**Barrington Pheloung
— Ampex Grand Master**

Recording Engineer Dave Hunt and Mix Engineer Chris Dibble. Also in receipt of a Golden Reel was Adrian Kerridge, MD of CTS Studios where the recording (and presentation) took place.

The *Morse* recordings totalled some 56 hours of music and involved the playing talents of a 33-piece orchestra. Recording took place in CTS Studio 1 on a customised 60-channel Neve *VRP* console with *Flying Faders* and *Recall*. Mixing was

conducted at Lansdowne Studios on a 48-channel Neve *VRP*.

In order to qualify for the Golden Reel award (now 16-years-old, and over 1000 presentations later), recording and mixing had to take place exclusively on Ampex professional tape. The award also qualifies its winner to nominate a charity for a $1000 donation from Ampex — in this case it was the Imperial Cancer Fund. ∎

# Contracts

● Recent completed contracts by Relflexion Arts studio design in Portugal include Registudio in Amadora province, a complex of seven rooms occupying the ground floor and basement of an apartment building and the Lisbon Planetario, that has had its sound system replaced, tailored by designer Philip Newell
● Studio 150 in Amsterdam has bought a DAR *SoundStation SIGMA* digital audio production system to cope with the increased business of their audio-for--video production. In accommodating the expansion Studio 150 has contracted The Audio design Group for a major rebuild.
● Recent Sony *PCM-3324S* digital multitrack sales in Germany include two machines to WDR in Cologne; machines to Audiorent in Recklinghausen; DAS in Munich; Musikhochschule, Wurzburg; and Walldorf Studio in Frankfurt.
● 3M *996* tape has been chosen by London jazz venue Ronnie Scott's for recording its sessions
● The South African Broadcasting Corporation (SABC) has recently bought a Solid State Logic *ScreenSound* digital audio editor for installation in the Corporation's new digital audio and video editing facility in Johannesburg.
● Yorkshire Television have bought two Avid *model 230 Media Composers* after evaluation of Avid, Lightworks, EMC

and Eidos systems.
● Recent sales of the *Optifile* mixing automation system include Sam Therapy in London; Carlton TV, London; and Blackwing Studio also in London
● The new premises of RTL in Bruxelles will be equipped throughout with Genelec *1019A* and *S30C* monitors. PWL new studios in Manchester have ordered Genelec *1034A* main monitors
● Sounds Good duplicators in Berkshire, UK, have chosen the *SADiE* hard disk editing system from Studio Audic & Video, to spearhead the company's move into CD mastering, digital editing and postproduction as part of its CD replication service.
● The BBC World Service have placed an initial order from Calrec Audio for 12 of their new *RQQ 2200*, 2-channel, 4-band parametric EQ units. Scottish Television have added another one of Calrec's *RQP3200* mic line preamplifier, compressor-expander-gate to their increasing total.
● NRK, the national broadcast organisation for Norway, has recently bought two *ScreenSound*s and a *SoundNet* for use on all its video and film audio postproduction work. Other recent SSL digital product sales include a *Scenaria* to ORF in Austria; *ScreenSound* to Carlton TV in London; two *Scenaria* production systems and a *ScreenSound* to Fox TV in Los Angeles; and a *ScreenSound* to STAR TV based in Hong Kong. ∎

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# YOUR NEXT CONSOLE PURCHASE MAY HAVE A PROFOUND IMPACT UPON YOUR ART AND BUSINESS. SO LISTEN CAREFULLY, CHOOSE WISELY, AND BE SURE THE CONSOLE'S BUILDERS ARE PASSIONATELY COMMITTED.

POPULAR MYTH HAS IT THAT success in the music and recording business is a matter of luck, a happy spin of Fortuna's wheel. Sure.

Fact is, if you've enjoyed any luck, an unwavering dedication to your work likely made it happen.

When it's time for a new console, however, lots of people will court you for your money, but few will share or understand the nature of your passion. Does that matter?

It should. For just like you, we at D&R are serious — passionately serious — about what we do. It shows in the way we design and handcraft our entire console range.

Mere words, we know. So we invite you to listen critically to any console from any of our worthy competitors. Regardless of the price. Then apply your ears to one of our consoles. You'll hear what thousands of D&R owners know.

Our zealous devotion to quality goes beyond sonic subtleties. It also manifests itself in

highly tangible ways.

For instance, we phase-correlate every audio stage. Our star-grounded circuit and module designs eat noise for lunch. And we surround the complete signal path with a custom-welded, fully shielded steel chassis.

These are just a few of the make-sense reasons why all D&R boards sound so quiet and so accurate. (Ordinary designs in an aluminium chassis can't touch this — and you can hear the difference.)

Naturally, being rather fervent in our passion, we have come up with some inspired features. Such as floating subgroups (which we pioneered, by the way). Full-band, "high-def" equalisers. Extensive sourcing. User-configurable aux send, monitor, channel, and EQ signal paths.

All of which will inspire you to work faster, more flexibly, and ultimately, more profitably.

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*This crustacean understands grounding better than some manufacturers — so don't be suckered. At D&R, we employ a unique starground system. On each circuit. On each module. On every console.*

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*D&R handcrafts a wide range of mixing consoles for recording, live sound, theatre, post-production and broadcast.*

# Spirit Automation



**Spiritual guidance from Steinberg**

Soundcraft's *Spirit Auto* console represents something of a world first by not only offering VCA fader and switch automation, at the lowest level in the market, but also for integrating the automation into the desk itself at this price bracket. The Steinberg *Automation* software required to run *Spirit Auto* at its highest level is optional but it does considerably enhance the desk's capabilities.

To recap briefly, *Spirit Auto* runs in three basic modes: first, as a stand-alone, computer-independent, 'ordinary' mixing console; second, as a standard MIDI, continuous controller generator, from its VCA faders, CHANNEL, and MONITOR-ON switches, for interfacing to a MIDI sequencer; and third, as a more evolved MIDI generator running Soundcraft protocol, double-standard, MIDI resolution for interfacing to the aforementioned Steinberg *Automation* software.

These statuses are communicated to the desk by the presence, or absence, of MIDI connections at *Spirit Auto*'s MIDI connection panel (under its armrest) and the position of a cluster of DIP switches (placed nearby). These set standard MIDI or Soundcraft MIDI protocol, message speed, Running Status, half-duplex or full-duplex communications (equivalent to a keyboard's local on or off status) and an Autochase mode, which causes the desk to send snapshots every three seconds to enable a chasing sequencer to pick up a mix more quickly.

Each of the desk's Channel faders and Channel and Monitor mute switches generates MIDI continuous controller data — moving Channel fader 7, for example, generates MIDI Continuous Controller #7 data, while the same Channel's ON switch generates Continuous Controller #39 data. Thus in standard MIDI mode, desk moves can be written into a sequencer at which point the user's aptitude and skill with the sequencer will dictate the degree of flexibility on tap. C-Lab users will benefit from using the sequencer's Real-time MIDI Generator (RMG) page templated for P-User events to display fader and single switch moves of groups of 16

channels simultaneously. However, 24-channel desk configurations and the MONITOR ON switch statuses will require an additional P-User template to be created. *Notator* and *Creator* users looking for fine editing control will have to harness the power of the program's Hyper Edit page to give fader contour and Switch activity displays by allocating incoming continuous controller data to the Page's Instrument Sets.

Steinberg *Cubase* user's can visualise fader and switch movement by using the sequencer's MIDI Manager page by creating a MIDI Mix part and building a MIDI mixer map of Objects corresponding to the incoming continuous controller data. In the context of handling MIDI desk automation data this page's graphical depiction of desk moves is perhaps marginally superior to C-Lab's offering by virtue of the former's Local, Write and Replace statuses and the ability to display more of the desk controls on one screen. C-Lab does, however, through Hyper Edit have the edge in the fine edit stakes.

As such both of the most popular sequencing systems in Europe work well with *Spirit Auto* but are not nearly as elaborate or natural as the custom-written Steinberg *Automation* software. Written for the Atari *ST*, users intending to run the program alongside *Cubase* on the same computer will require a minimum of 2Mb of on-board memory. Due to *Spirit Auto*'s potential for data generation port expansion would seem a sensible requisite in all instances.

*Mac* users have not been neglected with the news that JL Cooper has announced details of its *Softmix* package which will feature enhanced resolution, moving fader and switch displays, cut and paste editing, auto punch in and 16 soft sub groups. This seems like a similarly equipped *Mac* package to Steinberg's Atari software.

The Steinberg *Spirit Auto* package runs an 8-bit protocol as opposed to MIDI's usual 7-bit resolution, locks to SMPTE and MTC and is based around a main screen with a graphical depiction of the desk's

automated faders and switches. The performance and presentation of the package greatly exceeds the expectations of a product at this end of the market. It is well thought out and surprisingly able in recording mix information using a typical Steinberg arrangement of transport controls. Penalties are incurred in the absence of some of the more elaborate off-line editing possibilities and the fact that the desk only interacts with the system as the generator and the computer keyboard and-or mouse plus screen display have to be used for instigating functions and the reading of nulling points, for example. When in doubt, remember the price.

The system uses Write (based on a Write-to-End-type principle), Read and Update (relative movement of faders to recorded data) statuses for each of the Channel faders and switches and this can be sectionalised at will from all controllers to just faders, to any combination of switches and faders, all of which are selected by mouse. The system is also not immediately destructive of mix data and permits a previous pass to be retrieved from internal memory. Global mode controls are provided, channels can be named (albeit abbreviatedly) and soloed from the screen. Eight soft fader groups are available on screen to be driven by mouse, alternatively for hard control a Channel fader within a group can be designated master and run off real finger movement. Channel and Monitor Ons are also grouped.

Writing data in and out of Read can be adjusted for a ramp between otherwise disparate values and, thoughtfully, an Auto mode status is available which negates the need to find null positions by flipping a controller from Read to Write once the Read value is exceeded. This is particularly important as the screen fader blocks can get a little busy when real fader, recorded fader and VCA position characters are all running simultaneously. The automation also has a Snapshot complexion to it which is pseudo-cue-sheet based or can be shot

in live using a SNAPSHOT button placed in the bottom right-hand corner of the console.

The degree of editability will surprise some hardened automation users. Coming as this does from the 'musician' side of things, Steinberg have approached the business in a very graphical way, using horizontally scrolling contours to depict fader movements in a manner similar to C-Lab's Hyper Edit and the 'Jobs' principle of the Amek-Steinberg *SuperTrue* system. There is a down side in as much as a full desk's worth of fader and switch movements displayed on screen is a lot less than informative. However, zoom functions for the number of controllers on screen at once and the section of time (up to two minutes) being shown make things much clearer. The absolute position of the cursor is represented by a SMPTE display and this can be toggled to show bars.

In addition to Cut, Copy and Paste functions of Parts of a mix across individual or groups of channels, a Steinberg Toolbox allows fader contour or switch activity to be redrawn (freehand or as straight lines) or altered. A Smooth function does just that over a highlighted section of otherwise jagged and unrefined fader movement.

While it can be said that in puritanical terms there is no such thing as a natural automation system, once a system is learnt beyond the fundamentals it becomes of increasing value to the operator. This is certainly the case with the *Spirit Auto*-Steinberg *Automation* combo, because it becomes so integrated into a MIDI-orientated setup that it becomes almost essential.

There is no getting way from the fact that this automation system is very much more than some MIDI toy and for its market it is a well targeted and intelligently implemented package that does all it says it can, easily. Perhaps it is a little too geared towards the MIDI user to be of immediate appeal to top-end automation power users but then again they will not be buying it. This system is aimed clearly at the entry level and anyone else who can do with such a tool at such a price. There is a lesson to be learnt here. ■

**Zenon Schoepe**

**Soundcraft, Cranbourne House, Cranbourne Industrial Estate, Potters Bar, Herts EN6 3JN. Tel: 0707 665 000.**

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Akai *CD3000* stereo sampler

# Akai's next generation

Taking Civeden, the 'home' of the notorious Profumo affair as the venue for the UK launch, Akai recently unveiled their *3000*-series samplers.

The range comprises the *S2800*, *S3000*, *S3200* and *CD3000* stereo samplers all of which boast 16-bit, 32-voice, 8-output operation and include onboard digital effects and resonant filtering. Memory options begin with 2Mb as standard in the *S2800* expandable to 16Mb; the *S3000* and *CD3000*'s standard 2Mb

are expandable to 32Mb. The *S3200* has 8Mb as standard, expandable to 32Mb and also comes fitted with SCSI, SMPTE sync, digital I-O ports and a hard disk (and MO disc) recording facility, while the *CD3000* places the SCSI port alongside a CD-ROM drive and is accompanied by five 'free' CD-ROM sound discs.

Akai also announced availability of Sampletools *Polystar* librarian/sample editing software,and Sampletools *Parastar* real-time parameter and wave editing software for the *S1000*, *S1100* and *S3000* series — both packages run on the Apple *Mac* supporting *System 7*. Sampletools *Wavestar* real-time wave editing software for the same

samplers will run on both the *Mac* (*System 7*) and the Atari *ST-STE* computers. The company also released a version of *Wavestar* for their cheaper *S01* sampler (*Mac* and *ST-STE*), and *Proex 01* sample editing software for the *S01*-Atari *ST-STE*.

**Akai UK Ltd, Haslemere Heathrow Estate, Silver Jubilee Way, Parkway, Hounslow, Middx TWH 6NQ. Tel: 081 897 6388.**

# Francinstien

R&D company Perfect Pitch Music have launched the 1U *Francinstien* stereo enhancement system. Featured

are six controls which can be used to emphasise level, time-difference and tonal cues which the ear and brain need for spatial perception. The system works by matching the high frequency intensity differences with the essential low-frequency, time difference information. The unit combines this processing with low-end, mid-band and HF EQ to provide stereo enhancement.

**Audio Digital Technology, 178 High Street, Teddington, Middlx. TW11 8HU. Tel: 081 977 4546. Fax: 081 943 1545.**

# Silver 5L nearfield

Silver Productions wanted to design a nearfield monitor to the exacting standards of main studio monitors and add into the design the various time domain, phase, polar response and proximity parameters that

govern nearfield monitoring. The result is the Silver *5L* . The monitor has a power handling of 150W at 8Ω; will produce 88dB ±2dB from 130Hz to 18kHz and will produce an accurate bass response down to 74Hz. With the optional sub-bass units the extension is 36Hz and in-room response is 26Hz.
**Silver Productions (London) Ltd, 29 Castle Street, Wiltshire, SP1 1TT. Tel: 0722 336221. Fax: 0722 336227.**

# Spatializer

The *Spatializer* from Desper Audio Products is a real-time processor that allows the placement and movement of individual sounds in 3D space. Using standard stereo playback, the system can be used in every phase of audio production with the effect maintained throughout all stages of recording, rerecording and postproduction. The finished product is mono compatible, surround-sound compatible, and requires no decoding.

The *Spatializer* consists of a rackmounted processor unit and a portable 8-channel console unit with joystick controllers and control function.
**Audio Intervisual Design, 1155 N. La Brea Avenue, W. Hollywood, CA 90038. Tel: +1 213 845 1155. Fax: +1 213 845 1170.**

# Otari's *Concept 1*

The *Concept 1* Music and Production Audio Console is a new digitally controlled analogue desk; features include symmetrical dual-path architecture with 24 buses and 10 aux buses. Both audio paths in each module have their own 4-band EQ and 100mm long throw fader, and each individual audio path features full dynamic automation of its fader and mute. All *Concept 1* systems come complete with Otari's *DiskMix* dual path VCA automation system.
**US: Otari Corporation, 378 Vintage Park Drive, Foster City, CA 94404. Tel: +1 415 341 5900. Fax: +1 415 341 7200.**
UK: Otari UK, Unit 3, Elder Way, Waterside Drive, Langley, Berks SL3 6EP.
Tel: 0753 580777. Fax: 0753 42600.

# Digigram PCX5 card

The *PCX5* is a 2-channel audio processing card designed for *PC-AT* and based on DSP of the Motorola-*56000* range. The card integrates all the features of Digigram's *PCX* product range into a small format design. *PCX5* records and reproduces CD-quality audio, the compression-decompression algorithms are implemented to the MUSICAM ISO-MPEG-Audio standard. Digigram compression algorithm is also available but faster data exchange with the PC environment allows this board to run without compression extending its flexibility.
**Digigram, Parc Technologique, de Pre Milliet, 38330 Montbonnot, France.**
**Tel: +33 76 52 47 47. Fax: +33 76 52 18 44.**

# MKH times five

Sennheiser have expanded its *MKH* range of studio condenser microphones with the new variable polar pattern *MKH 80*. The five switchable patterns are omnidirectional, wide angled cardioid, supercardioid and fig-of-eight. Features include broadband directional properties within a wide angle area, low noise performance, wide range of modulation control and high dynamic range.
**UK: Sennheiser UK Ltd, 12 Davies Way, Knaves Beach Business Centre, Loudwater, Bucks, UK. HP10 9QY. Tel: 0628 850811. Fax: 0628 850958.**

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# Korg A4 guitar processor

Pedalboards offer a different approach to guitar multieffects processing than your average rackmount, because they generally offer a greater degree of performance orientation and consequently tend to appeal to a different type of player. Korg's A4 pedalboard disguises its digital capabilities well in presenting itself as a straightforward foot unit with footswitch selection of programs, a tuner and pot-type controls for parameter editing within a patch.

Patches are organised in five banks of six, and are duplicated for the 30 factory presets and 30 user presets. A Bank is called up on a dedicated footswitch which increments sequentially on a two-character LED display but a sound is only activated when one of the six, illuminated, preset footswitches within that Bank is depressed. The changeover is instantaneous to the point that decaying notes are processed across patch selections.

The two blocks of user and factory presets are toggled between on a small button which can, with a degree of ballet-like dexterity, be selected by foot. Larger footswitches are supplied for Bypassing the processing and for switching between the performance and edit modes.

The standard of construction is good although not confidence inspiring — there is something almost delicate about the moulded plastic shell and the way it creaks, while the exposed parameter programming pots look like they could only endure one all-out assault from a misplaced size 12. However, the under-foot feel of the all-important footswitches is reassuringly positive and the A4 is stable and squat on its base.

Connections are provided for the guitar input, individual left and right audio outputs on standard jacks plus two, programmable, external controller jacks (for external amp channel switching), a mini, stereo,



**Korg A4 — operational ecstasy**

phones jack and the 9V DC PSU. MIDI In and Out handle program dumps and loading and patch change receive and transmit.

Effects are arranged in fixed, serial order, blocks starting off with a compressor with sensitivity, attack and level; distortion with seven basic tones, gain, tone shaping; EQ latching to distortion selection, and EQ with fixed low and stepped 16-frequency mid and high. Pitch shift and delay can be linked or independent with ±1 octave, fine shift and high cut. Almost a second of delay is on tap, with feedback and top-end damping; the chorus-flanger runs with triangle or sine wave LFO, with adjustable speed, depth, positive or negative feedback and up to 75ms of delay. Reverb sports three basic characters, adjustable RT and high-end damping.

Also on board, and programmable per preset, is a switchable amp simulator, open and closing of the external control jacks, an adjustable noise reduction threshold and a preset's output level.

To edit an effects block, the Edit mode is entered on the EDIT-PERFORMANCE footswitch, the relevant effect is selected with MODE SELECT buttons at which point the six pots take on the parameter functions listed for that effect on the top panel. Overall comparison of an edited version of a preset to its original can be made and while editing a dot appears on the 2-character LED display whenever an edited parameter value equals that of the original. The desired combination of effects blocks are finally selected on

the preset selector footswitches and firmed up with a Store routine.

Input works on a 3-colour LED and thankfully the device is very unfussy about input strength. The A4 is also surprisingly quiet due mostly to the excellent response of the simple but effective noise reduction circuit. The straight-through tone of the unit is strong and wholesome and consequently works well with chorus-flanger-type tones.

If 60 presets is not enough, it is worth mentioning that because Edit mode is selected on a footswitch, recalled patches can effectively be edited live by stamping on footswitches and adding or subtracting effects blocks from a preset. This is far superior to the locked-out nature of many 1U-high rackmount units that require perhaps too much forward planning and restrict the guitarist's options once the music is under way. The A4 is likely to appeal to those who identify with the effects pedal approach and want to develop sounds as they go.

Disappointments include the overdrive tones which are a little too 'bee-in-a-jam-jar' apart from three of the possible seven timbres which have a quite unique and interesting mid-boost character to them and represent excellent starting points. Most also respond to a little less than maximum gain at which point decays become a touch *fizzy* and while this can be remedied with EQ, it is hard to get a convincing hard chord sound. Modern solo sounds are handled comfortably. The reverb and delay quality, while not sparkling (no bandwidth figures are supplied), is

very apt for guitar. The pitch shifter is not the fastest or most consistent thing on the planet yet as a result has a lot of charm. The compressor hits a 'brick wall' fairly early on but again is complementary. All these minor inadequacies do not stop the device adding up to considerably more than its constituent parts and the A4 has a very immediate and earthy feel to it.

The automatic tuner defies criticism. Stamping on the Bypass footswitch accesses the tuner but holding the switch down slightly longer also mutes the clean output allowing you to tune in silence. You strike a string, the A4 decides what it thinks it is and displays the note while giving a very reliable and stable indication of the direction in which it needs to be pitched. Excellent.

The A4 is extremely easy to use and comfortably demonstrates that this sort of processor does not have to look high-tech and behave in a convoluted manner for it to be adequate. It is easily as good as many more expensive but less accessible units. While the pitch shifter may not be the fastest and the delays not the crispest, there is simply no substitute for being able to get to a unit's capabilities quickly. ■

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# QUADRA 950

Taking the Apple Quadra as pro platform for audio-MIDI recording, Mike Collins looks at system requirements, the available hardware and software

The *Macintosh* computer first became popular with pro musicians in the USA, where it is designed and manufactured. Now it is finding its way into recording studios as a MIDI sequencing and digital recording device. As a computer, the *Macintosh* and *Quadra* ranges are also widely used for desktop publishing, graphic design, and a wide range of business applications.

Most *Mac* MIDI software originates in the USA, and currently the most active companies for both MIDI sequencing and digital audio are Digidesign, Opcode, Passport, Dr T's and Mark of the Unicorn. German company Steinberg also offer programs for the *Mac*, and C-Lab are about to add their support.

Several companies (Sonic Solutions, Digidesign, and Mark of the Unicorn) offer digital audio recording direct to hard disk via NuBus cards. Others use the *Mac* as a 'front-end' for control of their proprietary hardware — Otari offer their *Pro-Disk 64*, Roland have their *DM80*, and various systems are available from Yamaha, *DAWN*, Studer-Dyaxis, and others.

But just why does the *Quadra 950* appear to be the machine of choice for recording studio and professional musical applications? It will take a large amount of RAM (up to 64Mb), it comes with a high-speed high-capacity internal hard disk of 250Mb or 400Mb capacity, and it has space to add a CD-ROM drive, an optical disk drive, and a further large capacity hard disk drive all within the cabinet. The video circuitry will drive any monitor, up to 21 inches, in colour and, although the colour resolution drops from 24-bit to 16-bit on 21-inch screens, this is more than adequate for musical and recording work. The larger sized screens (16–21 inch) are better suited to editing waveforms and for use with screen-space-hungry software like Digidesign's *Pro-Tools*. The 33MHz 68040 processor, improved SCSI implementation, extremely fast internal hard drive all contribute to making the *Quadra* quick and responsive in use — an important matter when the clock is ticking in the studio. Compared to its predecessors in the *Macintosh* range, the *Quadra 950* has a faster processor, a faster hard disk drive, better video, better audio, and space for more peripherals inside the main computer casing.

## Audio

Certainly the *Quadra*'s internal audio facilities are an improvement over previous *Macs*, but they are still not great. Apple should really feature a built-in DSP processor (like the NeXT computer and the forthcoming Atari *Falcon*), so that CD — quality audio is always available. Stereo line inputs and outputs are provided and (conveniently enough) there is provision for the audio outputs of an internally mounted CD-ROM player to be fed into the internal audio circuitry. The internal speaker is larger than in previous models, but still provides relatively low-quality output.

There are three methods by which you can play sounds internally on the *Quadra* — by using the square-wave synthesiser for beeps or simple waveforms, by using the wavetable synthesiser to produce sounds based on a single complex waveform cycle (which you have to predefine numerically to represent the timbre you want), or by using the sampled sound 'synthesiser'. This last option is the most usable of the three, and you can either record sounds directly into the *Quadra* via the audio line inputs or replay existing sound files in AIFF format or *Macintosh* 'snd' format. A popular 8-bit digitiser is available (*MacRecorder*) which has a built-in microphone and line inputs, and comes with an excellent audio editing and synthesis application called *Sound Edit Pro*. This software has recently been upgraded to record and playback at the CD sample rate of 44.1kHz, to open *Sound Designer II* files, and to use the *Sound Tools*' Sound Accelerator card if available.

The *Quadra* allows you to play sounds continuously from disk while other applications execute, and features stereo outputs — unlike the compact *Macs*. The *Quadra* also allows you to compress audio files to save space on disk, and lets you expand these in real-time for playback. Using the sampled sound 'synthesiser' you can play back multiple channels of sound for layering dialogue, sound effects and music, for instance, whereas the other internal sound synthesisers only offer single-channel playback. The Apple Sound Chip used in the *Quadra* features 8-bit 22.25kHz sampling and sample-rate conversion. The message here is that these in-built audio capabilities are really aimed at multimedia users, rather than at audio professionals.

## MIDI and audio

Apple supply their own MIDI software free with every computer. Called *MIDI Manager*, this provides MIDI input and output from any compatible MIDI software, and also allows several MIDI programs running concurrently (under System 7 or MultiFinder) to communicate internally. Just about all professional MIDI software packages can use *MIDI Manager* — although most also offer their own software routines for more efficient data handling.

The basic *Quadra* typically comes with either a 230Mb or 400Mb internal hard disk, and with 4Mb or 8Mb of RAM. The internal hard disk will hold all your software, with space for plenty of MIDI files and such like. For recording audio you will use external drives which have the required large capacity. To set up a more versatile system you might add a RAM upgrade to, say, 24Mb, a 128Mb optical removable drive for backups, a CD-ROM drive to give you access to popular sample libraries, and a *LaserWriter IIf* printer for music scores or for session documentation. A modem could prove useful for sending or receiving technical information or MIDI files, and an ISDN card could be used for bringing in audio digitally from remote locations.

For digital audio recording and editing, Digidesign's *Pro Tools* 16-track system is highly recommended. This consists of four NuBus Audio cards, four 4-channel 19-inch racked Audio Interfaces, and a NuBus System Accelerator card. For working to picture you will need a 19-inch racked Video Slave Driver to resolve the system to house sync, and a SMPTE Slave Driver SMPTE reader-generator. You could use up to four 1Gb external hard drives (or more) for recording audio, and a 650Mb optical drive is recommended for mastering your mixes, as well as for general backup. ProArchive 2Gb data DAT or 5Gb 8mm tape backup units are available which let you upload or download in the background while you continue recording or editing. In practice, a JL Cooper *Fadermaster* hardware controller offers a much more usable alternative to the mouse for moving faders and so forth. *ProDeck* software provides an on-screen emulation of a traditional audio mixing console, complete with channel faders, EQ, automation, effects, and audio recording transport controls. Bear in mind that the more channels you are using, the more screen space you will need! *ProEdit* software offers an edit list for replaying audio files at specified SMPTE locations — which makes the system well-suited for working to picture. MIDI recording and playback is also featured, but is only very rudimentary at present. Finally, using the digital outputs of your *Pro Tools* Audio Interface and Digidesign's *MasterList PDS* software, you can record CDs directly from the *Pro Tools* system to the Yamaha *PDS Compact Disc Recorder*. Realistically, the *Pro Tools* system demands a *Quadra 950* and large screen for best results, so it is likely that these computers will find their way

**Apple's tower of power — the *Quadra 950***

into professional recording studios in the near future.

For MIDI recording, the choice of software is currently between *Performer*, *Vision*, and *Cubase*. Versions of these (*Digital Performer*, *Studio Vision*, *Cubase Audio*) are now available which will work with the *Pro Tools* hardware to allow you to record and edit hard disk audio tracks from within the MIDI sequencing environment .

For sample editing (as well as hard disk audio editing) you can use Digidesign's *Sound Designer II* software or Passport's *Alchemy* software which is optimised to work with most popular samplers (such as the Akai *S1100*). For synthesiser editing, there are just two choices — Opcode's well-established *Galaxy* software, or Dr T's *X-Or*. Recording engineers and producers may well appreciate Opcode's *Track Chart* software which lets you display 'timelines' indicating track activity for your MIDI sequencer and audio tracks.

All Opcode software can take advantage of Opcode's *OMS* MIDI software, which is somewhat similar to Apple's *MIDI Manager* software, and allows internal communications between *OMS*-compatible software running concurrently on the *Quadra*. *OMS* can allow you to run non-*OMS* software concurrently, and can also communicate with the *MIDI Manager* if necessary, but everything is more well-integrated if you stick to *OMS*-compatible programs. This system allows you to name all the MIDI devices in your studio, define which MIDI channels and output ports they work with, and then make these names available within *Vision* or *Studio Vision* sequencers. *Galaxy* also works well with *OMS*.

For music scoring, Coda's *Finale* software has the most professional set of features, and runs very well on the *Quadra 950* with a 21-inch monitor. Other packages worth considering are *Mosaic* from Mark of the Unicorn, and *Encore*

from Passport. A number of musicians and recording studios (such as the BBC Radiophonic Workshop) are using custom-written MIDI software to control their MIDI recording and synthesiser equipment. For those of you with the motivation to follow suit, there are two software packages which you should check out. These are Ear-Level Engineering's *Hyper MIDI* software which adds MIDI capabilities to Apple's *HyperCard* software, and Opcode's *Max* software which offers a graphical way of programming MIDI applications.

The better *Macintosh* MIDI interfaces all offer SMPTE read-write and have built-in MIDI patching, merging, and processing features. Models to note are Mark of the Unicorn's *MIDI Time Piece* (with eight separately-addressable Ins and Outs), and Opcode's *Studio 4* or *5* (with eight or 15 Ins and Outs, respectively). Mark of the Unicorn also offer the *Video Time Piece* for VITC synchronisation, and Opcode offer the *Studio A/V*.

# System considerations

You do need plenty of RAM to set up a professional MIDI-Audio workstation on the *Quadra 950*, and thankfully it can accommodate up to 64Mb. Your system software can easily eat up around 2Mb of RAM — especially with a liberal sprinkling of inits and DAs. These might include Apple's *MIDI Manager*, *MTP* or *VTP* DA's, *OMS*, *Studio 5 /OMS* Setup and so forth. *Pro DECK*, *ProEDIT*, *Sound Designer II* and *Studio Vision* all require 2Mb of RAM, *Galaxy* requires about 1Mb, crossfades in *Pro Tools* have to be computed in available RAM, looping in *Sound Designer II* uses available RAM, and so on. Taking these figures into consideration, 12Mb would be a reasonable minimum RAM requirement; double this would be

more ideal. The extra RAM would make everything work more efficiently, with program code not needing to be swapped between RAM and disk as often, and with a realistic amount of RAM available for crossfades, looping and so forth.

For audio recording, you need to use fast hard disk drives with access times of 18ms or better, and transfer rates of 600kBytes or greater. One hard disk drive per each four channels is recommended for *Pro Tools*, and each mono track uses 5Mb of disk space per minute at 44.1kHz sample rate, so you will need plenty of disk space. Digidesign's *Pro Store* high capacity hard disks are available in either 660Mb or 1Gb configurations, which offer high reliability with fast access. The *Pro Store* optical drive is recommended for backup and stereo mastering, but it will provide up to two channels of recording and playback with *Pro Tools* if you wish. Third-party drives are available from DAC and others, but you do have to check for ▶

## SUGGESTED READING

- **The Audible *Macintosh***
  by David M Rubin
- **The MIDI Manual**
  by David Miles Huber
- **Music Through MIDI**
  by Michael Boom
- **MIDI Programming for the *Macintosh***
  by Steve DeFuria & Joe Scacciaferro
- **MIDI Programmers Handbook**
  by Steve DeFuria & Joe Scacciaferro
- **The MIDI Resource Book**
  by Steve DeFuria & Joe Scacciaferro
- **The MIDI Implementation Book**
  by Steve DeFuria & Joe Scacciaferro
- **The MIDI System Exclusive Book**
  by Steve DeFuria & Joe Scacciaferro

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compatibility.

In practice, you will also need a compatible professional-quality DAT recorder to back up files for long-term storage, or to transfer source music into *Pro Tools* to edit or work on further. Cheaper DAT machines are not all compatible with *Pro Tools* for backup, and mostly feature SCMS which only allows one digital copy to be made from your DAT tapes — a highly unsatisfactory situation in a professional recording setup.

To allow you to work with synchronised video and audio and achieve rock-solid lock, you really do need to synchronise all your devices to an independent master house sync or video black burst source. For this you need to use the Digidesign *Video Slave Driver*, the MotU *Video Time Piece*, the forthcoming Opcode *Studio A/V*, or some combination of these.

One of the bottlenecks in any computer-based system is the speed of disk drive data access, which is currently limited by the speed of the standard SCSI bus. The *Pro Tools System Accelerator* is a NuBus Card which provides a SCSI II interface. This improves the workings of everything both by providing faster data access, and by taking responsibility for disk access away from the *Mac*, thus allowing the *Mac* to apply all its available CPU processing power to screen redraws and program operations. On the subject of screen redraws, I am advised that *Pro Tools* software does not use *Macintosh Quickdraw* routines to create the screen graphics, so the various screen accelerator cards which you can buy to speed up *Quickdraw* graphics will not produce any improvement in screen redraws with a *Pro Tools* system.

The *Quadra 950* can execute certain commands in *Sound Designer* software up to three times

faster than on a *Mac IIci*, for instance. The same applies to certain time-consuming operations in the *Finale* scorewriter. Processing requirements for MIDI sequencing, synthesiser editing, and for utilities such as *Track Chart* and *Cue* are much less demanding, so they pose no problem for the *Quadra 950*. Of course, if you are using Apple's *MIDI Manager*, you will get much better performance from a *Quadra 950* than from lesser models.

## Conclusion

For the well-heeled musician, or for the recording studio which must have the best available system, a *Quadra 950*-based MIDI-Audio workstation, including a selection of the components discussed here, makes a lot of sense.

With such a system you can control a large MIDI rig using custom MIDI software if required; you can edit the sounds in your synthesisers and samplers (with the benefit of a large monitor to see your synthesiser parameters or sample waveforms in detail) and you can record and edit MIDI sequences using your choice of the best sequencing packages available on any computer.

You can record and edit multitrack audio digitally, or edit your DAT or analogue 2-track mixes; you can originate, edit, and print out music scores; and you can transfer MIDI, audio, or any other computer files to and from remote locations via modem or ISDN telephone links. You can synchronise audio and video recorders with your MIDI system using SMPTE, and excellent utility software is available to assist you in putting music to picture. And at the end of your project you can master directly to compact disc.

A fast and versatile computer such as the

*Quadra 950* is definitely a sound choice to handle such a wide range of tasks, and the *Macintosh* software and hardware peripherals discussed here provide a much more well-integrated solution than that available on any other computer platform. ■
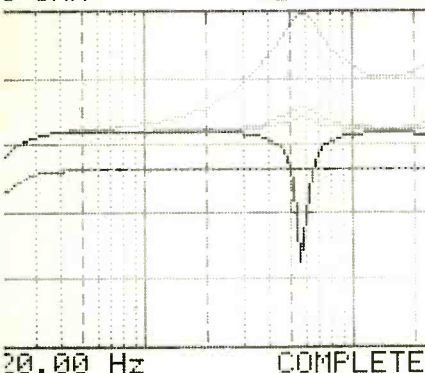
**Rebuilding East Germany — the Chemnitz opera-house looks on**

# DIGITAL OPERA

Opera lovers in the former GDR city of Chemnitz, 200km from Dresden, are this month celebrating the reopening of their 700-seat opera-house. And they have ample cause to raise their glasses, for the venue — the Neue Stadttheatre, bombed to a shell in 1945 and partially restored in 1951 — now boasts one of Europe's most sophisticated sound reinforcement systems.

The multimillion-mark refit, initiated before German unification and subsequently upgraded to state-of-the-art digital standards, encompasses a Renkus-Heinz distributed speaker system, a TOA *ix-9000 Virtual Console* and Klotz *Oak-Link* fibre-optic signal distribution.

Six hundred people work in the refurbished building, a dozen of whom are employed as technical audio staff, and the total audio installation budget came in at around DM3.5m–4m. Dr Wolfgang Ahnert, head of Acoustic Design Ahnert in Berlin, has been senior consultant to the project since its inception in 1987, working with Dr Schmidt of the Institut fuer Kulturbauen on the opera-house's interior acoustic design and personally overseeing the sound system specification and supplier tendering process.

'When the project started,' Dr Ahnert explains, 'we were planning with the audio systems then available in the GDR. We couldn't import from the West because of hard currency restrictions. The standard we had to work to was one you wouldn't really want to write about now.

'In the spring of 1990, with all the political changes happening, the opera's administrators called us to say "We're still alive!". By then the theatre had closed, some work had been done — rain was falling on the auditorium floor — and because it was felt that a town of 700,000 people could not be without an opera-house they could not stop.'

The scheme's financiers, the Chemnitz authorities and the State of Saxony, also felt the city (which, then known as as Karl-Marx-Stadt,

was one of the GDR's most important centres) merited priority investment over other venues' claims.

'So we ended up planning the whole thing twice, with new procedures and new products, finding a minimum of three companies to tender for the contract. In the autumn of 1990 we decided on Salzbrenner as installation, cabling and QC contractor in conjunction with Siemens, and Renkus-Heinz as the loudspeaker system.'

The Salzbrenner group, established in Hallstadt in 1963, embraces two companies. Salzbrenner handles audio installations and (among others) Philips and TOA dealerships, while Stage Tec, their distribution arm, distributes the *ix-9000* throughout Europe (although Austria is handled separately by Sumetzberger, which also made the sale to ▶

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## Mike Lethby visits the Chemnitz opera-house to learn about its newly installed digital sound system
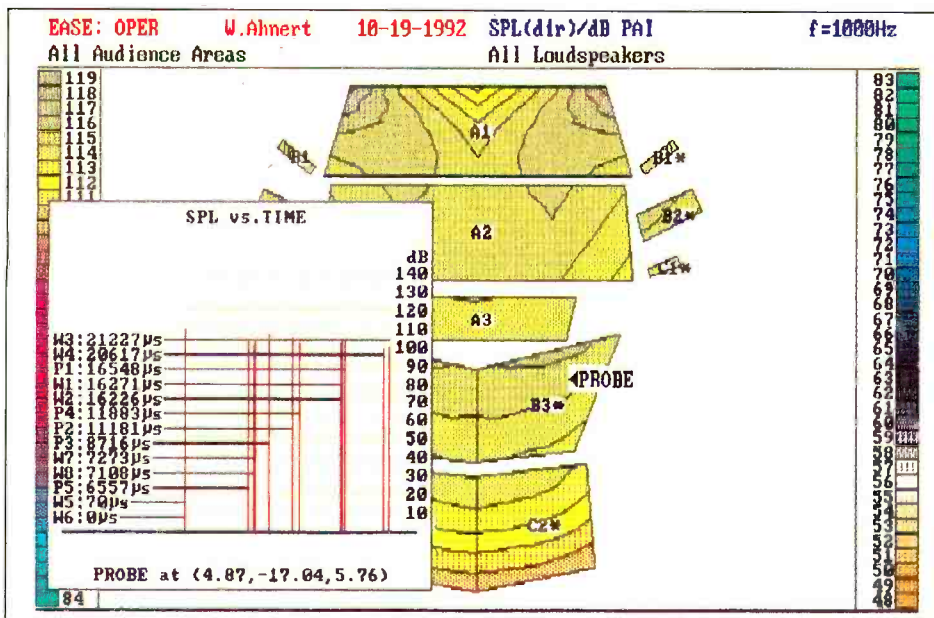
**Fig.1:** *EASE* display of direct sound distribution over the audience areas (parquet, 1. and 2. balcony) and time of direct sound arrivals of the speaker signals at the 'probe'-position in the 1. balcony
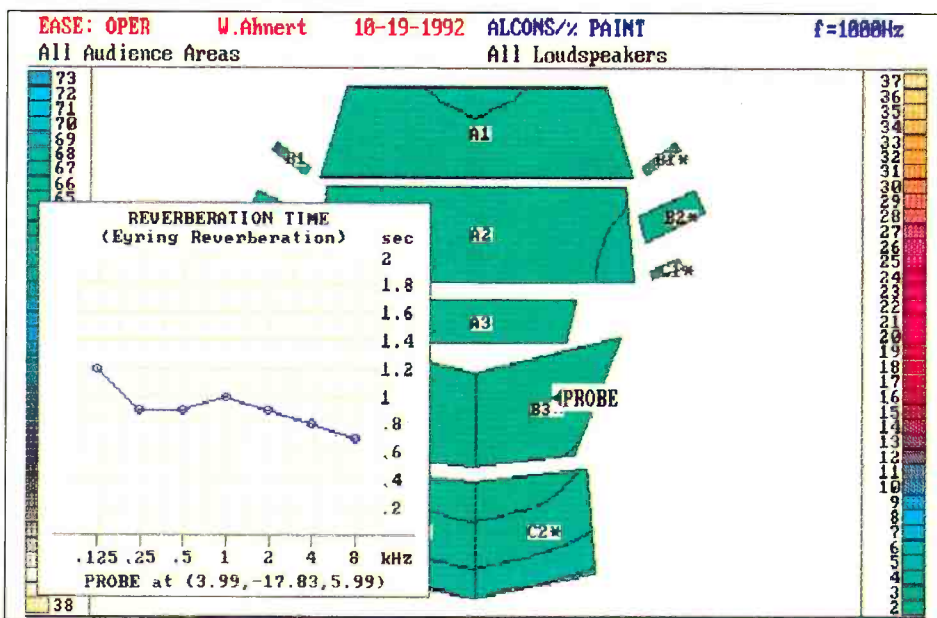


**Fig.2:** *EASE* display of alcons-distribution in the opera-house ( everywhere ≤7%, good). In the window there is shown the reverberation time vs. frequency

Chemnitz.)

Ahnert adds: 'We are very happy to have the TOA console in conjunction with the Renkus-Heinz and Klotz systems. It's the first time these have been used together in one major installation in Germany.'

## A new era

'The task,' continues Dr Ahnert, 'was to create a modern opera-house which could also serve different uses including ballet, drama, live performances and other "gala" events. The interior was completely remodelled — only the 100-year-old exterior shell remained from before.'

Within this shell is an impressive performance and production space. Behind the stage is a cavernous warehouse-like area 100m deep for props, sets and stores — room after vast room. A railway runs through the complex on which large sets can be propelled back and forth. Two floors down, massive machinery underpins the lifting and revolving stage.

Ahnert: 'We needed a system that would provide excellent coverage and good effects, all produced electronically with no mechanical aids. It had to provide the impression that all sound comes from the stage and not from loudspeakers, which required a good delay system so that original sound sources could not be perceived as coming from any loudspeaker. It all had to be as good as the room acoustics.'

Another major requirement was the ability to change, instantly, the PA's EQ characteristics to suit live music or spoken voice. The Klotz system, says Ahnert, makes it simple for the engineer to alternate between music and vocal EQ curves.

It was the same Dr Wolfgang Ahnert who invented *EASE* (Electro-Acoustic Simulator for Engineers), the PC-based room simulation programme for which Renkus-Heinz, coincidentally, is worldwide distributor.

*EASE 1.2*, therefore, was the tool he used to model the auditorium loudspeakers' dispersion patterns. It allows either the energy distribution (direct or direct plus reflective) or an 'intelligibility

projection' pattern over the whole auditorium to be viewed on-screen or printed as an isoline plot in full colour.

This new version of *EASE* also offers a simplified room-entry procedure based on five standard geometrical shapes which can be scaled and combined as required before modelling commences. Examples of the programme's analysis of the Chemnitz Opera are shown in **Figs1** and **2**.

The TOA *ix-9000* console was one of the last items to be chosen. Ahnert's tender specified that a desk should be able to save fader and EQ positions. Studer and Saje models were considered but TOA provided an offer which allowed him to buy into facilities that might otherwise have been beyond his scope.

'It was a very happy result for us and our budget,' he affirms. 'Now we have the possibility to save all settings.'

As for the Klotz *Oak-Link* component, Ahnert claims that fibre-optic technology is less significant than the computer-controlled signal distribution and processing it affords: 'The Klotz system is extendible — we can add as many units as we wish in future. The matrix is especially impressive; it combines a lot of distribution purposes at every point behind the A–D and D–A convertors including, for example, the rehearsal room, so it is easy to transfer signals from one room to any other.

'It provides input matrix and outside routing to, for example, a broadcast mobile and our own production studio. This also means that with *Oak-Link*, you can work without the console.'

The production studio, up on the third floor, is dedicated to recording tasks. There is an analogue matrix, a Soundcraft 32-channel *6000* console and a Fostex *G16S* multitrack machine with Dolby *S*.

## Sound reinforcement

Dr Ahnert's PA tender specification focussed on cabinet size, dispersion characteristics and efficiency.

'In Germany,' he explains, 'the major loudspeaker brands considered for this type of installation are currently d&b, Electrovoice, Meyer and Apogee. Although d&b was a close contender in matching our specification, Renkus-Heinz provided better efficiency and ultimate power at high SPLs — which we require for loud effects.'

The Renkus-Heinz auditorium PA system is divided into three major areas: the main 'Portal' [proscenium arch] system and two delayed systems covering the balcony and circle seating.

Permanently installed in the stylish auditorium are 13 Renkus-Heinz *SR-1A6* and 50 *CM-61* loudspeakers. The latter, powered by Philips *100V* amplifiers, provide a distributed system for 'panorama effects'. Their exact positions were based both on *EASE* data and Ahnert's experience in other theatres.

There is also a complement of 'mobile' speakers which can be moved around for individual productions as required, mounted over and around the back of the stage. They include 13 Renkus-Heinz *SR-1A6s*, 19 *SR-2As* and two each of the *LR-2A*, *TSC C-2* and *TSC C-2SUB* cabinets.

The *SR-1A6s* 60x40° horn makes it a long-throw version of the *SR-1A*, a trapezoidal design based around a 2-inch HF driver and a 15-inch woofer. The larger *SR-2A* has two 15-inch woofers and ▶

a 2-inch compression driver on a 60x40° horn. The *LR-2A* is a 2x18-inch sub-bass cabinet. All three are in the Smart Systems range.

The *TSC C-2* and *TSC C-2SUB* cabinets were the first of Renkus-Heinz's new *TSC* Series, introduced at AES Vienna this year. The former is a 3-way full-range horn-loaded trapezoidal unit, designed for large concert system arrays, offering up to 133dB SPL with two 15-inch woofers, a 10-inch cone mid and two 1-inch compression drivers mounted coaxially with the mid unit on a single constant beamwidth horn. According to Renkus-Heinz, its 'unique coaxial technology provides point source reproduction from 300Hz–20kHz'. The identically-sized *C-2SUB* cabinet has two 18-inch horn-loaded drivers, extending LF response to 30Hz.

Each cabinet has Aeroquip fittings for compatibility with the company's *ATM Series* modular flying-bar system.

Every loudspeaker also has its own processor (Renkus-Heinz *X22/PM22-15* units predominating) and its own amplifier, most of

which, here, are located under the stage. D–A conversion is performed either at the amplifiers' inputs or in the main console racks.

Cabinets located behind the stage front, Herr Salzbrenner explains: 'Give sound engineers the possibility to radiate effects in any direction for any type of production.'

Eight hundred metres of multicore cable are involved in all. Two fibre-optic cables handle the microphone inputs, 64 channels from the microphone preamps to the studio. Two more fibre-optics provide security, automatically switched in should the primary pair fail.

## The console

The *ix-9000*, built by the TOA Corporation in Japan, is one of the most ambitiously equipped live sound console designs to make it from drawing board to reality. The first 32-fader desk was developed in conjunction with TOA's Austrian distributor, Sumetzberger, for a prestige installation at the Vienna opera-house; British
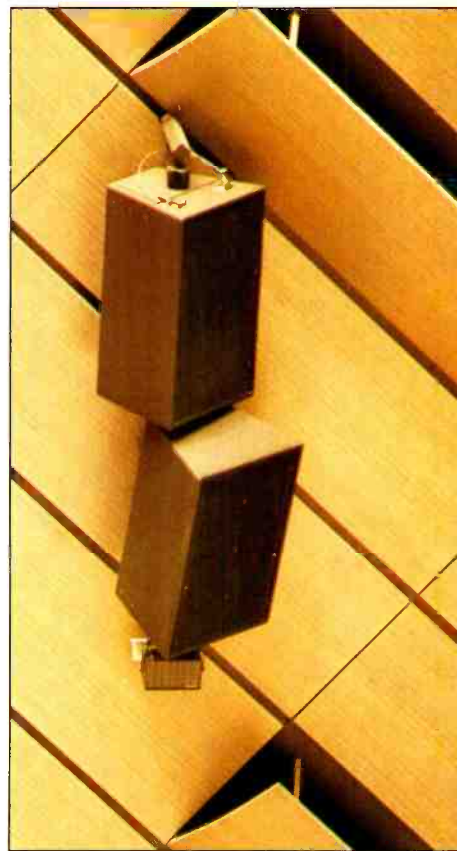


Detail of speaker installation

sound engineer Nigel Luby was also brought in as a consultant to the project. The 12-fader console installed at Chemnitz is TOA's second sale. According to Martin Sweeting of TOA UK, ten more European *ix-9000* contracts are in the pipeline. Meanwhile, back in Japan, an *ix-11000* postproduction version is already ready for trials — and a 1993 APRS launch.

Briefly, TOA have eschewed almost all the halfway houses on the road to automation. The *ix-9000* fully embraces the concept of the virtual console, in which a control surface handles no audio and is purely an interface between human being and the software processing, performed by custom DSP chips housed in remote processor racks.
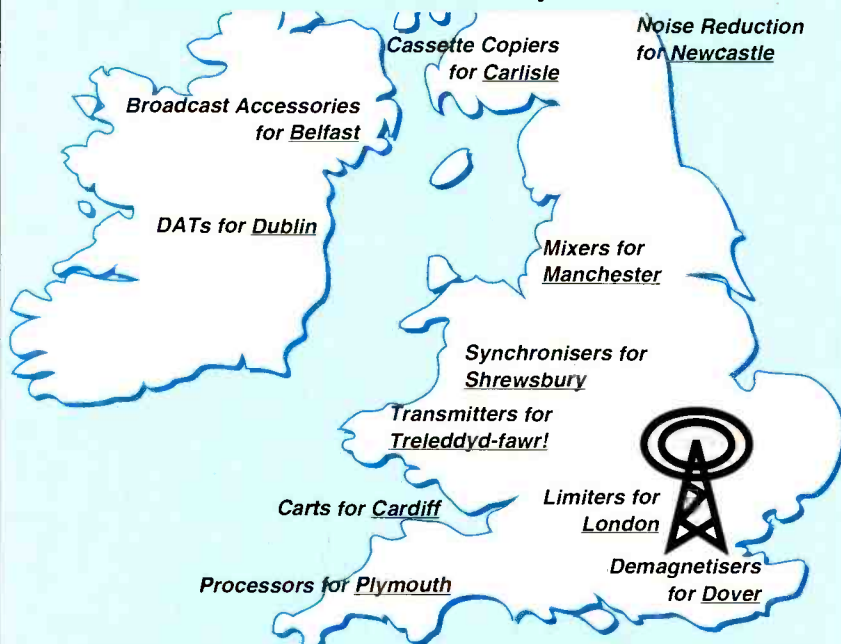
Desk and racks communicate via a digital bus and all audio is handled totally digitally by the racks, which interface with the analogue world outside via separate A–D and D–A convertors. The latter can in turn be sited at the closest practicable point to the original sound sources, minimising signal degradation. Moreover, the racks' processing functions are integrated with a comprehensive software-controlled routing matrix which can place every aspect of an audio system under central console command — from system EQ and loudspeaker switching to input routing and tape machine control.

The matrix offers a maximum of 256 I-O lines (the actual number depending on the number of processor cards fitted), and these can then be grouped in any configuration required under a system of 'pages' using the console's two touch-sensitive plasma screens.

In Chemnitz it is set up to handle any conceivable application that the venue might be called upon to provide, including direct recording onto the Akai *DD1000* or various tape machines, and it can function as an on-air console, with digital outs to the transmitter. The two screens allow two-person operation with the console ▶

**The Production Studio**

split into two independently working parts (and provide duplication should one screen fail). A separate monitor screen displays the complete routing status of the console for each performance sequence.

Wolfgang Salzbrenner: 'It means that while you only have 12 faders on the console, you only have to press one button to access all eight pages on the screen. On Page 1, for example, you can have the tapes; on Page 2, the microphones; on Page 3, your wireless microphones.'

Input connectors, switch assign are all accomplished instantly on-screen. Selection of most routing functions and their attendant values involves a combination of touch-screen, 'soft' (context-sensitive) push buttons and the familiar rotary input dial that is common to many digitally-controlled devices. The monitor shows whether any component is working or not, whether a loudspeaker is connected to the amplifier, and the setup also provides sequential

power-up for the amplifiers, each delayed by one second.

Inevitably, this short description of a complex idea looks complicated in cold print. However, when you get your hands on the console, it quickly becomes clear that TOA have done an excellent job of making most operations fast and instinctive in practice.

An assignable 'virtual' control surface, which by definition means you cannot see all of your settings all of the time, will not suit everyone. Trident's *Di-An* won few friends in studios; but TOA's new vision of this concept in the *ix-Series* looks likely to refute many preconceptions.

The first question about a 'virtual console' in live performance — what happens if you need instantly to fade down one singer's microphone during a show? is answered confidently by Salzbrenner: 'It's actually very quick. You simply enter the page, the microphone's reference letter. And there you are.'

The second question is: how do the installers justify such an investment in digital technology for a live performance venue?

'One, it's very simple for the Tonmeister [sound engineer] to work with this system; he can store everything he needs,' replies Salzbrenner. 'Two, as for analogue and digital options: if you make a complete digital system, then the price is nearly the same.'

## Investing in Europe

Chemnitz Opera marks a significant contract for Renkus-Heinz's European office (in fact, it is their largest installation to date in Germany). The California-based company is steadily introducing new products in order to strengthen its profile as a serious SR contender, and this was an opportunity to prove its mettle in the context of a quality-orientated system with an all-digital front end. Similarly, it is important for TOA who have invested considerable R&D funds in the *ix-Series* project — in which Chemnitz represents the second European sales contract.

That such prestigious projects continue to go ahead (if at a more cautious rate than before Black Monday) must be a small but nonetheless encouraging sign for the SR industry. The opera-house engineers were unequivocal in praising their new audio system's sonic quality. Chemnitz's citizens will have their own chance to judge the results starting this month. ■

**Distributors:**
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## STUDER
### PROFESSIONAL AUDIO EQUIPMENT

# APRÈS MIDI

## Chris Meyer evaluates at the application of MIDI as an alternative to present professional studio automation systems
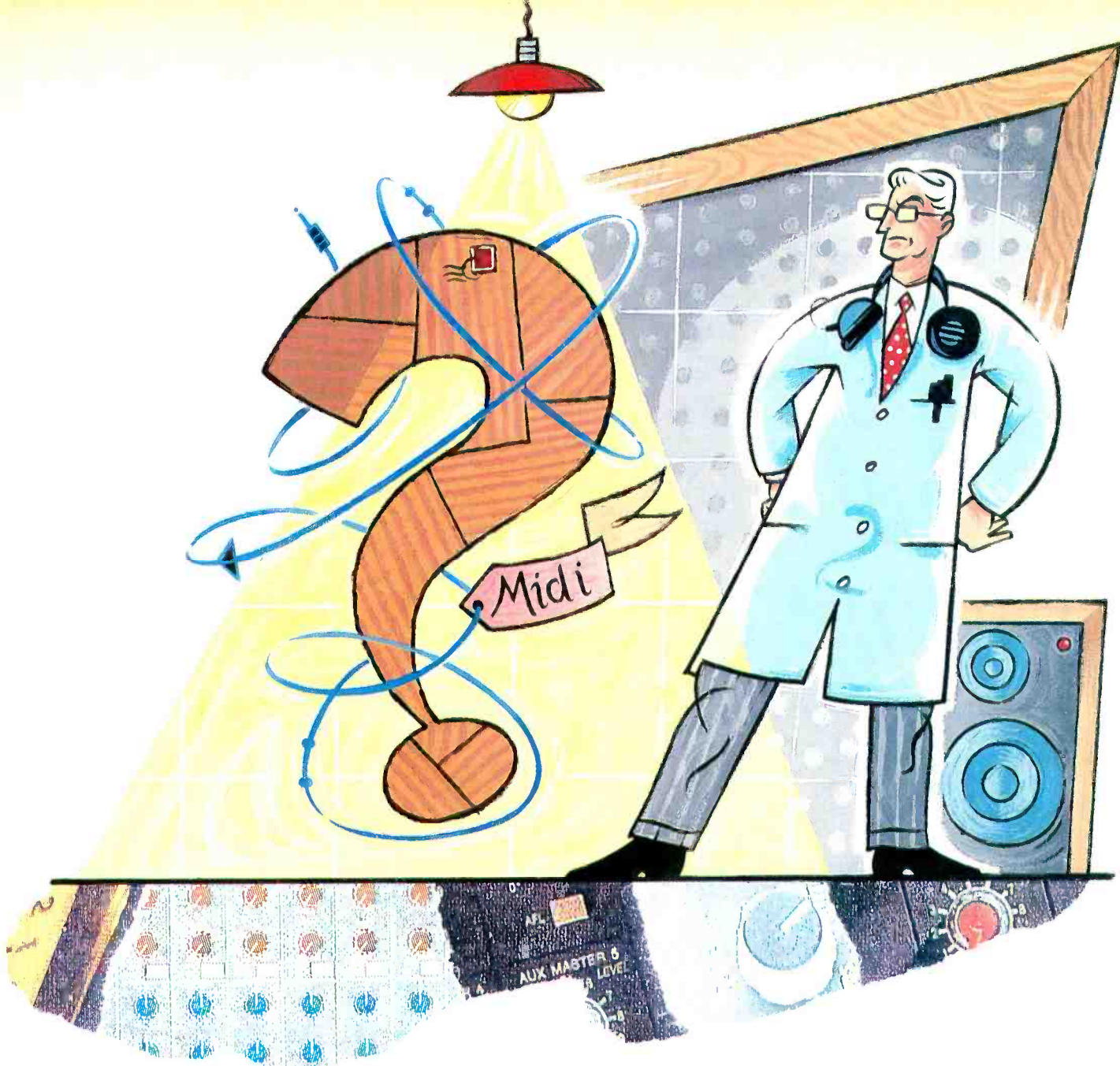
**M**any studio owners have invested heavily in automated mixing systems that record console settings and little else. Many of the same people own 'universal' tape deck synchronisers that require a new cable and software for each tape transport.

Despite the differences between the implied claims of such systems and the reality of their operation, no one would argue that these modern devices have made life in the studio a lot easier (especially during mixdown). But what if someone claimed they had a system that would automate most of the effects processors in a studio? Such an add-on would synchronise to your current SMPTE-based system, and change effects programs with timing as tight as a millisecond. In some cases, it could also automate the changing of the effects parameters themselves — such as noise-gate release time, chorus depth, or reverb predelay. With an additional investment, you could even automate the patching of the effects — which device received audio from which send and appeared at which return on your console (possibly even with automated levels). And as a final clincher, this system might already be compatible with most of the effects devices you already own — from virtually any manufacturer (no proprietary systems here) — and would be

likely to be compatible with most of the effects to be released in the next several years.

Do not go searching through the ads or new products section of this magazine; no one is advertising this revolutionary breakthrough. Why? Because it has been available for nearly a decade now. In fact, many so-called 'project' studios already use it; you just might not recognise it. It is called MIDI.

Yes, I know — MIDI is that thing the great

unwashed on the other side of the control room glass (musicians) use so that they do not have to learn how to play their instruments. Or maybe it is to satisfy a latent sexual desire to mate with a computer. The fact remains, one of the greatest secrets that have been kept from the pro studio world is that musical performance is only part of MIDI's capabilities, and MIDI can also act as a very intelligent GPIB or serial control interface to automate parts of your studio previously ▶

29

untouched by 'professional' solutions. If you already use it, you know what I mean; if not, go get a drink, forget all you have been told about MIDI's musician orientation and 'semiprofessional' nature, and read on.

## Effects automation

You have probably noticed that most digital effects devices these days have MIDI jacks on the back. At a minimum, these boxes allow you to change the effects preset remotely by using a part of MIDI known as the Program Change message. The MIDI language is capable of selecting from a minimum of 128 different programs per device (a recent 'Bank Select' extension code has expanded this to over two million, although few boxes currently support this), and talking to 16 different devices independently on one physical cable. At first, you may think this may not be of much use (since you have two hands — or better yet, a second engineer — who can do that for you), until you consider automating these changes. Aside from there being one less detail to remember (or one less second engineer to hire) during the mix, you also start to precisely time effect switches on downbeats or scene changes without needing to sharpen your video game button-punching skills.

Another trick some of these devices allow is to change the parameters of the selected preset using what are known as Continuous-Controller messages. This is a group of roughly a hundred 'soft knobs' that can have a resolution of 128 values (32 of these may be extended to double precision — over 16,000 values — but, again, few devices currently take advantage of this). A good number of these controllers have useful defaults (such as master volume, wet-dry mix, effects

> To automate all this you need a MIDI sequencer— the same sequencer musicians use for music also records all the commands needed for studio automation and allows pretty slick editing to boot

depth), and most devices will allow you to assign any controller number to any parameter it allows to be controlled. Creative possibilities run from riding the wet-dry mix to changing chorus rate or reverb depth at specific points in the song.

The third feature most commonly available on MIDI effects devices is the ability to store and recall banks of presets. There are two types of users in the world — those who prefer to just find a suitable factory preset to use, and those who prefer to whip up a customised effects program for each situation. For the former crowd, this store-recall ability allows you to collect more patches to choose from (manufacturers usually cook up more than they can fit in the box itself,

and will make them available to their users); for the latter crowd this adds the ability to save your handiwork in case the record company rings you up six months later wanting a remix and you have already wiped out all your old programs (having moved onto a new project).

To automate all this, you need a MIDI sequencer — the same sequencer musicians use for music also records all the commands needed for studio automation and allows pretty slick editing to boot. Sequencers come as stand-alone boxes or software programs run on personal computers, and occasionally as integrated systems such as that on Korg's *SoundLink*. Regardless of their physical nature, virtually all can be made to slave to SMPTE time code. There are also a few sequencers custom-built for studio automation, such as Digidesign's *Q-Sheet* program from the *Mac*.

There are, of course, downsides and shortcomings to all of this. For starters, not all signal processors and effects devices allow MIDI control. Digital processors find it easier to go the MIDI route — most of their parameters inside are being manipulated by a computer and numbers anyway, so letting MIDI instead of the front panel tickle these is easy. Despite this, not all digital processors allow you to control every parameter, and some glitch when you do. It is altogether more difficult with analogue devices such as equalisers and compressors; to alter the parameters of these devices, you need to find a way to map digital MIDI commands onto 'normal' potentiometers and resistors. In most cases, this is done with VCAs (frequently driving the cost upwards and the sound quality downwards). Nonetheless, you can still find a few MIDI-controlled EQs and compressors from companies such as Digitech and

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Drawmer. Although automating effects presets and setting is a useful addition to most of today's consoles, you can automate their routings as well: MIDI-controlled audio patchbays have been welcomed by many engineers. Most of these allow any input to be buffered and to go to any number of outputs. This means you can switch your limited number of console sends — or even a loop on each input, if you are feeling adventurous — to any of your out-board processors and route them back into your returns (or a few spare channels if you have them). This can save a lot of patching time and pays dividends if you are making the most of your studio and have several projects running simultaneously. To affect the levels of the effects sends, returns, and submixes, there are numerous MIDI-controlled VCA boxes available; there are also a couple of matrix mixers out there where each output can receive a unique mix of each of the inputs, which in turn creates a really flexible effects send system. There are even a few completely MIDI-controlled mixers that can serve as effects-return submixers (although most do not have specs up to professional applications). Patch all of the above together in the right order, and you do not even need a mixing console anymore — in fact, I have done just that in my own studio (but that is another story).

## Traditional territory

It is interesting that MIDI, often considered a plaything by professionals, can provide the missing link in a 'total' automation system eminently suitable for professional studios. Possibly more interesting is that MIDI is now starting to creep into previously professional-only territory — bringing with it a fresh approach. MIDI was founded on the premise that any properly-equipped musical instrument should be able to talk to any other one, without knowing beforehand exactly what it was going to be talking to. Conversely, an instrument being talked to should not care who is doing the talking — it should just take the commands and do the best it can with them. This approach is inherently non-manufacturer-specific, which is a significant different between MIDI and most professional studio equipment.

A recent extension to the MIDI specification was MIDI Machine Control (MMC). Based on the ESBus protocol, it provides a very complete set of transport and remote-control commands including optional bells and whistles such as macro commands and error reporting. A set of defaults exist for controlling analogue or digital tape — or disk-based recorders along with video decks and MIDI sequencers. The first MMC applications to appear allow MIDI sequencers to control analogue tape decks as if they were just an integrated set of parallel tracks. There has always been a bit of an awkward wall between sequencers and tape, and MMC is finally helping to bring it down. In this way, sequencers can easily wrap their operating systems around tape deck control. However, there is no reason why the tables can not be turned such that tape deck synchronisers can be used to control MIDI sequencers — along the lines of the automation on the new Trident *90* — aiding integration into more established professional studio practices.

The next threatened conquest of MIDI is in the area of professional-level mixer automation. Would it not be useful if you could mix and match your console to the automation system you desire? Or exchange mixes between studios that had different equipment, and still make some sense of them? MIDI users expect similar things of their sequencers and musicals instruments, and expect the same to be possible with their mixes. There are, of course, certain barriers — getting the resulting hoard of MIDI messages down a MIDI cable in a timely manner is one (although many manufacturers have solved this by using multicable systems), and just plain mediating between different manufacturers' approaches is sometimes another. But do not count MIDI out: many of those who have are currently losing business to lower-cost project studios.

We take it as our God-given right to be able to route an audio signal anywhere we like in a studio — it may take some connector and signal level matching to achieve (and digital audio has thrown a curve into it), but we can do it. There really are not many reasons why we should not have the same capabilities with control signals in our studios as well. Getting manufacturers to work together may seem tricky, but the MIDI crowd have been doing it for a decade now. It is time for the pro studios to stop growling at MIDI and project studios and instead integrate it into their own rooms — no one will send you to jail for it, and in the long run it will make your life (and mixes) better off. ■

**Chris Meyer is Chief Engineer at Roland R&D in Los Angeles. He is also a member of the MIDI Manufacturers' Association and has written various aspects of the MIDI specification.**

# GOING TAPELESS

## Yasmin Hashmi reports on new developments in tapeless audio and integrated tapeless-MIDI systems from AES, San Francisco

American computer manufacturer **Atari** chose last year's San Francisco AES Show to unveil their new *Falcon 030* computer (which costs around £900 in the UK for a system with a 65Mb hard disk and 4Mb RAM). **D2D Systems** have designed disk-based recording-editing software for the *Falcon* called *D2D-EDIT*. This provides waveform editing, an event list, marking on the fly and primitive audio scrub. The system will lock to MTC or LTC via a Steinberg or C-LAB interface, and future enhancements are planned.

A simulation *RADAR* (acronym for Random Access Digital Audio Recorder) system was on show from **Anatek**. The system is aimed at replacing analogue multitrack, will have a simple user interface and support up to 24 tracks of recording-playback. It currently uses three 1Gb hard disks to support around 20 minutes of 24-track recording and Anatek plan to allow up to eight machines to be connected together to provide a maximum of 192-track playback. They recommend DAT for backup purposes and will offer an optional *DataDAT* system for fast and intelligent archiving. *RADAR* should be in production by March 1993 and will cost around $15,000 for a 24-track system.

**Tascam** were showing a prototype of their new *RA-4000* which supports two inputs and four outputs. The system has an optional dedicated remote called *RC-4000* which provides editing keys, a small LCD, a locator and a jog wheel. VTR emulation software is planned as well as digital I-O and P2 protocol serial control. The cost will be around $5,500 and $1,500 for the *RA-4000* and *RC-4000* respectively and the system should be in production by March this year.

Two new packages designed to work with existing products are the *Digital Waveboard* from **Mark of the Unicorn** and *SAW* from **Innovative Quality Software**. The former is a 4-channel hardware card for the *Mac II* or *Quadra* computers which supports two inputs and two outputs. It runs with Mark of the Unicorn's *Digital Performer* software and costs around $1,500. Conversely, *SAW* is a software package which runs on the *CardD* from **Digital Audio Labs**. It provides graphic editing with waveform displays (calculated while recording takes place), simple MIDI sequencing and the ability to preview a mix. *SAW* costs around $600 (available in the UK once distribution is finalised).

*Metrotracks* from **Metrosoft** is a new software package which is still under development and is designed to run on a *NeXT* computer. It supports editing, mixing and effects and eight internal channels across 32 virtual tracks with two inputs and two outputs. Up to eight tracks can be mixed down in real time and dynamically automated with real-time EQ and effects. The system supports MIDI and reads and generates time code. It does not, however, currently support audio scrubbing or an event list. *Metrotracks* costs around $600 (excluding the *NeXT* and A–D conversion, for which the *A-D64X* from **Singular Solutions** is recommended).

## Multichannel developments

**Digital F/X** have merged with both **Hybrid Arts** and **Waveframe** and were demonstrating the *Waveframe 401* and the *Digital Master EX* (a new version of *ADAP IV*). The latter is aimed at the music industry and runs on an Atari *520* upwards. It supports 4-channel operation with 16 virtual tracks and nonreal-time mixdown. It also supports full LTC chase-lock and a complete system (including Atari) costs around $6,000, although the hardware and software can be purchased separately for use with an existing Atari system (including the new *Falcon*). The *Waveframe 401* provides 8-channel editing, mixing with EQ, full synchronisation and machine control and costs around $15,000 for a complete system with 40 minutes of recording.

*AudioFile PLUS* from **AMS** now supports 24-track operation which can be provided by a combination of both hard and optical disc. The system will also control the Pioneer laser disc recorder for random access picture control and operates with the *Logic 1* mixing console which now supports 4-layer operation, giving 48 mono or stereo inputs.

**Otari** demonstrated a 24-track system in the form of the *ProDisk 464*. On show was the new control console sporting a mixing section (complete with faders) and an editing section. The new editing software displays all 24 tracks on screen, with waveform displays and global editing of any selected cues or sections. The system also offers autoconforming, although the only format currently supported is CMX. By using removable

disk drives, off-line loading can now be performed using the new **ProDisk** Backup Station which runs on a *Mac SE*.

The *408 OMX* from **Augan** now also supports CMX format-only autoconforming and can control a video machine or a time-coded DAT, recognising all eight audio flags. New developments also include biphase in and out and the system on demonstration was controlling a Pioneer laser disc system via RS422.

*DAWN II* was unveiled by **Doremi**. It has a new custom controller which provides track select keys, edit keys, a jog wheel and transport controls. It also has VTR emulation software and can control external machines using Sony and ▶ Ampex protocols. Software developments include on-screen transport controls and an enhanced screen display which allows cues of a similar nature to be identified by assigning them a unique colour. **Doremi** also announced that they will be developing a production system in conjunction with **Pacific Recorders & Engineering Corp** which will consist of *DAWN* and a mixer control surface.

Enhancements to **SSL**'s *Screensound* include a fourfold increase in the speed of editing operations, multi-input recording, VTR emulation and extended machine control (the system can simultaneously control up to four different machines, each with its own offset). The system now also supports a 'Super MO' option which provides fast copying functions for library purposes and can be positioned near to the user by using an *Ethernet* extension. In addition, it can be used as a working drive in place of a hard disk, but supports four channels rather than eight.

**Studer** were demonstrating a 48-channel *Dyaxis II* and were previewing their comprehensive autoconform software which operates in the background. *Dyaxis II* is *Mac*-based and supports up to six processors, each controlling eight channels of audio with each channel having five bands of EQ. The system uses virtual tracks for editing and has a 'virtual mix' feature which automatically performs a mix should the channel capacity of the system be exceeded. A warning appears on screen and the system takes one seventh of real time to perform the mix. Each processor costs $29,000 and includes everything required except the *Mac* (which must be better than a *IICi*). Studer also plan to offer Dolby's *AC-2* compression in conjunction with a 3.5-inch optical for a cost of $6,000 per processor.

Following a dispute with **Soundtracs**, Australian-based **Soundfirm** have renamed their *Soundtracker* system *EDI-Tracker*. The system uses a touch screen and is geared towards film sound editors. Enhancements include improved audio scrub, intelligent global insert and cut features (useful for adapting last minute changes made in the picture) and selective printing of dubbing charts and the events list. Also new is nonreal-time mixdown and an optional software package called *Copycut*. This can be used for automatically generating an EDL in order to ▶

**MFX2 from Fairlight**

autoconform time coded DATs. It does this by recognising discontinuities in the time code and by reading the user bits (which have been used to give each DAT a unique identity).

The *MFX2* from **Fairlight** has a new time compression-expansion feature which, by operating in the frequency domain rather than

time domain, allows compression of up to 50% to expansion by as much as 200%. The system also supports Fairlight's new *Take Software*. This allows multiple takes to be stacked on the same track and in the same place. For auditioning purposes, the takes are listed along with their respective layer number in the cue list.

**Roland** announced their new track manager software for the *DM-80*. It runs on any *Mac* and provides waveform displays with global editing, an events list and metering. Control between the *Mac* and the *DM-80* is via MIDI and an optional locking resolver is available for locking to wild LTC. The cost of the *Mac* software will be $650 plus around $100 for the MIDI interface.

**Digidesign** announced their *DINR* software which runs on *Sound Tools, Pro Tools* and *Sound Tools Pro Master 20*. *DINR* costs just under $1,000 and provides hum cancellation and broadband noise reduction. It works in conjunction with Digidesign's DSP module which supports 20-bit recording and 24-bit internal processing. Enhancements to *Pro Tools* include new software which runs features from *Pro Edit* and *Pro Deck* under one operating system. Digidesign will also be introducing *TDM* which supports 256 channels of digital audio and will operate with third party cards from companies such as Lexicon, Apogee and Opcode.

**Lexicon** were demonstrating v.3.0 software for the *Opus*. Among its features are CMX autoconform software and the ability to use colour displays. Lexicon are currently working on ADR software which will include features such as take listing. This will allow setting punch-in and out points with the VTR automatically rolling back for a retake. **Korg** were also demonstrating new their software for their *SoundLink*. This supports enhanced mixing and editing functions as well as external machine control and digital effects.

**Spectral Synthesis** unveiled software and hardware enhancements for their *Audio Engine*. The system now provides non-sequencer-dependent automated mixing, machine control, CMX EDL autoconform and same track ▶

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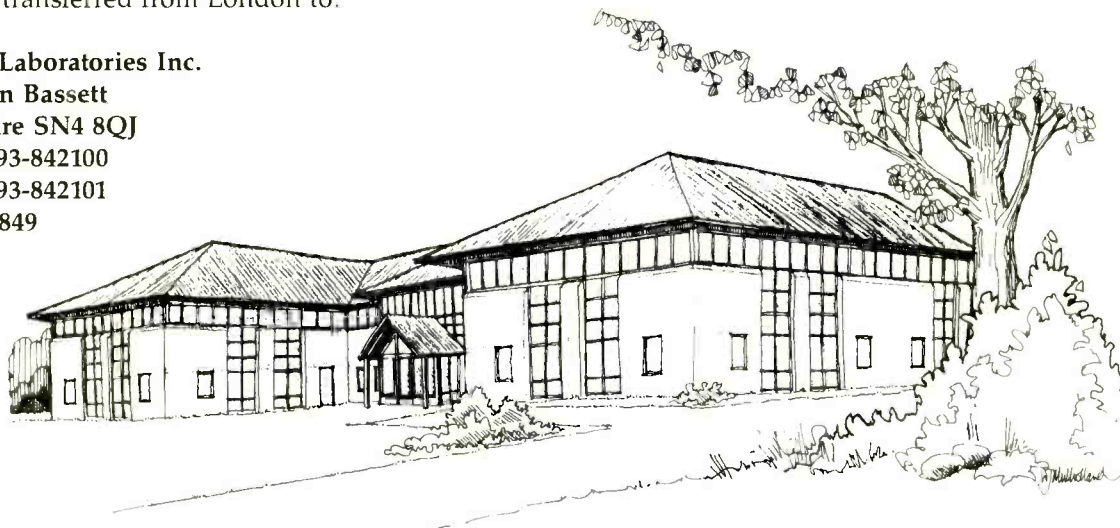
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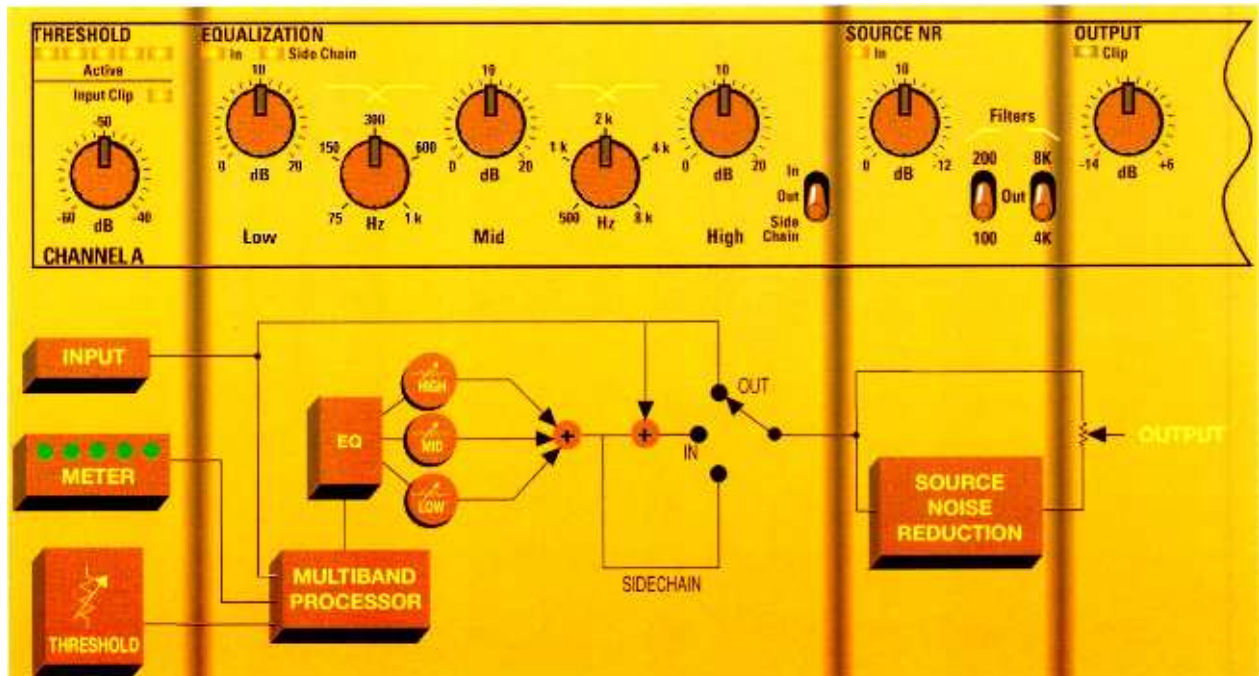


## SPECIFICATIONS

### Layout
1 U-high unit containing two independent channels of Spectral Processing. Stereo link switch for use with stereo material.

### Operating levels
Switch on rear of unit selects between High (+4 dBu) and Low (-6 dBu) line-level operation. Maximum input levels: High 24 dBu, Low 14 dBu. Maximum output level: (High or Low) 24 dBu. Input and output clip LEDs indicate when maximum levels have been exceeded.

### Inputs/Outputs
XLR connectors. Balanced, floating. 10 kohm input impedance, 20 ohm output impedance.

### Threshold
Adjustable from 60 dB to 40 dB below the nominal operating level. Five LED meter array indicates when processor is active.

### Equalization section
Three-band filter section with adjustable crossover frequencies. Low - Mid crossover adjustable from 75 Hz to 1 kHz, center detent at 300 Hz. Mid - High 500 Hz to 8 kHz, center detent at 2 kHz. Maximum boost of low level signals in each section: greater than 20 dB, with crossover controls set to detent positions.

### Operating modes
In: processing active, main path and side chain are added within the unit.
Side Chain: Side chain signal only to allow external manipulation.
Out: all processing bypassed.

### Source NR
Sliding band noise reduction section reduces noise present in the input signal by up to 12 dB.

### Filters
Selectable high- and low-pass filters prevent unwanted signals from entering the low level processing stage. High-pass: 100, 200 Hz. Low-pass: 4 kHz, 8 kHz.

### Output
Overall channel gain with processing out adjustable from -14 dB to +6 dB.

### Dimensions
44 x 483 mm rack mounting (1.75 x 19"). Maximum projection behind mounting surface 248 mm (9.76"), plus a further 65 mm (2.5") for the connectors.

### Power Required
User-selected voltage nominally 100, 120, 220, 240 V. Consumption 36W approx.

### Operating Temperature
0 to 40 degrees C.

When developing our famous noise reduction systems, we learned how to process low-level signals while leaving high-level signals untouched. Now we're putting that special knowledge to work in a new kind of dynamic equalizer.

The two-channel Dolby Spectral Processor lets you raise low-level signals in three frequency bands by as much as 20 dB without affecting high-level signals. It's like a magnifying glass for sonic details. You can emphasize ambience, lift harmonies, and bring out

otherwise inaccessible subtleties - all without squashing transients, increasing overall track level, or disturbing the overall sense of dynamics.

For maximum effectiveness, you can adjust the threshold below which processing occurs, the amount of boost in each band, and the crossover between bands. With noisy material, you can also switch in a gentle sliding-band noise reduction circuit. From adding that extra touch of presence on a vocal track to sweetening a final stereo mix, the Dolby Spectral Processor provides the kind of EQ you've always wanted. Contact your Dolby Professional Products dealer soon for a demonstration.

"I was able to achieve sounds and manipulate program in a way that I never could with an equalizer or compressor."

"On a vocal it can help brighten and bring out a section where the vocalist backed off the microphone."

"It makes a big difference to the sound, and you can control its effect very convincingly."

"It's great for pulling ambience and room sound from a track."

"I'm blown away by it! Depending on the track, it can open every thing up or tighten up the bass."

The

# EQ

## You've Always

## WANTED

punch-in-out. It supports recording to optical and the use of JL Cooper's *CS-10* for functions such as multitrack scrub. Spectral announced two new products for use with their disk-based systems, namely the *ADAX-8818* for eight analogue inputs and outputs as well as stereo AES-EBU and SPDIF and the *Synclock*, which converts multiple clock sources and accepts composite video or black burst, LTC, AES-EBU, word clock, oversampled word and tach.

**Sonic Solutions** demonstrated a host of new features for their *Sonic Station II* which are aimed at expanding the system's applications for sound to picture. In addition, Sonic have agreed an exclusive relationship with JL Cooper for a new moving fader hardware controller which will use Penny & Giles faders and will have expansion modules. The *Sonic Station II* now supports between 10 and 14 internal channels as well as 20-bit recording and the *Sonic Station II PLUS* can be expanded in modules of four channels. Sonic also announced two installations of their new *SonicNet* networking system which supports 100 channels of 44.kHz audio using fibre-optic cable.

## Multichannel Systems

Also on show from **Sonic Solutions** was their *CD Printer* (which writes at double speed and supports 80-minute CDs) and their CD premastering system. This supports DCC, MD and CD-ROM premastering and provides audio compression for A-B comparison purposes.

**Akai** presented two new versions of the *DD1000*, namely the *DD1000i* and the *DD1000s*. The former includes all of the usual *DD1000* features plus external machine control and time compression-expansion. The *DD1000s* is designed to be a slave unit without the front panel control. It has the same functions as the *DD1000i* and can be controlled by the *DL1000* remote or the *Mac* option.

**Digital Expressions** were showing their *Mac*-based *SoftSplice* system which has two digital inputs and outputs and four internal channels. The system will run on any *Mac* and provides waveform editing with an events list, 2-band EQ and user-definable gain and pan envelopes. LTC chase lock is supported and future plans include automated level and panning.

**Studio Audio & Video**'s *SADiE* was being demonstrated on **Aras Inc**'s stand. It featured enhancements to editing displays and synchronisation to LTC and MTC (with chase lock

due for imminent release). Also shown was a prototype of scrub and shuttle functions and the system is now capable of replaying from two disks simultaneously.

On **Studer**'s stand were *Dyaxis I* and *Dyaxis Lite*, both operating with their new custom controller. *Dyaxis I* featured enhanced *MacMix* software and can now control the Studer *D740* CD-R system for Red Book Standard CDs. *Dyaxis Lite* operates on a *Mac Classic* and now supports 3.5-inch optical disc as the primary recording medium. For short format applications such as news items and interviews, Studer announced network transfer between all of their *Dyaxis* systems via *Ethernet* or *AppleTalk*, taking 1.5 and ten times longer than real time respectively to transfer audio files.

**Plasmec**'s *ADAS-SA* was on show on the Penny & Giles stand in both rackmount and desktop versions. The complete system (£3,500 in the UK) provides playlist management of jingles, waveform editing, analogue and digital I-O and LTC chase and trigger lock.

The *56K System* from **Turtle Beach** featured new developments including LTC chase lock and time compression-expansion. Turtle Beach also announced two new products, namely *Multisound* and *Wave*. The former costs around $600 and is designed to meet multimedia needs. It works under *Windows 3.1*, will accept 48 channels of MIDI and is capable of both synthesis (behaving just like a *Proteus* synthesiser with 32-voice polyphony) and stereo disk-based recording at 44.1kHz. *Wave* also runs under *Windows* and is basically the *56K System* without time code slaving or time compression-expansion.

**Opcode** were demonstrating *Studio Vision 1.4* software which works with *Sound Tools, Pro Tools* and *Audio Media*. It now features loop record and edit, real-time editing of audio and MIDI information with markers and locators and 4-track playback. An unlimited number of virtual tracks can be used with up to 16 tracks being displayed on screen along with MIDI information.

The *CardD* was on show from **Digital Audio Labs**. It is an *AT*-compatible card which costs around $800 and provides stereo disk-based recording with sampling rates of 32kHz, 44.1kHz and 48kHz. Their *EdDitor* software costs $250 and provides waveform editing and their *I-OCardD* provides S-SPDIF I-O for just under $300. The system does not currently slave to time code (although this is imminent), but can map to MIDI and be triggered by a sequencer which could run on the same platform. The new software runs under *Windows 3.1* and features auditioning between edit points and a cart display.

Unfortunately the controller of **FEG**'s *Audio Solution* had been mislaid, so it was not possible to have a demonstration of the system. It was, however, shown at Photokina and now features 4-track recording direct to optical. The system has been ported over from the **Atari** platform to a 6820, 32MHz processor with a VME bus structure in order to allow the use of proprietary cards. ■



**The 408 OMX from Augen**

# One stereotype you can't ignore.

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# 20/20 VISION

## Lexicon's Jim MacArthur discusses the design philosophy behind the Lexicon *20/20 AD* 20-bit convertor

The world of audio production, which previously has been satisfied with 16 bits, is gradually changing over to higher resolution formats. Every month sees the introduction of a new 20-bit recording system. Mid-priced, disk-based digital audio workstations are now available with 20-bit resolution (such as Sonic Solutions' *Sonic System*) and many recently announced, high-end digital consoles also support 20 bits.

As for the 16-bit limitation of compact discs, keep in mind that the encoding methods to be used on MiniDiscs and Digital Compact Cassettes claim to make use of up to 18 bits of incoming audio. Even without this encoding, it stands to reason that the performance of the production medium should comfortably exceed that of the distribution medium.

### Resolution versus dynamic range

Most manufacturers classify convertors by their resolution, meaning the number of bits the convertors spit out but this does not mean that all of the bits carry valid information. Generally, the lower few bits contain no musical information, only convertor noise.

A more accurate measure of convertor performance is its dynamic range. In the analogue world, dynamic range is defined as 'the ratio between the maximum signal and the noise when no signal is present.'[1] However, virtually all convertor IC (and most convertor box) manufacturers call this ratio the signal-to-noise ratio (S-N). Their definition of dynamic range is the 'ratio of the full-scale signal to the broadband noise (0Hz–20kHz) measured with a -60dB signal.'[2] This is similar to S-N ratio, but adds certain types of low-level distortion (such as zero-crossing distortion) that S-N ratio measurements in the absence of signal can miss.

This definition of dynamic range provides a basis for determining a convertor's Effective Number Of Bits (ENOB). This tells you how many of the convertor's bits work for a living, and how many are along for the ride.[3]

$$ENOB = (Dynamic\ Range - 1.72) / 6.02$$

or

$$Dynamic\ Range = (ENOB * 6.02) + 1.72$$

Using the first equation, we can see that the convertors on a typical DAT recorder, which have a dynamic range of 90dB, have an ENOB of 14.7 bits. In other words, 1.3 of the 16 bits you are recording are noise and distortion. The second equation tells us that in order to guarantee 16 bits of music, we need convertors with 98dB of dynamic range. Very few 16-bit convertors offer dynamic range anywhere near this figure, hence the need for higher-resolution convertors.

### Dither

So a 98dB convertor makes sense in the 16-bit world. How does the Lexicon *20/20 AD* (at 112dB) justify its existence? One way we take advantage of the extra performance is with dither. Dither can be loosely defined as a low-level signal which is injected into a system to reduce distortion. In an audio system, the signal is random noise, and the distortion is quantisation error: the error created by a DAT when it throws away all but the first 16 bits of an audio signal. This concept of improving a signal by adding noise smacks of magic, but you have probably used a mechanical form of dither countless times and thought nothing of it. The use of dither dates back at least to World War II (and probably earlier). Back then, bombers used mechanical computers (boxes filled with hundreds of precisely milled and interconnected cogs and gears) to perform navigation and bomb trajectory calculations. Engineers noticed that the accuracy of these computers fell off sharply when they were removed from the aeroplanes. After some research, they concluded that the vibrations of the

aeroplanes were necessary to reduce the error caused by slip-stick motion — the tendency of mechanical parts to move over each other in a series of short jerks instead of a continuous motion.

The solution was to mount small motors onto the computers to replace the missing vibration. This vibration was called dither from the Middle English verb 'didderen' meaning 'to tremble'. Every time you tap a mechanical meter, you are increasing its accuracy by adding dither. No magic involved. **Figs1a –1d** illustrate how dither works in the audio domain. **Fig.1a** shows the frequency spectrum of a -60dBFS (decibel referenced to digital full scale) 1kHz sine wave digitised by the *20/20 AD*. The spike at 1kHz is the sine wave itself. The noise floor of the system is roughly -145dBFS. All 20 bits of the output are being used. The typical dynamic range of the *20/20 AD* is 112dB, so the ENOB is 18.3.

**Fig.1b** shows what happens when the same signal is truncated to 16 bits with no dither. Note the sharply increased harmonic distortion (spikes at multiples of 1kHz) and elevated noise floor. Not pretty.

We can improve the signal by adding dither in the form of low-level white noise to the signal before truncating it. Although the dither has raised the noise floor to -127dBFS, it has removed the harmonic distortion. It is possible to filter the quantisation noise in such a way that almost all of the noise in the frequency range where human hearing is most acute (2kHz–4kHz) is shifted to higher frequencies. This is called psychoacoustically optimised noise shaping, because it takes advantage of the variations in the ear's sensitivity to low-level noise over the 20Hz – 20kHz range. **Fig.1d** shows the astonishing results that can be obtained from this technique. In the critical region, the noise floor is -142dBFS, only 3dB higher than the nontruncated convertor. In other words, this dither allows you to record 18 effective bits onto 16-bit media with no loss of information in the critical 2kHz – 4kHz band. Naturally, there are limitations. The price we pay for a depressed noise floor in the critical band is a severely elevated noise floor at high frequencies. Theoretically, this noise floor is less audible than the flat one in **Fig.1c**, but it sure looks scary.

Also, as Wannamaker points out in his paper on the topic [4], you cannot reap the advantages of a dithered 20-bit A–D convertor unless you have a D–A convertor of comparable accuracy at the output of the audio chain. Those consumers who have paid for high-end D–A's to play their CDs through will finally be able to hear something besides A–D convertor noise. ▶

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# Compression

One of the aspects of digital recording that audio engineers have the toughest time with is the lack of headroom in the conventional analogue sense. If a convertor signal exceeds 0dB full-scale (dBFS), it clips in a very harsh and unmusical way.

Manufacturers of high-end convertor boxes have recently begun to address this problem. One manufacturer adds a nonlinear element in the analogue domain before the convertor. This reduces the sharp edges caused by clipping, and simulates the soft knee of analogue tape
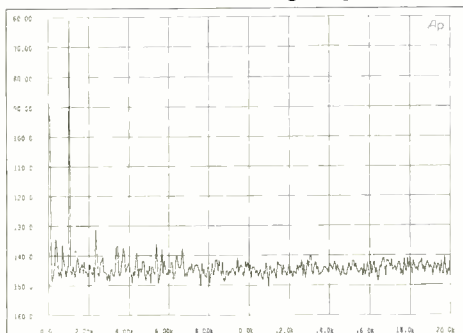


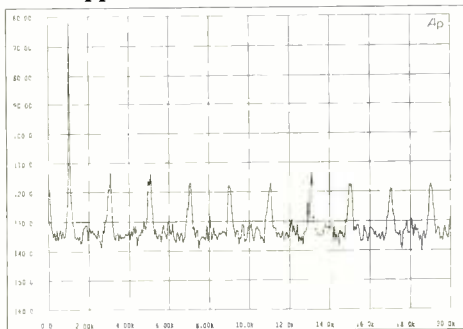**Fig.1a: Spectrum of the *20/20 AD* in 2-channel mode with a -60dBFS sine wave applied**



**Fig.1b: The digital signal from the *20/20 AD* has been truncated to 16 bits without dither**
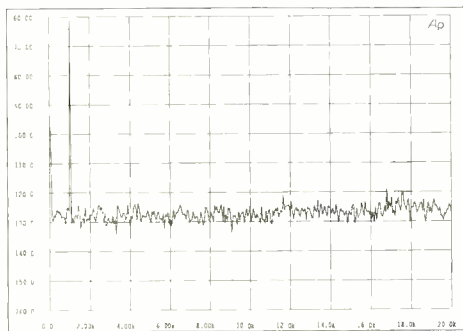


**Fig.1c: Wide-band dither removes harmonic distortion, but raises the noise floor**
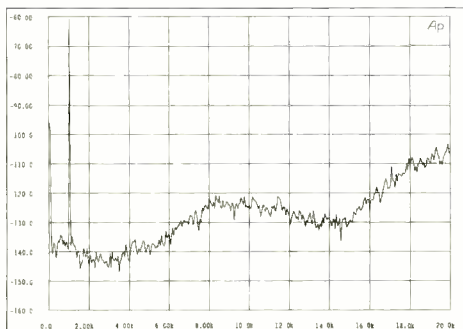


**Fig.1d: Psychoacoustically Optimised Noise Shaping lowers the noise floor in regions where hearing is most acute**
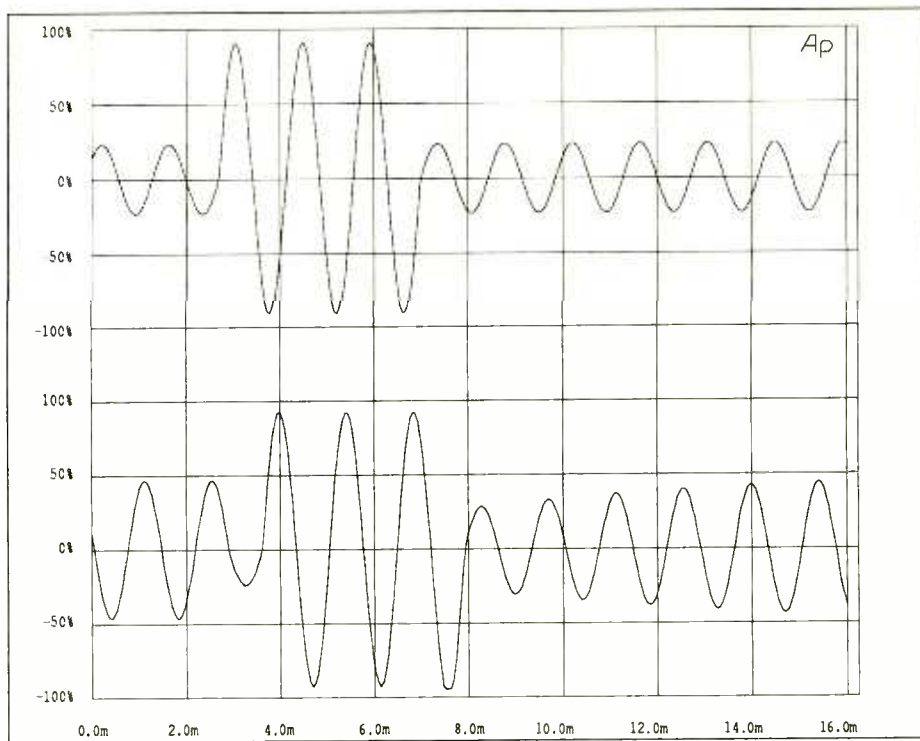
**Fig.2: Digital look-ahead compression in the *20/20 AD***

saturation. Lexicon's solution is to add a look-ahead compressor in the digital domain. Its operation is illustrated in **Fig.2**. In this example, the engineer adjusts the level so that peak signals are -6dBFS, or 6dB below clipping. He then sets the gain of the compressor to 6dB, so that the peak signals are boosted back up to 0dBFS.

Between 0ms and 3ms, the input signal (upper trace) is below the compression threshold, and the output signal (lower trace) shows a 6dB gain (as well as a 1ms delay). Starting at 3ms, a transient on the input signal peaks near clipping. The compressor reacts by reducing the gain on the output signal to keep it from clipping. By the time the transient has made it through the delay to the output, the gain has been reduced almost to 0dB. After 8ms, the gain slowly recovers to 6dB.

Both the analogue limiting and digital compression techniques trade off spare dynamic range for insurance against clipping. While both methods offer substantial improvements over digital clipping, they both have drawbacks. The analogue method creates distortion in signals whose levels are high but not in danger of clipping. The digital method prevents this form of distortion, but its finite look-ahead time can add distortion to the leading edge of transients that would otherwise clip.

The best solution is to record all 20 bits to a suitable medium, and then apply digital compression to the signal in the recording studio, where problem signals can be attenuated with a minimum of added distortion. However, in the absence of a 20-bit recorder, some form of compression in the convertor box is your best hedge against digital clipping.

## None of the above

There are times when you will find yourself recording a source with such limited dynamic range (for example, a close-miked bagpipe) that even 16 bits represents an absurd amount of overkill. In these cases, no amount of dither or compression is going to help, and all of my arguments fall flat.

However, because I hate to lose an argument, I added a special mode to the *20/20 AD* which doubles the number of channels to four, at the expense of reducing the dynamic range from 112dB to 100dB. This way, you can record an orchestra in glorious 20-bit stereo, or push a button and get four channels for the close-miked bagpipe (allowing one channel for each drone, plus the chanter).

All of my arguments boil down to one basic fact: the piece of your digital recording equipment with the lowest dynamic range is probably your A–D convertor (unless you have a seriously noisy mixer). If your current recording system sounds a bit grungy at low levels, it is probably because your current A–D convertor is having its way with the LSBs. A well-dithered 20-bit convertor offers an easy way to widen the dynamic range bottleneck without retiring your 16-bit recorder. A convertor with built-in compression will add insurance against overload.

The only way to know if the improvement is audible enough to justify the investment in a high-end convertor is to get your hands on a 20-bit convertor and do low-level listening tests. Only then will you be able to determine whether my two years would have been better spent perfecting my pizza recipe. ∎

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3. Proakis, John G., *Digital Signal Processing Principles, Algorithms, and Applications — 2nd Ed.* p 32. New York: Macmillan, 1992.
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Sennheiser is one of those names that seems to have been around the pro-audio industry for years. And it has: for nearly 50 years. The company also give the impression of being involved in areas other than the obvious pro areas of microphones and headphones. And they have: Sennheiser are currently active in almost all forms of transducer activity (except speakers) and allied technologies including wireless and infrared. Their activity spans half-a-dozen distinct business areas yet they can still come up with surprises such as the DM 20,000 (approximately £8,500) *Orpheus* headphone-valve amplifier system.

Over the last two years, Sennheiser have also created surprises in the business field by acquiring the, much older, fellow German company of Neumann as well as rearranging its own distribution throughout much of the world into Sennheiser companies. Clearly there is a strategy here and as it now involves two major names within the studio world it seemed an ideal time to look more closely at both companies.

Earlier this year, following the takeover, rumour had it that Neumann's console division was to be sold off, and that the Neumann name was going to disappear. Sennheiser took the unusual step of issuing statements to deny that this was the case, but one could logically assume that this was one of the options open to Sennheiser. I put some of these matters directly to Professor Jorg Sennheiser, son of the company founder and CEO of the company.

## Acquiring Neumann

'We took over Neumann at the beginning of 1991 but had been talking before that,' says Jorg Sennheiser of the takeover. 'Neumann had been a well respected competitor over the decades and in earlier times the competition was very stiff. Following the death of Georg Neumann, there was nobody to immediately succeed. We could see good products on the microphone side, a fading of the cutting lathe business and considerable investment in the console business, the details of which we knew less about.

'We entered talks with Neumann because we felt and still feel that the two names together span the complete breadth of professional microphones giving us a very strong position. In the area of mixing consoles Sennheiser has been keen to enter this market and has had success with small portable types but little in the really professional recording business. We felt that Neumann had a good concept for consoles in that field and here was a good opportunity to enter this area as well'.

When Sennheiser took over Neumann they were prepared to invest and work on the console side of the company which they saw as having a good future. However, it took much longer to sort it out to their satisfaction and, they hope, to the future satisfaction of their customers.

'At a certain time the rumours spread across the whole audio community that I was trying to sell this activity to somebody else. However, it was clear that that if the Neumann brand is not Neumann, it loses its attractiveness. I had to become very involved myself in the changes. But I am certain that the Neumann design approach — consoles with analogue electronics and digital control — is the best technological solution in the present and for the coming years.'

During the reorganisation Neumann closed their doors on console orders. What has emerged now is a new approach; away from total custom design and towards a more standardised product that can be still be customised but at extra cost.

'Console orders were accepted again from the

spring of this year and it appears that our revised approach is working. Neumann can now provide consoles at a price that is reasonable for the console quality'.

But what of the microphone side? Are there to be any restrictions on Neumann's activities in this area?

'There will be no restrictions at all. Neumann is a separate brand for separate customers. Those who buy Neumann for a certain task would not use a Sennheiser for that task and vice versa. They sound different and we are offering the sound engineer a choice.

'At certain points we may have to interleave our products in areas like stereo mics and shotgun types but right now we have different customers in the same market for both'.

There has been considerable interchange of information at the R&D level where there are fundamentally different principles used by the two companies. Neumann's is strictly a low frequency condenser design with polarisation while Sennheiser offer dynamics and the RF approach for condensers. Maybe in the future a combination of these principles will lead to many different types of microphones. But that is not a key factor right now. The microphone manufacture in both companies runs well.'

There have been numerous changes in Sennheiser distribution around the world recently.

Sennheiser. We don't have plans to acquire any more agencies but I am always open for discussion'.

## Marketing and R&D

Sennheiser have invested 10–11% of their turnover in R&D for years. They see this as essential, and such research should also include modern production technology which is vital is you wish to manufacture in an expensive country like Germany. This investment has produced some unique product areas.

'We were the first to manufacture active speakers and produced them for more than 20 years stopping about ten years ago. We were actually a little too early with the concept.'

Some products that Sennheiser became known for were the results of the company in the past being totally technically driven. In a then much smaller industry, engineers were able to develop products and then find a market for them. An example of this approach would be the introduction of rifle mics. Of course such products were not produced in ignorance but there was little or no market research done prior to the R&D other than the feeling that if they could be made there should be a demand for them. In the case of

will pay for it?'

But what about two recent introductions — the *Orpheus* headphone system and Neumann *U67* promotion? How does this fit with the philosophy?

'There is no contradiction here. The *Orpheus* was never meant to be sold but developed to show what can be done with today's technology when merged with yesterday's. It came out so beautifully that there appeared to be a demand for the product even at that price — at least enough to sell a limited edition. It may seem odd that we produce one product in 100,000 of units and another in tens but they both are designed to meet different demands.

'The Neumann *U67* relaunch is also just responding to a market demand. We have found that is was possible to produce a limited number at a reasonable price. It fits well with our principles although we have to remain flexible in production techniques.'

Interesting enough, there are demands for reintroductions of old products in other areas.

'There is a demand for the original *HD414* headphone which by today's standards is old fashioned — certainly not an objective transducer by today's standards. Over the years however many people became used to listening through an *HD414* in an objective manner to evaluate quality. There is therefore a continuing replacement demand. So maybe we will one day supply this requirement.'

Visiting Sennheiser there are signs that the company is about to develop quickly in a wider range of market areas. The new marketing arrangements, the acquisition of Neumann and the enthusiasm for the mixing console markets; the extensive developments in the wireless microphone fields and the sheer amount of R&D undertaken on new products are very impressive. On another level entirely, it is also the cleanest factory I have ever visited.

It will be very interesting over the next few years watching how the Sennheiser-Neumann combination develops. Outwardly very different companies — in products and philosophy — will actually both benefit from the acquisition. I also feel quite safe in saying that the Neumann name is in reliable hands with Sennheiser. Their ▶

# THE MIC MECHANICS

## Following the fall of the Berlin Wall, Sennheiser set tongues wagging by buying out fellow German mic men Neumann. Keith Spencer-Allen crossed the border for a field report

Was this a policy of gaining more control of marketing or were there other reasons?

'I think that a certain stage it is necessary to have a more direct access to the market and the customer. Using representative companies we have good but limited access and no control. So rather than build up a separate operation we have acquired many of the representative organisations that considered selling to us, complete with the sales team. This is what we did in the US, Canada, the UK, Belgium and France. One of the benefits of this arrangement is that Sennheiser will now be able to distribute products from other manufacturers.

'We also look for better feedback from customers as they will be dealing directly with

this product, however, it found a home in film production in Hollywood and gradually, from there, in other areas around the world. Success for intuition in this case was proven when in 1987 Sennheiser's founder was awarded an Oscar for Scientific & Engineering Achievement for the *MKH816* rifle mic.

These days the markets have changed and Professor Sennheiser describes the company as striving to be totally market driven.

'It is mandatory for us to concentrate on product development where a market need has been identified and the customer can dictate the features. The quality standard also has to suit the needs of the customer. It is easy to produce a product to as high a quality as possible but who



**The Neumann *U67* — revived**

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The 'nostalgic' *HD1000 Charleston* headphone with its stand

The spares store is large, aiming to meet the wish to service and repair any mic they have ever made! Although the intent may be there, unfortunately there are some that just cannot be repaired due to exhaustion of spares and no remaining sources outside. One of the casualties of this situation is the valve *U47* — that valve has not been available for many years and so the only option is a new electronics package.

It is clear, however, that Neumann are proud of their old microphones (as well as the current range). The reception areas houses a permanent display of 'antiques' including the original Neumann 'bottle', the *CMV3*, the mic that started it all off.

As well as totally new ranges of mics such as the *KM100*, Neumann have recently been looking to past products to redevelop to meet modern requirements. The *TLM50* is an example of how the popular, but long out of manufacture, *M50* technology has been resurrected and updated to meet modern requirements.

Another example is the special run of 'original' *U67*s currently under construction.

'We will be supplying the *U67* with original *EF86* tube in its original shape and nickel finish in a limited run,' commented Neumann Marketing Manager Wolfgang Fraissinet. 'We had been asked by so many customers if we would consider making a U67 again that we decided to see if it was possible. Although we keep a stock of *EF86* valves for mic repair there was not enough. We set about finding small quantities of valves all over the world, testing them to ensure they were of suitable quality, until we had enough. Our dealers were asked to survey their customers for interest and we were very surprised by the positive response.

'This will be a limited run and we will never be able to do it again. We are looking at other old designs to see what possibilities there are, but ▶

## Neumann today

As well as being one of the longest established professional audio manufacturers anywhere, Neumann also is probably the name that comes to mind first when thinking of German companies in this field. Despite being such a industry name, they have remained quite private and most know little of the full extent of Neumann products outside of the German market.

Established in 1928 by Georg Neumann, the Neumann company have always been located in Berlin. Production of microphones and disc cutting equipment both commenced shortly after the company's formation. The move to the present site, within a stone's throw of the site of Checkpoint Charlie, where the Berlin Wall used to snake through central Berlin, was in the early 1960s. The removal of the Wall has left Neumann in the unusual situation of being right in the middle of the city.

Mixing consoles were added to the product line in the early 1960s and this area of their business has become increasingly important. The company quite readily admit that, to date, most of their console products went into the home market and Eastern Europe to meet demand in broadcasting, theatrical and concert use. But following the influence of Sennheiser, this will not remain a restriction for the future.

Spread over two floors, it is clear that Neumann are an engineering company. I have rarely seen such extensive facilities for working with metal, much of which is completed in-house. For the consoles, many of which were custom designed, the on-site metal work aids construction time and aids design. For the microphones it helps monitor quality.

Microphone capsule assembly is all by hand, in clean areas, at one of seven workstations. Each station handles a different capsule type but the staff have the ability to move between any of them. At another end of the building there is a room dedicated to the assembly of the wedge-shape grille made famous in the *U67, U87* and *U89* — three layers of mesh attached to a ring and hoop, painstakingly assembled by hand.

Microphone testing is equally rigorous with each microphone undergoing a full anechoic chamber test, a subjective test — speech, shaking, banging and moisture, and lastly the LF test in the Neumann 14-metre LF stiff-walled tube to check response down to 10Hz.



*U47* a casualty of spares exhaustion

so far it is just the *U67*.

As one might expect, the vinyl-disc cutting equipment side of the business has declined dramatically over the last five years. There is no demand for new lathers, although as with the microphones, Neumann retain specialist staff to undertake repair work on heads and lathers, and will continue to do so. The quite amazing work on mechanical cutting of compact discs undertaken with Teldec in the mid-1980s has also been shelved.

It is, however, the mixing console division that is now Neumann's major interest, with two-thirds of the workforce being employed in this area.

To date Neumann have four ranges of console — *N20, N40, N5000* and *N7000*. The first two are for standard requirements in broadcasting, theatre and music recording and can be assembled to meet specific customer requirements in size. It is, however, the *5000* and *7000* that are perhaps of greater interest.

Both are digitally-controlled analogue consoles of a sophisticated design. The *5000* is assembled to order from standard 'cassettes' to produce a console that is fully automated with complete static or dynamic storage, and reproduction of all console facilities including moving faders, EQ, pan, output paths, dynamics, monitor faders and pan.

Design work on the *5000* series started in the early 1980s with the first unit being installed in 1985. Construction and size is flexible — a recent order of five units for the German Army Media Centre included an 80-channel *5000*, while I was shown a wraparound design, one of three for the Austrian ORF. Customisation abilities extend to colour as well — the complete console, frame and modules with the ORF order being dark blue all over while there was mention of a totally pink design delivered to a Munich facility.

The *7000* has certain similarities with the *5000* but adds a new control surface and software enabling the status of the console to be controlled on screen.

Another new departure is the concept of offering components developed in these consoles for interfacing with other manufacturer's equipment. Components currently available include the automated EQ system, a delay unit, a sophisticated mic amp, the screen-console software package, and the moving fader automation.

As Wolfgang Fraissinet added, 'Offering



**The *HD414* design created in 1968**

components to interface with other makes of consoles is a new field for us but it has been well accepted'. It also forms the first part of a plan to move into the international console market with new console designs based on the *5000/7000* approach 'but on a basis that is comparable in price to the competition' said Fraissinet 'but of course we will have some surprises'.

## SENNHEISER: A BRIEF HISTORY

**Professor Sennheiser had been working at the University of Hanover until it was bombed in 1943. He then moved to this farmhouse north of Hanover**

Founded in 1945 by Professor Fritz Sennheiser, Sennheiser Electronic KG came into existence following a set of circumstances that could have been easily been quite different. Professor Sennheiser had been working at the University of Hanover until the facility was bombed during 1943 where upon they moved the department to an old farmhouse about 12 miles north of Hanover. Towards the end of the war as the allies moved through Germany there were severe restrictions placed upon any occupation that could be construed as having any military application. Sennheiser's work had been in the area of vocoder and scrambling research and as such fell within this category and the farmhouse was sealed with threats of shooting anyone crossing the threshold.

With the passing of time it became clear that the farmhouse had been forgotten about; searching for a means to earn a living, Professor Sennheiser started making millivoltmeters out of the materials to hand in the farmhouse. The entry into acoustics and microphones followed a request from the local Siemens company, who were cut off from their manufacturing facilities in Austria, to make a copy of a microphone. This design was improved upon and manufactured to requirements on an OEM basis. This aspect of the business grew with more demand for mics for dictation machines and tape recorders.

At this time the company was known as Laboratorium Wennebostel or more briefly Labor W. High-quality power amplifiers were added in 1949; the *MD3* miniature stand-mic, with separated head and body, in 1950; AF transformers in 1951; and miniature

magnetic transducers in 1952.

1954 saw the introduction of the *MD21* Reporter microphone, a microphone that has had continuing success and is still a current product. The *MD21* was also the first product to bear the name Sennheiser rather than the previous Labor W which was increasingly inappropriate as the company was now far more manufacturing orientated rather than a laboratory.

Products then followed thick and fast including the *MD82* shotgun mic in 1956, the first wireless mic systems in 1958; the *MD421* dynamic mic in 1960 which has sold 350,000 to date. Answer machines came in 1961, active speaker systems in 1965, broadcast mixers in 1967, the *HD414* open headphone in 1968 whose production had passed the million mark six years later.

Infrared technology was added in 1976 for headphone systems. There has been a continuous development of products out of these core business areas and the company now spans six identifiable market areas — home entertainment, hearing aid systems, industrial applications, communications, music business and popular music recording, and the Studio group for all broadcasting, film and classical recording.

The original farmhouse has been renovated and sits alongside the manufacturing complex built since the 1950s and now employing 1000 people. In 1982, The son of the founder, Professor Dr. -Ing Jorg Sennheiser assumed active management of the Sennheiser company after having been its Technical Vice President for more than seven years. Currently, the company remains privately owned. ■

Neumann have welcomed the wider opportunities that the connection with Sennheiser now gives them outside of the home markets. There is a firm belief that there is no conflict between the ranges of mics they both market as they are addressing largely separate needs while remaining complementary. But Fraissinet admitted that with the R&D departments in close

liaison there will have to be care that there are not similar developments in both companies.

With basic mic design philosophies so diametrically opposed there would, thankfully, seem to be little chance of, or wish to see, Sennheiser-badged Neumann mics in the future but there is certainly room for some interesting hybrid designs. ■

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# As Air Studios' new Lyndhurst installation nears its opening date, Patrick Stapley reports on its progress

We're now definitely on the home stretch,' claims a weary Dave Harries, 'but there is still a hell of a lot of work to do.' In fact the workforce at Lyndhurst Hall has been tripled to meet deadlines now dictated by firm studio bookings, and the building is literally buzzing with activity both day and night.

One deadline that was met recently, was the Princes Trust Gala in the Main Hall which officially opened the studios in December. The event, which was a musical version of *Under Milk Wood*' years ago, was organised by Martin and directed by Sir Anthony Hopkins. A 22-strong cast including Mary Hopkin, Tom Jones, Jonathan Price and Harry Secombe supported by the Wren Orchestra and a Welsh male-voice choir performed in the presence of Prince Charles and an audience of 600. Autograph Sound Recording were called in to look after the sound design, sending splits to the Control Room where the performance was recorded by Air's Jon Jacobs. The evening was also recorded by Chrysalis Television for future transmission. George Martin described it as 'like putting on a West End musical for just one night,' but said it went without a hitch, which was remarkable considering the only full rehearsal was on the day of the performance.

The Main Hall is now all but complete, some finishing details remain such as fitting the Huppe system of sliding and sealing partitions to the front of the isolation booths and installing variable acoustic panels around the Hall itself. The wood block flooring has been relaid using the
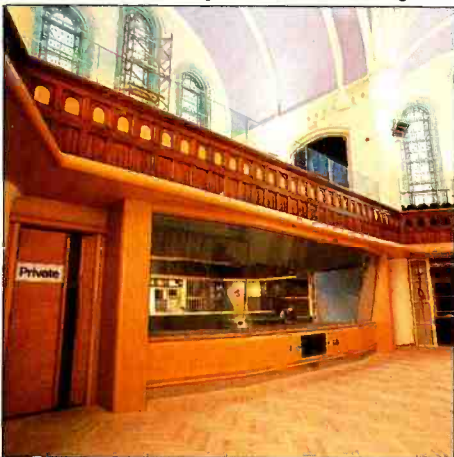
original 3-foot blocks cut in half to make up numbers. The ceiling, which was damaged when a high pressure concrete hose burst, has also been repaired and repainted.

The Hall has now undergone its first serious acoustic test: a rehearsal with 90-piece London Symphony Orchestra and 130-piece chorus. George Martin gives his impressions of the sound.

'It was very interesting because we deliberately hadn't done any acoustic work on the Hall, because we wanted to wait until we could hear a big orchestra in there. The string sound was absolutely gorgeous, the actual quality of sound was lovely although the reverb time is too long — around four-and-a-half seconds which is almost cathedral-like — but the nice thing about it is that it's a very even tone, a very warm sound. The dynamics are quite superb and there is no compression which halls can sometimes suffer from with very loud music. All we need to do is reduce the RT, and that's being done with a system of variable acoustics.'

The original plan for controlling the RT was to suspend a tent-like structure from the ceiling that could be winched up and down, but this has been replaced by the idea of three acoustic canopies that stretch out over the Hall on horizontal wires. These will be remotely controlled from the Control Room, and there are also plans to add a reflective lid structure.

'Once we've sorted out the reverb time, this will be a first class location for classical work,' continued Martin. 'There is a great dearth of really good classical recording rooms in the UK: Abbey Road's Studio One is in great use, but not everybody likes it, and places like the old Kingsway Hall, although they have lovely acoustics, suffer from awful noise problems which is unacceptable with today's technology. At Lyndhurst we can offer an excellent acoustic environment plus complete isolation.'

The Main Hall will be a truly multifunctional studio offering five recording areas, and full provision for scoring and TV work. Eight camera points have been positioned around the studio (more can be catered for if necessary) plus TV lighting. The camera lines run on triax cables to an external connector box at the side of the building where a TV truck can simply plug in. Also supplied are 12 audio ties, 12 video ties and two power supplies for technical power and air conditioning.

Around the edge of the galleries, a wooden 'eyebrow' has been fitted to conceal cables, and lift-up covers are provided to allow additional cables to be neatly dropped in and hidden.

Mic line wall panels have been placed around the Hall in strategic positions which include MIDI tie lines, stage box ties allowing 16 lines per box, audience voting tie lines, MD cue system and so on. There are also some mysterious holes in the walls between the studio and Control Room which studio manager Malcolm Aitken considered an essential feature.

'Never build a Control Room without several holes in the wall! If you don't put them in, you'll find engineers running cables under the doors into the studio with the resultant damage to both doors

and cables. It also solves the problem where a guitarist wants to play in the Control Room but is convinced the sound will change because he's not using his favourite guitar lead — all you do is run it through the wall.'

The Control Room window has finally been installed after a number of attempts to get it right. The two panes of 10mm and 12mm laminated glass had to be replaced four times ▶

**Main Hall Control Room**

Air Lyndhurst's Main Hall nears completion

# AIR MOVEMENTS 6

**Interior of the Main Hall Control Room**

due to wrong sizing, scratches and ripples. Another problem, that nearly set the Main Hall back three weeks, was caused by a breakdown of isolation between the Control Room and the gallery above Dave Harries explains.

'An inspection showed that plaster board on top of the Control Room was touching the underside of the gallery. To put it right would have meant taking up half the gallery floor and reconstructing the roof below. We had several meetings with various companies who all quoted us three weeks to do the job. In the end Angus McPherson persuaded one of the carpenters, who was quite small, to go in under the gallery floor and crawl along by air ducts and take a look. He managed to cut out all the offending pieces of plaster board with a Stanley knife and it only took him four hours.'

The Control Room has been fitted out with preformed wall panels made from perforated hardboard. Originally the panels were to be fabric covered but the general consensus is that they look better without it. A large acoustic 'cloud' has been placed above the console to help break-up the sound as well as containing some bass absorption and concealed lighting.

In the corners of the room are four large, semicylindrical, grilled structures that look almost like acoustic diffusers; in fact they are air conditioning vents. Sadly the company dealing with the air conditioning contract, Delby Services, went into liquidation recently, giving Air another headache.

'We were Delby's first customers when they went into business in the 1970s and we've used them ever since,' said Dave Harris. 'It was a real struggle finding another company who could do the job in time, to the same standard and for the right price, but we have found a replacement and we're now back up to speed. The system in the control rooms is an air displacement system rather than a traditional air dilution system, what this basically means is efficient air conditioning without drafts or cold spots.'

The 72-channel Neve *Legend* with *Flying Faders* was installed at the end of November and has gone into service very smoothly. Due to the lack of space for a dedicated automation monitor, ideas are being banded about for a 'head up' display which would reflect onto the Control Room window; another idea is to install a hydraulic lift to raise the monitor from behind the console.

As with Air's previous studios, the console's patch bay has been installed remotely. Chief maintenance engineer Rob Haggas gives the reasons.

'Air haven't had patch bays in desks for ages, because we feel it's much better to have them mounted vertically and preferably somewhere near the centre of the stereo image so you can accurately hear what you're plugging in. The strips are all connectorised as we've found that after five or six years they will need replacing and this is obviously a much quicker job without a soldering iron. The whole of the patchfield is gently pressurised with cold filtered air that blows any contamination out of the sockets.

'Another idea we've incorporated is a system which remembers the position of patch cords — a patchfield 'total recall' system if you like. Because this is a totally balanced installation complying with 16th Regulation earthing which requires all metallic equipment to be earth bonded, it frees up the sleeves of the jacks so at the end of the session a computer can actually scan through the top rows of the jacks raising each one a couple of volts in turn and then read on the insert points to see where the cords have actually gone. When you come to recall the patch, which is saved on disk, LEDs associated with each socket will flash to indicate patch positions, this is done in pairs until all the patches are correctly reinstated. It gets around the maintenance man's trauma when he comes in first thing in the morning to plug something up and can't quite remember if he's left the patchbay as he found it!'

The majority of out-board equipment is coming over from Oxford Circus where it will fit into recessed, pull out trolleys which mate on DL connectors. Each trolley will take up a complete strip of patch and the idea is to have identical arrangements of out-board in each studio to provide continuity throughout the facility. Extra out-board gear gets racked up into spare trolleys and there is additional patching at the back of each to cater for any stacked gear. Plenty of line level interfaces have been included to deal with keyboard rigs and so on, rather than relying on mic DI boxes — the aim is to get the signal at the correct level and as clean as possible at source.

Acoustic tests were run in the control once all the equipment was in place, and this disclosed an unforeseen problem.

Rob Haggas: 'It was interesting, because an RT plot of the room made it appear virtually perfect, but the old test of actually running some music through the speakers and actually walking around showed there was a problem frequency. What we found was that since all the out-board equipment had been positioned in the back of the room, it was blocking some of the wave fronts to the bass absorbers behind. Sandy Brown Associates came up with the idea of venting the floor in front of the racks and this seems to have cured the problem.'

The Machine Room can fit four multitrack-sized machines, but the studio also has access to the central Apparatus Room on the postproduction floor where additional machines both analogue and digital can be interfaced. A special water cooling system has been installed that runs throughout the building and can provide up to 12kW of chilling in any one room.

Also housed in the Machine Room is an Audio Precision *System 1* measuring system which will check all inputs and outputs to and from the desk as well as running a monitor 'health check'.

'It's something that Japanese studios have been doing for sometime now,' says Haggas. 'There are MLSSA programs within the Audio Precision system, and we can get very precise information about any changes to desk performance and the monitoring. Once the room has been set up it will report any changes such as driver deterioration ▶



**Studio 2 Control Room with 'acoustic cloud' on the floor awaiting fixture to the ceiling**

**Commemorative plaque in the Entrance Hall**

or crossover irregularities. By regular testing, we should be able to catch faults in their very early stages, and hopefully before engineers bring them to our attention.'

The Anteroom-come-Artist Room that connects between the Control Room and one of the large isolation booths, has been altered to comply with council regulations. Because the room provides an exit to the car park, its doorway has had to be enlarged due to the studio being used for public performance. This has resulted in further isolation problems which Air are resolving, but when not in use the doorways will be covered by soundproof panels. Other doors in the studio have been supplied by ADW in High Wycombe, who built the studio doors at Air Monserrat. The doors in the Main Hall are finished in steamed beech and are half-an-hour fire rated with 45dBs soundproofing — they cost £2,000 apiece.

Although most of the activity has been concentrated on the Main Hall since our last visit, other areas have also progressed. The postproduction floor is taking shape with all the rooms now plasterboarded and isolated, including a vocal booth at the back of the *Logic 2* Room. Both *Logic 2* consoles were delivered to Oxford Street during November where they have been enthusiastically received, both by engineers and clients.

The lift is now installed and working. The atrium is complete with glass canopy and york stone paving with underfloor heating, and the kitchen and restaurant areas are finished.
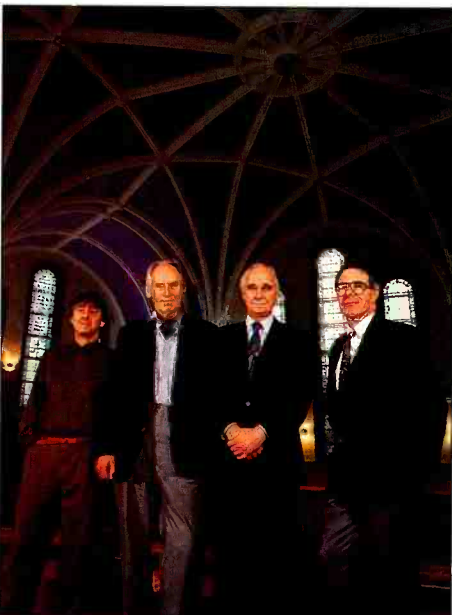


**The pantheon revealed. Left to right: Malcolm Aitken, George Martin, John Burgess and Dave Harries**

A transformation has occurred at the front of the building where instead of builder's rubble there is a new car parking space, for 20 cars, with walls, iron railings, electronic gates, Victorian-style lamps and bollards. But not all was plain sailing as Dave Harries describes.

The railings caused all kinds of arguments. There were terrible rows going on between the contractor, builder and the railings man; it was all heading towards a crisis, because it had to be finished for the Prince's Trust Gala. Anyway the architect was up here when they were having all these problems getting the railings and the gates to fit properly; he said, "I don't understand you blokes, all you have to do is work to the drawings and it will be perfect." The railings man turned round to him and said, "listen mate, the only similarity between those walls and your drawing is the address!" It worked like a charm, after a moment of shocked silence everyone collapsed with laughter and from that moment on the whole thing went like clockwork. It's amazing what a bit of humour can do.'

The next phase of intense activity will be directed at Studio 2 which has to be finished for a two-month block booking by Dire Straits. At the time of our visit in mid December there had been little progress made and the studio was still very much a shell, but by the time this article appears Studio 2 will have been operational for two weeks. At the same time the Main Hall will be well underway with its first commercial booking — scoring sessions for Henry Mancini. It all goes to schedule Lyndhurst Hall should be complete by March. ∎

## MONITORING

The monitoring at Lyndhurst Hall has been designed jointly between Andy Munro and Air. I found Munro analyser in hand in the Main Hall Control Room and he explained the philosophy behind the systems he is installing.

'It all started a couple of years ago when Dave Harries and I sat down and conceived what we would do if we were to build no compromise monitoring. We agreed on several things, firstly that it should be digital if possible to give us control of all the parameters as well as enabling direct digital output from a digital console rather than going through an analogue process.

'The intention was to use this almost as an R&D platform and try new ideas evolving the monitors mainly as a result of the Air Engineers' response to them. There's been a kick-back against big monitoring systems and despite what anybody says most people tend towards trusting small speakers more than big ones. So what we've tried to do is remove all the acoustic anomalies present in big monitoring systems such as the room interfacing problems which we hope to correct using a digital crossover.

'We systematically tried out different drive units and modified some Dynaudio drivers to give more power handling, but at the same time keeping the smoothness in the response. We ended up with a completely new high mid-range unit and a modified standard low mid unit that has special ferrofluid to keep it cool. The bass drivers we're using in all the rooms apart from the Main Hall are a special version of the 12-inch driver that we use in our *M3* and *M4* systems, but we couldn't physically fit them into the soffit in the Main Hall's Control Room, so we've used TAD bass drivers there instead. The crossover we're using is the Yamaha *D2040* which is the first of their full 20-bit devices, and the power amp is a *Chameleon 2200* which we chose because of its 115-volt rails which provide us with very high headroom. We've used the MLSSA system to set the whole thing up.

'All the rooms will be capable of full Dolby surround monitoring and by using the Yamaha crossover we can do some interesting things by adding delays — for instance we can make this room sound like a 200-metre viewing theatre by putting a digital delay into the ambience channels. You can put delay on the front channels as well so you can create the effect of being 30 or 40 feet from the screen — you can literally steer the monitors backwards and forwards relative to the picture. Traditionally people have mixed film from quite a distance from the screen and this has not just been to get the right sound picture but also to compensate for the delay aspect. The crossover allows you to store presets, so it would be very easy to recall various room simulations.

'Another thing we've done is store three crossover programs — one for classical music, one for rock and roll, and another somewhere between the two. This is the wonderful flexibility you can get with digital processing. When we do our analogue crossovers we have to take the point of view that the things going to be fairly bulletproof and what we do is set them up with fourth order networks to band limit the amount of power going to each driver. The problem with using steep filters like this is that it messes up the impulse, but what we can do with the programmable crossover is actually program in another configuration that uses far more gentle filtering that will enable us to do proper time alignment.

'Now, the trade-off using gentle first-order filters is that you spread the power to the drive units over a wider bandwidth so you can't for example shut down the mid-range units to stop bass getting into them. This means that the mid-range units have to be capable of handling a certain amount of low frequency. Now, luckily most Dynaudio drivers are actually designed to take quite a lot of bass, so they cope with spillage without adversely affecting the sound, but what you don't get is good power handling. This is absolutely fine for classical recording but totally unsuitable for rock, so with the presets on the Yamaha you simply press button B and you're configured for perfectly time-aligned classical recording, press button A and you're set up for rock and roll, press button C and you're somewhere in between with a second order network. People will be able to tailor their power versus quality requirement.

'This is the first time this has been done successfully because until now no-one's come up with the combination of the hardware and software in the same box. I think this is the start of a very exciting new phase for large monitoring systems.' ∎

In late 1990 Philips ran some technical-looking adverts for the company's CD players. Unfortunately, the technical content was badly inaccurate, quoting figures that would have made the playing time of a CD add up to less than a second. The adverts soon disappeared in a flurry of embarrassment.

Now Philips have got it wrong again, this time with technical-looking adverts for the company's new Digital Compact Cassette system. This time the advert is a double-page spread, headed 'Mysterious ways by U2', and showing two pages full of 'digital' ones and zeros. 'What you see here represents only the first 4.7 seconds of the piece,' reads the ad. Unfortunately not.

Philips own technical literature gives the audio bit rate for DCC as 384 kilobits/second, which puts the number of bits needed for 4.7 seconds of pop group U2's music at 1.85 million. The number of bits shown in the advert is a paltry 42,000. So the numbers are out by a factor of 44.

Philips say that they checked but 'felt that the original message had the right style, creativity and impact – in the end it just sounded right as originally scripted. We do not believe that the exact number of ones and zeros would be relevant to the average, healthy, happy minded individual.'

Perhaps not, but, when concocting a 'technical' advert, would it not be just as easy to get the numbers right, instead of wrong, and say the double-page spread shows a tenth of a second of music instead of the completely incorrect 4.7 seconds?

Remember Brad Kay and his fascinating theories on old stereo? In 1985 Kay, a Californian musician and record collector, suggested that record companies such as EMI have true stereo music recordings in their vaults which date back to the 1920s. These recordings were made in stereo only by accident and are, consequently, not labelled as being stereo. Many actually predate the first music test recordings (made intentionally in stereo) by Bell Labs.

In the 1920s and 1930s, before the use of magnetic tape, studios cut recordings directly onto wax discs. They played safe by simultaneously cutting two discs, one as a spare. Brad Kay discovered that they sometimes played extra safe by using two microphones — one for each cutter — instead of simply splitting the output signal of one mic and feeding it to both cutters. In these cases, the result is a matched pair of recordings, each with a different sound perspective.

Kay found matched pairs of recordings and used one as the left-hand channel of a 'stereo' recording and the other as the right-hand channel. The stereo effect he achieved is very impressive.

As a result of these experiments, the recording industry quickly became polarised — especially the engineers at EMI. Some pooh-poohed the idea, arguing that the perceived stereo effect was created by phase shifts due to slight inaccuracies of synchronisation between the two recordings. In other words, it was accidental automatic double tracking. Other engineers, however, found the stereo effect too convincing to be the result of ADT. For many months after my writing on the subject I was caught in the crossfire between factions with

## Barry Fox

# Upset over 'proto-stereo' recordings; no climb-down over DCC advertising

surprisingly strong convictions.

EMI Records have access to recordings of Sir Edward Elgar made in 1928 and 1933 which some engineers believe to be in true — but accidental — stereo. In the late 1980s, one EMI engineer tried blending the company's matched pair of master discs of Elgar's recording of the *Kingdom Prelude* cut at Abbey Road in January 1933. For a while afterward there was a plan to release the recording without any mention of its 'blended' source to see if the music buffs spotted anything unusual about it. Then the game plan changed; the results were sufficiently impressive to inspire EMI's Classical Division to promise an official stereo release in 1993.

The plan was unveiled in spring 1992, when EMI Records' Classical Division released the first of several volumes of cleaned-up Elgar material.

'*Volume 3*, to be released in 1993, will include a previously unpublished true stereo version of *Kingdom Prelude*', promised EMI. The promise was still being publicised by EMI's press office in early October 1992, when the company won the prestigious *Gramophone* award for Elgar *Volume 1*. But, by then, the powers that be inside EMI had quietly abandoned these plans to include the stereo restoration, admitting that they were 'airbrushing' the published promise.

Said Richard Abram, in charge of compiling all the Elgar Material, 'We are enormously disappointed, and mildly embarrassed, but we know what the result would be. It would only be fake stereo.'

'Know' is a very strong word, usually only used in an area of doubt after conclusive test have been carried out. EMI, however, appear to have carried out no such tests.

Engineer Michael Dutton worked on the prize-winning recordings and was preparing to start work on the stereo material when — shortly before the awards — he was taken off the job. But he still wants to carry out tests which would settle the dispute once and for all.

# We do not believe that the exact number of ones and zeros would be relevant to the average individual

'If EMI will provide the original master recordings,' he says, 'I will spend my own time analysing whether they are, or are not, true stereo.'

Michael Dutton had planned to use the CEDAR and Sonic Solutions *No-Noise* digital systems at Abbey Road to lock the two masters in perfect sync, thereby eradicating any phase errors and revealing any true stereo content. He also planned to process matched discs of Elgar's *Cello Concerto*, recorded in March 1928 at the Kingsway Hall, in the same way.

Richard Abram says that the decision to kill the project was taken on the strength of a technical paper written by another EMI engineer, Anthony Griffith. This paper argues that EMI only ever used a split feed from a single microphone in their recordings.

The question remains: would it not be worth EMI's while to settle the matter using scientific tests — by using digital technology to eliminate phase shifts and listening to the result?

'We saw no point,' says Abram. 'Whatever the result will be, it won't be stereo. We are bitterly disappointed, but it is a snare and a delusion.'

If you are out there, Brad Kay, let us hear from you. Your idea is every bit as controversial now as it was in 1985.

*Volume 2* of the Elgar material already relies on some very clever audio processing. Some of the discs were actually test pressings taken from the composer's personal collection. When he recorded one version of the *Serenade Mauresque Opus 10 No 2*, he took the tempo so slow that the disc ran out and the turntable had to be switched off during the last chord. The result was a recording in which the pitch of the last chord zooms upwards (as the turntable slowed down). The recording was not originally issued, but EMI's engineers have since been able to use the digital processors at Abbey Road to stabilise the pitch of that last chord. I raise the question again: would it really hurt EMI to use the same technology that allowed them to achieve this stabilisation to sync up the two *Kingdom Prelude* masters and then to listen to the result?

There is a sad footnote to the sorry tale of how the BBC's Radio Publicity Department once again missed tricks when publicising the BBC's excellent Radio Show ('Business', November 1992, *Studio Sound*). Because the Department Head had got so huffy when I published criticism of the poor publicity for the last Radio Show, at Earls Court, I wrote personal letters this time before going into print. There was no reply, but I thought at least that someone, somewhere inside the Department might have taken some interest.

Not a bit of it. When the BBC extended the Show's run by two weeks, I learned about it only by spotting an advertisement which the BBC had placed in a newspaper. Despite my letters I heard nothing from the BBC's Radio Publicity people. Neither did others. I phoned and the woman who had been dealing with the Radio Show had changed her job.

All this coincided neatly with reports of more cutbacks on BBC staffing and a speech given by Jonathan Powell, Controller of BBC1, in which he estimated that 20% of the BBC's income from licence fees is wasted on inefficiency. ■

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With apologies to those computer sales people who are both honest and knowledgeable — it is frightening to realise many are not. It is also true to note that, although computers have become more complicated, their pricing has not reflected this. The result is that extraordinary levels of sophistication are available at 'commodity prices'.

In response to these factors, here are some suggestions intended to make the computer acquisition experience for the audio practitioner as positive as possible.

My first directive must be never to 'buy the computer store'. Instead, buy the specific product you want from the retail outlet that will give you the best support at the most acceptable price. It is important to remember that the lowest price is not always the most important consideration, since real hands-on support before and after purchase is almost never available at the lower profit margin levels. If you can find a musical instrument-audio dealer who carries the brand of computer you want, you may also find the dealer more familiar with the software you are likely to need and more sympathetic to your needs in general. No matter what the problems in that particular dealership, there will be more knowledge about what 'you' do in audio and recording and what you will expect of the computer.

The 'boutique' computer store in the United States becomes an increasingly endangered species, as Uncle Ed's Mega-warehouse becomes the normal mode of distribution for commodity computer sales. Down the endless aisles, along with the five-pound boxes of frozen fried chicken, the 48-roll packages of toilet paper and the AM-FM microwave ovens, you will find a *486SX* computer with a 230Mb hard disk. And the guaranteed lowest price in this or any other alternative universe. If you need service, it will come from the same person who helps you find the toilet paper that is 'soft as a cloud'.

Instead of 'buying the store', try buying the computer manufacturer. After a very painful few years, computer makers who all have had to resort to commodity channel distribution, have taken support back in-house with 800 telephone numbers and trained support personnel. This still may not work as well in practice as it does in theory, but it is a lot better than it was a few years ago. The mail order and commodity price wars that marked the end of the 1980s have seriously depleted the numbers of dealers dedicated to high quality service and removed the profit incentive for those still extant. So the manufacturers are trying to fill the gap.

Remember that the failure rate for small computer stores and even for medium-sized chains has been quite high. Do not leave a large deposit with them if possible, or if you must for a short time frame, use a credit card where you have the protection of federal (or equivalent) law. The same thing goes for service. Make your dealer quote a date; if it is longer than a week or so, ask the dealer to order the needed parts and return with your computer when summoned. This is far better than losing your computer in whatever state of repair and your becoming an unsecured creditor

## Martin Polon

# Bringing a computer into the recording studio can present more problems that those readily associated with machines and music — first you have to buy it

should the dealer suddenly file for bankruptcy.

An old, but reliable, line advises 'never take the word of a sales person'. Yet while the advice is sound, it should not be taken as a blanket denunciation of all sales persons. In fact the computer manufacturers themselves have a lot to answer for when it comes to inaccurate information dispensed at the dealer level. Let us assume that company X has upgraded their *DDT* computer. The new machine, called a *DDT-X*, is basically the same as the old *DDT* except for the transition to a different processor and a software architecture with more user options. Company X does its usual minimalist job of communicating the changes to its dealer network. The company does not want to fuel unwarranted expectations about the new machine. Curiously, by saying as little as possible, the manufacturer ensures that the I.W.T.W.D.T (I Wish This Would Do That) gossip system is energised — with more user nodes than old Mother Bell (ATT). Everyone starts to fantasise about features and speed and channel bandwidth, and the first thing you 'know' is that the *DDT-X* has more power than the computers on the Space Shuttle. The dealers are only human and they often pick up on the same rumours that everyone else, and it is not until the computer magazines reverse engineer the new product that the truth finally emerges. Sure, the *DDT-X* does more things than the *DDT* but it is not what the 'rumour mill' suggested. The bottom line is just as it was in Julius Caesar's time: 'Caveat Emptor' (Buyer Beware).

Always try to anticipate future needs. In other words, try to buy the 'most' computer you can afford, no matter what the intended initial application. Nobody will ever make a mistake buying a machine with the newest, fastest

## Always try to anticipate future needs

processor they can find. Ditto hard disk and RAM cache size, on-board RAM, plug-in sockets and so on. The only consideration is not to purchase 'beyond the curve'. That is, do not pay a penalty in size, price or function to get a technology so new that it has not become practical for mainstream applications.
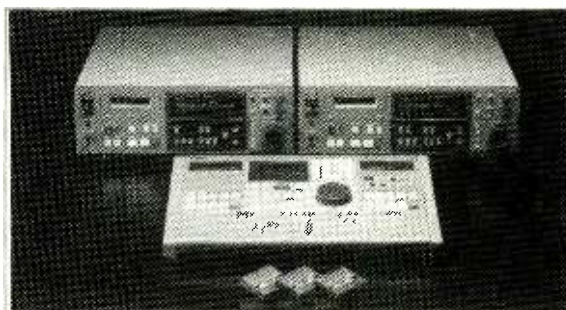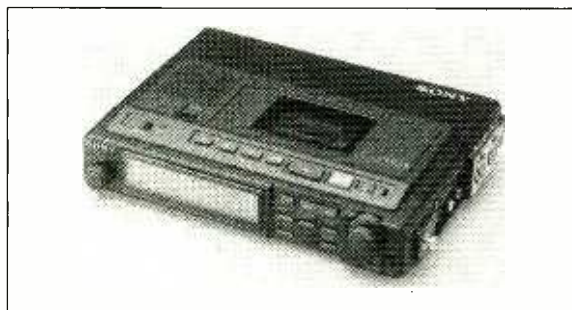
If you do not want to overextend on the initial purchase, then it is wise to be sure that a machine is upgradable. It is helpful that chip makers such as Intel have provided options for upgrading the 486 processors and the forthcoming P5. The next generation 'P' family of microprocessors will be plug-in replaceable as newer units are developed. Apple Computer has also established clear paths to upgrade its mid-range *Macintosh* units to the *Quadra* performance level. Other computer manufacturers offer similar options. So-called third party vendors make plug-in accelerator boards for many popular systems.

A point almost too obvious to make is that of avoiding 'bargain systems' and 'special packages.' The old saying, 'you get what you pay for' has lost none of its power in being translated into the language of the computer business. Do not let a sales person pressure you into buying something because it is a bargain or a special package. The extreme level of competitiveness found in computer retailing today means that any sale or package price will be repeated again in a relatively short time frame by the dealer you are 'dancing' with or one of his competitors. Any product being 'spiffed' because it is being discontinued will almost without fail be replaced by something better and most likely at a lower price than the original product in question. Do not be afraid of bargaining with a dealer over price; especially over used or discontinued products.
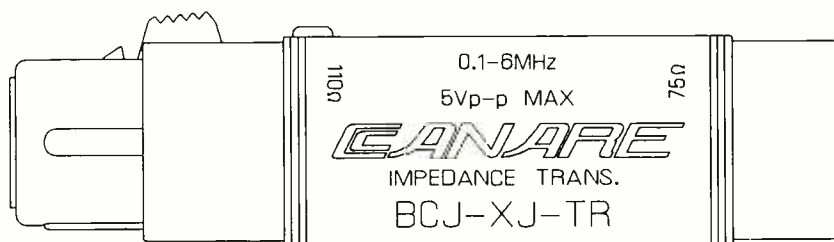
Possibly the easiest trap to fall into is believing you can wait for new technology or lower prices before buying. In the case of delaying a purchase hoping to get a better 'deal' or more 'bang for the technological buck,' you enter the computer world of the existential. You are essentially *Waiting For Godot*, and if you remember your Samuel Beckett, Godot never comes! That is to say that the growth and progress of computer technology never simply reaches a plateau and stops. The pricing decline of a given level of technology would be the corollary of the above. The price for existing technology just keeps declining as newer technology supersedes it. The best option is to shop carefully and commit to a purchase. Consider it as an investment in education as much as anything else. The learning experience will be worth the admission price — which may well be rather steep. When you move up to an improved system as your demands change, consider that your ability to utilise more significant technology is the result of your previous purchase. And never — but never — read the advertisements in the newspapers and magazines and conclude that you were taken for a ride because the price has fallen since your purchase. That is the nature of the computer business and you must remember that you have had the use of that technology for all of the time it has been depreciating. If you cannot live with this principle, take the best advice I can give and get out of the business. ■

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# POWER PLAY

## Sam Wise examines the requirements of power amplifiers for wide-band and limited-band audio

In two previous reviews the performance of the Hill *Chameleon* (*Studio Sound*, March 1992) and Australian Monitor *AM1600* (*Studio Sound*, April '92) power amplifiers were evaluated. These two amplifiers exemplify two very different design philosophies. The *Chameleon* design is based on the principle of supplying high levels of dynamic, or short-term, power with a lower 'continuous', or long-term, power rating. The benefits of this approach are reduced size, weight, cost and steady-state power consumption. The *AM1600* design presumes that the user wants or needs to have the same high power on tap continuously. It delivers practically the same 'continuous' output power as it will produce on the short peaks of dynamic signals.

## Bandwidth

Initial tests produced results for these two amplifiers which compare their power performance using a standardised laboratory test signal and also for full audio bandwidth music signals. However, amplifiers are not always required to handle the whole audio band as many loudspeaker systems (both studio monitor systems and sound reinforcement systems) make use of signal splitting in the form of controllers or crossovers.

As a final evaluation of the relevance of dynamic power, the music programme signals used in the original wide-band tests were passed through a modern Linkwitz-Riley active crossover having 24dB/octave slopes. The audio band was divided into four bands: 25Hz–160Hz, 160Hz–1.6kHz, 1.6kHz–8kHz and 8kHz–20kHz. The music test signals originally used for full audio bandwidth tests were passed through these filters and recorded as band-limited test signals. These were then replayed through the *Chameleon* amplifier at a level which just failed to illuminate the CLIP LED, as we had found this to be highly accurate. Unfortunately, the *AM1600* has been returned to the suppliers, but we have assumed that it will maintain its maximum peak power since it has almost identical continuous and dynamic power ratings. The results are shown in **Table 1**. Measurement accuracy is somewhat limited, the possibility of ±0.5dB error. This means that the amplitude of a 1000W test result

could actually be anything from 891W to 1122W.

At first this table appears bewildering. Why is there such a variation in peak power levels? Examination of the sampled amplifier output signals suggest two likely reasons for this. For example, on the Miles Davis recording, the peak signal which results in a 1008W peak output power from the broadband test lies within the 1.6kHz–8kHz band. When band limited, this peak can actually be higher since the amplifier's average power is reduced in this band. In the 160Hz–1.6kHz band, the peak power appears low, but this signal actually has a very flat amplitude characteristic and is driving the amplifier close to its continuous output power level — that is, there are no peaks in the signal within this frequency band. In the lowest frequency band, the predominate sound is that of the bass guitar, which has a transient attack peak and a lot of sustain.

The My Bloody Valentine recording is interesting because is consists largely of a 'smear' of instrument distortion. This is evident in the high proportion of the signal being produced in the upper frequency band. Also, the balance between frequency bands is quite even.

With the exception of the 160Hz–1.6kHz band in the Miles Davis recording, all of the output power performances are well above those of the *Chameleon* amplifier's continuous rating of 600W into 4Ω with both channels driven. The *AM1600* could be expected to deliver about 900W under all conditions — that is, under certain operating conditions the *Chameleon* will produce a higher peak output level, while under others the *AM1600* would produce a higher peak output.

## Limiting effects

Unfortunately, signal controllers not only split an incoming signal into various frequency bands, they can also introduce various forms of dynamic modification — including compressing and-or limiting the signal levels. Since each controller is different in performance, it is difficult to assess simply the requirements they impose upon an amplifier. In some cases, controllers reduce the dynamic range individually in each frequency band and prevent large signal peaks from reaching the amplifier. Usually this is intended to prevent burning out speaker driver coils or to prevent over excursion of the speaker cone or diaphragm. In most applications, the correct

choice and use of an amplifier with high dynamic power and more limited continuous power performances will protect the speaker voice-coils while allowing the peak level to remain high. Typically, compressor action will have the effect of reducing the peak-to-average ratio of a signal.

If an amplifier is being pushed to its performance limits, raising the average signal level relative to the peaks will place more stress upon the power supply. In the case of an amplifier with a continuous rating such as that of the Australian Monitor *AM1600*, there will be no reduction in peak power since peak and continuous ratings are the same. Amplifiers with a large dynamic headroom may actually experience reduced peak output power as more compression is applied. Regardless, the average level will increase. The details of these effects depend mainly upon the threshold, compression ratio and attack time constant of limiter-compressor circuits and upon the details of the design of an amplifier's power supply — specifically, how long it will sustain a high output peak level.

## Summary

It is evident that passing different types of music through a frequency dividing network will result in different peak output levels appearing in the various frequency bands. There is, as should be expected, a change in the peak-to-average (mean) ratio required of the amplifier as a result of splitting the signal. As can be seen from **Table 1**, there will be both increases and decreases in this ratio. A marginally decreased benefit from an amplifier having a higher dynamic performance in certain bands can result, but this is only so if the musical content of the signal is more or less consistent. This may be important when choosing amplifiers for use with a touring band using consistent instrumentation and playing a consistent style of music throughout. Otherwise, the variety of possible musical signals strongly suggests the use of amplifiers with a relatively high peak-to-mean performance ratio in all frequency bands.

Limiting and compressing a signal will reduce the benefit of a high-dynamic power amplifier, but only if the process is performed on the whole signal rather than on individual instruments, as is usually the case.

Therefore, it seems, a well-designed power amplifier with high dynamic headroom is likely to give better 'bang-for-the-buck' than an amplifier with a high continuous rating. In most cases, consideration of audio quality being equal, there will be no audible difference, given that the EIA RS-490 dynamic power level is the same. ■

| TABLE 1 | | | |
|---|---|---|---|
| Signal Bandwidth | Miles Davis *Tutu* | Prokofiev. Finale from *Symphony in D, Opus 25* | My Bloody Valentine |
| 2Hz–20kHz | 1008W Peak | 920W Peak | 870W Peak |
| 160Hz–1600Hz | 630W Peak | 860W Peak | 809 W Peak |
| 1.6kHz–8kHz | 1156W Peak | 725W Peak | 987 W Peak |
| 8kHz–20kHz | NA | NA | NA |

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# Seeing Red

Dear sir, I read with interest Mike Lethby's report on The Red Square Invites festival ('Ten days in Moscow', *Studio Sound*, November 1992) and all its attendant tribulations.

Having been well acquainted with this festival, I thought your readers might find some further background interesting — as well as giving a bit more insight into the problems of dealing with the ex-Soviet Union.

In fact, the idea for the festival dates back to 1989 when we (Cinac Systems) were starting the installation of the new sound studio complex at Mosfilm in Moscow. Omari Sokhadze of Intertheatre had the interesting idea of a large opera festival on Red Square to celebrate the 'opening up of Russia', and we were asked to conduct a feasibility study as to whether a decent reinforced sound could be obtained or not (due to the notorious slapback echo in the Square).

Like most problems in life, this could be solved if the budget permits the correct solution to be applied, and we enthusiastically pronounced it possible.

There then followed literally years of discussions, changes of opinion, production companies came and went — as did sponsors — and it was all too evident from early on that while Mr Sokhadze might have good ideas, he did not have the experience to carry them out. However, he was speaking to the right people, and we had a top-notch team lined up.

By the beginning of 1992, the festival had already been put back two years. A major production company had pulled out due to lack of enthusiasm on the part of the backers (the situation in Russia was hardly encouraging), and the run-in time to properly organise and publicise a festival of this type for 1992 was considered too short. Cinac Systems were still '110% in the team' and had a contract of intent which had been signed several years previously. Based on the probable stage layout, we had an L-C-R main system design with surround for the seating area and delay systems for the video screens for the rest of the Square. A Saje *Memory* console would be used together with the appropriate processing gear and full stage monitoring. The final system would be worked out and budgeted once we had a fixed programme.

Several production companies later, we were informed that Birchwatt were now handling production and that all was 'going ahead'. This was still the situation at Easter 1992 and, although time was now very tight, it was still possible to stage the event from our point of view. All that was needed was the money (and the artists).

We then heard nothing more, and faxes were not returned. We assumed that the festival was off. However, upon arriving in Moscow in May for a 'tuning up' visit to Mosfilm, we found that the festival was going ahead after all.

We immediately called Mr Sokhadze, who reluctantly came to the telephone and dealt with us in a very offhand manner (virtually 'who the

hell are you?'). We called it quits.

As well as giving some background to the run-up of this festival, this letter is also intended to outline some of the difficulties involved in working with our ex-Soviet friends. Our experience with other projects has shown that there is something approaching resentment of the fact that you are 'coming in' to do work there, and it is very hard to get enthusiastic (or even normal) cooperation out of people. You can never do enough, and it is always your fault if anything goes wrong (and I thought we were on this project together). This can be lived with but it is all the more frustrating when you know how much better things could be.

In the case of the Red Square festival, we did a lot of work, consulted with a lot of top people and generally did all we could to move things along before being dropped like a hot potato. Okay, so that's life; you can't win 'em all. However, it is a shame that, when reading about all the problems that Birchwatt-Britro had to deal with, I know we had a team that was both international and local, involving real interpreters, technical support (including the sound department of the Bolshoi) to help overcome the inevitable, but anticipated, difficulties.

Judging from Mike's article, it looks as if the whole crew did an excellent job and, fortunately, had either local contacts or experience to rely on. However, I sympathise with their sentiments entirely in terms of being let down and of what the festival might have been.

Please don't misinterpret this letter as 'sour grapes'. Rather, I am trying to demonstrate that working with the Russians is going to remain difficult as, even when you are told that everyone wants to learn from you, they nearly always do things how they want, and you have to live with it.

Mick Kluczynski is so right when he says that his crew learned a lot from the Russians but that it was not a two-way process. Historical and social reasons dictate that this is likely to remain the case for some time to come, and it is best for foreign visitors to accept the situation and to try to work with the situation rather than wonder why things are so different.

**Terry Nelson, Cinac Systems SA, PO Box 17, 1422 Grandson, Switzerland.**

# Square Dance

Dear sir, I read with interest your article on the recent Red Square Invites production in Moscow, and I would like to take this opportunity to inform your readers of an earlier and equally enterprising event which demonstrates that the Russians are more than capable of teaching us a thing or two about concert promotion.

On June 21st 1992, a huge crowd (estimated at more than 150,000) enjoyed live folk and rock music from top Russian and French artists — the first ever privately-organised concert to take place in Red Square, and in near-perfect blazing June weather. The concert's organisers, Orion Service Company and Moskovsky Konsomolets newspaper, had wisely decided not to charge for

tickets, figuring the three or four million roubles wouldn't be worth a whole lot by the time they got it in the bank and exchanged it for dollars in the face of a free-falling exchange rate. Instead, the Russian promoters were very successful in attracting quality sponsorship — from Coca-Cola among others — and the entire 3-hour show was televised and transmitted to households all over the Russian states.

Coincidentally, this type of sound reinforcement production could only have been made possible by the use of Turbosound's long throw *Flashlight* system, which is capable of covering distances of up to 200 metres without significant loss of high frequencies, and hence without delay towers.

The front of house PA consisted of a mere 36 *Flashlight* high-low pairs (about eight tons), flown left and right of the 60-metre-wide stage, an covering Red Square perfectly. Orion's sound crew got first sight of the equipment on the day before the show and had the system flying in a matter of hours. The only potential hitch occurred when, 20 minutes before the start of the show, mains power (rumoured to be courtesy of the Kremlin) still had not been restored following the afternoon rain shower.

**Martin Reid, Sales and Marketing Manager, Turbosound Ltd, Star Road, Partridge Green, West Sussex, RH13 8RY.**

# On the Tube

Dear sir, thank you for the constructive review of the Hughes & Kettner *Tubeman* (*Studio Sound*, December 1992).

We appreciate your comments are favourable to the product. We do wish, however, to advise that the product is not an effects unit like a DOD or Boss pedal, and has not been designed for this purpose. It is very much a valve preamp that provides true valve sounds when used in conjunction with a solid-state amplifier.

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**John H Skewes, Managing Director, John Hornby Skewes & Co Ltd, Salem House, Parkinson Approach, Garforth, Leeds LS25 2HR.** ■

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ERRATA
Two errors appeared in '10 Days in Moscow', November '92, *Studio Sound*: the photograph on page 35 shows Alan Roberts, not Bob Birch, with Mick Kluczynski and Colin Rowell, and Birchwatt Productions is jointly owned by Bob Birch and Mick Kluczynski. Apologies for any inconvenience caused.

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# Mixing the bright, dead and live

There are times when modern recording techniques can be so sophisticated that we miss out on the possibilities for making a track unique. As little as 10 years ago it would not have been unusual for the mix engineer to have only two or three reverb devices to hand. The 1970s had shown that recording instruments by direct injection meant that everything was so up-front in character it was almost touching — very flat, lots of detail but desperately lacking in depth. We knew that a solution would be the ability to treat each sound with a different acoustic character and so break up the one dimensionality. The limited amount of mix processing to hand meant that signals often had to share reverb settings and this was not the complete answer.

In recent years we have been blessed with more low-cost-but-good digital reverb systems than we know what to do with and there is little to prevent a different processing on each signal. However, I think we may be missing out on some of the sound character that the restrictions of old carried with them. (There is just no pleasing some folks!)

While not advocating the return of the 'just like being at a live performance'-style of recording, there can frequently be much to gain from a sense of cohesiveness in the sound character — as if some of the signals had a relationship in their ambient quality to each other. Whether this is in acoustic ambience or artificial may not matter but such techniques can give an 'attitude' to a track that may not be present if other mixing techniques have been employed. This could be considered as the antithesis to mixing technique where signals are balanced against each other in level and frequency so that all the right emphasis is there but lacking in an overall sonic character. Neither is right and neither is wrong — just different.

The key to the creation of a sonic character may lie with the producer but the tools have to be to hand. And it is some of those tools that I want to talk about.

In many cases we might just be talking about the right recording environment. Battling with too live or too dead acoustics just saps creativity. It may work for some titles but not for all. Having worked in overdead studios for many years when they were more popular as a concept, you soon develop techniques

for adding ambience in the immediate vicinity of the recording. Your predominant problem is the fact that a room may be dead sounding but usually outside of an anechoic chamber that is simply an illusion. Normally there is still reflection from surfaces around you, only that with the number of the reflecting surfaces reduced those remaining sound more prominent and frequently less pleasant.

While we may be able to overcome much of the unwanted character of a dry signal with EQ and artificial reverb, there is little that can be done to remove ambience from an over-live signal. Flying in the face of these points, however, the optimum room for average applications would preferably be on the live side. Such a room would naturally sound better than a drier acoustic, giving the musicians a better feeling and in turn a better performance. Also in general it is easier to damp down an acoustic in small areas rather than liven up a dry room — in small or large areas.

Acoustic control of small areas is a powerful tool in track laying. Some studio designs build in special acoustic areas but these may not always be as extreme as you may need. Experimentation can produce some interesting results.

In the early 1970s one of the effects that the engineer was meant to be able to recreate was the 'Stephen Stills Guitar Sound'. It was one of those sounds that really required Stephen Stills, his Martin acoustic guitar and his unique playing technique but some of the elements were achievable. The guitar had to be closely miked and then compressed heavily with fast attack and medium slow release so that the sustain was immense. So that the guitar sound

retained the same character as the compressor released, it was necessary to make the area around the guitar as dead as possible so that when the gain rose on sustained notes there was no increase in the ambient sound relative to the guitar and hence 'distance' in the resultant sound. It took weeks of experimenting not to achieve this until Stills happened to mention in an interview that the answer was cushions — as many as you could manage. Armed with this information we built a mini anechoic chamber — cushions on the chair, under the chair, over the floor and in a tight circle around the musician to about shoulder height. There was just about enough room to gently move but not too much. The musician's head is above the cushions so he is not likely to suffer from the claustrophobia induced by very dead acoustics. It takes some preparation and a musician who wants to achieve an end result and it works very well.

I have used this technique for many instruments and a few voices and it works. You have build it up each time around the sound source — it would require too many compromises if you permanently constructed an acoustic cage — remember it needs to fit closely around the instrument.

Taking the building concept further, during the LA AES Convention in 1990, there was an exhibit from a company under the name of Pelonis Sound & Acoustics (Downey, California) which presented great possibilities. Named 'The Edge', it was a broadband sound absorber that became increasingly more reflective with increasing frequency. It was wedge-shaped — about four feet on the angled dimension, depths of 21 and 7 inches at each end and an

overall length of about eight feet (a sort of giant Toblerone). It was not just stuffed with absorbent material but had tuned multiple chambers that would diffuse what energy they did not absorb. The interesting thing about these units was that they were stable in whatever face they stood on. Theoretically it looked as if it would be possible to create quite complex recording environments using these units as giant building blocks, tuning areas of specific reasons and then moving it all around the following day.

This was the only time I saw that company or heard from them so I have no idea what they are up to now. Unfortunately for them, it was such a good idea that it made you want to go away and work on your own designs rather get your money out for theirs.

Another technique that perhaps has limited application but sometimes can work if you follow it through is variable distance miking. If you are in the lucky position of knowing how the final stereo image on a title should sound then you could increase the impression of depth by the use of distance between the mic and the instrument — not just on a few, but all the instruments, deciding what should be close, what at the rear and using positions to achieve that. I have used it on small bands with multiple overdubs to some effect but if everyone is not on your side there may be problems.

You can always partially recreate this by replaying a signal through monitors in the studio and rerecording the ambience with differing mic and room positions and distances. This allows you to work the event which may be more convenient but less effective. You will probably have to record the ambience signal in stereo to achieve the result but all the individual signals can be sub mixed once the balance and image position is sorted.

None of these techniques are possible during the mix stage and need to be completed during track laying. However you do need to experiment before session time or you may find anxious faces, or worse, bored faces wondering why you are taking so long to achieve an effect that they do not understand. And in such cases, if the clients are unsympathetic, it does not matter if you have created the greatest track and mix structure ever — you wasted their time which they rarely forget — unless it wins them a Grammy. ■

ILLUSTRATION: CARL FLINT

"MUSIC ANNEX's first AMEK purchase was in 1977. Our HENDRIX film console for Studio IV San Francisco is our ninth AMEK. Film mixer Will Harvey specified a fully-featured equalizer at the front of the module and a flexible bus matrix with true LCRS panning. We found out this is standard equipment on HENDRIX. In addition he required a powerful automation system and a small footprint so that he could work alone. Also standard. He got what he wanted and I got what I wanted: a remarkable console for a reasonable price."

David Porter, President, MUSIC ANNEX Inc, San Francisco, California President, SPARS, 1989 - 1990

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HENDRIX

HENDRIX

THE HENDRIX THE SPARS 90
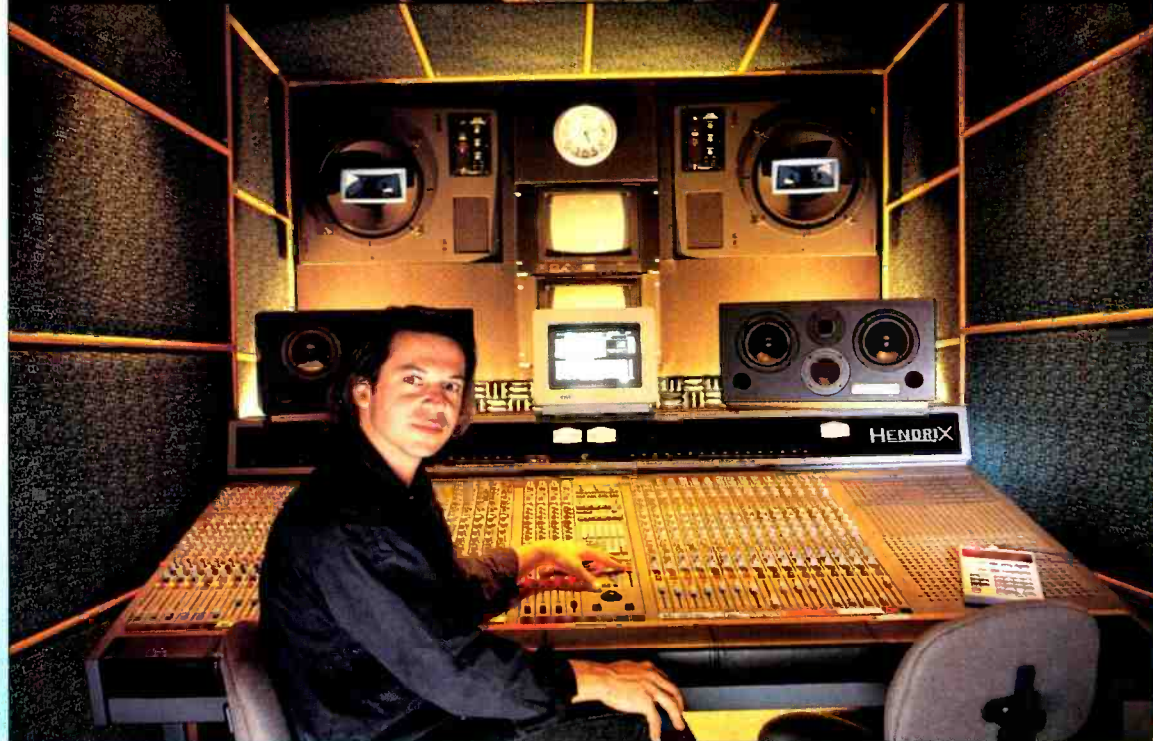
"Over the course of 15 years (and more than a dozen gold and platinum albums), ARTISAN RECORDERS has built a very demanding client base who expect nothing less than the extraordinary. Therefore when the time came to build our new RED TRUCK, we had to think long and hard about our choice of a mixing console. I knew of AMEK's wonderful EQ from previous work on an ANGELA console , and the company has a well-established pedigree with some of the best producers and artists in the business.

In HENDRIX , I found a console whose brilliant ergonomic design fits the parameters of mobile recording, and whose first rate sonics and automation would satisfy my most demanding and valued clients. This truck had to be exceptional in every way - and it is." Peter Yianilos, Owner, ARTISAN RECORDERS

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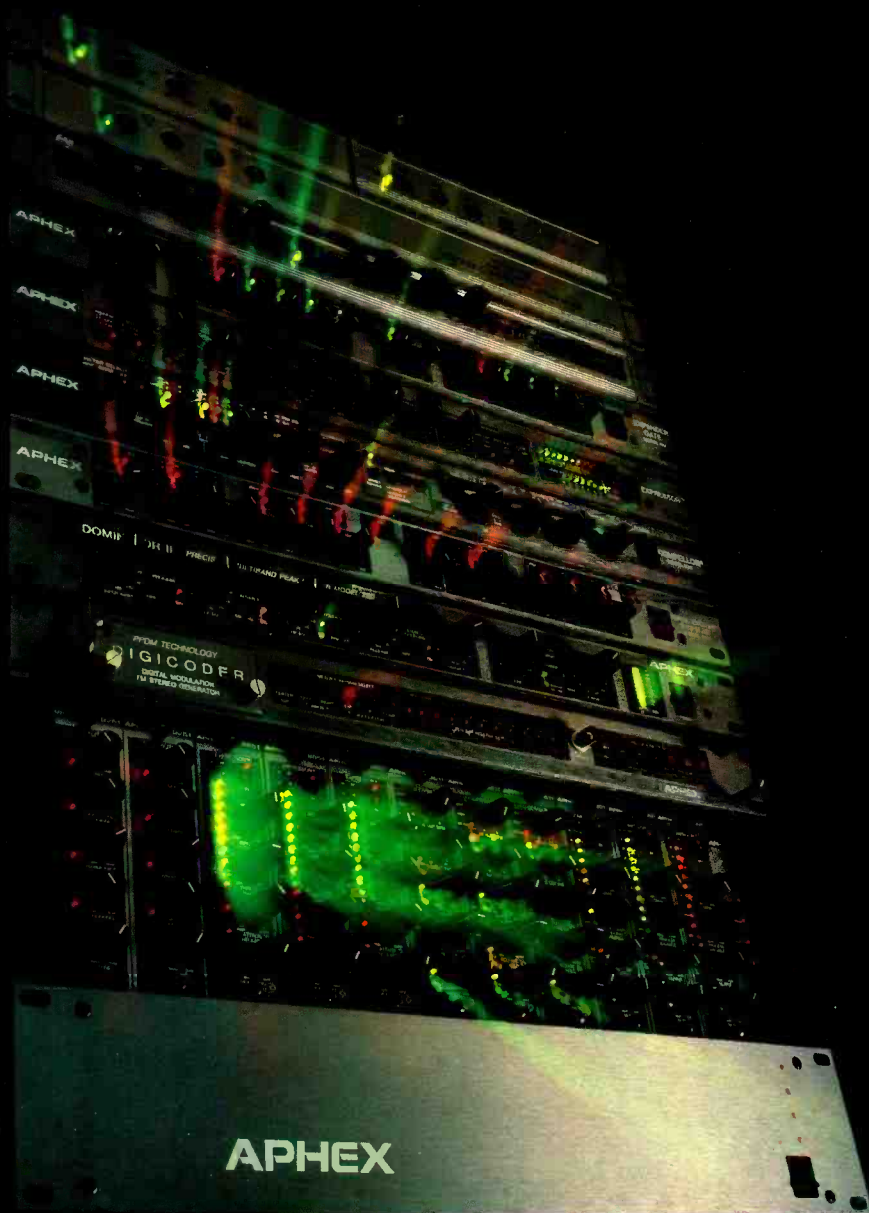AMEK/TAC US Operations: Telephone 818/508 9788 Fax 818/508 8619.

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