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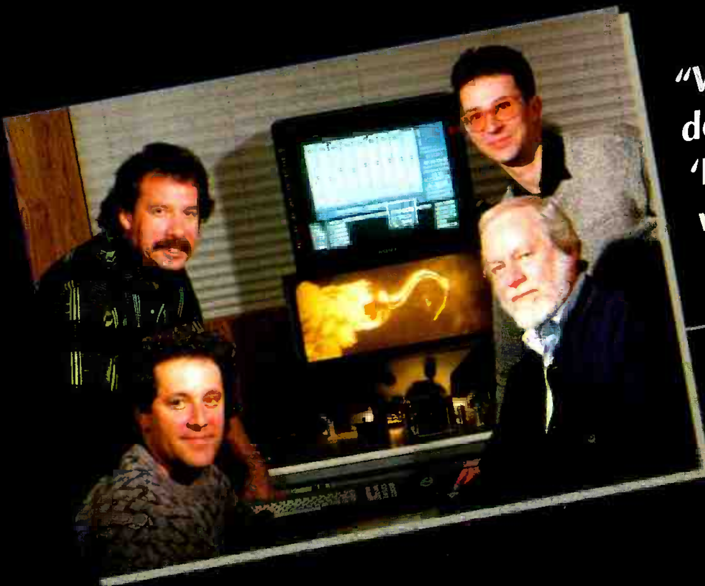
ATC monitors — see 'In the Driver's Seat' page 53

USER-FRIENDLY, CLIENT-FRIENDLY



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Speakers' corner

There has always been an implicit conflict between the objectivity of audio equipment design and the subjectivity of actually listening to music. On one hand you have, for example, the (general) consistency offered by electronic components and the reassurance of accurate measuring devices. On the other you have such indefinable qualities as 'feel' and 'catchiness'. It is, of course, the stuff of audio witchcraft.

One particularly acute area of subjective-objective conflict is that of audio monitoring. While it is relatively easy to represent the performance of a monitor speaker design in the form of figures and plots, the whole exercise is severely compromised once that monitor is installed in a working environment. In fact, the whole process of evaluating monitor performance is so subjective that monitor 'reviews', as such, are not an accepted part of the format of *Studio Sound*.

As we are all aware, the acoustic of the listening environment is a sufficiently significant factor that auditioning different monitors in the same room can give a more consistent result than auditioning the same monitors in different rooms — when the monitors are positioned sufficiently far from the listener to allow the room characteristics to register. It would, then, make considerable sense to position the monitors such that the listener hears the direct and indirect components of sound in a proportion which renders the reflected component insignificant. What we are talking about is midfield monitoring.

Nearfield monitoring is hardly a new concept, but its main application to date has been more the emulation of domestic listening conditions than elimination of room acoustics. Its other value has been that of standardisation — nobody's about to praise the performance of Yamaha *NS10*s, but they do provide an invaluable reference point between the studio and the outside world as well as between studios themselves. But the question remains: why not use high-powered, close-proximity monitoring in an effort to raise listening standards?

For the motion are certain of the monitor manufacturers who have designed and produced speakers precisely for this purpose — at least one such design goes as far as to allow for reflections from the surface of the mixing desk in its design. Against it are the old opponents of conservatism and bad practice.

In spite of its associations with youth culture, the recording industry is now mature enough to have developed its conservatives. While this is neither surprising nor entirely unhealthy, it is certainly not entirely healthy. On the downside, people's attitudes regarding studio monitoring necessarily address the issue of commerciality — and, like it or not, many studio bookings are still taking place on the strength of the equipment they can list. Placing yourself in the position of the studio manager, therefore, you have to assess the relative merits of the best technical monitoring system against those of the most commercially attractive system. That is not really such a shock until you start to assess the 'on air' times of the main monitors against that of the nearfields. Judged this way, 'serious' monitors represent a frighteningly high investment. It is a sad reality that more cost-effective, and potentially more accurate, listening environments are available, yet their use is being prohibited through ignorance.

In an industry so preoccupied with progress it is curious that monitoring remains such a subjective experience. But then it is a curious industry whose business is so dependent upon subjectivity. ■

Tim Goodyer

Cover: DynaudioAcoustics *M4M* main monitor
Photography at The Hit Factory, London, by Nick Milner

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Autumn digital audio times 1992

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STUDIO
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Sting to-go

The complete SSL portable studio can be unpacked and ready for use in two hours

The adage that the best ideas are simple ones, is something that SSL have been successfully demonstrating for years. They have also shown that giving the client what they want, rather than what the company think they want, is generally a good way of sustaining sales. So it was with intrigue that I took a trip to SSL's Begbroke headquarters to see a product that qualifies in both respects.

A few days before my visit, I had received a rather cryptic phone call from SSL's marketing director Colin Pringle, who talked of *Portastudio*s and the like — not something one immediately associates with SSL. However, after arriving at Begbroke and being ushered through a door with 'Strictly No Visitors', scrawled on it, things began to fall into place. The room was filled with flight cases, in the midst of which was a *G series* console. However, closer inspection revealed this to be no ordinary desk; with feet firmly fixed in flight case bases, and prominent dividing strips separating channel sections from the centre section, it became clear that this console was designed to go! In fact the 64 - channel console splits into three flight - cased sections – two 32-channel blocks and a central block. The stand-alone patchbay is encased separately as are the computer rack and video monitors — and it does not stop there. A whole studio's worth of equipment including power supplies, power amps, speakers, outboard racks, digital multitrack (Sony *3348*), two-track tape machines (R-DATs and cassettes), assorted cables microphones and spares have been packed away into 24 custom-designed flight cases from American company Nashville Cases. This totally portable, 48-track studio will neatly load into the back of a 10-ton truck, and SSL claim that two people can assemble it in under two hours, and break it down in half that time.

The concept of a portable console has been in SSL minds for some time, in particular for broadcasters who have a temporary requirement for additional audio production facilities. But there also appears to be an increasing trend amongst artists now to work away from the studio — Def Leppard, U2, Stevie Nicks, Billy Joel, Rod Stewart, and Mike Oldfield are just a few names that have recently rented houses and-or equipment. So when the American hire company Steerpike approached SSL with the idea, they were only too pleased to oblige. Apart from the major modifications to the *4000 G* console, SSL became involved in all aspects of the project, feeling the only way to truly ensure its success was to design, construct and test the entire system themselves.

Steerpike is owned by Sting, who will also be the first to use the portable studio to record and mix his next solo album. Sting's last album *The Soul Cages*, was recorded in various studios in France, Italy, America and the UK over a nine-month period. Coproducer and engineer Hugh Padgham, explains how the idea for a portable studio evolved from that experience.

'We took the Voyager mobile to house near Pisa for part of the last album, and we found it was a very relaxed way of working — it took a lot of the pressure off, there was none of that red light syndrome you get in the studio. The only problem using a mobile, though, is that you're very cooped up in this small area, so the idea with the portable studio is to gain all the advantages of a mobile without the space restrictions. '

Padgham was also involved in specifying various elements of the setup.

'When the idea first got banded around, I came on the scene and said I wanted the desk to have things like VU meters rather than bargraph meters, and *E Series* equalisers rather than *G series* equalisers. The outboard equipment was put together with a view of it being a package that contained most things that people would want, although of course it can be varied to suit the client. The whole setup represents a package with everything you need for tracklaying or mixing, including *Ultimation*'.

Main objectives were that no case should weigh in excess of 400 pounds, and that everything should fit through a standard, single doorway — this was successfully put to the test in negotiating corridors and doors at SSL HQ. The console cases themselves are on wheels and have a depth of just 27 inches a feat that has been achieved by building in a pivoting mechanism into the frame allowing each section to tip backwards from the horizontal to the vertical. In this position the section is locked into place and the top covers are secured ready for transport. Each case is heavily padded with medium-density foam rubber to cushion against any shocks, and the console frame has been redesigned in places to add greater strength. The meters have been mounted slightly differently to ensure extra stability, and card retainers have been fitted where necessary.

Another important consideration was that interconnections should be kept to a minimum and be as straight forward as possible. All audio interconnections are made via substantial 150-pin Amp connectors; each 32-channel section has five cables connecting it to the remote patchbay, and one connecting it to the centre section. The centre section itself is connected to the patch by three cables. Mini Delta ribbon connectors are used to connect computer and machine control functions, and a video multicore books up the console TV and flight cased RGB monitors.

Each section has separate power distribution via a 48-pin connector carrying audio power plus motor power for *Ultimation*. Power distribution is handled by Furman auto-sensing multi voltage power conditioning units, and a total of 11 are used for the entire setup. Being auto-sensing (90 to 250V) there is no need to reset voltage for different countries.

The patchbay has connections for two 48-track machines; multitrack 2 sends are normalled from multitrack 1, allowing two *3348* machines to seamlessly overlap during live recording. SSL are supplying a 48-way stage box which plugs directly into the back of the patchbay rack; another 16 lines are provided at the rack itself. The various outboard racks connect directly to the patch, as does a Nemesis 8-way cue mixing system – the first from the system's new manufacturer Shep Associates.

The whole portable studio package, which has taken four months to complete, was delivered to Sting's new Wiltshire home on August 8th where it has been set up in a large living room, doubling as both studio and control room.

Early reports suggest that all is going very well. ■

Patrick Stapley

Steerpike. Tel: +1 201 481 5797. Fax: +1 301 481 3148.

Xeric Studios in Limerick — see Contracts

Contracts

- Xeric Studios in Limerick have bought a Soundtracs *In-Line* 48/32 production console with *Tracmix* automation and Akai digital recorders, making it the only 48-track digital studio in Ireland.
- BOP Recording Studios in Bophuthatswana have chosen FM Acoustics *801A* power amplifiers. The amplifiers are driving Kinoshita monitors in a special way; one side of the *801A* drives the crossover for the bass frequencies; the other side drives the crossover for the treble frequencies. Two 801As are therefore used upfront and a third is used to drive two infrasonic speakers that have a frequency range of 10 to 25 Hz.
- The Finnish State Opera have selected Dynaudio Acoustic monitors for all of their monitoring requirements at the new State Opera House in Helsinki. Main monitoring will be provided by the new *M4M* 4 x 12in four-way active system, which the Opera team first heard at the Vienna AES.
- Recent Nexo contracts include a *PCLine* based system to the Centro Cultural 'Caixa Terrassa' in Spain. Also in Switzerland distributor Zap Sono has supplied a *PCLine* system to the Theatre Pour Enfants, a theatre intended purely for children's productions.
- Studio design consultancy Harris, Grant Associates have been appointed to provide acoustic, design and architectural consultancy for the comprehensive refurbishment of Studio 3 at the Hitokuchi-Zaka recording complex in Tokyo.
- Lexicon have announced the sale of a complete *OPUS* system and an *OPUS/e* editor to Woodholly, a leading postproduction facility in Hollywood, CA.

Update

Some slight adjustments are needed to our Audio Recording feature of last issue. In the DAT Players entry, the Sony *PCM2500* is a model no longer made; the *SV3700* and *SV3900* are made by Panasonic; the Casio *DA-7* is now a discontinued model; and a player we failed to mention altogether was the popular *HHB 1 PRO* professional DAT recorder available from HHB Communications. Tel: 081 960 2144.

In-brief

- For the last few months, Richard James Burgess has been in Orinoco's Neve room recording and mixing a new album by Praise using *Q-Sound*, a system that enables the user to place sounds outside the normal stereo field. This should be the first complete album produced in this country using the technique.
- A new pro-audio hire company, FX Rentals, has been launched in London. Says Richard Conway of FX Rentals, 'Customers in the current economic climate are primarily requesting a high standard of service with a focus on value for money. We have the people, the knowledge and the equipment to satisfy those needs.' FX Rentals, 213 Kilburn Lane, London W10 4BQ. Tel: 081 964 2288. Fax: 081 964 1910.
- Beginning this October, all new cassettes released by BMG Classics in the US and Europe will be recorded with Dolby *S*-type, Dolby Labs have announced.
- Pop music impresario Pete Waterman, has divulged plans for a major recording studio in a converted Manchester chapel this September.

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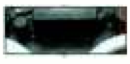
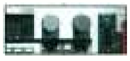

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ATARI FALCON

When the US-based Atari Corporation launched their *ST* range in 1986, it signalled a significant change in the role of the microcomputer. While the *ST* was initially intended to be a powerful games machine, it was subsequently supported by many professional software houses which provided it with a plethora of high-quality applications in word processing, desktop publishing, graphics and, of course, music. The latter was due in no small part to the inclusion of built-in MIDI ports.

Nothing lasts forever, and the *ST* is now a decidedly dated technology. While there have been a few enhanced versions of the machine (including the *STE* and *Mega-STE*), Atari have floundered where new products are concerned. Many projects failed to make it past the design stage while others have been ill-conceived — such as the *Notebook*, a recently-released portable utilising technology from 1988. Even the *TT*, which was intended to compete at the higher end of the market, has only enjoyed real success in Germany.

Rumours of a new machine first began to filter through some months ago. When a supposedly official specification was posted on *Usenet* towards the end of Feburary this year, it was sufficiently impressive to be greeted with general amazement. But with a few slight changes, the spec proved to be accurate. Originally codenamed the Sparrow, the *Falcon 030* is based around a Motorola 68030 processor running at 16 MHz, making it similar to the lower-end Apple *Macintosh* machines (*Classic II* and *LC20 II*). This chip uses independent 32-bit data and address buses for increased parallel performance, and an optional Motorola 68882 floating-point coprocessor can be fitted to assist processor-intensive operations. The operating system will be a multitasking system, called *MultiTOS*. Memory takes the form of SIMMs (Single In-Line Memory Modules) and comes in three configurations: 1Mb 4 Mb and 14 Mb. The internal disk drive is of the 1.44 Mbyte variety with an optionally-available internal IDE hard drive. Externally, the machine will initially use the same casing as the *ST*, although it will be coloured dark grey. It is, however, anticipated that this will be changed towards the middle of 1993.

Graphically, the *Falcon 030* is Super VGA-compatible and will handle 16-bit colour giving a palette of 262,144 possible colours, up to 65,536 of which are simultaneously available. The *Falcon* also has a 15-bit overlay mode for video titling and will accept an external video signal for GenLocking.

What is most interesting is the inclusion of a Motorola 56001 Digital Signal Processor running at 32 MHz (the same chip is used in digital audio boards such as Mark of the Unicorn's *Digital Waveboard* for the *Mac*). This was originally intended to be an 'optional extra' but Atari felt including it as standard would open up the computer's potential applications.

As might be expected, the *Falcon 030* has MIDI In and Out ports to facilitate its use with MIDI software — much of which is likely to be updated from its *ST* version. Other connections include a 16-bit DMA (Direct Memory Access) port which will handle up to eight channels of CD-quality audio — in fact, the sampling bandwidth is 50 kHz. To complement this there are 16-bit stereo A-D and D-A ports (although these appear on 3.5 mm jack sockets). Also available is a software-driven subsystem matrix to allow users to interconnect the inputs and outputs from the DMA chip, A-D and D-A convertors. The inclusion of a SCSI II port will come as a relief to Atari users accustomed to using SCSI-to-DMA conversion cards. Considered in tandem, the DSP and SCSI II ports open the door on direct-to-disk recording systems unhindered by ancilliary hardware. In fact, they should permit various different kinds of real-time musical applications — digital synthesis and creation of sample waveforms are just two such possibilities.

How many different tasks the *Falcon 030* will be able to perform concurrently will be down to the efficiency of *MultiTOS* which will allow you to prioritise certain tasks over others, facilitating background working. Applications will also be able to communicate with each other, so permitting data sharing between programs.

The potential of the *Falcon 030* is significant. From the digital audio standpoint, direct-to-disk recording will be possible without the use of any additional hardware (apart from a fast hard disk connected to the SCSI II port). The introduction of Peavey's SMDI, a protocol allowing the transmission of MIDI data via SCSI, will also have a part to play if it is accepted as a standard and this may lead into a 'workstation' application for the *Falcon* where digital audio, MIDI and graphics are all handled by the same low-cost computer. UK prices will be £399 (about $750) for the 'domestic', 1 Mb RAM version and £499 (about $948) for a 4 Mb RAM version (inc VAT).

From the launch of the original *ST* back in the early 80s, third-party software support for Atari has been readily forthcoming and, judged by the reaction of programmers attending a recent development meeting, the *Falcon* will be well received. Its launch is set for the Dusseldorf Messe on the 21st August (arrival in the UK before the end of the year). You have been warned. ■

Vic Lennard

Atari Computer Corporation, Tel: +1 408 745 2000. Fax: +1 408 745 8800.
Atari Corporation UK Ltd, Atari Tel: 0753 53 3344. Fax: 0753 822914.

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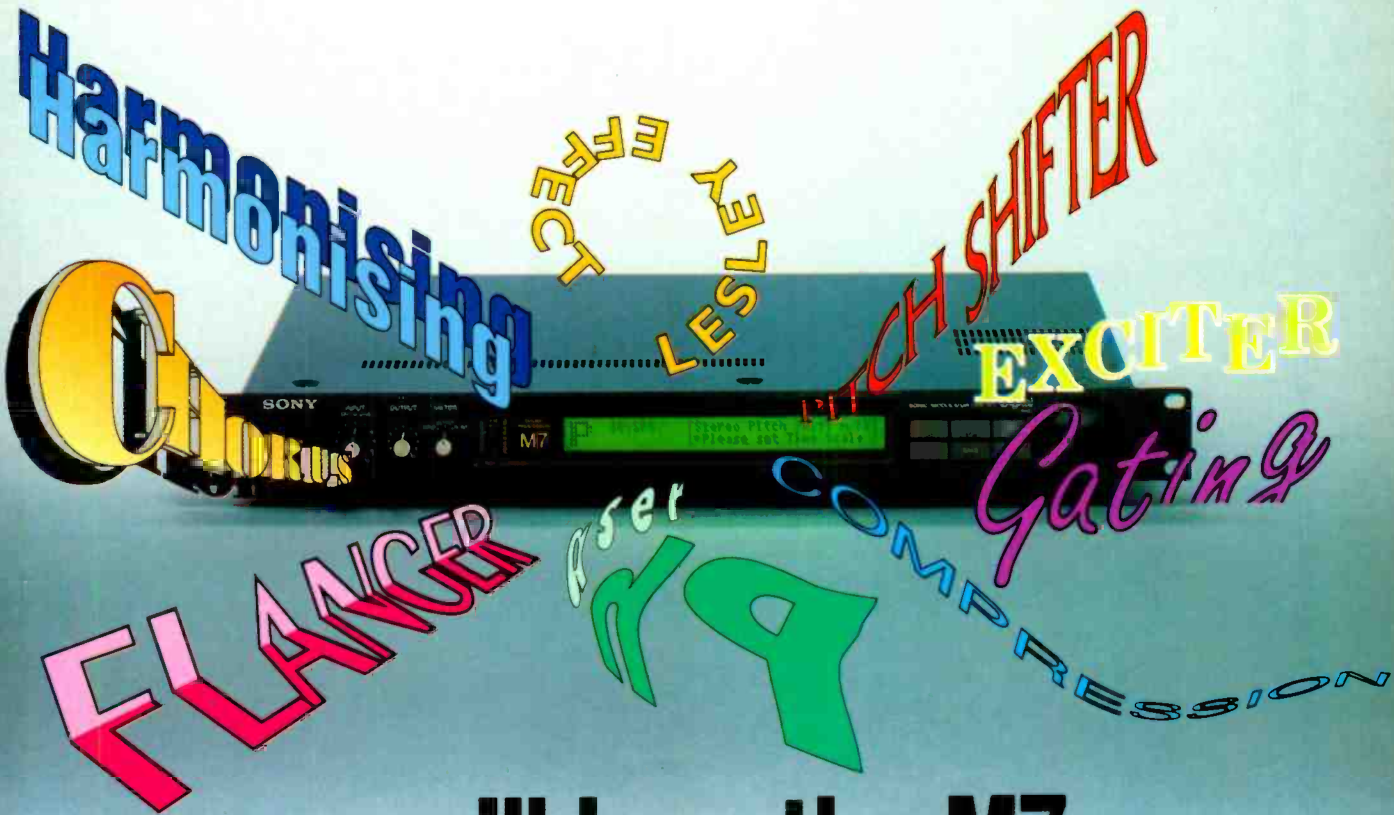

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Middle East, Geneva 022 7336350 Africa, UK 0256 55011

Sony Broadcast & Communications

VOCE DMI 64 Mark II organ module

Searching for a modern synth module to convincingly take care of organ sounds is not an easy task, yet it remains a very real need. While most contemporary synths carry a collection of usable organ sounds in their repertoire, using them for this purpose in a rig often brings the keyboardist to a logistical problem as dedicating an expensive unit with the potential for some unusual and individual tones of its own to a fixed-velocity run-of-the-mill sound is also wasteful. The situation is further aggravated by the fact that most modern synths run a narrow selection of organ sounds, yet the classic tonewheel organ that they try to replicate had rather broader capability. VOCE's *DMI64* Mark II is a no-holds-barred attempt at replicating tonewheel sounds via sampled waveforms and additive synthesis, all brought down to earth with good analogue crunch. It has 64-voice polyphonic, 16-channel multitimbral and uses a heavy dose of real-time MIDI to bring it all alive.

Improvements over the Mark I include front-panel display with data-entry buttons, simulated tube overdrive and an extra 35 presets (bringing the total to 99). Otherwise the 1U rackmount is pretty similar and spartan: front-panel power button, a comms LED that signals MIDI activity, and four-digit LED with two pairs of increment-decrement buttons either side.

The back panel houses a socket for the external PSU, MIDI In, Out and Thru, and two audio outputs which can be addressed individually from the preset programming routines. Presets are recalled from the front panel with the leftmost increment buttons or via MIDI patch change and the patches and can then be edited from the front panel or via software available for IBM *PC, XT, AT* and 386 or compatibles and Atari computers. The software is free of charge but requires the buyer to apply for the disk direct to VOCE. This was unavailable for the purposes of this review but whatever it does it has to be an improvement over editing from the front panel which, while certainly possible, is tedious beyond belief.

Parameters are called up with the increment-decrement buttons to the left of the display and the values adjusted with those on the right. An indifferent button response, cyclical scrolling through parameters and a general lethargy of operating speed combine to make the process irritating. But it is almost unimportant because of the strength and variety of the presets . Before long it becomes easy to dip into the editing mode to tweak a chorus speed here, and a vibrato depth there, providing you keep the manual open on the pages that tell you which digits correspond to which parameters.

The core tone of a preset is dictated by one of 63 waveforms. Eight of these are programmable from a base waveform by a process of additive synthesis which allows the harmonics represented by a tonewheel organ's nine drawbars to be altered. The process is slow (nudging values mutes the output as the internals compute the new waveform), fiddly (digital access is a far cry from the handful of plungers approach on a real tonewheel), but it works as it should.

There are no envelopes to adjust, no filters to scale and no cross modulations to attempt. Just straightforward fixed-velocity triggers. What remains of the editing functions after the basic MIDI channel, MIDI mode, multitimbral operational parameters, tuning and volume adjustments and distortion switching are taken care of is down to the control of what can loosely be referred to as the effects section.

Thus we encounter the adjustment of rotating speaker acceleration, deceleration and their individual speeds, vibrato depth and rate, chorus depth and rate and percussion waveform, volume and decay. Only one effect can be used in addition to distortion and percussion. In all instances, parameters can be adjusted to give a background level for the effect, which will always be present in the sound, in addition to a switchable effect level. Thus a gently warbling vibrato can have extra wobble thrown in at the stomp of a switch. Other titbits include key-click volume, to replicate years of contact corrosion, and low frequency tonewheel leakage — or hum. VOCE *could* have gone the whole hog and got the *DMI64* to 'smell a bit funny' after its been left on for a while.

But it is the percussion possibilities, click, hum and effects, and how they all combine that make the unit so special and leagues ahead of a modern synth's best attempts. Even in its Sunday, church-going, best the *DMI64* still sounds dirty.

Wire up some external MIDI control and you will forget that it is a 1U rackmount and it all gets extremely organic (no pun intended) and playable because of the amount of interaction. Control-change information can kick in the vibrato, chorus and rotating speaker with variable rate and depth plus switching in the percussion and altering its volume, decay and waveform, and switching in the distortion circuitry. With this the *DMI64* transforms into a very satisfying instrument to play. There is a lot of movement in the sounds with the peculiar beating and harmonic shifting that you need. The device is not true stereo but spreads the effect over both legs as an enhancement.

Just about every classic tonewheel sound, that this reviewer can think of, is represented in the 99 presets in some way. There is some duplication, but are there really 99 radically different sounds on the real thing anyway?

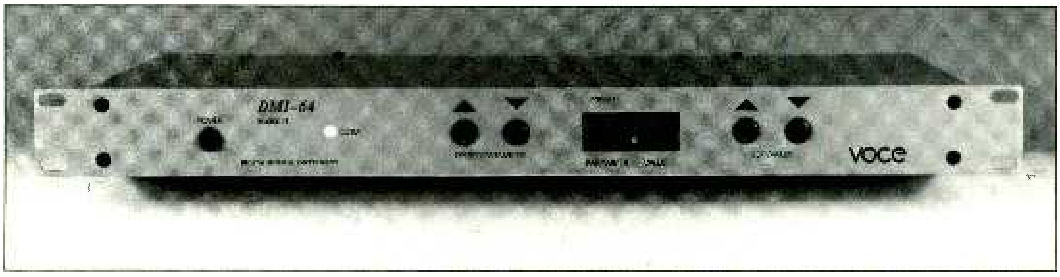
Most impressive are the glassy percussive phased tones which synths never get near — delicacy with dirt. The overdrive circuitry works astonishingly well on certain presets giving that wheezy left-hand rumble that always reminds me of jet aircraft engines. External processing through fuzz boxes and amps helps those that fizz a bit too much, but it is an exercise that is always worthwhile anyway. The *DMI64* also has a healthy handful of excellent cheesy 'brothers and sisters' tones so it is by no means dedicated solely to the pursuit of power chords.

Versatility of such recallable nature is one of the unit's strongest points as you can flip from a mellow pad to a raging solo at the tap of a button, or set up a split of two sounds on one preset. The 64-voice polyphony also means that if you resort to using your forearm, it will sound like it. On the downside, the lack of an easily-accessible volume control (volume along with the effects parameters are written into a preset) and the ability to tune quickly are missing. The mono signal is insipid — output for a preset can be addressed to either output individually, collectively as mono or in pseudostereo. It would have been handy with the polyphony to be able to stack sounds from the box without resorting to external trickery, because the result is phenomenal.

If you want that tonewheel sound and are considering investing in a modified original, I would strongly recommend that you give the *DMI64* Mark II a listen as it might be what you are looking for. If you need a dedicated source of organ sounds, look no further. This device's ability to integrate into a MIDI setup and its outstanding authentic sounds simply put it in a class of its own.

US: VOCE, 111 Tenth Street, Wood-Ridge NJ 07075.
Tel: +1 201 939 0052.
UK: MCMXCIX, 708A Abbey Road Tudor Estate, London NW10 7UW.
Tel: 081 963 0663

Studio Sound's Music News is compiled by Zenon Schoepe



VOCE's *DMI64* Mark II is a no-holds-barred attempt at replicating tonewheel sounds

The Box

The Box, the stereo sound stage display device consisting of a diamond of multicoloured LEDs, is available once more in a new guise, simpler and less expensive than the original. Its 10 x 10 array of LEDs displays information about a stereo signal in a manner similar to an X/Y display on an oscilloscope, and the resulting shifting patterns clearly indicate stereo balance, width, position and phase relationships. Problems such as lopsided images, narrow width and out-of-phase components can be observed, making the unit an ideal complement to a less-than-perfect monitoring situation. Diagnostic applications include tape head azimuth adjustment, where any errors can be seen even with a 1 kHz tone.

The input requires something around line level or a headphone output to fully drive the display, and built-in compression ensures that low level signals fill as much of the display as possible. Gone are the absolute level measurement facilities and wooden case of the original model, but the new price reflects the changes. The new manufacturers are prepared to discuss custom versions; units have already been supplied fitted with PPMs and with multiple matrices for Ambisonic sound stage display.

ITZA, 44 Challacombe, Furzton, Milton Keynes MK4 1PD, UK. Tel: 0908 502836.

AB1 from PMC

Martin Audio

The *EM* Series from Martin Audio is a range of six enclosures and two system controllers designed specifically for smaller scale installation and corporate use. Ranging from the *EM25*, a passive design combining an eight-inch cone transducer with a 30 mm compression driver, to the *EM250*, a high powered sub-bass unit using 2 x 15 in cone drivers.

Martin Audio, Cressex Industrial Estate, 19 Lincoln Road, High Wycombe, Bucks, HP12 3RD. Tel: 0494 535312. Fax: 0494 438669.
USA: Martin Audio America, 22930 Miller Road, Chicago Heights, IL 60411. Tel: +1 708 758 0652.

Professional

The Professional Monitor Company have launched the *AB1* 2-way transmission-line monitor intended as a midfield monitor for broadcast and video post-type applications. Also new is a passive version of the large *BB5* Studio monitoring system, the *BB5-P*, offering the benefits of the *BB5* system but at a budget price by avoiding the need to buy the complex amplifier package specified with the full-blown active *BB5* systems.

The Professional Monitor Company, 27 The Avenue, Highams Park, London E4 9LB. Tel: 081 531 5305 Fax: 0582 579278

Apogee Sound

Apogee have now designated their well-known *AE-1* monitor as the *AE-1S2*. The system has been redesigned as a true nearfield monitor, the enclosure has an appearance similar to the Apogee *SSM* loudspeaker with a perforated metal grille and the familiar Apogee tapered front. The new unit features a titanium-domed tweeter with output to 25 kHz and a long-excursion eight-inch cone driver. The cabinet construction is of 100% Finland birch ply and frequency range is quoted at 63 Hz to 19 kHz (± 3 db).

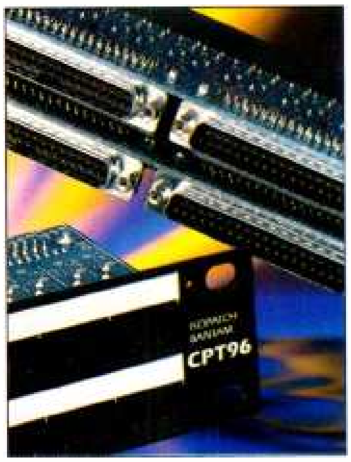
Apogee Sound Inc, 1150 Industrial Avenue, Petaluma CA 94952. Tel: +1 707 778 8887. Fax: +1 707 778 6923.
UK: Apogee UK. Tel: 0932 772241.

The new EM series from Martin Audio

New Signex

The latest addition to the Signex range of connector panels is the *Isopatch Bantam* panel which has been developed in response to requests for a cost-effective alternative to conventional *Bantam Jack patch* panels.

The *Isopatch Bantam* has rear connections via 37-way D-series multiway connectors which are fitted to the unit as standard. The unit has been designed for use with balanced lines (unbalanced signals can be handled). All sockets are supplied completely isolated but any pair of vertically adjacent sockets can be normalled in either half normal or full normal mode by simply bridging the gaps in the relevant programming pads on the top PCB with solder.



Isopatch Bantam

Isotrack, PO Box 747, Poole, Dorset, BH12 4YG. Tel/Fax: 0202 747191.

Dolby

Dolby's Spectral processor brings a unique approach to dynamic control and is based on techniques used in Dolby *SR*. Dolby claim the device allows quiet parts of the signal spectrum to be magnified without affecting the louder sounds or squashing transients. The process is level dependent and a threshold control sets the point below which detail will be magnified. A familiar three-band EQ-like section is then used to tune and boost the part of the spectrum by up to 20 dB. To deal with noisy source material, a gentle, sliding band, noise reduction section has been included. Dolby say that unlike 'exciter' products. Spectral Processing will not introduce artificial distortion into the program.

UK: Dolby Laboratories, 346 Clapham Road, London SW9 9AP. Tel: 071 720 1111. Fax: 071 720 4118.
USA: 100 Potrero Avenue, San Francisco, CA 94103-4813. Tel: +1 415 588 0200. Fax: +1 415 863 1373.

Audio Design

Audio Design have released the *Copy-Rite* SCMS defeat box. The unit is designed to prevent the Serial Copy Management System (SCMS) from being invoked by a DAT equipped with this copy prohibit system. By switching off the prohibit bits during copying, the copied tape can be used without any copy flag restrictions. The unit works at any sample rate without adjustment.

Audio Design, Unit 3, Horseshoe Park, Pangbourne RG8 7JW. Tel: 0734 844545.

AXYS nearfield

The *NF-1* nearfield monitor is the latest addition to the AXYS *Repro* series. The *NF-1* consists of a 6.5 in woofer-tweeter midrange speaker and a 28 mm ferrofluid-filled soft-dome tweeter. The *NF-1*s are a self-powered design and come as a matched pair, featuring an input gain control, an adjustable low-frequency roll-off switch as a 'domain' filter allowing adaption to the acoustic environment. A time-align circuit is provided which is able to bring the acoustical centres of both woofer and tweeter on one vertical line.

Duran Audio BV, PO Box 2055, 5300 CB Zaltbommel, The Netherlands.
Tel: +31 4180 15583
Fax: +31 4180 18077

Tannoy

Tannoy's new *CPA5* sub-bass unit complements the *CPA5* and other compact reinforcement speakers where extended bass or additional power handling is required. Comprising four 5-in sub-bass drivers in a compact box (22 in x 12 in x 8 in) the unit is capable of delivering up to 112 dB, accurately reproduced down to 45 Hz.

Tannoy Ltd, Rosehall Industrial Estate, Coatsbridge, Strathclyde ML5 4TF, Scotland.
Tel: 0236 420199.
Fax: 0236 428230.

Waveframe

The WaveFrame *401* is a new cost-effective digital audio recording system incorporating computer, monitor, recorder, sync facilities, inputs, outputs, hard disks, cables and software starting at under $15,000 (about £8,100). The 8-track system includes facilities such as VITC, digital I/O, mixing and multitrack punch recording as standard, and utilises 'tried' software that has been refined over the last four years. Projects are interchangeable between the *401* and other WaveFrame products allowing added flexibility.

WaveFrame Corporation, 2511 55th Street, Boulder, Colorado 80301. Tel: +1 303 447 1572.
Fax: +1 303 447 2351.
UK: Stirling Audio, Kimberley Road, London, NW6 7SF. Tel: 071 624 6000. Fax: 071 372 6370.

Enhanced BASE

A new BASE spatial sound processor has been released featuring balanced line operation. The *101C* has selectable balanced inputs and outputs at either +4, +8 or +10 dBm, as well as retaining the original unbalanced (–10 dBm) operation. Also included are separate input and output ground lift switches, and LED indication for signal present and peak. The upgraded unit incorporates an overrated power supply said to give better transient response as well as the ability to handle higher outputs. In addition the fuse is now mounted externally, and the unit has a hard-wire bypass switch. BASE say that unlike other spatial processors, their's is not a sound localiser but produces a highly defined and realistic sound image.

UK: Sound Base Ltd, 4 City Road, London EC1Y 2AA. Tel: 071 256 8716. Fax: 071 256 9279.
USA: Gamma Electronic Systems, 25020 West Avenue Stanford, Suite 80, Valencia, CA 91355.
Tel: +1 805 294 1380.

Klark Teknik

Klark Teknik have released two new products. The *DN728* digital delay line and the *DN800* active crossover unit.

The *DN728* is a dual input, six output, user configurable delay offering 1.3 s of total delay (upgradable to 5.2 s). The device, which is primarily intended for sound reinforcement applications is also suited to video post and synchronising audio for satellite transmissions. The 18-bit unit displays settings in either microseconds, metres/feet, or video fields (PAL or NTSC), and these can be set as absolute or relative values. In distance mode, a temperature compensation facility is incorporated. Configurations and settings can be stored in 64 nonvolatile memory positions, and quickly recalled from front panel controls. Various retrofittable remote control interface options are to be made available.

The *DN800* crossover has four balanced inputs and eight balanced outputs, which can be configured as a stereo four-way, stereo three-way or a four-channel or quad two-way system. The 1U unit utilizes plug-in frequency cards for a choice of 12, 18 or 24 dB/oct slopes with Linkwitz-Riley, Butterworth or Bessel responses — and band overlap is possible. Other features include Mono Bass for subwoofer systems, trimmers for phase adjustment between bands and switchable phase reverse for each output. Options include VCA limiter cards, output transformer balancing and fixed EQ cards.

AXYS *NF1* monitor from Duran Audio

Klark Teknik PLC, Klark Industrial Park, Walter Nash Road, Kidderminster, Worcs DY11 7HF. Tel: 0562 741515. Fax: 0562 745371.
USA: Klark Teknik Electronics, 200 Sea Lane, Farmingdale, NY 11735, USA. Tel: +1 516 249 3660. Fax: +1 516 420 1863.

Publison

The French company Publison have introduced a 16-track optical recorder and editor. Based on four optical drives (four tracks per drive), the system offers six track-hours of recording, plus an additional 16 hours from hard disk, and can be used in optical mode, hard disk mode or a combination of the two. All editing functions found on the *Infernal* Workstation have been included.

Publison Audio Professional, 18 Avenue de la Republique, 93170 Bagnolet, Paris, France. Tel: +33 1 43 60 8464. Fax: +33 1 43 60 8031.

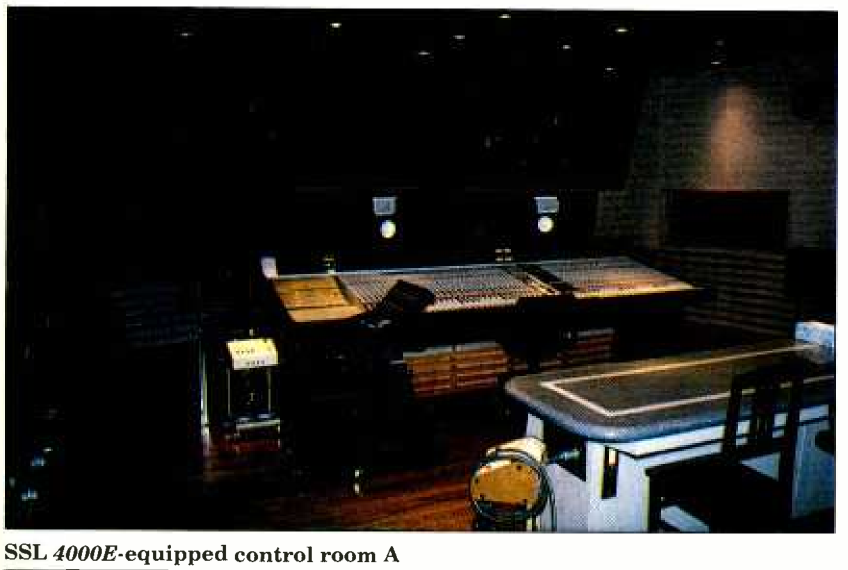
STUDIO TERRA

Zenon Schoepe visits the Tokyo-based house recording facilities of Toshiba-EMI

Any doubts about the shortage of usable land in Tokyo are laid to rest with a visit to Toshiba-EMI's Studio Terra complex in the Bay area. As you queue to use a car elevator that whisks you to a rooftop parking area some six floors up you realise that the roofs of the neighbouring buildings are similarly thatched with Toyotas and Nissans. Other contraptions regularly pressed into service in Toyko to maximise the use of each square metre include car ramps that hang cars stacked like so many sides of beef into the spaces between buildings. It is thus shocking to contemplate the expenditure associated with the large floor space occupied by the Bay area's Studio Terra, built some four years ago when Toshiba-EMI's more central HQ, housing studios of its own, became too cluttered and forced expansion. A total of five studios are on tap for the company's projects, Studio Terra being essentially a house facility with two rooms (Studios 2 and 3) supported at the Akasaka Minatoku HQ and three (Studios A, B and C) supplemented by numerous suites placed at the new site in Higashishinagawa Shinagawaku.

The Bay area complex was built by the construction team of Nitobo to Toshiba-EMI's specification and design and uses the ample space ingeniously with the usual tasteful tapestry of exotic wood and interesting stone most evident in the main 161 m² recording area which is shared by Studios A and B. Each control room additionally has a dedicated booth and is arranged as a mirror image across the large live area surrounded by smaller isolation booths peppered with custom-built artist foldback systems.

Studio B's 60-channel *V* Series Neve was originally fitted with *Necam 96* but has since been replaced by GML automation which Studio Junior Chief Manager, Isao Itoh, refers to as a more reliable system in terms of software. Studio A houses a 56-channel SSL *4000E* with *Total Recall* but ususually no automation. 'To many young engineers the SSL is a more familiar desk and they can work faster on it, so we have to have one,' explains Itoh, adding that the two studios are effectively being used as a tracking package to feed the Neve *V60* and GML equipped mixdown room of Studio C.



SSL *4000E*-equipped control room A

'Our aim is to offer consistency in our rooms,' says Itoh of the Studios A and B. 'Only the desks are different, everything else in the control rooms down to the *FM1000* power amps is identical.' This is supported by the presence of Mitsubishi, Sony *48* and Studer *A800* multitracks in both rooms. Rey Audio *RM-4B* monitoring and 72-channel by 60-channel mic routing matrices.

Studio Terra lives up to its responsibilities as a record company house studio with eight CD mastering suites, CD-I, AV and video editing rooms for the big business karaoke market plus a vinyl cutting room — an unusual inclusion these days in Japan given the nation's wholehearted swing behind CD as a playback medium. Equally unusual for Japan is Studio Terra's employment in these rooms of three women engineers. Itoh responds: 'They have very good ears and they are popular with the clients.'

Neve Custom *8078* with *Flying Faders* in Studio 3

Maintenance at Studio Terra is widely regarded to be of as high a standard as the Sony test-bed of CBS-Sony studios in Tokyo with both having the capability to fix and service everything in house without too much recourse to the manufacturers.

Studio Terra's other site being the original Toshiba-EMI studio in Akasaka, Minatoku has less of a Western air about it exuding instead the decidedly Japanese office-type atmosphere that permeates through many of the country's larger studio complexes. Housed in the same building as the Record label's administrative offices are two studios, one housing a custom Neve *8078* and the other a computerless 56-channel SSL *4000E* stated to be aimed at 'budget' recording.

The old Neve is retrofitted with *Flying Faders,* a feature that was installed in the desk long before Neve made the retrofitability of its automation package to non-*V* series desks a well-known possibility. The desk is coupled to a 179 m² live area that is heavily used for classical orchestral recording. The room is stated to be unusual in sporting a decidedly live acoustic, something that was not commonplace in Japanese studios 10 years ago when it was constructed. In this instance its acoustic character resulted from Toshiba-EMI staff travelling abroad and encountering similarly-featured rooms in foreign studios.

Outboard equipment in Japanese studios, while ample, is surprisingly sparse by Western standards but there is a lot of similarity in what earns a place in the racks, as Itoh explains. 'The Japanese always want the latest technology. Engineers are

The orchestral live recording area of Studio 3

always talking and rumours and ideas spread very quickly. If someone hears that a new piece of outboard equipment is doing well in a Los Angeles studio then very quickly everyone wants to buy it in Japan. That's the Japanese mentality.'

This explains the presence of Keepex *804s,* Urei *1176s,* Roland *SDE 3000s,* Korg *SDD3000s* and Yamaha *Rev 5s* in every Toshiba-EMI control room. Interestingly, the Toshiba-EMI engineers themselves think that the outboard selection is more than adequate, particularly veteran Chief Recording Engineer, Yoshihisa Watanabe, who started his work in the mid-1960s on mono film sound. He is now well respected in the field of Enka — a type of popular crossover Japanese folk music. 'Because I record Enka, I'm interested in capturing pure natural sound and generally use very few outboard effects.'

He does, however, acknowledge that the Japanese's, almost wholesale, adoption of digital multitracks has spawned new working practices. Sentiments that are echoed by Chief Recording Engineer, and comparative novice, Seijo Okumura who has a mere 12 years at Toshiba-EMI under his belt having graduated through the traditional tea-boy route and who states his average working week ties him to the engineer's chair for a minimum of 50 hours including Saturdays and Sundays.

'I only work about 5% of the time on analogue with most of that being drums which I then transfer to digital. I would like to work like that more often but these days I don't have the time to.'

While the Japanese obsession with tube kit for digital recordings has been instigated by the desire to warm up and counter the purity of the medium, Okumura states that there are practical advantages to digital that are often overlooked in the philosophical arguments concerning sonics. 'A lot of Japanese engineers like the sound of digital and there is a strong CD tradition here. However, one of the biggest benefits of using digital machines is their speed of operation with fast access and locate times plus you've got so many tracks to work with. These are the real considerations that make it hard to go back to analogue.'

It is exceedingly rare in Japan to encounter producer-engineers and the overwhelming trend is for house engineers as opposed to freelancers guaranteeing career development and on-the-job training for the fortunate. However, the fundamental problem facing the recording industry in the country seems insurmountable. Land costs make it difficult to start a studio without serious collateral forcing these studios to assemble in what by Western standards are in many cases artificially low stratum for the facilities and professionalism they offer. Large record companies like Toshiba-EMI protect their own interest in the time-honoured fashion of investing in their own dedicated studio facilities but even these cash-rich giants are not immune to land costs.

Land costs effect raw talent — rehearsal facilities are also expensive and oversubscribed as are the venues, as Sam Machida of the Domestic A&R department explains.

'It makes our job harder because it means we have to invest time and money in developing promising acts just to find out what they're like. That makes the producer's role even more important when it comes to recording, particularly when it comes down to money control.'

Machida's throughput of work, along with that of his colleagues, means that he has to resort to using studios outside of the Toshiba-EMI empire defeating the object of owning house studios in the first place and falling back into the land-costs trap again. 'I would certainly like to see more rental studios in Tokyo. From my point of view we could do with even more studios belonging to Toshiba-EMI.'

Large partitioned live area shared by control rooms A and B

In such an environment it seems that having cash is still not enough. ■

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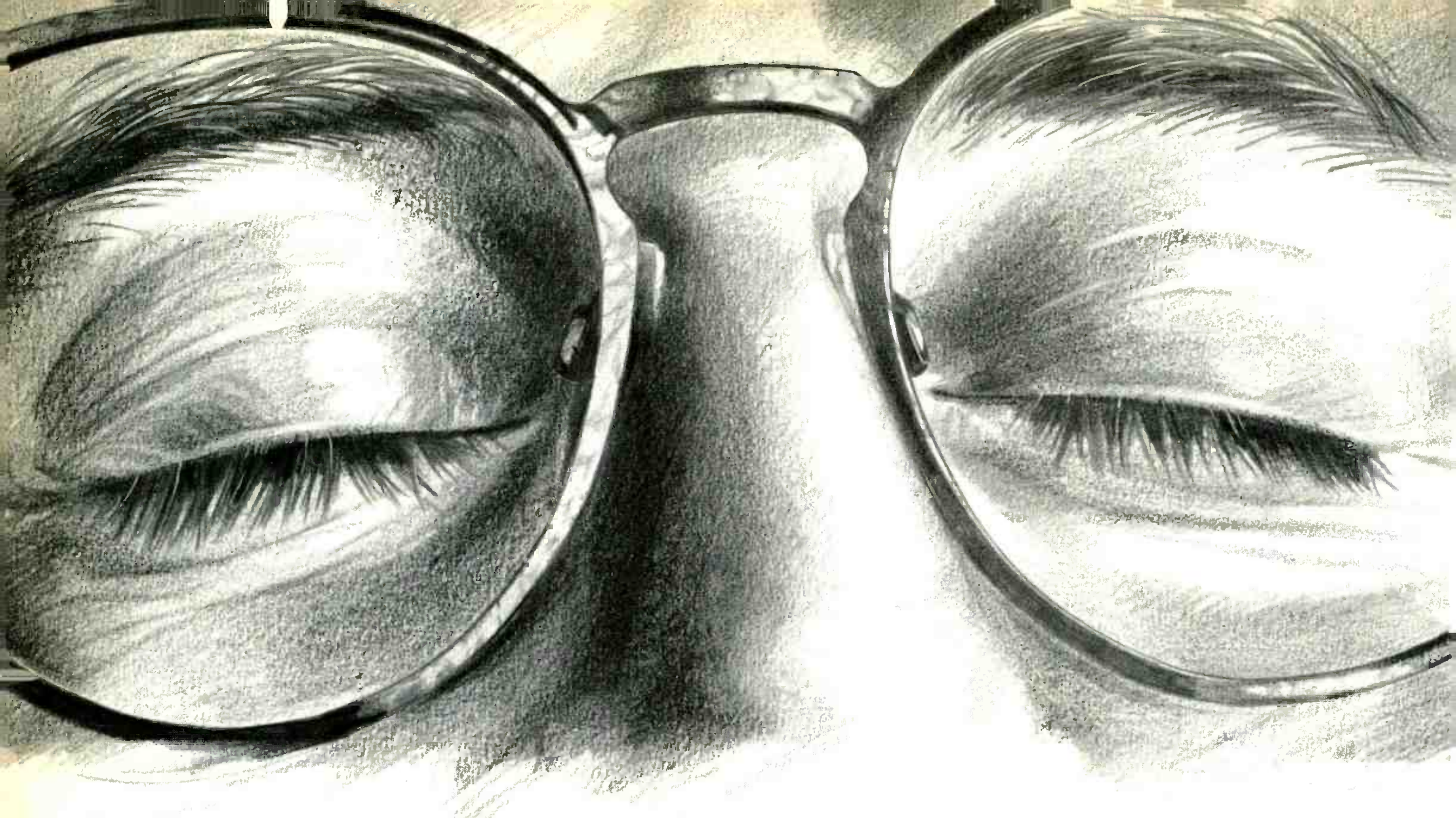

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SON OF BOXER

Ben Duncan looks at the latest version of Discrete Systems' *Boxer* monitoring system, its technical background and development.

Over the past seven years, Discrete Systems have installed over 40 *Boxer* monitoring systems in some the largest and finest control rooms worldwide. The company, based at Pinewood Studios, 30 miles northwest of London, works in conjunction with sister company, acoustics research and design consultancy Harris, Grant Associates, who design the control rooms and execute installation, up to complex refit projects, for film, television and recording studios. Recent clients include The Hit Factory, in both London and New York; BBC Maida Vale Four & Five, in London; Sony Classical, New York and Hamburg; Red Rooster, Munich; and Hitokuchi-Zaka, control rooms Three and Five, Tokyo.

Neil Grant, the main man behind both Harris, Grant Associates and Discrete Systems, believes this 'global' involvement is the only way to produce a monitoring system which merges seamlessly into the control room: 'As control room designers with a quest for excellence, we would be at the mercy of the monitor speaker manufacturer and a host of other agents, unless we could control not just the speaker sources, but the entire monitor path'.

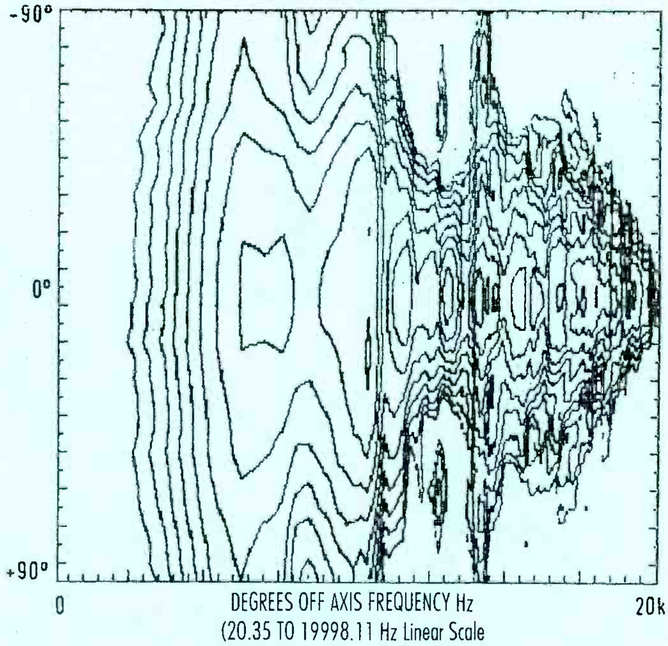
The *Boxer* system is exceptional because, with the sole exception of the LF drive units, every part including the active crossover, is developed 'from scratch', and exclusively manufactured. The decision to develop a whole system in this way in order to shoot for perfection is highly risky. Grant hails from Scotland, and emphasises both points with a quote from the revered Glaswegian architect, Charles Rennie Macintosh: 'There is hope in honest error, none in the icy perfections of the mere stylist'. To control quality and assure consistency across every installation, specially selected amplifiers and cabling form part of the system. Grant explains: 'This avoids the situation — which actually happened in the early days — when a client mentioned that he had four old Crown amplifiers, all of different flavours, lying around . . . use those. The problems in resolving polarities, gains, and sonic qualities along with taking responsibility for disparate maintenance lead us to never again allow ourselves to be in this situation. Hence each system is sold with everything — including spares, tools and manuals'.

System overview

To arrive at the the foundations of the *Boxer* system, Discrete Systems reasoned that division of the frequency spectrum is critical. Typically, with two-way systems, the crossover point ends up in the 1 to 3 kHz region, and with systems capable of adequate SPLs (historically a 15 inch/380 mm bass-mid cone driver and HF compression driver), directivity is severely mismatched in this, critical, frequency region. **Fig.1** shows the resulting, flawed directivity in 3D. Subjectively, the outcome is a change in tonal balance as you move around. One can escape the directivity mismatch by dividing into more bands. However, crossovers are inherently unsavoury devices, particularly as at least one crossover point inevitably lies in the midrange where the ear is most acute. So one wants as few crossover points as possible. Three ways were chosen as the optimum, with crossover points equally spaced at the decades 250 and 2.5 kHz, with an overall three decade, -6 dB bandwidth from 25 Hz to 25 kHz. Now to fix directivity matching, the reasoning goes that the mid device has to fit broadly midway between the size of the LF and HF piston sizes. It will need to be around 3 to 6 inches (75 to 150 mm) in diameter. But to meet SPL requirements, the voice-coil will need to be much bigger than the 1 or 2-inch (25 mm or 50 mm) size commonly fitted to cone drivers of this size. The answer is to use a dome driver, which has a voice-coil as big as its diameter. Choice of driver size is now linked to voice-coil size, as well as to excursion capability, and the optimum for both turned out to be 4-inch (100 mm). **Fig.2** compares the SPL and required excursion capability of different sizes of dome radiator. With descending frequency, the position has to move increasingly far to fulfil the SPL required of it. The graph shows excursion in millimetres for the 4-inch dome for 135 dB SPL. If cone area were doubled the excursion requirement would halve, but then we would have to start again (see above).

Grant determined that the *Boxer* system should be capable of generating 135 dB SPL *at the mix console* without stressing the drivers. Excursion has been covered — but can a 4-inch direct-radiating device produce this kind of SPL without melting? Grant considers that the midrange device ▶

Fig. 1: Directivity of a two-way monitor wystem comprising a 15-inch bass driver and HF horn, showing isobars of equal sound pressure. The 'neck' where the 15-inch directivity narrows, before the horn takes over, occurs two-thirds of the way up the frequency scale, and can be seen most clearly by turning the graph on its side



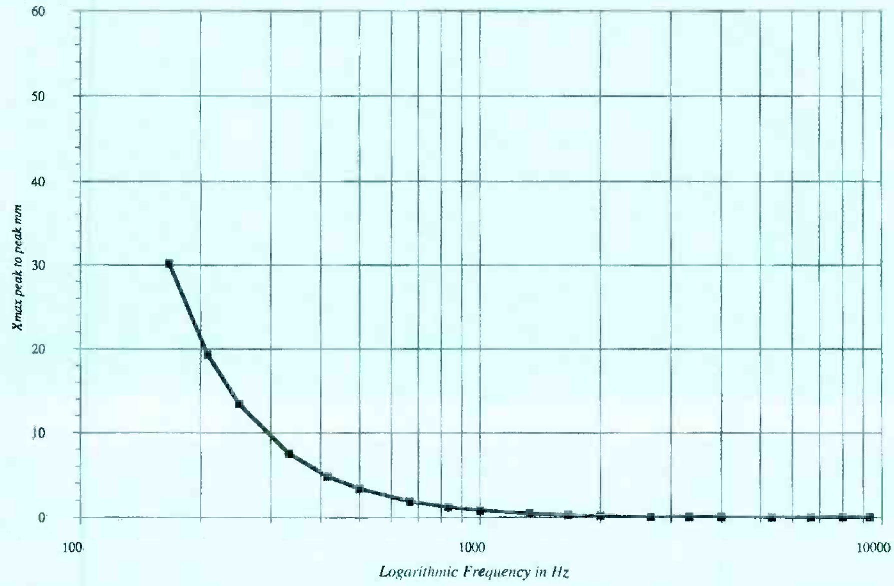


Fig. 2: The peak-to-peak excursion ('Xmax') needed to attain the maximum 135 dB SPL capacity from the *SD100*'s 4-inch dome, against frequency. Only a slightly smaller dome would have to move up to a double the distance for the same SPL, at a given frequency

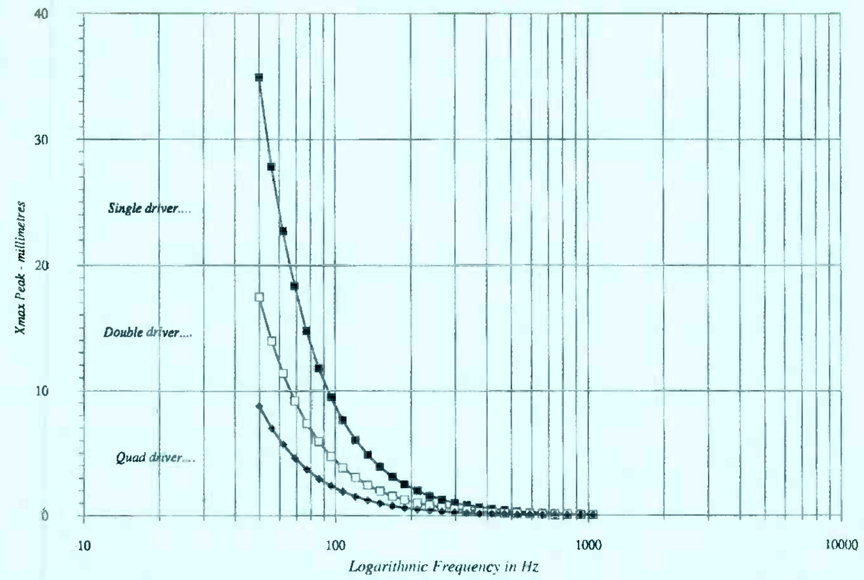


Fig. 3: Excursion ('Xmax') distance in mm, required from 1, 2 and 4 times 15-inch LF drivers, to attain 135 dB SPL, versus frequency

has to do the hardest work in this department. Other manufacturers have taken the line of the least resistance, employing multiple drivers in the band to handle the power. It is a pragmatic solution, but one that inherently causes virtual source problems, hence 'phasiness', and excess directivity that is inappropriate to the control room. In stereo, the subjective outcome is mobile and incoherent imaging.

For the LF section (below 250 Hz), we enter the maze of design equations with the requirement for a 135 dB SPL capability (at the mixing console), to be -6 dB down at 25 Hz, referred to 100 Hz, with a low and critically damped Q of 0.5, which Grant regards as an essential (but rarely achieved) criterion of monitoring quality. Easier-to-achieve damping factors of 0.7 are more the norm. The graphs in **Fig.3** show the excursion needed to achieve the low bottom-end response for a single 15-inch, and for two and four units, giving a large piston area. Using more than one driver *is* permissible here, as even at the highest frequency in this band, the drivers are mutually close by comparison to the wavelength. Using multiple drivers is also desirable, even mandatory, for good transient response. Grant explains: 'Imagine pushing one large piece of cardboard with a single, powerful but quite heavy voice coil. It will work fine, except the system will not start and stop as quickly as pushing four smaller pieces of cardboard each a quarter of the size and mass, with smaller voice-coils'. The outcome was a decision to use 4 x 12 inch (310 mm) drivers in the large *Boxer* system, and 2 x 12 inch in the smaller system. Some would argue that multiple bass transducers should be separately enclosed. This is recommended 'If,'says Grant sagely, 'resonant frequencies aren't matched by the maker, one unit out of step with the others will be beaten to death by its chums.' The downside is potential interaction between the individual resonant chambers and ducts working in parallel. But by paying attention to driver matching, Discrete Systems have found that mounting them in a common enclosure is safe, and it substantially reduces the complexity of the LF enclosure — as well as the complexity of the electrical loading on the amplifiers.

Drive-unit developments

Having decided on a broad specification for the drive units, the challenge was to develop them, as no off-the-shelf units existed. To this end, Grant recruited Stanley Kelly, who has a pedigree as a resourceful designer. Working in radar and telecoms before 1945, and then in audio, he is best known for developing the ribbon loudspeaker, beginning with the Decca units in the '60s. More recently, he has also been involved with Apogee in the US, the renowned hi-fi ribbon speaker manufacturer.

To achieve superb transient response, the design of the *Boxer's SD100* midrange and *SD38* HF drivers focused on achieving very high, almost terminal, flux densities in the voice-coil's gap, around 1.75 Telsa (T) for the Mid and a significantly higher 1.95T for the HF. The columar Alnico magnet used to achieve this density in the *SD38* reverts to a past age. Since the collapse of the cobalt mining industry in Zaire (formerly the Congo) in the late '70s, nearly all loudspeaker makers have had to compromise their wares with ceramic magnets. The first permanent magnets were made pre-1910 from carbon steel. ▶

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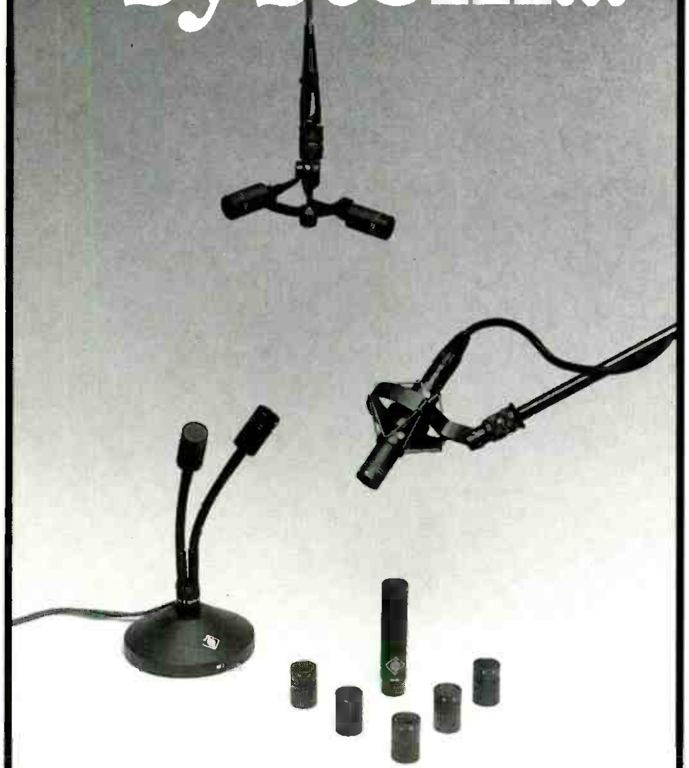

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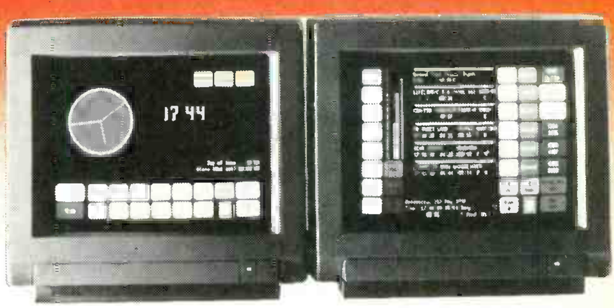

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SD38 **high frequency driver**

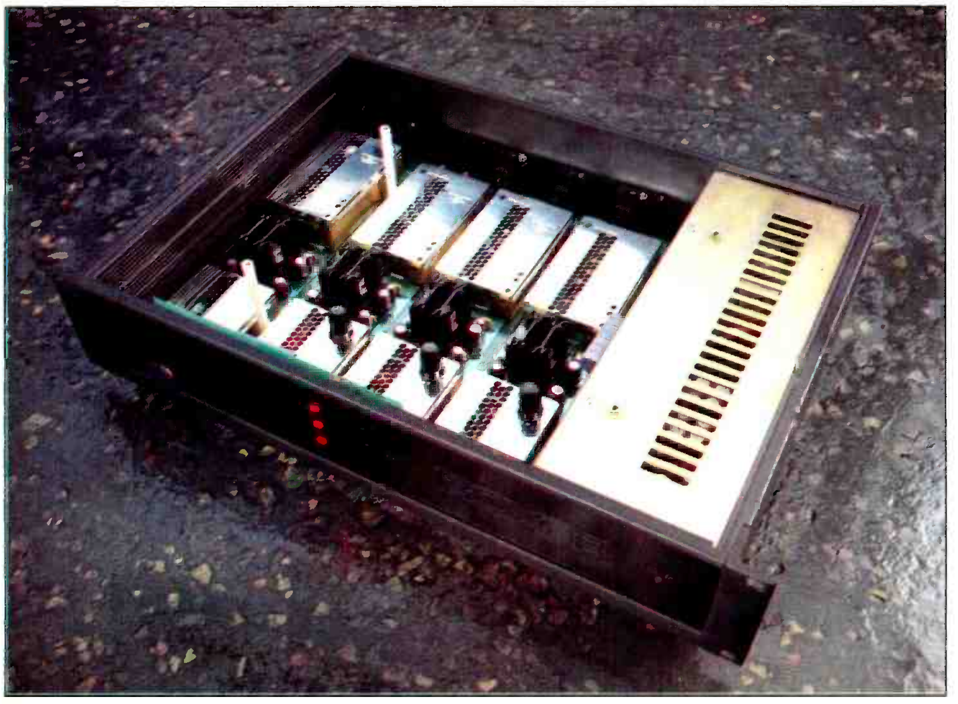
Adding Tungsten raised their power. In the '20s, Japanese metallurgists invented cobalt steel, and in 1934, much more powerful magnets were made in Sheffield, England, by alloying cobalt (Co) steel with nickel (Ni) and aluminium (Al). Copper was subsequently added, leading to AlNiCo. The columnar magnet used in the *Boxer* drivers was invented in 1948 by cooling Alnico in a suitably-aligned magnetic field, yielding 40 times the magnetic flux density of the original carbon steel variety. The cost of this is in the magnet, for which the raw-parts budget in the *SD38* is £80 — compared to £5 to £10 for the average ceramic magnet in a high-power dome tweeter. Achieving the flux densities also meant tracking down the purest Swedish iron, from which the top plate and pole tip are made. The *SD100* presently uses a large diameter ferrite magnet, but the Columax version is under development to push flux density to be similar and eventually even higher than the HF driver's.

The stability of the dome, considering its substantial excursion, was solved by designing a dual suspension with 'inverted polarity'. This means one suspension pushes while the other pulls , cancelling potential bell mode resonances. These would cause the dome to rock, quickly destroying the voice-coil. Damping behind a dome driver is important, but rarely considered. Long-fibre wool between the magnet and the dome controls the air mass, damping decay 'tails' from the TDS response (see later), helping to build a neutral response.

Few of the hundreds of watts pushed into monitor speakers emerge as sound. About 95% of it warms your studio — if it does not burn out — speakers first. To achieve high power handling without brute force, namely a heavy, fat voice-coil, and the damage to transient response this would entail, a great deal of thermal analysis was done. A laminate of glass cloth and aluminium alloy, bonded with polyamide adhesive, was specified for the voice-coil former of the *SD100* mid driver. This forms a rigid, light former which can handle very high temperatures, while being a good thermal insulator, it also keeps the heat away from vulnerable glue joints. Thermal coupling to the magnet is so good that it is worth blacking the outside of the magnet to improve heat radiation. To the same end, the *SD38* HF driver employs an aluminium-magnesium alloy for its voice-coil former, for the near ultimate product of stiffness versus lightness. Having the driver survive is only part of the story. Efficient heat transfer greatly reduces audible distortion caused by thermal ▶

THERMAL COMPRESSION

When a voice-coil heats up, its resistance increases, reducing the power you can push through it (for a given amplifier headroom). The change in resistance alters the drivers' individual and mutual tonal balance. It is often this effect (rather than impending deafness!) that cause many engineers to wind up the treble when monitoring loudly. Thermal compression also makes speakers act (and sound) like a soft-knee compressor. Three of them! Big drive units generally have longer time constants than small ones, so the sonic result of inadequate SPL capability and-or voice-coil cooling is akin to passing bass, mid and treble through three very different compressors. With stereo, compression also causes programme-dependent image shift, when, say, a loud instrument focuses on one channel. ■



***X3MB* mono crossover, as used in Boxer series 2 systems**

compression (see sidebar).

The outcome is that Discrete Systems are one of very few monitor system manufacturers never to have a had a midrange driver burn out. They also claim to have produced the world's most sensitive, direct-radiating dome tweeter. The company's confidence in ruggedness is reinforced by the specification of an amplifier capable of 62 V rms to drive it. The bought-in ATC 12-inch and TAD 15-inch LF drive units have voice-coils edge-wound with copper ribbon on a former made from a laminate of metal Kapton. A curvilinear cone is used for rigidity and again, distortion is minimised by making the voice-coil short compared to the gap.

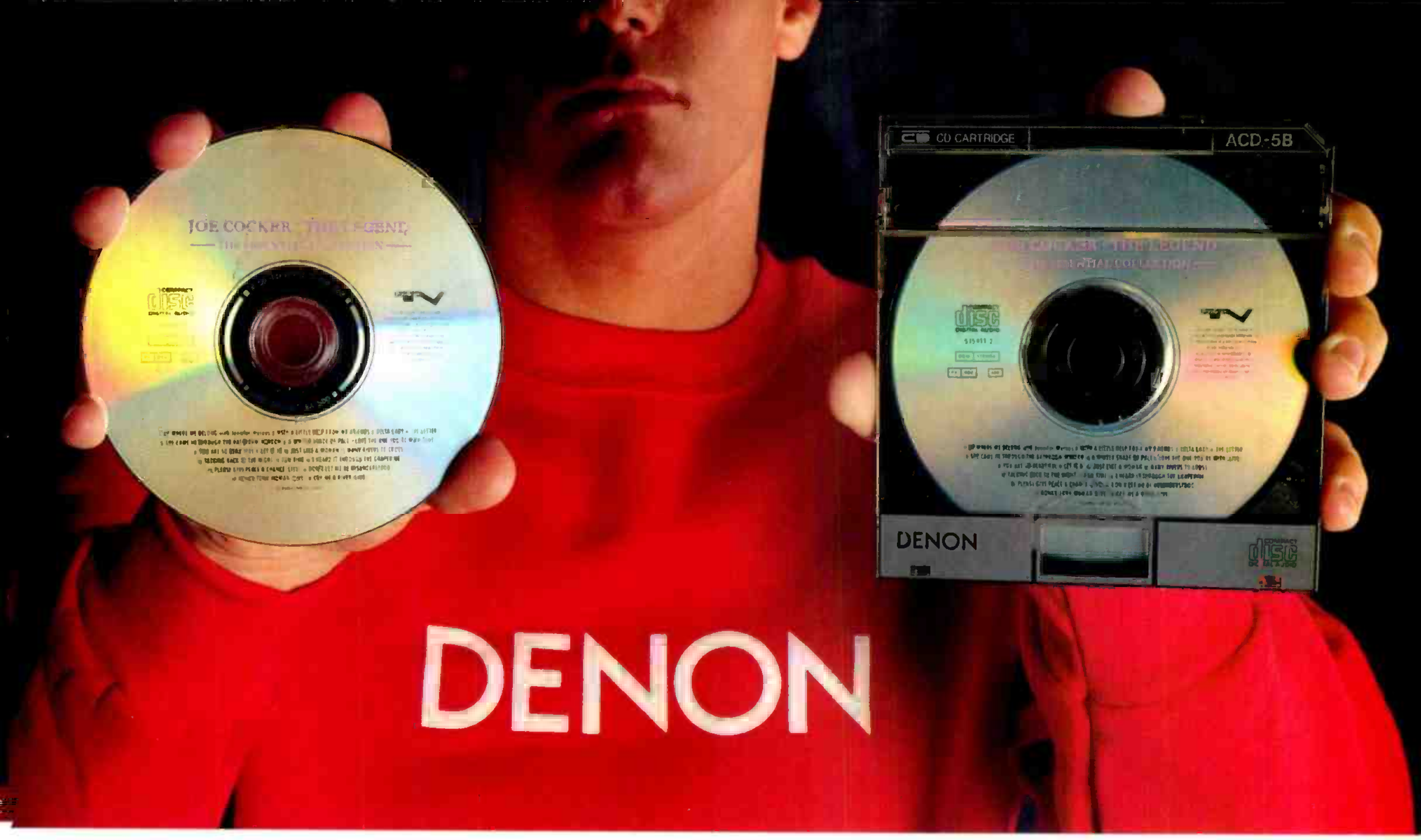
The crossover

The first *Boxer* crossover was a stereo three-way unit, designed by Matt Dobson, Tim Isaac and Bill Whalley. In 1991, Grant gave the author a near-enough free hand to redesign it, with more headroom and a lower noise floor. It was agreed that the revised crossover would be mono, avoiding some potential ground problems and allowing each crossover to be adjacent to its respective amplifiers and monitors. The limiters' range needed extending, it was felt that the input transformers should be avoided, and a level control independent of the band trims was sought for stereo balance.

The input transformers were replaced by an Analog Devices IC providing a true differential input and capable of operating on high voltage rails, providing a generous >+28 dBu input capability before clip. The IC's low noise (in excess of requirements) was traded with a small attenuator for added input headroom, and a +2 dB gain capability for balance trim, in ½ dB steps. The crossover proper comprises a State Variable or 'Good' filter with low-pass, band-pass and high-pass outputs, followed by Sallen & Key high-pass and low-pass postfilters. To reduce noise pick-up, and having selected low noise bipolar op-amps with 600 Ω drive capability to replace the original Bi-FET types, it was possible to reduce all the resistor values by a factor of about 10. It was also necessary to convert to very close tolerance E96 values, while capacitor values were normalised to the E3 range, for ease of sourcing tight (1%) tolerance parts. After computing these changes, *MicroCAP III* analogue simulation was then used to check that the crossover frequencies, phase and group delay relationships agreed with the specification. They did. The simulation was then used to look at the interchannel phase and group delay relationships after moving the LF protective high-pass filter out of the mid and HF path.

Tolerances were then appended to all the gain and frequency determining parts in the simulated circuit. While counter intuitive, the fairly convoluted circuitry of a sophisticated crossover varies less than one might think, as predicted by the central limit theorem. Monte Carlo data enables realistic production limits to be set for band responses from the outset, rather than guessing, then iterating acceptance limits with successive production batches. At the output, Analog Device's new and unique, 8-pin balanced driver IC was considered essential to save on the space otherwise occupied by the standard hydrid circuit, with its duo of fussy trimpots. Our aim to overcome any trimming was not entirely satisfied, for while output CMR proved good and consistent, the IC exhibited a high and unit-to-unit varying DC offset — which was solved by trimming out!

The remaining work concentrated on sonics. The signal path was converted to be direct coupled, with the inevitable exception of the high-pass filters' series capacitors, and input blocking (using Japanese electrolytics expressly designed for 'good audio tone'). Input coupling is set with an exceptionally low –3 dB point of 0.008 Hz, so the LF phase response isn't unnecessarily compounded [1]. Specific types of UK-made metal-film resistors, French-made polypropylene and class 1 ceramic and polystyrene capacitors were chosen, based on several years of listening tests. A Harris op-amp was chosen for its excellent all-round specifications, having lower %THD, noise and DC offset than the *5534*, and faster slewing (25 v/µs), more gain-bandwidth and drive capability than the *TL071*, as well as improved sonics [2]. Noise pick-up — particularly hum harmonics — are a nuisance in any high-quality filter or EQ circuits where the plates of physically large frequency-determining capacitors extend the area of sensitive nodes. The original crossover already employed a toroidal mains transformer. In the new unit, noise from the mains (including 50 Hz and harmonics up to 10 kHz) was reduced by over 20 dB, by individual regulation, by specifying ▶



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interwinding shields, and by housing the power supply in a separate enclosure. The final touches were to devise the optimum combination of star and signal-flow grounding. The finishing step was to take all the theoretical work and convert it into a physical unit, which combines high standards of electronic circuit assembly, with the sculpture-like exterior 'feel' that approaches the high-end of domestic audio. This was done in-house by Discrete Systems' Matt Dobson, who heads the company's production and 'industrial' (packaging) design.

Amplification

While it was logical to have the crossover as part of the *Boxer* package, it would be easy to leave the power amplifiers to the client. But Discrete Systems were keenly aware of the dangers of this approach. After a great deal of research, the company struck up a formal relationship with Electro Acoustique Applique, based in Nozay, France, with whom they developed a special version of their *Square 1000*, a PA amplifier. Both subjective quality and ruggedness of the original unit were extraordinarily good, with it employing bipolar output transistors and being intended for PA. It represented the engineering peak of its particular line. The derivative made for the *Boxer* system, which powers *all* the drive units, is rated at 450 W/ch into 8 Ω, with capability down to 2 Ω and 1,200 W. Using an amplifier of this capacity for HF drivers seems foolhardy, but in fact, reliability is enhanced, transient capability is extraordinary and as all the amplifiers' voltage limits are the same, there are no dynamic spectral shifts.

It is easy for power amp makers to *claim* their products are reliable. Discrete Systems are in the happy position of being able to report that over 100 of their special units are out in the field with very few problems. These have mainly been teething, and noncatastrophic revolving around the very low (100 V and below) mains supply in Japan, and the unusually high 240 V mains in the UK — compared to 220 V in Europe. In common with a number of other high-quality designs, the *Square 1000* made for the *Boxer* system features independent, generous-rated power supplies comprising 2 kVA per channel transformer windings and reservoirs totalling 64,000 μF/ch. In common with rather fewer units, *all* the power devices and all PCBs plug-in and have gold-plated contacts. The crux of the unit's reliability is the housekeeping. A transformer isolated analogue computer monitors the output current, voltage, temperature and signal frequency, anticipates overload (including excursions outside the output transistors' Safe Operating Area or *SOA*), and prevents clipping and other stresses.

TEF-ology

Harris, Grant Associates purchased the very first TDS (Time Delay Spectrometry) analyser in the UK in 1984 — Crown's *TEF-Techron* system. TEF stands for Time Energy Frequency, and TDS enables very clean measurements to be made of those parameters that are in both the time and frequency domains (see [3]). This 'TEF' device has been the principal measurement and analysis system for the company ever since. Unusually, it is one item of '80s instrumentation (and dreamt up by Heyser back in 1969!) which grows more useful as time passes and the rest of the world catches up, especially now that its output data can be postprocessed by *Mathematica*, a powerful engineering maths program.

The TEF machine has been widely used by Discrete Systems in the development of the *Boxer* system's cabinets, transducers and the interface with the room. To develop the *SD100*, measurements were made of rocking and bell-mode behaviour of the dome and voice-coil, using a ceramic hi-fi cartridge as the input transducer. The resulting data was used for diverse 3D mappings of X-Y movement against frequency. Similarly, the TEF machine has been used for various 3D mappings of individual distortion harmonics vs amplitude and phase, against time and frequency, for both the HF and MF drivers. Another use is to map the half-space polar (directional) response, then calculate the power response, to verify driver and crossover integration.

In developing the *Boxer* enclosures, panel resonances were mapped with an accelerometer connected to the TEF, which was set to map the integrated displacement against frequency and time. The resonances could then be seen as well-defined tails in the amplitude spectrum. Bracing could then be designed to couple appropriate points of the cabinet surface, and mass increased to move the resonant frequencies. As sound travels faster through denser mediums, Discrete Systems have found that solid fixing of the *Boxer* enclosures to the lightweight class of control room shells leads to early radiation of the LF from the floors, walls and ceiling of the room itself. Decoupling the cabinet at a very low resonant frequency is an effective filter, excluding audible frequencies from early structural transmission, as well as dramatically improving isolation experienced in adjacent studios and outside the building. The TEF machine was used of confirm structural LF coupling, using sandwiched pairs of accelerometers as input devices, with data output derived from the vector differences.

By creating drivers from scratch, Discrete Systems have been able to design-in an unusually friendly approach to servicing the *SD38* and *-100* diaphragms. These are easily replaced in the field, as location is on precision dowel pins, and tools are provided. Knowing Murphy's law ('Take your wellies and the sun will shine.') going to this length it seems like a good omen that servicing will not often be needed. LF drivers are serviced by exchange, and complete subsystems are loaned should return-to-base repairs need to be undertaken.

Concluding comments

Discrete Systems continue to research into improved products and processes. They have an enviable position with access to the latest in acoustics research from its sister company, as well as to some of the largest acoustically-treated spaces in Europe, at Pinewood Studios, set well away from the rumble of traffic.

Additionally, the company receives feedback from some of the world's busiest and most highly regarded studios. Steady improvement is by degree, rather than a stream of new products. This is the nature of systems where every part is handmade by craftsmen rather than being tooled for mass production. Matt Dobson, the director responsible for development and production has the last word on *Boxer's* evolution 'We are bringing more of our production in house, rather than subcontracting; we want, and have nearly achieved, the capability to carry out all the processes — not necessarily to eliminate subcontracting — but to have the capability to try out our ideas quickly and learn from the problems'. The concept of the minimum feedback loop. 'New ideas are essential parts of the process, but turning them into a producible and *saleable* product is a major part of the effort. While there are improvements to be made in all areas — we're still far from the ultimate studio monitoring system — I believe we're closer than anyone else, at this moment'. ■

References

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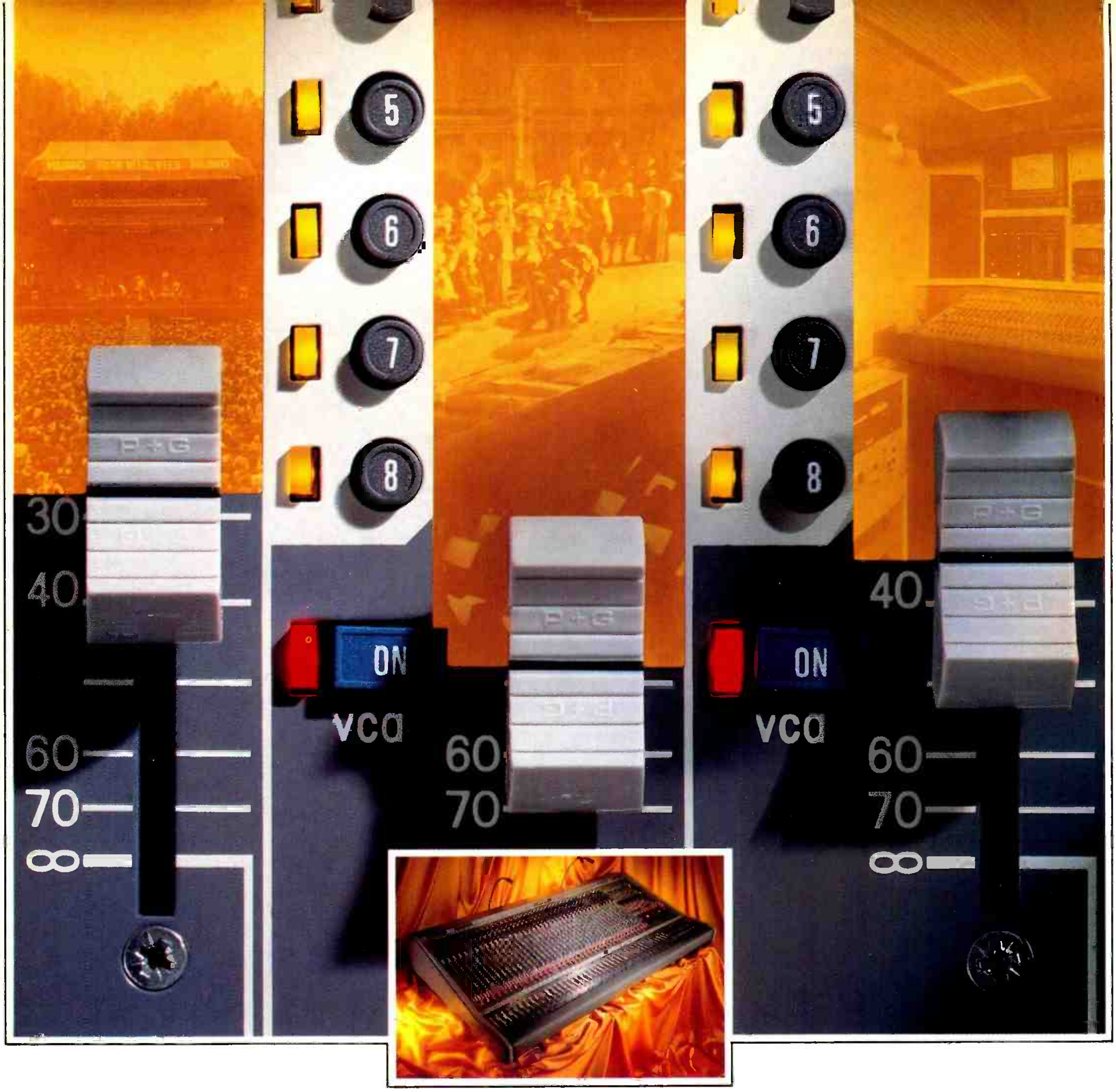

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Customisation is possible by moving windows around on screen, enlarging or reducing them

AUDIO ENGINE

Yashmin Hashmi assesses this PC-based recorder from Spectral Synthesis

American company, Spectral Synthesis have been in business for around five years having started by designing high-end-PC-plug-in cards and software for sampling. Their policy of continually developing their systems led to the addition of disk-based recording capabilities under the name *Digital Studio*. This was one of the first tapeless systems to allow up to 16-track recording and was a breakthrough in terms of its pricing. However, the operational software at the time left something to be desired, but one of the advantages of using a PC (as opposed to a dedicated controller) is that displays and control features can always be modified and this is what happened to the *Digital Studio*.

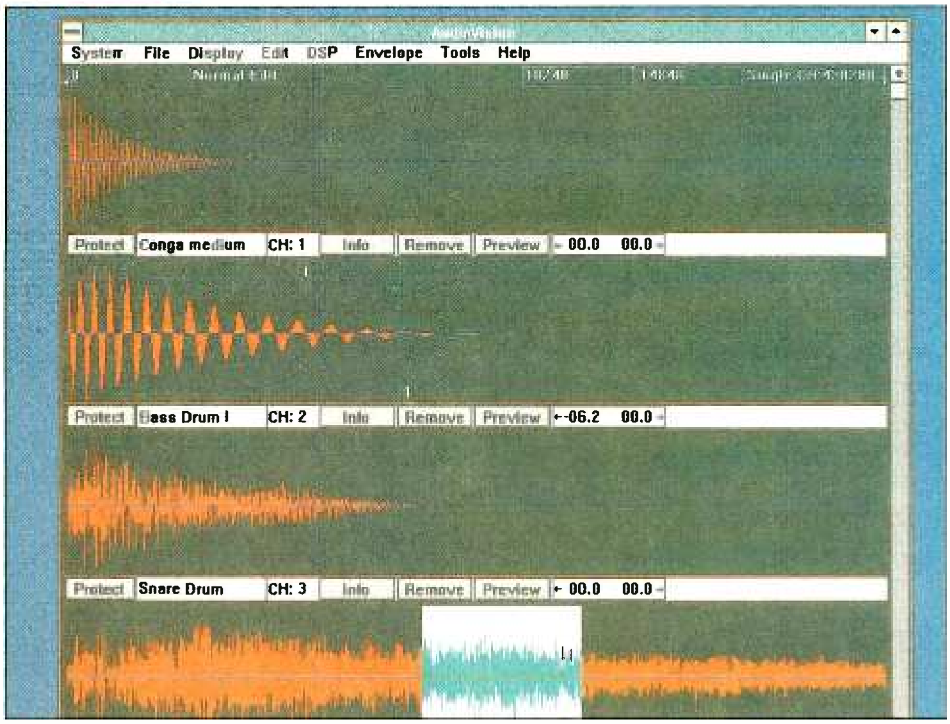
The *Audio Engine* represents a further development which provides a common hardware platform for both sampling and disk-based recording and has done relatively well in the USA. However, until recently, Spectral did not have European support for their products and many will not have been aware of the system's existence. Now that a distribution network has been established, a wider audience will have even greater choice when it comes to multichannel systems.

Hardware

Although the *Audio Engine* is PC-based, it is not designed for recording to the hard disk of the user's existing PC. The system consists of a main hardware rack with any IBM compatible (provided it is actually compatible) as the user interface. The PC should have a colour monitor and run *Windows* and is used as the user-interface only — multitask processing is provided by the rack which contains up to seven DSP chips. Operation can be supplemented with third-party controllers such as JL Cooper's *CS-10*, any *Windows*-compatible device (such as a tablet and pen) or any MIDI compatible controller.

The three main expansion cards are the *FlyBy* card (a digital audio bus controller which controls the timing of up to 140 simultaneous audio channels), the *Synth* card and the *Digital Studio* card. The *Synth* card is used for sampling and digital effects. It supports eight-voice sample replay and comes with two outputs and 4 Mb RAM. Up to four *Synth* cards can be supported, providing 32-voice sample replay with eight outputs and 16 Mb RAM. The *Digital Studio* card is used for disk-based recording and is capable of DSP functions such as crossfades and mixing.

The basic system is capable of channel expansion simply by adding further disks (up to a maximum of four). Each disk is guaranteed to provide at least four simultaneous channels. However, it is possible to achieve more than six simultaneous channels per disk, depending on how long cues are and how fragmented the disk has become. Thus a four-disk system, although guaranteed to provide 16 simultaneous channels, would usually provide more. ▶



Four tracks can be scrubbed at the same time

I/O

Balanced and unbalanced analogue inputs and outputs can be purchased in modules of two up to a maximum of 16 — thus allowing multitrack recording and replay. Each module also provides two AES/EBU and SPDIF interfaces as standard. A time code interface (which reads and generates all standard flavours of LTC) is optionally available as well as a MIDI interface.

Operation

The disk-based recording software includes modules and windows for recording and editing, mixing, routing and digital effects. A certain amount of customisation is possible by being able to move various windows around on-screen and enlarging or reducing them.

The sampling and sound design software is completely separate from the disk-based recording and editing software, although samples can be internally transferred to hard disk. It can perform mono or stereo sampling and provides a wide range of processing features.

There is on-screen patchbay for routing between any of the system's modules as well as inputs, outputs and tracks. When using the patchbay, the cursor turns into a small jack plug and routing simply involves selecting the devices to be connected — the system will then automatically draw the connection.

Disk-based recording and editing uses a virtual track display of up to 256 tracks, any number of which can be displayed on screen.

Recording

Recording is 16-bit with a choice of sampling rates of 32 kHz, 44.1 kHz and 48 kHz. Any input can be routed to any track and there are 16-segment on-screen level meters (one segment per bit) for each input. Each hard disk in the system is automatically assigned a unique colour, which is useful for quickly identifying and selecting which disk is to be recorded to. If the user chooses to allocate a specific duration for the recording, the system will automatically drop out of record after the allocated time. Once recorded, audio appears as blocks in the track display.

Editing

The colour of a block is the same colour as its source disk and this can be useful in quickly identifying whether an arrangement is trying to exceed the channel capacity of any disk — whether or not the total channel capacity of the system is being exceeded. So, for example, if there are more than four red blocks arranged simultaneously, it is immediately obvious that the red disk is being requested to play more than four channels at the same time.

The system will perform audio scrubbing and can scrub any four tracks simultaneously. Blocks can be quickly split into cues on adjacent tracks, cues can be snapped to a particular time code or moved to any track and overlaid on other segments with non-real-time crossfades. To assist with finding edit points, any track can be expanded as a waveform and there are 10 levels of undo.

Cues can be saved-in and auditioned from the directory, which can be searched and listed in a variety of ways.

Mixing and Effects

The 16-channel on-screen mixer provides real-time digital level control, panning, mute and solo. Faders can be ganged for stereo control and although the mixer is not internally automated, it can be automated using an external MIDI sequencer which itself would be driven by the *Audio Engine's* own time code.

The effects module lists a large selection of effects such as reverb, stereo compressor, noise gate, parametric and graphic EQ, filters, oscillators and even a test tone generator. Effects which have been selected from the list appear as graphic units stacked in a rack. Each unit has controls which can be controlled by the cursor and all effects are real-time. The selected effects also appear on the patchbay screen for routing and again, can be automated by an external MIDI sequencer. In addition, there are a number of time compression-expansion algorithms to choose from, with labels such as 'slow backgrounds', 'medium music', and 'high FX'.

Useful features

There are several features which are particularly useful such as the 'nuke it' option which allows the user to delete a take from the disk rather than just from the project-sequence. Another feature allows the user to quickly select the desired part of a take, while automatically deleting the rest. In addition there is a help system which gives information on selected features and procedures — rather than having to look it up in the manual.

Back up

This can be to optical disk, tape streamer or DataDAT (which allows faster than real-time transfer of multichannel audio). Cues can be auditioned from optical in real-time, which is useful for sound library applications.

Conclusion

Due to its layout and the intelligent use of colour, the *Audio Engine* is fairly straightforward to understand and operate. The patchbay, virtual tracks, choice of windows and open architecture also make the system very flexible and it seems that wherever possible, Spectral have intentionally avoided restricting the system's functions, allowing the user to find their own way of working rather than being forced into a particular pattern of operation. This extends to the choice of whether to display cues as blocks or as waveforms, although waveform editing alone can be frustrating due to the time taken to redraw.

However, a certain amount of operational discipline can be helpful, hence the use of modules and optional windows within them. Nonetheless, the sampling module, although very flexible, could perhaps, be made more powerful if completely integrated with the disk-based recording and editing side — such that moving from one module to another were transparent and samples or cues could be sequenced on the same display. It could be argued that this might provide too much flexibility and lead to confusion, but it would be preferable to have the choice.

When one considers that the same hardware platform can be used for both sampling, sound design and disk-based recording in addition to the comprehensive capabilities of the system (including digital mixing and effects), the *Audio Engine* provides good value for money and is far more powerful than its price range implies. ■

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Roland

Dear sir, Mike Collins is to be commended for his astute review of the Roland *S-750* and *S-770* samplers in the June issue, and his comparisons of the units with Akai's high-end samplers was particularly useful for those many studios making the difficult choice between one family or the other. There were, however, a couple of points that called for clarification.

I was particularly appreciative of his praise for the *S-750* manual, in as much as I wrote it, but his statement that 'it is a great improvement over the *S-770* manual' is a bit puzzling. When Roland released the *S-750*, they at the same time upgraded the software for the *S-770* to version 2.0 (now superseded by the operationally-identical 2.1), and commissioned a brand-new manual for the *S-770* as well. The two current manuals, therefore, are nearly identical; the major differences deal with the presence or lack of digital 1/0 and internal hard disk. (I also hope that Mike was not being disparaging of the version 1.0 *S-770* manual, which I also wrote, and which most folks I talked to thought was pretty good. Perhaps he was comparing the new *S-750* manual to the original Japan-produced *S-770* manual, which was, shall we say, weak.)

The other point concerns the video monitor problem. Yes, it is unfortunate that Roland no longer makes an appropriate monitor for its samplers, but an Atari-compatible RGB monitor (which I found for $200) works just fine, and an appropriate cable can usually be obtained from Roland main dealers. The placement of the monitor relative to the sampler is also not as big a deal as Mike perceives. If you have a good monitor, the mouse, and especially the optional *RC100* remote controller, you can pretty much put the sampler itself *anywhere* – except for tweaking the input levels while recording samples, there's no particular reason to have it within reach at all. The higher priority, therefore, should be putting the monitor in a comfortable position, while the sampler's position is essentially irrelevant.

Paul D. Lehrman,184 Palmers Street, Arlington, MA 02174, USA

AIDS

Dear sir, I read with interest Martin Polon's short article on AIDS. I fully support the idea of AIDS research and also the support of those who find themselves HIV positive.

Could I draw attention, though, to something Martin Polon wrote. He says, 'It is time that we accept that the disease is out there and that it must be stopped at all costs.' Martin implies that the cost is financial. The cost to find relief and cure might be financial, but the real cost to stop the disease is a change in lifestyle. I accept that it is possible for disease to be transmitted to innocents, that is blood transfusions and to babies, but one sexual partner for life, not just so called 'safe sex', would soon stop the disease. Sometimes it is easier to put a hand in your wallet than to change a way of life, but this is the price that needs to be paid.

Dave Aston, The Digital Audio Co, Croft House, Mill Lane, Steeton, West Yorkshire, BD20 6NS.

Russian AES

Dear sir, I want to inform you that in Russia the national section of the Audio Engineering Society is organised. The presentation of the Section will be held in September 1992 in Moscow, but just now more than 30 scientists and sound recording engineers participate in its activity.

If you are interested in the details of sound recording activity in Russia please inform me.

Alexander Gorodnikov PH.D., Head of AES Russia, RAMIS/AES (R), Office 722, Zemlianoiy val, 64, kor.1, Moscow, Russia.

Bad output

Dear sir, the undersigned has warned about the very mediocre input and output stages of even the best existing mixing desks and has long recommended the use of precision line drivers, for example on the monitor and main outputs of the mixing desk. This was done in 1986, and few paid any attention then (and even fewer even tried).

But even by just looking at the circuitry of these mixing desks, a good designer can instantly see the obvious compromises that have been made. Most of these output circuits cannot drive real-world loads, cable lines of several meters (and the sometimes rather low impedance, high capacitance inputs of monitor amplifiers).

Manuel Huber, FM Acoustics Ltd, Tiefenhofstrasse 17, CH-8820 Wadenswil, Switzerland.

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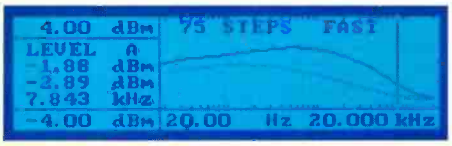

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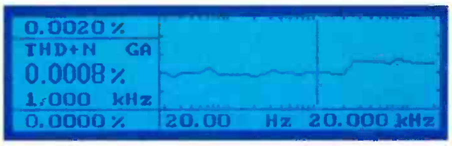

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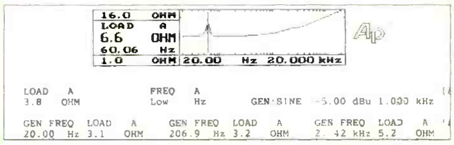

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MONITOR EQUALISATION AND MEASUREMENT

Philip Newell discusses the theory and practice of setting up a monitoring system

Probably all studio designers are asked from time to time by studio personnel to make one loudspeaker, room, or monitoring system to sound like another; at least in terms of certain characteristics. It is all too frequently expected that adjustments to the 'frequency response' will bring two dissimilar units into line.

If two loudspeakers are to be expected to *sound* the same, then effectively they must not only *be* the same, but must be auditioned under similar circumstances. The same requirement of similarity also applies to rooms, which again must be of similar construction in order to produce similarly perceived performances. In terms of classical acoustics, the pressure amplitude response, the phase response, and the presence of nonlinear distortions can be shown to determine the total performance of any loudspeaker or system, but the key question is to just how close a degree must those performance characteristics match before two units or systems can generally be deemed to be similar?

The response graphs published by many manufacturers are highly sanitised versions of reality; in many cases, marketing departments would not relish publication of more detailed response plots. Ironically, if nobody used 'smoothed' plots, then little would change in the order of things as everybody would just have shifted on to a more sensitive scaling. What such scalings *would* show however, would be just how far from perfection we remain.

Along with numerous other studio designers, I ceased using monitor equalisers for 'voicing a room' at the end of the 1970s. There are many 'old wives tales' which linger long after the truth of a situation is known. Judging by the number of people who still ask 'Which graphic do you use?,' or 'When are you bringing in the spectrum analyser?', whenever and wherever I am completing a monitor installation the myth of all 'professional' studios using monitor equalisation is obviously still alive and well.

Monitor equalisation *per se*, is not an absolute taboo, but there are only three common sets of circumstances where it may be used. It must never, however, be used to correct for reverberation problems as this would be attempting a cross-domain correction which can only end in tears.

Suitability

The three sets of circumstances where monitor equalisation *can* be used are: 1, to correct for driver discrepancies where such deviations from the desired norm are of a minimum phase nature, 2, to compensate for 'room gain' where a loudspeaker is placed somewhat near to a corner thus producing a bass build up, and 3, to apply a 'desired curve' where a studio operator requests, say, a roll-off of a shallow slope above a certain frequency.

All of the above cases are amplitude-frequency aberrations so a frequency-domain correction is in order. Reverberation time discrepancies are in the time domain and hence cannot be remedied by frequency-domain fixes. My practical solutions to the first three problems have been to: 1, design wide-band systems of sufficient linearity, consistent with sonic acceptability, so that high or low-frequency boosts or cuts were not required to linearise the systems, 2, to site the loudspeakers such that any 'room gains' were allowed for at the design stage, and 3, to adjust to 'desired curves' as much as possible by level changes of the tweeters.

My approach to the irregular or undesirable reverberation time problems have been either fix them acoustically within the rooms, or to learn to live with them.

The problems are based on the correlations between what first passes the ear, and what frequency balance is then left in the reverberant hangover. In our first three cases, we are dealing with adjustments to be made to the output from the loudspeaker. A minimum phase roll-off (or boost) is one which can be equalised in a way that will restore phase accuracy *with* amplitude accuracy. Should that roll-off (or boost) be nonminimum phase, then equalisation of the amplitude response will *not* restore the phase response and hence will distort transient waveforms. If the loudspeaker output *is* rolling-off with a minimum phase characteristic, then application of equalising boost will restore both amplitude and phase response. Unfortunately, however, each 3 dB of boost will *double* the power delivered into the loudspeakers when signals arrive at those boosted frequencies. It is, hence, all too easy to run into headroom problems either in the amplifiers *or* loudspeakers when such boost is present. Furthermore, it is almost unheard of for an equaliser to *exactly* mimic the inverse of the problem, so one frequently merely *changes* the problem. Third-octave or sixth-octave analysis is far too crude a tool on which to make any judgements of similarity of response. Somehow, the regularity of the bumps and dips in an applied corrective equalisation always appears to be more noticeable *as* equalisation compared to the bumps and dips of an unequalised system.

I believe that it is the regularity of split-octave equalisation which tends to make equalised monitor systems *sound* equalised, even when equalised for 'allowable' reasons. In the case of response irregularities where the problem is of a *nonminimum* phase nature, such as found at loudspeaker crossover points, any attempt to correct for amplitude response irregularities will inevitably cause aberrations in the phase response, which in turn will almost certainly impart an unnatural character to the overall sound. Such problems are to some degree inevitable in a large system but must be addressed as far as possible at source. They cannot and must not be dealt with by split-octave equalisation.

When a loudspeaker generates low frequencies in a largely omnidirectional pattern, constraining the angle of radiation by means of walls, floors and ceilings will concentrate the power in a forward direction. Most complete loudspeaker systems for use in rooms, to some degree or other, incorporate this property to augment bass output. Again, when loudspeakers are built into walls, the power concentration is usually included in the design criteria. From time to time, however, loudspeakers are positioned less than optimally in rooms not designed for them, and here, a rising or falling amplitude response can become apparent. Such a rise in the overall response of the room should not be confused with the room-reverberation-time-response anomalies. The room gain by constraint is both minimum phase and possesses the same time-decay characteristics as the loudspeaker system. It is therefore a frequency-domain problem, affecting all relevant frequencies in the same sense; it *can* therefore be equalised to give a linear overall response. Room reverberation, on ▶

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the other hand, is a function of the wavelength of reflected energy in the room modes. Such reflections are of a nonminimum phase characteristic; they also cause both peaks and dips in the overall response which will not coincide with discrete frequencies of conventional equalisers.

The major problem with attempting to make electronic equalisations compensate for reverberation time problems is that, unlike an omnidirectional microphone measuring a steady white or pink noise within the room, our ears can easily discriminate between the first pass of the sound and the reflected after-sound. When an over-long reverberation time exists at certain frequencies, the reverberant energy will make the room seem subjectively louder at those frequencies. Unfortunately, if the response of the monitor system is attenuated at those frequencies, then as a transient first passes the ears, energy will have been removed from that initial wavefront *at* those frequencies. The amplitude and phase characteristics of that transient will thus have been distorted and hence a 'natural' sound is almost certainly unattainable as transient integrity has been lost.

After a period of acclimatisation, the ear and brain soon learn the general character of the room from speech and other noises. Unless this reverberation is gross, when it can mask much of the detail in the sound, it is quite remarkable just how well we can adjust our perception to take the room into account. Were we to attempt to equalise the monitors to the steady-state performance of the room, then the leading edges of the sounds can sound *very* unnatural. Not only are they amplitude and phase distorted, but our ears and brains are still adding our automatic compensation. Those responsible for the aberrations of the 1970s when rock equalisation was still the norm, still have a lot to answer for.

Bear in mind that I am discussing analysis of studios. There *are* situations where it is necessary to compensate for time by frequency adjustment, such as in concert halls. When a band performs a one-night stand in an auditorium with a five-second reverberation time, the acoustics cannot be dealt with either in terms of cost *or* time. Here we may have a gross situation where, if at certain frequencies the hall rings and masks much of the detail for seconds afterwards, something must be done. In a live performance it is more desirable to lose transient integrity and maintain some definition rather than the other way round. It is therefore often necessary to reduce the offending drive frequencies, but such situations should never occur in a studio, and if they do, there is time to fix them acoustically. The old techniques of pink noise, a measuring microphone, and a spectrum analyser are clearly outmoded and potentially dangerous in the wrong hands, but what else is available?

Time-frequency reciprocality

Well, time and frequency are linked in the definition of frequency — the rate of change of phase with time. Were we to look at a single frequency on a spectrum analyser we would see a single-column display. The spectrum analyser represents a plot of amplitude against *frequency*. A remarkably similar picture would be displayed if we were to look at an impulse on an oscilloscope, which displays amplitude against time. A sine wave is a point in frequency, an impulse is a point in time. Conversely, were we to look at a display of a sine wave on an oscilloscope, we would see that it is a continuum in time, a full display across the screen. Again, were we to look at an impulse on a spectrum analyser, albeit briefly, we would see all columns energised, the impulse would represent a continuum in frequency. (See **Fig. 1**.) Strictly speaking, a true sine wave can never be turned on or off. It exists in perpetuity — for all of time.

The interrelationship between the spectra of the impulse and the sine wave is the Fourier Transform (named after the French mathematician who first described the mathematical relationship between the time and ▶

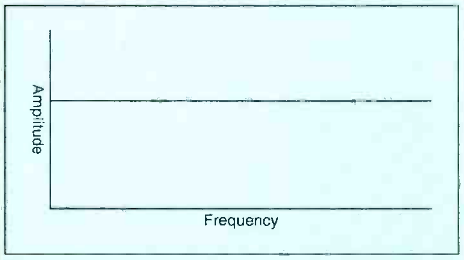

Fig.1a: Step function

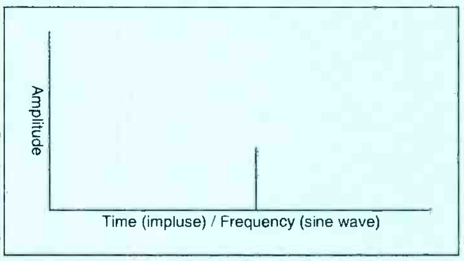

Fig.1b: Sine/impulse

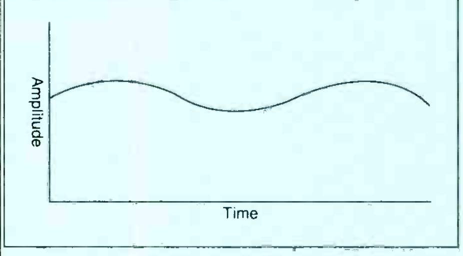

Fig.1c: Sine wave

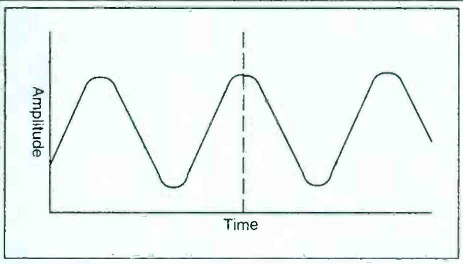

Fig.2a(i)

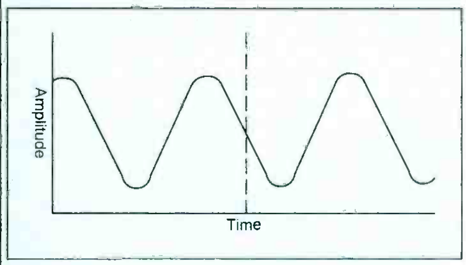

Fig.2a(ii)

-90° phase shift causes crest of (ii) to move with respect to (i) but the same overall time frame is still filled. Only the *pattern* moves along the axis

frequency domains). A Fourier Transform must be carried out frequency by frequency, and hence for a wide-band signal is a singularly tedious exercise. The advent of computers brought the possibility of the FFT (or Fast Fourier Transform). By use of such FFT analysers, impulses can be generated from the noise signals such as white noise, pink noise or a pseudo-random binary signal (PRBS). PRBS contains discrete frequencies of, say, 20 Hz intervals, which allow the analysers to 'lock' onto the signal with greater speed than waiting for each frequency to eventually, randomly, appear in the white or pink noise, which, as in the case of the impulse, contain all frequencies.

A further function of the reciprocity of impulsive and steady-state signals is their characteristic relationship in the terms of their phase properties, which is one reason why the phase integrity of a system is so important where 'natural'-sounding reproduction is sought.

As the definition of frequency is the rate of change of phase with time, only an individual frequency can have a phase shift. A sine wave, phase shifted, will show a displacement of its pattern along its time axis. (See **Fig.2a**.) A transient impulse cannot have a phase shift as it has no single frequency, but contains all frequencies simultaneously. A movement of the display along the time axis would represent another point in time when that impulse occurred. (See **Fig. 2b**.) That movement in time would represent not a phase *shift* but a phase *slope*. Whenever we meddle with graphic equalisers to correct such nonminimum phase distorions such as room reverberation, (where to improve amplitude response, we destroy phase accuracy) we tend to get further away from reality, not closer to it. So if we are going to attempt to assess what is really happening in a monitor response, we must be able to look simultaneously yet separately at both amplitude *and* phase with respect to frequency.

By means of Fast Fourier analysis and the related MLSSA technique (Maximum Length Sequence System Analysis) the phase and amplitude information can be assessed, *in situ*, and either looked at in terms of loudspeaker performance, room performance, or the performance of the combination. For FFT analysis, either our familiar random-noise sources may be used, or alternatively, impulse noise sources. Each have advantages and disadvantages.

Practical test signals

Any half-cycle of a square wave contains a very great number of frequencies in very specific phase and amplitude relationships. As any waveform distortions will upset the time-frequency balance of a system, no accurate square wave reproduction is possible if such distortions are present. In order to accurately reproduce a square wave therefore, an accurate pressure amplitude response, commonly referred to as 'frequency response', and an accurate phase response must be prerequisites. In view of this, accurate square-wave responses of loudspeakers should be the ultimate goal, as if a ▶

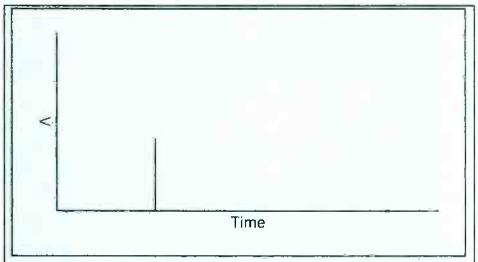

Fig.2b(i)

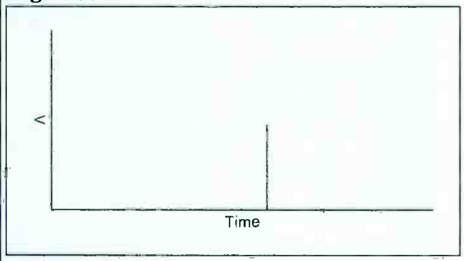

Fig.2b(ii)

Time shift of impulse causes movement along axis. The pattern, however, remains the same

Fig. 3: Comparison of step function responses of three loudspeakers

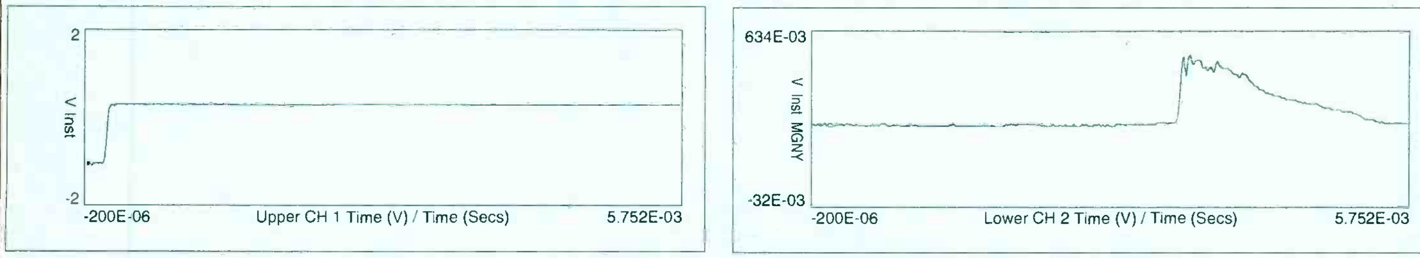

Fig. 3: Input Step

Fig. 3a: Response of electrostatic loudspeaker – tail off due to limited low frequency response

square-wave *can* be reproduced accurately, then *all* waveforms should be reproduced accurately. For testing purposes, the two extremes of square waves are frequently more convenient signals to use. At one extreme, the delta function or pulse, is a square wave with an extremely short duration, and is usually accompanied by a short mark-space ratio. At the other extreme is a step function, a half of a square wave with long duration 'on' cycle. In order to prevent comb filtering of the measured results due to truncating the data as a conventional square rises then falls, the test signal should have either an infinitesimally small 'on' time, or an infinitely large 'on' time. The delta function and the step are practical realisations of these requirements. During the subsequent signal processing, either can be regenerated from the other by a process of differentiation or integration, so the choice is down to convenience.

The pulse, or delta function, does have certain practical drawbacks. Firstly, it is difficult to generate accurately without overshoots or ringing. Secondly, as the burst of power is so brief, it is difficult to either hear or see (by means of an oscilloscope) whether any part of the system is clipping. If mechanical clipping of any drive unit did occur, nonlinear distortions, while possibly imperceptible aurally, would render the measurements useless. Thirdly, the low-frequency content of a delta function is so low that the bass drivers really do not get a chance to be put through their paces, nor are they given the power to excite any resonances to any measurable degree. A step function is easier to generate, works the bass drivers much harder, and is generally much more easily judged in terms of the 'loudness' of the impulse, giving the person performing the test a good indication of the overall power level being fed to the loudspeakers. There is thus less chance of clipping the system or encountering accidental gross, nonlinear distortions.

By means of Fast Fourier Transform, a white, pink, or PRBS noise source can also be used to produce an impulse or step-function fingerprint, but unless complex gating techniques are applied, an anechoic chamber is required, as the effects of the rooms on the steady-state signal cannot be separated from the responses of the loudspeaker systems themselves. The use of simple gating techniques allow a step to produce its characterisitic time-history fingerprint *in situ*, along with both pressure amplitude and phase responses via FFT analysis. Except in the smallest of rooms, where the first reflections appear before the low-frequency component of the step function can be fully integrated into the FFT, there is no requirement for the monitor systems under test to be removed from their daily working surroundings, and hence from their 'real world' loading conditions. Remember, the room is itself a part of a loudspeaker's air loading. Changing the position of a loudspeaker cabinet from being flush mounted in a wall, to being freestanding in a room, or even to being placed in a corner, may significantly affect the character of the air loading. In a location where the low-frequency drivers have more air to push on, more work can be done. Where this extra work manifests itself as a change in acoustic output, effectively, the performance of the loudspeaker has been altered. This is our previously mentioned 'room gain' which *is* correctable by appropriate equalisation. Irrespective of room reflections, a loudspeaker is not the same under all conditions. It is necessary to measure a loudspeaker in its particular working environment for accurate assessment, especially of the low frequencies. It is for this reason that the impulse and step-function fingerprints are more practically achievable from an impulse source, than from noise sources, conventional forms of which would require the loudspeakers to be removed to anechoic conditions if the room effects

Fig. 3: Comparison of step function responses of three loudspeakers

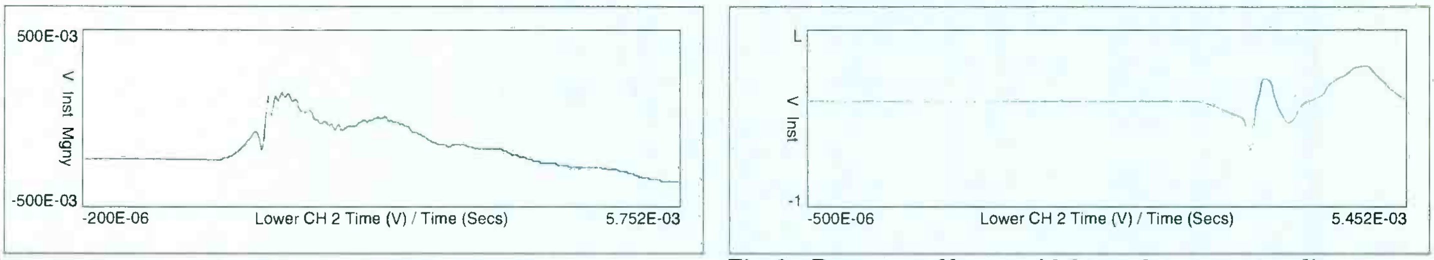

Fig. 3b: Response of well-known dual-concentric

Fig. 3c: Response of large, widely used, two-way studio monitoring system

were to be separated out.

Perceived drive unit differences

In a representative sample of around 30 high-quality midrange systems, while most sounded very similar on signals with a highly-tuned content such as resonant drums, sine waves or smooth enveloped tonebursts, *not one* like-sounding pair could be found when listening to white noise or a recording of a waterfall: not even some identical drivers from the same batch of the same production line. Ironically, less expensive, lower-quality drivers could be matched more easily, as inherent flaws tend to cause an easily recognisable pronouncement of a certain reponse characteristic. Such characteristics, together with other response limitations often tend towards a matching process of considerably greater simplicity. The more accurate a midrange unit becomes, the more the differences appear to stand out.

Taken to an extreme, a dozen loudspeakers of different shapes, sizes, and of differing materials, all with a response from 1,000 Hz to 1,002 Hz, when subjected to a noise signal will all give a remarkably good impression of a 1,001 Hz sine wave. Widen the response band, smooth out the major irregularities and increase your problems of subtle consistency!

People often mention that recordings of a certain well-known band always seem to sound good, again irrespective of on what, or where they are played. By chance, having been looking at an EPQ, band-splitting visual monitor, it was noticed just how strong the fundamentals appeared to be on those recordings. There was a great predominance of 'clean' pseudosinusoidal waveforms. Such, narrow-band or sine-wave-orientated signals produce a distinctive fingerprint which is very difficult to upset. The opposite extreme is the tendency for a wide-band linear system to allow *differences* in wide-band signal response anomalies to predominate over the similarities. The better the system, the more the discrepancies will show.

The point to be made here is that the present conventions of using pink noise or swept sine waves is of little relevance beyond a certain point, when very high-quality drive units begin to produce very similar response graphs which have no bearing upon the humanly-perceived discrepancies in their sonic character. From the work done to date, I believe that the step-function fingerprints do relate to perceived sonic performance, with the units which visually produce a recognisably more accurate output response sounding more 'natural' under representative listening conditions. **Fig.3** shows the step-function time-history responses of three widely-used monitoring loudspeakers. Although they all have acceptably flat 'frequency responses', the three systems *sound* very different. From these time-history waveforms of their step responses, it is little wonder that they sound so different.

It must be added here, however, that although the manufacturers published data on the above three loudspeakers is commendably 'flat' in terms of frequency response, a certain amount of averaging has been performed which is generally considered to be acceptable. If one decides to look at an unsmoothed spectrum as derived from an impulse, the *actual* differences in the amplitude and phase responses are readily apparent. These are the differences which conspire to produce the gross waveform differences seen in **Fig.3**. The amplitude response anomalies between **Fig.3a** and **3c** are shown in **Fig.4**. Third-octave analysis just cannot show this degree of detail. It is often quoted that the ear, in general, perceives third-octave▶

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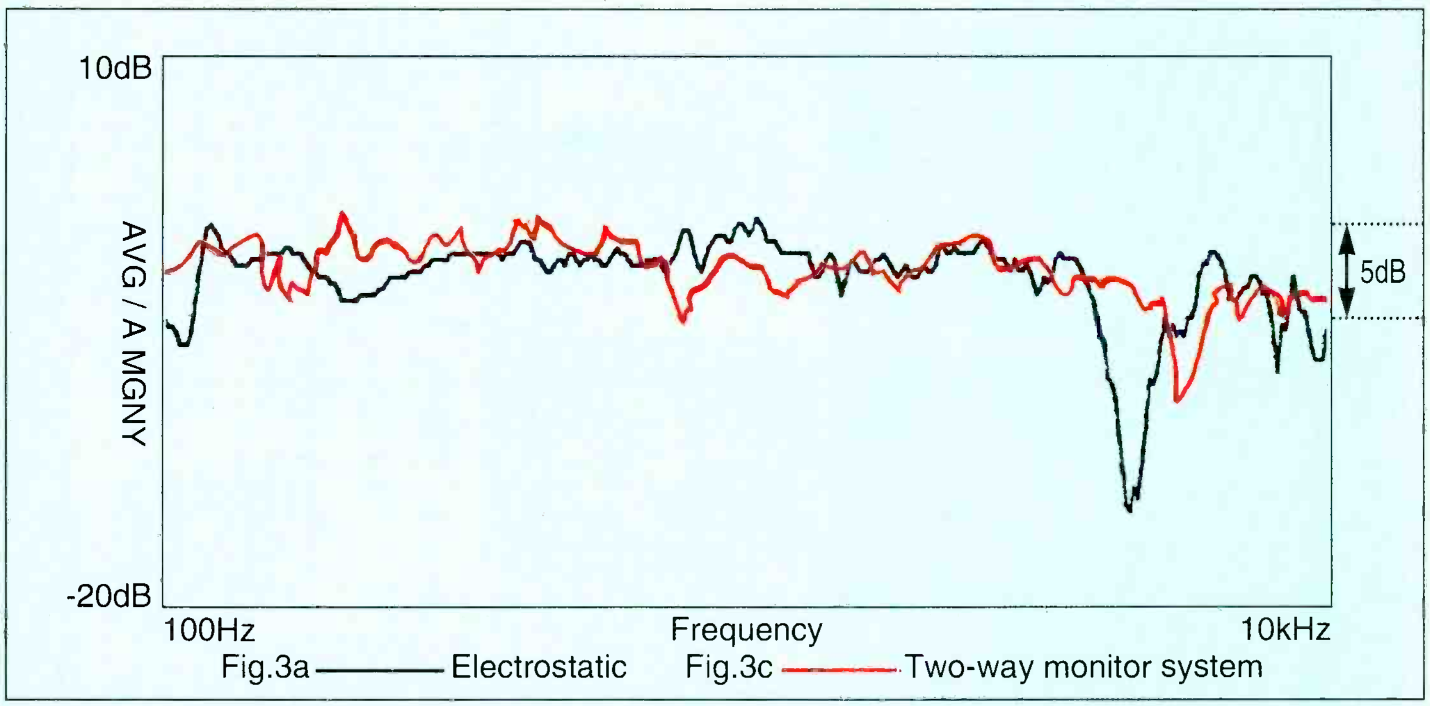

Fig.4: Both respectably performing within ±2.5dB over most of their range

power bands, and that minor discrepancies in two signals would not be perceived if the average power level in that third-octave band remained the same. Unfortunately, the compounding of these discrepancies, not only in terms of amplitude, but also in terms of phase and hence waveform, conspire to make performance differences which are clearly audible but do not show on third-octave analysis. Further to this, some manufacturers publish idealised performance plots with accompanying statements of permitted production deviations. While these specifications may appear to be commendable, production differences at the tolerance extremes may represent 4 dB difference in the output of two drivers at any one frequency for units with a ±2 dB tolerance. Such specifications are of great use to system designers and users, but they do *not* represent any clear statement on sonic compatibility between different drivers with similar specifications. Albeit largely in narrow-band disturbances, it has been our experience that impulse testing has been exposing performances more in the order of ± 5 dB rather than the ± 2 dB often quoted by manufacturers. Once again, I am not accusing the manufacturers of 'sharp pencil' practices as they have been using accepted techniques. I am implying, however, that the currently accepted techniques do not relate to sonic performance similarities. None of these problems are addressed by conventional split-octave analysis and equalisation. Such equalisation may, in some circumstances, produce the result of a more pleasing sound, but such adjustments rarely produce any greater degree of 'accuracy'.

Combination responses

When given a highly-linear monitor system, a room will probably dominate in the perception of the overall sonic performance, hence the reverberation time characteristics of the room will assume very great importance. The complication here, however, is that the knowledge of the reverberation time of a room, when calculated from an omnidirectional source such as a starting pistol, is of little consequence unless the loudspeakers were to energise the room in a similar manner. Without any fixed standard of directivity, it is impossible to say precisely what would be the 'best' reverberation time characteristic for any given room. Only when considered as part and parcel of the monitor system can the Rt requirements of any room be specified or judged. (see the 'nonenvironment' rooms discussed in *Studio Sound* November 1991.)

Impulse testing of the loudspeaker and room combination can, by Fast Fourier Transform, give a graphic representation of the actual, overall steady-state response of the whole monitor combination. Only when the steady-state response and the gated, on-axis impulse response produce one and the same, overall, linear response graph, can the response then be deemed to be in any way 'accurate'. Tailoring of the response by means of equalisation may then be possible to suit individual requirements based on average listening levels, fatigue problems, or other considerations, but at least any adjustments will enable the overall perceived responses to 'track' those adjustments.

Where an off-axis reponse becomes irregular or lobes, excitement of the room modes will not be uniform, even for minor differences in the position of the loudspeaker. Again, by the monitoring of the step-function response of the loudspeaker at various positions, it can be determined to what degree a room is being driven in each direction, and reverberation times at problem frequencies can be adjusted, either acoustically in the room, or by repositioning the loudspeakers to drive the problem modes to a lesser degree. While many people have put forward proposals for omnidirectional loudspeakers to mimic more ▶

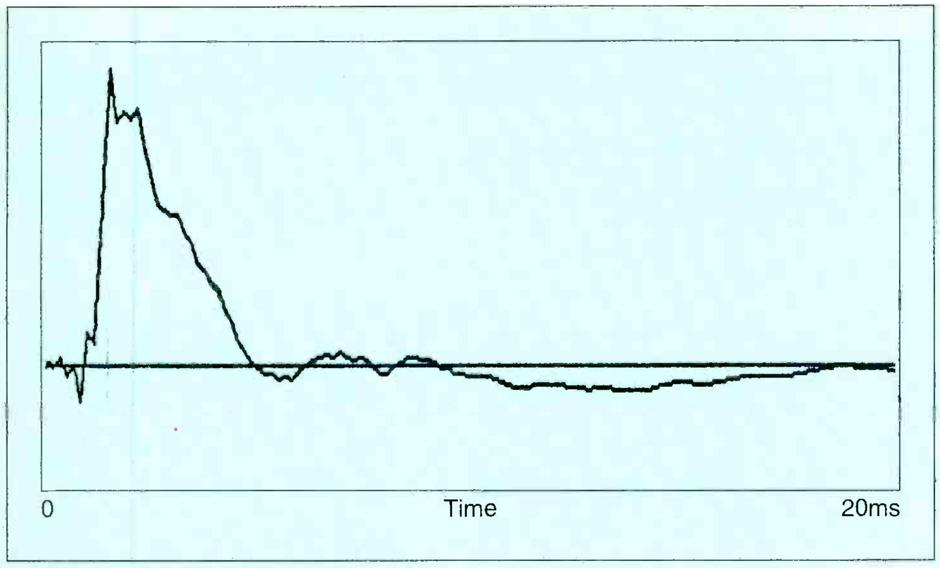

Fig.5: LMH impulse response at 2 metres

closely the characteristics of the original recording space, such polar distribution would cause an inordinately high direct-to-reflected ratio in the listening room, blurring the stereo imaging. About 60° to 80° would seem optimum for the horizontal directivity of the mid and top of a monitor system. Having said this, however, the impulse and step-function fingerprint at a 30° off-axis position should be as close as possible to the on-axis fingerprint for an even response while moving around the room. Furthermore, one can never *truly* reproduce the sound field of an instrument, as only in loudspeakers is the sound source distribution dependant entirely upon frequency due to the spacing of the high-frequency and low-frequency drivers. It appears that we are, in the foreseeable future, always going to be looking at compromise in loudspeaker design. Without common goals and standard specifications, however, it will prove very difficult to approach the most realistic compromise, as many manufacturers pull in widely disparate directions and merely try to promote the strongest assets of their own products.

Fig. 5 shows a practical application of the technique. The trace shows the step-function response of a large, two-way monitor system, in a suitably-designed control room, measured at a distance of around two metres. The rapid rise time, well behaved decay and conspicuous lack of resonant overshoot are all reminiscent of an electrostatic. The system is in fact capable of producing over 120 dB behind the mixing console and is horn loaded above 1 kHz. Many experienced listeners have deemed this system to be audibly very similar to typical electrostatics, but of much greater potential SPL. It is certainly by far the closest to the electrostatic of **Fig. 3** in terms of its step-function fingerprint.

When I first put forward these proposals for the publication of step-function plots in loudspeaker specifications, many people asked why I should propose again, something which had been largely rejected in the past. It had been generally considered that although they were all dissimilar, they showed little information in terms of what those differences actually were. My feelings at the time were that too many loudspeakers were too 'wrong' to produce fingerprints close enough to display much meaningful information: it was the *loudspeakers* which were failing to deliver, not the technique!

As **Fig. 5** shows, when one begins to close in on the target, the picture suddenly begins to make sense. The amplitude and phase-response plots of the system are shown in **Fig. 6**. Obviously, these must be commendably flat in order to have any hope of producing a step response such as **Fig 5**. The point is, however, that such a response, in the room at the listening position, was *not* achieved by any electronic equalisation. The horn (shown in the November 1989 edition of *Studio Sound*), was specifically, acoustically matched to the desired drive unit, and also to the properties of the intended room. Room reverberation time problems were treated entirely acoustically within and without the room. The only significant kinks in the curves lie at the crossover frequency but as this is a nonminimum phase nature, correction by conventional means could not restore both amplitude *and* phase to linearity. It was considered that attempts to linearise any part of this region would ultimately be detrimental to the sonic neutrality and natural musicality of the system. In any event, the band of disturbance is so narrow as to be deemed virtually inaudible and is best left as is for the time being. The drive units themselves were selected for their inherent wide-

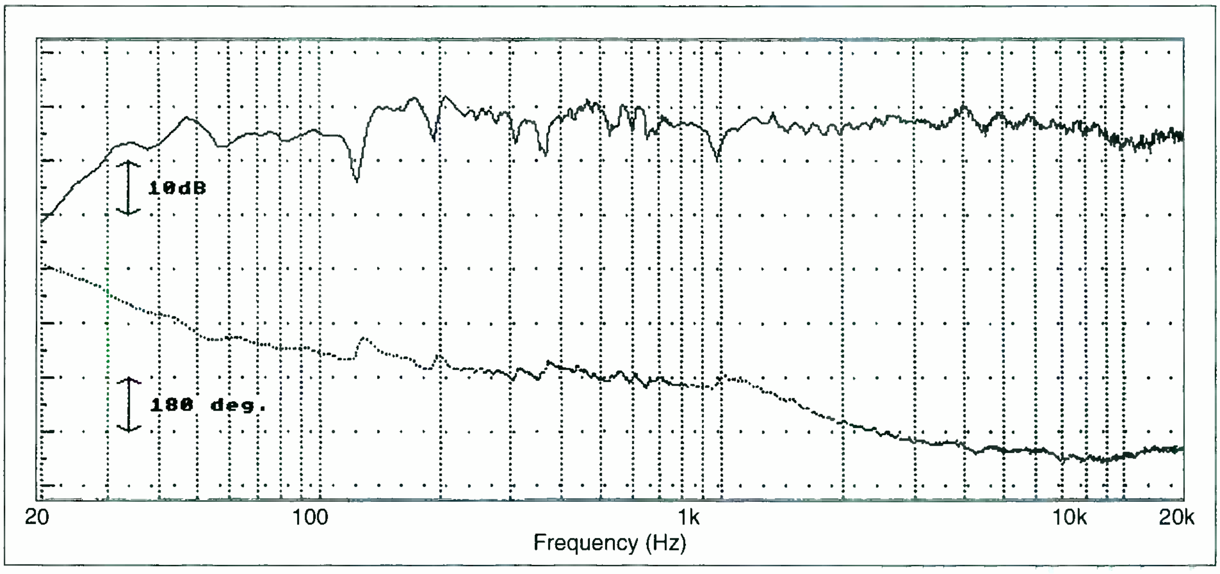

Fig.6: Two-way monitor measured on-axis at 2m *in situ* in LMH

frequency-response range and their subjective audible neutrality. The two by no means necessarily go hand in hand.

I would strongly recommend that step-function fingerprints and their Fast-Fourier-Transform derived phase and frequency-amplitude graphs should become the accepted reference standard for all loudspeaker, and loudspeaker and room, combinations which are intended for accurate studio monitoring. The general drift of technological advance should be towards more faithful reproduction of such step functions, after which everything else should begin to fall into line.

One interesting outcome of the aforementioned listening tests carried out on the midrange units at the ISVR in Autumn 1989, was that the perception of steady-state signals appeared to be more sensitive to amplitude-frequency discrepancies, while the transient sounds were most perceptibly different when smeared in terms of the amplitude-time plots by poor phase accuracy in the systems. In the frequency domain, a phase lead or lag can be introduced by relatively simple, conventional electronic means, but in the time domain, a time lead, if required, would necessitate an acausal correction. In other words, the correction would have to be applied before the incident to be corrected, took place. The obvious problem is how to sense an event in real time, before it occurs.

Digital signal processing technology has provided a practical reality to the concept of active control of the entire time, phase, amplitude convolution. By the introduction of a modelling delay between the signal source and the loudspeaker system, then by allowing feed-forward signal paths around that modelling delay, response errors can be detected, inverse modelled, fed-forward, then recombined with the original signal, before the total delay time of the error-correction system has elapsed. Such active control may well be the way forward in impulse reproduction accuracy. One thing *is* certain, however; given the complexities of the interrelationships, a spectrum analyser, measuring microphone and a graphic equaliser is such a grossly oversimplistic assessment and correction system that except in a very few specialised circumstances over the years, they have probably done far more harm than good. Indeed, if a general high or low lift is all that is required, few equalisers, if they *must* be used, sound more natural than a simple Baxandall-type 'bass and treble' control. ■

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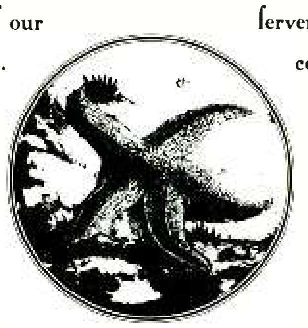

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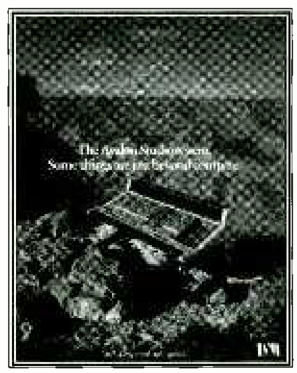

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One would imagine there was precious little that the Pope and the American all-male striptease act The Chippendales could have in common, but surprisingly enough both are professional ATC users. In fact the list of ATC clients is expanding rapidly with an impressive, if not diverse, collection of names from recording, broadcast, live sound, film, and research backgrounds.

ATC (Acoustic Transducer Company) was founded in 1974 by Bill Woodman. Woodman had worked in his native Australia as a loudspeaker design engineer for Plessey Rola, but feeling the job potential was exhausted, he relocated to the UK where at the tender age of 24 he secured a job as Principle Design Engineer with Goodmans, with responsibility for special products including the Audiomax professional range of loudspeakers and the development of soft-dome tweeters. Following disagreements over the marketing of Audiomax, Woodman left after four years to join Gale Electronics but only stayed for a matter of months, anticipating the company's difficulties and subsequent collapse. ATC was then set up, and the first product was a 12-inch sound-reinforcement speaker with a difference, as Woodman explains.

'All the midrange in large PA systems up until that time had been horn-loaded, and we developed a direct radiating speaker that Martin Audio took on. They then designed a system for Supertramp,

SCM *50* and SCM *100* monitors

IN THE DRIVER'S SEAT

ATC's soft-dome approach to monitor design has found them friends worldwide. Patrick Stapley talks to founder Bill Woodman

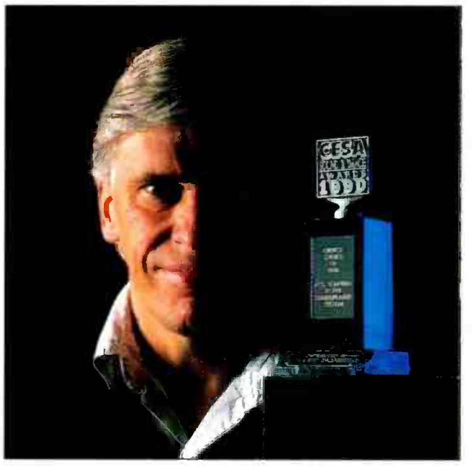
and it was this system that got ATC started as a PA manufacturer.'

Over the next few years ATC reputation grew and they became established in the sound reinforcement market. But Woodman was keen to broaden the company's appeal and set about designing a midrange soft dome unit. Through his work at Goodmans with soft-dome tweeters, Woodman had become excited about the advantages of the design and the possibilities of applying it to midrange drivers.

'Midrange drivers had always been the weak link in loudspeakers, and it had been in my mind for a long time to design a soft dome that would operate over the most critical region of the ear, 300 Hz to 3 kHz, offering the advantages of excellent dispersion characteristics due to the very small diameter (75 mm) of the transducer, and high power handling because of it being edge driven by a very large diameter coil, thus avoiding the modest power handling of traditional direct radiating systems.'

The midrange soft dome soon went into production and a range of large-diameter-coil bass drivers were designed to go with it. Both types of transducer featured short coils operating in a long magnetic gap, resulting in improved power handling and reduced distortion. Although not a new approach (JBL pioneered it in the early '50s), it became something of an ATC trademark along with the soft dome, and huge magnet assemblies.

'ATC's significance is that we actually bridged the gap by producing transducers that offered the high acoustic performance associated with the British direct-radiating designs, but also the huge dynamic range of the American horn-loaded systems which were favoured by the majority of studios at the time. However in those early days (1976) it received a poor response from the marketplace, because its advantages were not immediately obvious through analogue recording techniques. It's true to say that the soft dome came into its own with the introduction of digital recording about six years later, which was a bit of a shock because I thought the advantages would have been obvious right from the start. It wasn't until producers and engineers got themselves into trouble and had to go through the learning curve ▶



Bill Woodman

SCM10

of dealing with a medium that was much more telling, that they did something about traditional monitoring systems.'

The '70s had seen a flurry of R&D activity among UK manufacturers culminating in designs such as the *KEF 501* and *B&W 801*. Although these systems did a great deal to raise monitoring standards and were generally well received, Woodman had misgivings.

'There was this belief regarding the low performance of those speakers, that you required an optimally-flat bass response under anechoic conditions and they were given quite high Qs. The effect when you put them into a reverberant room was that they always sound overblown and underdamped. It's well documented that the most significant form of masking is from low frequencies, and consequently the clarity of the midrange suffers because of the bloated bass. This approach still exists in Britain to a large extent, and if you read the current text books they still recommend Qs of between 0.5 and 0.7. We run Qs between 0.3 and 0.4, and what we attempt to do is tailor the low-frequency performance to mirror low-frequency room gain, by doing this it produces a more natural bass response, plus it doesn't mask the all-important midrange.'

In 1982, Woodman's partner, Tim Issacs, developed an electronic crossover with phase correction and momentary gain reduction circuits; around this time, ATC were approached by Roger Quested (then studio manager of DJM Studios in London) to install a monitoring system.

'The DJM was the first big monitor system that we'd built — it was a 2 in by 15 in system with one of Tim's first electronic crossovers. Anyway the studio closed shortly afterwards and Roger Quested decided to go into monitoring systems. We produced designs and supplied him with our components including the electronic crossover, and away he went. That relationship continued for about three years, until in 1985 we were requested by Danish Radio to submit an active loudspeaker and out of that was born the *SCM50A* and *SCM100A* which the BBC subsequently used. Prior to that we'd only been making passive loudspeakers (*K50* and *K100*), but we were so surprised at the results from this active design, that we decided to seriously market it, and in particular aim it at the top-end domestic market.

'We called Roger in and offered him the range, but explained that it had to be marketed as ATC and not Quested. In the end no agreement could be reached, so we parted company. Quested carried on for a while using our components but once he felt we were in direct competition with him, he moved away from ATC components altogether. In fact as ATC started to build its own brand monitors and gain a higher profile, other companies that had been using our components, such as Genelec, stopped doing so.

'ATC's strength lies in the fact that it's an engineering company founded on the manufacturing of transducers, so we have control of the very things that are most important in the system. None of the competitors who tend to use the same sort of technology have that facility, they're all buying components in. The other aspect, is that unlike most other manufacturers of studio systems, ATC have a very large domestic market — for every speaker we sell to the professional market, we sell 24 to the hi-fi market, and this allows us to make investments in transducers and tooling that we couldn't possibly make otherwise. Today our business is equally split between manufacturing complete loudspeaker systems, and supplying components.'

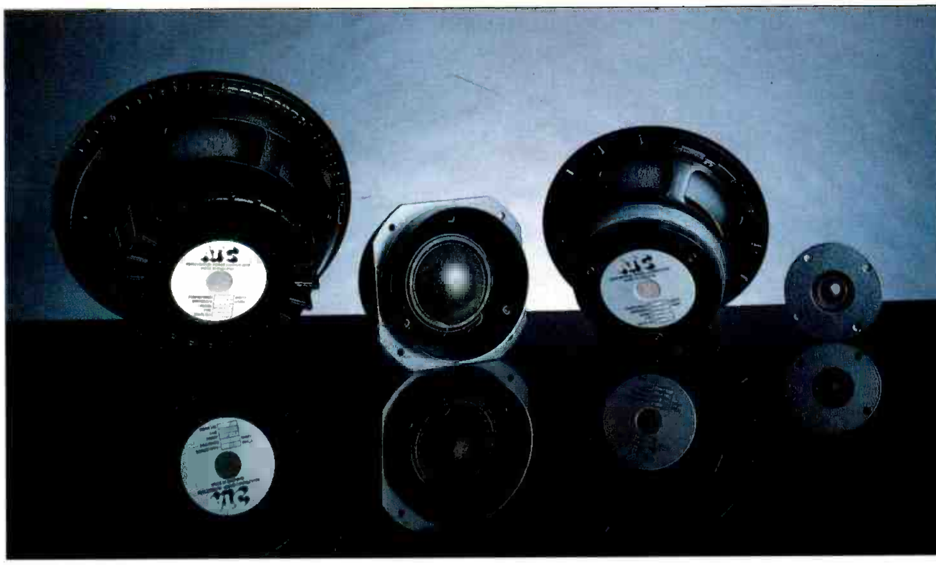
The only components that ATC do not make themselves are tweeters.

'We don't make tweeters because there is a choice of hundreds manufactured around the world, and if we were to manufacture one it would be very similar to what we buy from the Danish company Vifa anyway. We looked very closely at all the alternatives and in our opinion it's the best ▶

***SCM10* drivers**

CURRENT RANGE

- *SCM 50:* passive or active, three-way, max continuous SPL 112 dB @ 1M, size 716 X 304 X 333 mm (passive version).
- *SCM 100:* passive or active, three-way, max continuous SPL 115 dB @ 1M, size 832 X 398 X 424 mm (passive version).
- *SCM 200:* active, four-way, max continuous SPL 118 dB @ 1M, size 830 X 730 X 440 mm.
- *SCM 300:* active, four-way, max continuous SPL 120 dB @ 1M, size 884 X 925 X 476 mm.
- *SCM 20:* passive, two-way max continuous SPL 108 dB @ 1M, size 440 X 240 X 310 mm.
- *SCM 10:* passive, two-way, max continuous SPL 103 dB @ 1M, size 380 X 180 X 255 mm.



***SCM 50* drivers**

and most consistent high-performance tweeter available, although it's from a commercial manufacturer.'

Woodman does not differentiate between the hi-fi and professional markets. His view is that high fidelity is as important to the serious listener at home, as it is to the recording engineer.

'The truth is that many hi-fi enthusiasts have more discerning equipment than you quite often find at recording studios, and they can be incredibly discerning as individuals too. The studio industry tends to be very conservative, and a lot of studios are using very old equipment that they really shouldn't be using if their aim is to produce the best possible sonic results.'

Woodman includes horn-loaded monitors in this redundant technology, and blames them to a large extent for the hardness and harshness that has crept into mixes over the last decade.

'Without exception, horn-loaded systems have poor dispersion in the midband. Take the dual concentric design; it starts off with very broad low-end dispersion, but as the frequency increases the dispersion decreases until at the upper to midband it becomes very constricted, and then the high-end transducer will start radiating and it will suddenly become broad again. Now, the problem with this is if you're doing a mix, particularly of synthesised music where you have no reference — as opposed to acoustic music where you know what things should sound like — the tendency will be to equalise the midrange to get a balanced sound in the control room, and consequently you end up with a hard sound. If you have a speaker that correctly excites the reverberant field in the midband, you won't do it because it will take your ears off!

'Another inherent problem with the dual concentric design is the large low-frequency diaphragm. Because of its instability, you get bell moding as the diaphragm literally wobbles, which means the potential maximum SPL is limited by the mechanical integrity. Bell modes produce enharmonic tones and subharmonic ones at that, which are terribly tiring to listen to, much worse than harmonic distortion, and they can occur at quite low drive levels. We make a dual concentric for PA use, and although they deliver much higher SPLs than other manufacturers' studio versions, there comes a point where the mechanical problems are just not resolvable. I don't think dual concentrics are suitable for studio monitoring — basically they're past their sell-by date. In my opinion the only reason they've hung on so long is

Cut-away of *SM75 - 1505* soft-dome midrange driver

ATC DESIGN CRITERIA

1. Linear Amplitude Response — typically ±1.5 dB 100 Hz to 10 Hz.
2. Low Harmonic Distortion — typically 0.3% above 200 Hz @ 1W, and 1% 200 Hz @ 25W.
3. Low Intermodulation Distortion.
4. Low Time-Domain Distortion.
5. Well-Behaved Impedance Characteristic.
6. Linear Phase Response.
7. Large Dynamic Range.
8. Broad Dispersion.
9. Manufacturing Consistency.
10. Reliability.

because they are relatively good value for money; rather ironically in the Far East they're very popular because people look on them as being a good, solid British design.'

Not far removed from the concept of the dual concentric, is the mid-bass driver used in ATC's *SCM 20* and recently introduced *SCM 10* compact-nearfield monitors. The design is a hybrid incorporating both bass cone and midrange soft dome into one unit.

'When I designed the original soft dome, part of the design included a flared flange which acts as a phase correcting medium. It suddenly struck me that one could replace the flange with a diaphragm, and end up with a dual unit. The philosophy being that the diaphragm is a mirror image of the dome in the centre. The drive unit effectively becomes smaller in diameter so you hear this very even dispersion of frequency, and with both the *10*s and the *20*s you can move 80% off axis and they will only be 6 dB down at 10 kHz which is staggeringly broad. The advantages are that the stereo hot spot broadens considerably and you become less aware of the speaker — they disappear if you like.

'Another aspect of these speakers is the sheer volume they produce. The *SCM 10* (440 x 240 x 310 mm) produces 103 dB continuous SPL at a metre, and people find it hard to comprehend that such a small speaker can produce that much level without any trace of distortion. Because the ear perceives distortion in relation to loudness (the effect can be demonstrated by listening to a cheap transistor radio) the lack of distortion can give the impression that the output is quieter than it really is, and to start with it's a good idea to listen at an artificially low level and build up slowly until you get use to it. This is actually the case with all our speakers.'

Having just introduced the company's smallest monitor, ATC are currently working on their biggest. The *SCM 400* features a 4-inch soft dome and will deliver a continuous SPL in excess of 120 dB. It will be suited to large rock studios, and high quality PA applications, and should be available towards the end of the year. Also in the pipeline is a series of ATC power amplifiers.

With all this activity, ATC are presently extending their Gloucestershire premises, which considering Bill Woodman's other passion, vintage tractors, is probably a very good idea. ■

Barry Fox

The future; Digital Audio Broadcasting and the future of home music delivery

Every time I write about the DTI being a Mad Hatter's tea party, where as soon as anyone gets a grip on a job they move to another one, someone else who has had to deal with the DTI tells me another madcap story.

This year the record industry's world trade body, the IFPI (International Federation of the Phonographic Industry) held its annual council meeting in London. There was much rejoicing that Britain now has its own minister responsible for recorded music, jolly ex-teacher and tenor singer, Robert Key, Parliamentary Under-Secretary at David Mellor's Department of National Heritage, better known as the Ministry of Fun. Previously responsibility for recording has bounced around the DTI.

John Deacon, Director General of the British Phonographic Industry told how, in his 13 years at the BPI, he has had to deal with 23 different DTI ministers, as well as an endless stream of civil service advisers who move on just as they have begun to understand what Deacon was talking about.

Some years ago he went to one meeting at the DTI where two different ministers turned up. Each had previously thought that he alone was responsible for the record industry. Together they decided which one was not and could leave. Soon after the meeting the Minister who had stayed moved on to another job anyway.

With such a muddle at the top of the DTI it is hardly surprising that the Press Office gets in such a mess. Only a few weeks before Deacon's revelations it had really excelled itself.

The British Patent Office recently made a nice video to explain copyright law and its pitfalls. The BAFTA cinema was hired for a grand launch but the Patent Office was disappointed when only a very few journalists turned up.

As part of the government's competitive policy, the DTI's Press Office handles, and charges for, the Patent Office's press relations.

The screening began at 12 noon on a Thursday. The DTI, who had been talking to the Patent Office about the event for more than a month, mailed out invitations on the Wednesday, asking people to attend 'tomorrow'.

So, at best, people received the invite a few hours before the event. Most, like me, got their invites several days after the event.

For the sake of the record industry, let us hope that the Ministry of Fun is rather more businesslike then the DTI. I had gone to the IFPI meeting, not expecting much news. After the Minister's speech, and speeches from all round the world saying how lucky Britain was to have a minister responsible for recording, it was time for a seminar. Delegates from a thousand record companies in 65 countries sat back and heard Michael Tyler, Vice President of business consultants Booz, Allen & Hamilton, talk about the new technology of digital audio broadcasting. He proceeded to tell the record industry that DAB technology could well rewrite the rules of record retailing, and make the conventional record store a thing of the past.

Unlike previous pie-in-the-sky proposals for electronic home delivery, which rely on technology not yet available, the scheme which BAH has been working up for unnamed clients in the electronics and broadcast industry relies on existing technology and pragmatic simplifications.

The DAB system developed as a Eureka research project works in two stages. First it compresses stereo code, using the technology on which DCC's PASC coding is based. Then it spreads the compressed code between a large number of narrow radio channels. This makes the broadcast signal very robust. The system can squeeze five hi-fi stereo radio channels into each a 1.75MHz slice of radio spectrum.

The original idea was to use dedicated DAB satellites and broadcast conventional radio across countries and Continents. But fearing that satellite frequencies for Pan-European DAB service cannot be cleared internationally until the next century, the BBC is proposing a 20-channel terrestrial service in a 7 MHz slice of the VHF band.

Booz Allen Hamilton is proposing something much more radical.

Several hundred DAB channels will be broadcast simultaneously, by taking over unused TV channels on cable or satellite systems. One TV transponder on the Astra satellite could, for instance, carry 30 or 40 DAB stereo programmes. Astra have so far been unable to sell all its 32 channels for TV use, but is already committed to the launch of two more satellites.

'We may well see an audio Rupert Murdoch.' Michael Tyler told some, by now pretty perplexed, record company executives. 'If you put your heads in the sand you will regret it because whatever you do, your conventional retail channels will be eroded. People will copy what is broadcast. But, on the whole, they would prefer to be legal. The 'Armchair Record Store' gives them the chance.

> 'This means a totally new system of distributing music as an alternative to record sales.'

You still have a chance to mould the DAB concept, and turn disadvantages into opportunities. This is why we are proposing the Armchair Record Store now, on the basis that if you, in the record industry, cannot beat them, then you should join them'.

An ARS system will devote a cluster of DAB programme channels to a cycled repetition of the most popular new albums, with delivery on each channel staggered so that listeners need only wait around half an hour for their choice to come round. Less popular music would cycle on other channel clusters, but with a delay of up to 24 hours between selections.

For 'electronic browsing' more channel clusters will deliver an ever-changing choice of music. But two channels will simultaneously carry the same material, with one channel running half a minute behind the other. This gives customers the chance to listen, and then select music to purchase on impulse. Other channels could be interactive, with customers able to call up a telephone number, and suggest music selections for broadcast.

The 'black box' will be a digital recorder, which uses whatever are the currently-favoured digital disc of tape formats, for example recordable CD, Sony's MiniDisc, or Philips' DCC. The music signal is scrambled to allow preview listening only. Full quality recording is enabled by pressing a BUY button, like the PURCHASE button on a hotel room pay-tv control.

As the recording is made, copyright royalty details are logged from digital identification codes buried in the music signal. This information is either sent down a telephone line or written into a smart card in the recorder. At the same time the user is debited, either by automatic phone connection to a bank, or by deducting credits from a card. A unit similar to a VCR timer would control automatic recording of selected music.

The International Standard Recording Code, which IFPI has already approved as a means of logging conventional broadcast royalty payments, can also be used for ARS logging and to automate the transmission system and cut operating costs. There would be no disc jockeys to pay for chatting between records.

Jay Berman, President of the Record Industry Association of America, says he agrees with BAH's prediction, that DAB will attack the fundamental structure of the recording industry.

Says Berman, 'I would only quarrel with one thing Michael Tyler says, I think he may have underestimated the potential of cable radio.'

Michael Tyler says. 'What I am really saying is that the record companies should get into the cable and satellite delivery business.'

Jay Berman adds, 'This means a totally new system of distributing music, as an alternative to retail sales. But I realise that once a new technology has been set adrift, it is virtually impossible to catch up. The future is not rosy.'

Everyone I have since spoken to, hates the idea of ARS. The truth is that we are all suckers who like to browse in shops and buy records we never knew existed until we chanced on them in a rack.

On second thoughts, perhaps I like the idea of ARS because it will save me a lot of money. ■


All you'll ever need in a tape.
And then some.
GRAND MASTER
AMPEX
456
GRAND MASTER Gold
AMPEX
499

PERSPECTIVE

Martin Polon

CD Packaging may be a closed book to many people but in America it has become a controversial subject. Our US columnist reports

For several years now, the US record industry has been more than aware that its divergence from the 'norm' for CD packaging has attracted the attention of the American environmental lobby. Since the beginning of US distribution and merchandising in the early '80s, the record retail community has favoured the outsized and costly outer packaging called the longbox. This is a compromise designed by those selling music in the mid-'80's to ease the financial pain of adapting the retail fixtures in record stores from the large square package of the LP record. It also provided record retailers with a CD package large enough to deter shoplifting.

Unlike the rest of the world's record sellers, who had no choice but to accept new and different fixtures and better customer 'shrinkage' controls, the American record merchandisers had the Labels packaging to suit their existing tubs and bins. This put off the costly conversion to smaller compact disc fixtures in the mid-to-late '80s. Curiously, the US has continued to go-it-alone with the outsized wrap into the '90s.

From the American record merchandiser's retail perspective, the longbox is an ideal package. It allows the large LP bins to be used with dividers for CDs, thus eliminating new fixture costs. It is obvious in 1992 that through normal attrition, most of the record fixtures in place in the mid-'80s should have been replaced. It seems that these LP bins are virtually indestructible and a sizeable percentage of fixtures dating back to 1985 are still in use. In addition, the longbox has much greater possibilities for graphics and display space than the plastic jewel box — an important factor in impulse buying.

Additionally, from the persepective of many record retailers the longbox keeps the customers, frequently teenagers, from running off with the CDs under their clothes. No issue so endears the longbox to the American retailer as its supposed potential for theft prevention. Crime in US record stores is always cited by record retail experts as the real justification for the longbox. Yet the reality is that in comparison with other kinds of retail operations, the problems found in CD 'shrinkage' usually reflect on the inability of in-store personnel to control the record store space. In fact, one recent study pinpointed the existing tubs as being part of the theft problem. It seems that the very size and placement of the fixtures prevents the staff from fully viewing customers' action! It is also curious that in Canadian music retail locations, the 'shrinkage' rate is favourable with the traditional international shrink-wrapped CD box. And based on the staggering loss estimates for US record retailers, the longbox does not show a remarkable resistance to loss over its smaller cousin! It may be that the success with the smaller package north of the American border may be due to the fact that the Canadian retail locations surveyed generally paid their employers a higher wage and attracted a more dedicated sales force.

Since the late '80s, certain environmentally active recording artists have been aligning themselves with various 'green' movements, demanding socially acceptable conditions for their music — including the use of nontoxic inks and artwork and 'safe' packaging for their albums. On a much more significant scale. Various nonentertainment industry groups have been lobbying the news media about the waste and damage caused by the extensive outer wrapping on CDs. Legislation has actually been introduced in California to 'Ban The Box', although it would seem that the California Legislature might better serve its constituents by overseeing the Los Angeles Police Department rather than the record industry.

A curious competition has sprung up between groups lobbying for and against various packaging options for the CD. First we have the Ban The Box Coalition (BTBC), they have existed since 1990. As its name implies, the group advocates the elimination of the existing paper and plastic CD outer garment. A counter lobbying group, Jewel Box Advocates and Manufacturers (JAM), have existed for nearly as long to defend the plastic jewel box itself. They count among their members the several plastics and chemical companies that supply the polystyrene used in the production of the box. We should mention the National Polystyrene Recyling Company (NPRC), set up by many of the above plastic and chemical groups. They propose to completely recycle existing jewel boxes.

We must also acknowledge the Earth Communications Office (ECO), a Hollywood-based group who have been approached to endorse some of the so-called environmentally sound packaging systems. This brings us to the National Resources Defences Council (NRDC) and Californians

> No issue so endears the longbox to the American retailer as its supposed potential for theft protection

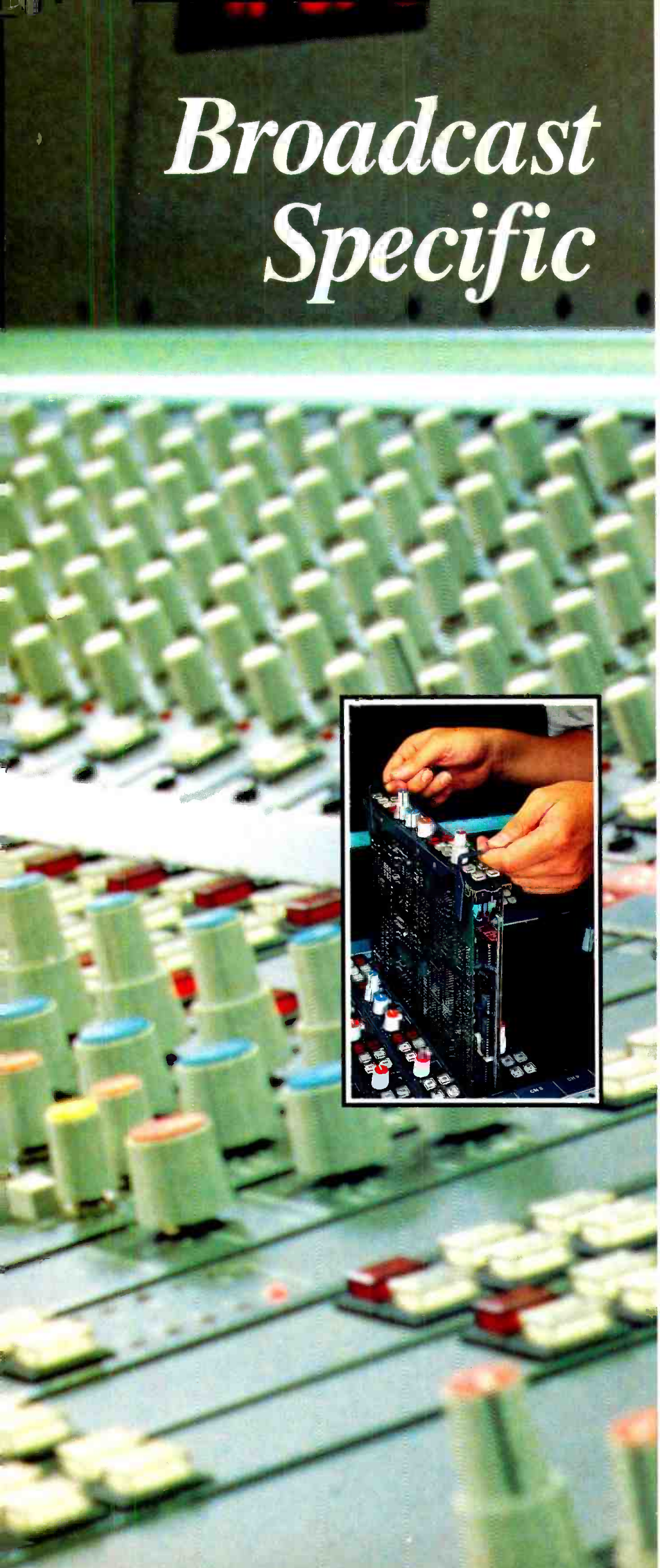
Against Waste (CAW) — these agencies are supposedly allied to the ECO to help draft 'standards' to make the record industry more environmentally responsible. Now add to this list the Entertainment Packaging Council (EPC) which comprises the innovators of the Eco-Pak and DigiTrak paper CD box systems and those suppliers and licensed box makers who will participate in producing the containers. The EPC are rather like the paper equivalent of the plastic supporters in JAM. Then there is the National Association of Record Merchandisers (NARM), a long-term lobbying and support arm of the record retail industry. This group have taken initially as their Holy Grail the retention of some kind of packaging to take advantage of the two-abreast stacking options for CDs in the LP bins. NARM also wanted retention of some similar dimensions (roughly 6 x 12 inches) in whatever new package is agreed upon for theft prevention and to accommodate and better display merchandising art. If you are confused by all of this, do not feel badly. So am I. On detailed inspection, the whole conundrum appears to have nothing so much as the 'spin dynamics' of the American political system in the 1992 presidential campaign.

Among the record labels themselves, the gossip has focused on doing away with the longbox — but the emphasis is on profit rather than the environment. According to an old record-label hand, 'The labels have long concluded that small children — and larger record executives for that matter — would not perish with the demise of the CD longbox. The cost savings inherent in eliminating the oversize CD packaging and the gains in placing more shrink-wrapped jewel boxes in the same-sized shipping cartons used for the longboxes spell increased profitability. Perhaps even more important, would be the efficiency of using the same packaging system and shipping containers as the rest of the world. CD pressing plants worldwide could exchange product to meet demand. All of this is very attractive to the labels. There has been voiced some concern that shrink-wrapped CD packaging would offer the label creative and artistic community. But all-in-all, the response has been affirmative from within the record companies themselves.'

More than a dozen packaging systems for the compact disc have been brought forward for sale within the United States, since the current controversy began. Some of the systems are discussed below. The following are plastic and are based on the current jewel box or are similar to it:

The CD longbox system
The jewel box and a paper and plastic outer-wrap packaging in use today. The shrink-wrapped jewelbox — the format in use today in the rest of the world. (Wrapped in the closed position.) The blister pack jewelbox — packaged open in a tough plastic skin to provide greater stocking dimensions and also favoured by record retailers.

Transition JAM-Pak
A heavy mylar film shrink-wrapped on an open jewel box to enlarge the package, as above. (A ▶



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The success of MBI's dedicated approach is typified by our securing one of the most important recent project contracts awarded to any broadcast console manufacturer by RAI, Italy's national broadcast network. We have already supplied consoles to the BBC at many of their central broadcast facilities, over a dozen U.K. local radio stations and the national networks of several European and other countries worldwide, many of them as part of a total MBI turnkey project.

Series 30

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H A Harman International Company

proposal from the JAM grouping.)

JAM-Pak
A heavy mylar film, shrink-wrapped on a closed jewel box. (An ultimate product, environmentally desirable assuming polystyrene recycling and the use of a recycled and recyclable mylar wrap.)

Inch Pack
Formerly known as the Stack Pack, a jewel-like box with a sliding drawer for easy access and interlocking capability similar to children's play blocks. (Could be sold wrapped open.)

Laserfile
A stacking plastic case similar to the jewel box that has a sliding, downward-hinging drawer and is to be sold wrapped in the open position.

The following CD packaging systems are based on the usage of cellulose-based paper and cardboard (and-or) plastic wrapping, much of it, if not all, recycled:

Eco-Pak
A folding card package, wrapped 'open' around a plastic CD tray. The pack will fold down to standard CD box dimensions for storage after purchase. It is supported by the Warner Music Group.

DigiTrak
A multipanel paperboard and plastic package shrunk open and stiffened with disposable plastic rails. Can be folded into a CD box with wraparound closure.

DigiPak
The predecessor of the DigiTrak, used for CD Maxi-Singles.

C-Case
An all recycled paperboard product with a pocket for both the CD and a companion booklet. The package uses a 'belly band' which can double as a business reply card to stiffen the 'open' shrink-wrapped pack.

Now, the bottom line here depends upon whose bottom line you want, if you talk to the record labels, you find that they have more or less decided to use the existing shrink-wrapped jewel box. In fact, through their trade association the Record Industry Association of America (RIAA), the six major labels in the US have committed themselves to a closed format with the 5 x 5½ inch dimensions of the original 'red book' jewel box. The longbox is dead: long live the jewel box.

By April of 1993, in theory, new releases of music will be available in the smaller format. Since BMG-RCA, Capital-EMI, MCA-Matsushita, Philips-Polydor, Sony-Columbia and Time-Warner-Elektra-Atlantic sell more than 90% of all music retailed in America, their agreement is nigh well binding. It may be that cardboard packaging is used in the closed position for certain projects on certain labels, but the vote is certainly in. The ecology lobby is reasonably pleased since the despised longbox packaging has been discarded once and for all. The jewel box itself can be recycled, as can the mylar film used for shrink-wrapping. All of which bodes well for the environment. Pressing plants and jewel box makers can breathe a sigh of relief since their expensive machinery will not be made obsolete. That leaves the record retailers. The wrath of this group has yet to be felt, but the record companies have made vague noises about some kind of assistance. It is interesting a note that all of the systems preoffered as jewel box replacements were designed as both 'open' and 'closed' systems. With those options eliminated, the record retail community are going to have to bite the bullet and install up-to-date fixtures for shrink-wrapped CDs. They can certainly look to Europe and the UK for a lead on how to do that, but whether they will actually do so remains the multimillion dollar question for the US record industry. And the following coda really does apply here: if the record producing, distributing and retailing communities had worked *together* to find a solution to the CD box problem in the US, the thousands of trees cut down to make paper for all of the reports, articles, groups, lobbying units could have been saved! ■

The longbox is dead: long live the jewel box

TALKING TAPE

A comparative review of the current and previous generations of analogue recording tapes by Sam Wise

Unexpectedly, analogue recording continues to thrive, bolstered by recession-hit capital purchasing budgets, and the limited success which digital editing systems have had. The impact in recorded quality provided by Dolby *S* and *SR* (see *Studio Sound*, August 1992) have also threatened the early supremacy of digital recording formats, since analogue can be made to come close in performance to 16-bit digital. But what of the latest analogue tape product launches by 3M (Scotch)-*996* and Ampex-*499 Grand Master Gold*? Are there audible performance benefits? How do these new products compare with each other and with the previous generation of tape sold by 3M, Ampex and BASF? Are they suitable for all applications? These are the questions which we hope to answer in this review.

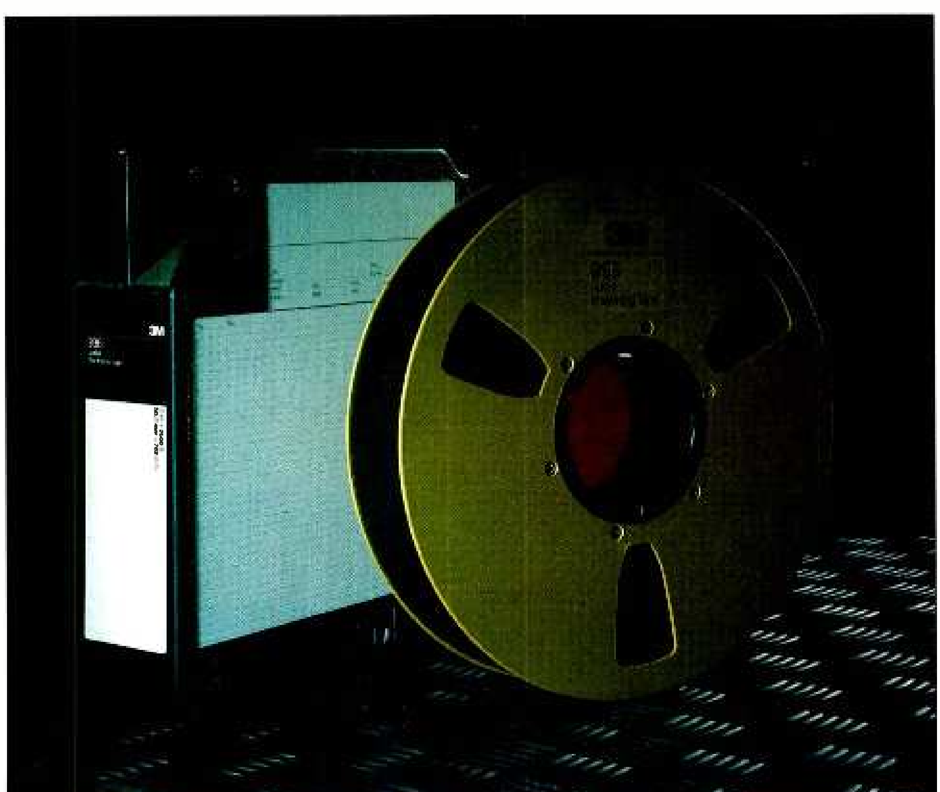
The test machine

Since we are doing a review of both the Tascam *238S* 8-track cassette machine, and the *MSR-24S* 1-inch 24-track recorder, it seemed not too bold to ask them to loan me a good quality ¼-inch stereo machine from their range to use for tape testing. One requirement was relative ease of access to the machine's setup controls, since each tape will require bias, level and HF adjustments prior to testing. This is reasonable on the *BR-20*, since access is under the front control panel which remains connected for easy operation while adjustments were made.

A limitation on the *BR-20* is in the bias adjustment, which only just achieved the required level for 3M *996*. Otherwise things went well using this machine. The results of the tests will therefore reflect the performance of the machine as well, and may differ when used on other recorders. However, the results we give are comparative, and should remain so on any quality recorder.

The heads and guides were thoroughly de-magnetised and cleaned before each tape was tested. Head azimuth was checked only at the beginning of the measurement session, but head mounts seem stable and therefore unlikely to affect measurements.

The evenness of tape wind is affected by the quality of the transport mechanism, its state of repair (in this case almost new), the speed of wind, and the properties of the tape material and its back coating. A comparison of *BR-20* wind quality with other machines was not possible, so comments on evenness of wind are entirely subjective and may only relate to this machine.



Packaging

Like other products, tape is now subject to packaging enhancements, sometimes adding actual value to the products, sometimes just intended to indicate that they are something special (or just different).

3M tapes type *996* and *226* came in moulded plastic boxes, retaining the tape reel not only by the centre, but also by the edges. This helps to prevent the tape end from coming loose in the box and getting damaged. Two plastic clips hold the box shut, reducing the risk of the tape falling out if the box is accidentally dropped. Inside are self-adhesive labels with space for marking all details of the recording. These can be easily attached to smooth areas on the outside of the box, including the edge — visible labelling when the tape is shelved.

The *996* spool is gold anodised, with a machined, well finished moulded-plastic centre hub. Six self-tapping screws on each side secure the flanges to the hub. The hub was a good fit to the NAB hub on the recorder, much less loose on the machine hubs than typical tape-spool centres. This will improve wow and flutter on some machines due to a reduction in movement of the reel on the recorder's platen. The *226* spool was a more conventional silver anodised finish with injection moulded hub and three bolt-though fixings. This fits in the more normal relatively sloppy fashion onto the machine. Ampex tapes types *499* (*Grand Master Gold*), *456* (*Grand Master*) and *478* (*Low Print*) all arrived in conventional but robust cardboard boxes having a centre locating ring and a hinged edge. The spools for these are of conventional construction like the 3M *226*. The *499* has grey anodising, while the others are silver. There is writing space on the back of the box, but not on the edge. BASF *911* (*Studio Master*) was provided on a flangeless hub — an accident of communication with one of their staff. This was fitted with flanges for test purposes.

The box is of similar construction to those provided by Ampex. All tapes arrived with flat reel flanges. I do not remember that in my studio days.

Tape handling

Unexpectedly (for me at least), there was a radical difference between the handling characteristics of the tapes. They seemed to fall into two groups — those which wound evenly at high speed (BASF *911* and Ampex *478*) and those which weaved at high speed, creating a pack which might easily get ▶

damaged (both 3M tapes and Ampex *499* and *456*).

Performance tests

A standard recording level approximating to 520nWb/m was used for all relative measurements. This is the standard recording level presently used within the BBC (representing PPM 6), and is also widely used in independent recording and broadcasting studios. The absolute accuracy of this is somewhat prone to interpretation, but it will be within 1dB of 520 nWb/m as measured according to most any local standard. Between each tape type, the bias was adjusted according to manufacturer's recommendations. The recorded level was then matched within 0.2dB at 900Hz, and the HF at 16kHz set to within 0.2dB of the 900Hz signal using an input signal level 20dB below the reference level. While setting up the bias and level calibration for these tapes, it became very clear that they are not very compatible if the best results are desired. Bias levels varied considerably, as did the required input drive. The average studio chooses a tape, gets the machines lined up for it, then purchases only that tape. In broadcast, things can be different. Organisations like the BBC prefer to hedge their bets, and the volume of tape they purchase is enormous. Therefore, it is desirable for them to get a minimum of two suppliers on line who can supply tape which is near enough a match. This prevents the potential problem of having to simultaneously realign hundreds of tape machines. Ampex *478* is one such tape, designed not so much for ultimate performance as for consistency with a standard.

The following results cannot be regarded as having absolute accuracy, since tape measurements are notoriously difficult, time consuming and dependent upon accurate machine performance and alignment. However, they can be relied upon to accurately depict differences between the tapes under test.

Bias noise

Bias noise is the noise generated by the tape on playback due to recording bias only, that is, there is no signal input. In this area the tapes are very similar, ranging only over about 2dB. Measurement error can be expected to be about ±0.5dB on this test. See **Table 1**.

Maximum operating level is found at 900Hz or 1kHz by inputting a signal which will produce a third harmonic distortion level of 3%. **Table 2** shows the results of these measurements.

In this comparison it is clear that Ampex *499* has the best dynamic range, with 3M *996* close behind. These are both of the new generation of tapes, having about a 3dB advantage over the previous generation — represented by 3M *226*, Ampex *456* and BASF *911*. In their data sheets Ampex specify a signal-to-noise ratio of 81.5dB for *499* compared to the 3M specification of 79.5dB, giving Ampex a 2dB superiority. While I have not been able to duplicate their values, my measured difference of 1.4dB in favour of Ampex confirms their relative performance. **Fig.1** gives comparative maximum recording level versus frequency graphs for all six tape types, in this case indicating the 3% THD level at each ⅓-octave band centre. Clearly some tapes will tolerate more level at 900Hz, confirming **Table 2**. Note that the relative high-frequency performance also varies between tape types.

HF saturation level

According to IEC94 Part 5, using a signal frequency of 10kHz, saturation level is defined as the input recording level at which no further increase in recorded level occurs. In addition, it is interesting to note the level at which compression on tape begins to occur. For the purposes of this review, the onset of compression is defined as ▶

TABLE 1

Tape type	22Hz to 22kHz RMS	'A' weighted, RMS	CCIR 468-3 wtd, RMS
3M *996*	-61 dB	-65.7dB	-53dB
Ampex *499*	-62.2 dB	-66.7dB	-53.7dB
3M *226*	-61.6dB	-66.1dB	-53.2dB
Ampex *456*	-60.9dB	-65.4dB	-53.7dB
BASF *911*	-60.3dB	-64.7dB	-51.8dB
Ampex *478*	-62.3dB	-66.8dB	-54dB

Table 1: Bias Noise in dB below 520nWb/m

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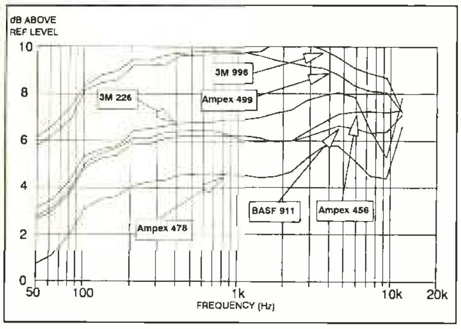

Fig.1: Maximum operating level with 3% THD

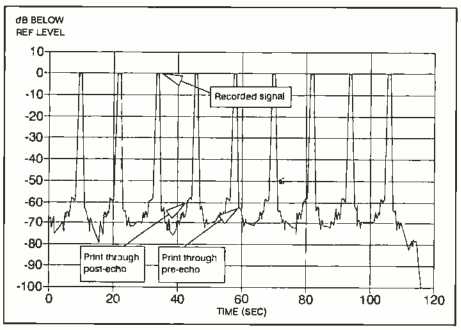

Fig.2: Print through test on Ampex *478*. One wrap of tap is recorded with a signal at reference level, the tape is stored for a period without rewinding — in this case 48 hours. The pre and postecho levels are measured. These are reversed in the graph since the tape is running backwards

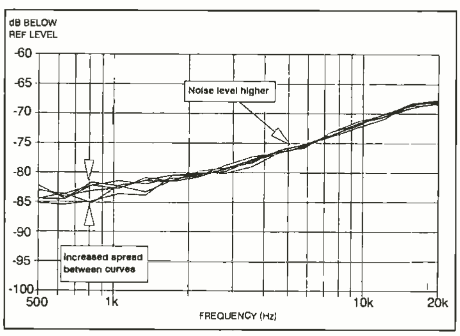

Fig.3: BASF *911* - curves of noise generated by 60Hz stimulus tone at six levels from -20 to -80db, re 520nWb/m

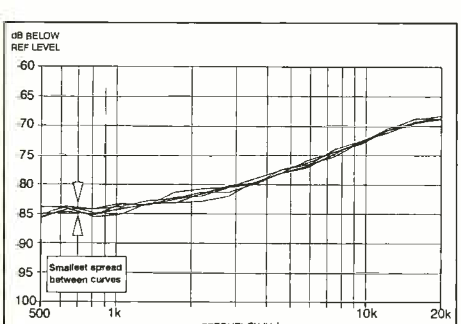

Fig.4 Modulation noise of 3M *996* tape

the point where 0.2dB of compression has occurred. Levels are referred to the reference level of 520nWb/m. Both of these measurements are given in **Table 3**.

Print through

While it may not be too important in many music applications, good print through performance is vital for speech. Print occurs when a recorded layer of tape induces its signal onto an adjacent layer, producing an echo effect. If this echo comes before the wanted recording, it is called pre-echo and is particularly disturbing. Print was measured on these tapes by two techniques. The first involves recording approximately one turn of the tape with a nominal full level signal — in this case 520nWb/m. This is played back and the resulting levels before, during and after the recorded turn are measured.

The second method is similar but opposite. In this case a blank section of tape is recorded within an area where adjacent tape loops are recorded to the reference level. This is a worst case since both pre and postecho occur together.

Fig. 2 shows a typical pre and postecho measurement. The print through test results are given in **Table 4**. Levels are given in dB below reference level. Here the clear winner is Ampex *478*, which is specifically designed for low print performance. Next are 3M *996* and BASF *911* which are worse by about 3dB. The other tapes are similar in performance, giving a further deterioration in print-through level of about 2dB.

Modulation noise

Modulation noise was very evident 20 years ago, providing one of the most aurally irritating sources of background noise since it altered with recorded signal level. With pure sounds like solo flute, the results were nearly unbearable —particularly on some types of tape machine. Modern tapes are much better.

Two methods of measurement were used, the first is similar to that which we use to assess the modulation noise characteristics of A-D and D-A convertors. A low frequency tone is recorded at several signal levels typically reducing by 5 or 10dB each time. In our case, a 60Hz tone is used, with levels dropping from -20dB to -80dB in six steps. On playback, the signal is passed through a ⅓-octave band-pass filter and further reduced by a high-pass filter at 400Hz, leaving only noise products evident. A graph is produced showing the noise under the filter as it slides from 500Hz to 20kHz. A spread of level between the graphs gives an idea of modulation noise. The greater the spread, the more the noise alters with recorded level and the more audible it will be.

In this test similar results were obtained with all tapes except BASF *911*. This result is shown in **Fig. 2** and has an increased overall noise level averaging -82.5dB, and the worst spread ▶

TABLE 2

Tape type	MOL -Maximum Operating Level	Bias Noise 'A' wtd from Table 1	Dynamic Range = MOL - Bias Noise
3M *996*	9.4dB	-65.7dB	75.1dB
Ampex *499*	9.8dB	-66.7dB	76.5dB
3M *226*	6.9dB	-66.1dB	73.0dB
Ampex *456*	6.2dB	-65.4dB	71.6dB
BASF *911*	6.3 dB	-64.7dB	71dB
Ampex *478*	4.5dB	-66.8dB	71.3dB

Table 2: Bias noise and dynamic range or signal-to-noise ratio

TABLE 3

Tape type	SOL - HF saturation Level	Onset of compression
3M *996*	+9.3 dB	0dB
Ampex *499*	+8.8dB	0dB
3M *226*	+8.3dB	-1dB
Ampex *456*	+7.9dB	-1 db
BASF *911*	+7.9 dB	-2dB
Ampex *478*	+7.7 dB	-3dB

Table 3: HF saturation level and onset of compression relative to the reference level of 520nWb/m

TABLE 4

Tape type	Pre-echo	Post-echo	Both
3M *996*	-58.5 db	-55dB	-50.5db
Ampex *499*	-57dB	-53.5dB	-49db
3M *226*	-56.5dB	-53dB	-49dB
Ampex *456*	-57dB	-53.5dB	-49.5db
BASF *911*	-59 dB	-55dB	-51.5dB
Ampex *478*	-62dB	-58dB	-54.5dB

Table 4: Print-through measurements in dB below 520nWb/m reference level

TABLE 5

Tape type	Scrape Flutter	Maximum Level Variation (2 minutes)
3M *996*	0.08%	0.34dB
Ampex *499*	0.09%	0.6dB
3M *226*	0.06%	0.58dB
Ampex *456*	0.075%	0.34dB
BASF *911*	0.12%	0.25db
Ampex *478*	0.12%	0.48dB

Table 5: Scrape flutter and two-minute maximum level variation

between curves, being about 4dB at 500Hz. The lowest spread is shown in **Fig. 3**, from 3M *996*, being about 2dB at 500Hz, while Ampex *478* gives marginally the lowest overall level of noise as shown in **Fig. 4**.

The second method of measurement uses FFT techniques, giving an analysis of the noise which is added around the recorded tone, in this case 2kHz. **Fig. 5** shows the results from Ampex *499*, which are almost identical to 3M *996*, Ampex *456* and *478*. The overall noise level is low and the spreading of the skirt at 2kHz is minimal. Compare this to **Fig. 6** for BASF *911* which give marginally the highest average noise level, and **Fig. 7** for 3M *226* which has low average levels, but a very much broadened noise skirt about 2kHz.

Scrape flutter

Scrape flutter is a measure of both tape transport performance and the smoothness of the tape itself, giving another indication of noise produced due to tape action, in this case due in part to the longitudinal vibration of the tape material as it passes over the heads. In this test 3M *226* gave the best performance at 0.06%. Other results are shown in **Table 5**.

Instantaneous level variation

Tape dropout occurs when there is muck on the tape, lifting it slightly off of the recording or playback head, or when there is a deficiency in the tape's magnetic coating. In order to evaluate dropout a section of tape was recorded for 2m, then played back while sampling the tape's output level at 30ms intervals.

Over this period, none of the tapes gave what could be called a dropout, since no audible loss of signal could be detected. However, all tapes gave some kind of level variation. The figures given in **Table 5** indicate the maximum peak-to-peak signal level variation over the two-minute period. Two of the curves are of particular interest, both shown in **Fig. 9**. 3M *996* has a cyclic level variation which must be due to the physical tape coating or a slight weave introduced by tape slitting. Ampex *499* shows several distinct level drops. None of this is likely to be audible, but does give an indication of variations in tape manufacture.

Summary

The newer tapes, 3M *996* and Ampex *499* do give improvements in recordable dynamic range, with Ampex *499* giving the best overall result. At HF however, 3M *996* scores more highly, while also having better print characteristics.

If smoothness of high speed wind and print through are the most important criteria, then Ampex *478* must be the choice.

For a slightly lower price, either 3M *226*, Ampex *456* or BASF *911* offer good value for general recording use, with 3M *226* being marginally ahead on dynamic range.

When used with Dolby *S* or *SR*, an audible improvement in apparent dynamic range of between 15 and 20dB will be achieved, giving the high dynamic range tapes an effective signal-to-noise ratio of 90 to 95dB — very similar to 16-bit digital. The Dolby systems also are very effective at reducing distortion on tape, though not matching digital performance. But when overload occurs, it is gradual, unlike the brick wall effect of digital systems. One remaining limitation is wow and flutter — this digital has cured forever. Has anyone got a really good tape transport to close this final gap? ■

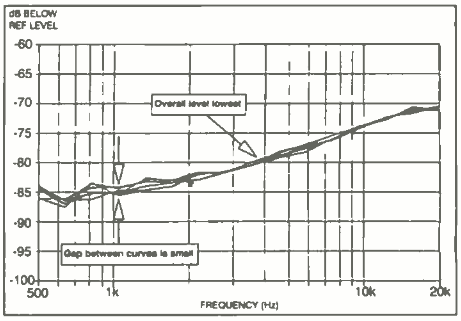

Fig.5: Modulation noise of Ampex *478* tape

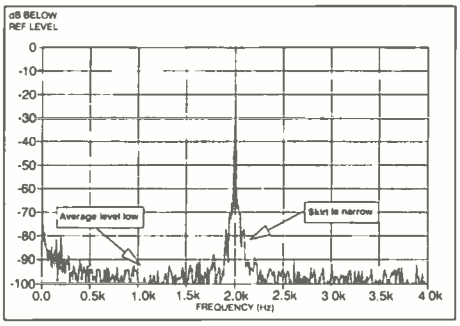

Fig.6: Modulation noise spectrum of Ampex *499* tape – 3M *996* is similar

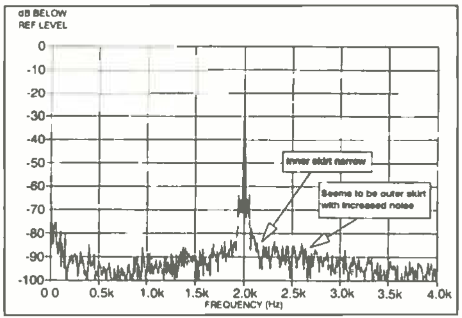

Fig.7: Modulation noise spectrum of BASF *911* tape

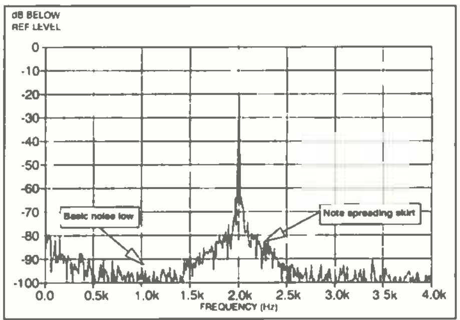

Fig.8: Modulation noise spectrum of 3M *226* tape

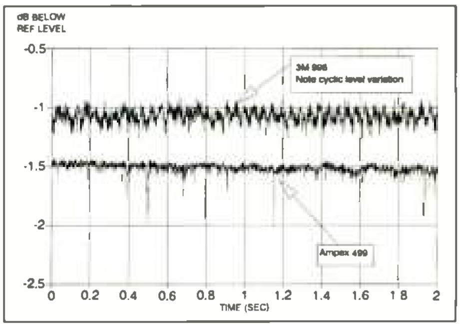

Fig.9: A constant 1kHz level at reference level is recorded on tape. The instantaneous playback level is stored and measured for level variation. Note that the range of level variation is small

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Keith Spencer-Allen on miking economy

Recently I found some notes I had made years ago on mic techniques which worked well enough to make me note the details. Many of these related to techniques for miking largish numbers of musicians which were developed in one particular studio but as soon as I realised they did not always work equally in other rooms I stopped making them. At that time there was a great deal of interest in making a dozen string players sound like two dozen, four brass sound like ten and so on. Saving money (or making the most of a limited budget) was the producer's aim and while sometimes doubling was used, frequently they would turn to the engineer for help. Doubling orchestral parts is not always very satisfying because the two sections do not really gel and it still sounds like two sections. In stereo there may be more spread because there are different signals on the left and right of the picture, but the effect is rather shallow. If you flip the stereo image on the double it frequently becomes confusing. There is definitely something superior about events that happen simultaneously and blend in the air.

Returning to our budget-ridden sessions — the only solution was multiple miking. As many mics as possible had to be prepared prior to the session which involved knowing the exact line-up.

The key was to try to find sounds within the room that were discrete enough in character that the ear was fooled into thinking these were not from the same source. There are three factors that can 'fool' the ear — tonal differences, attack time differences and time delay.

The first move had to be a close mic. You needed a signal as close as possible without any room tone — but that, of course, depended on the instrument. It may even be possible to have two close mics if the instrument generated significantly different sounds over its body. For example, a large stringed instrument such as a double bass (or sometimes even a cello) can generate several sounds over its length. Quite different tonality is found at the bridge, over the bowing area, in the region of the f-hole and very occasionally from the neck. The differences in attack and tone between the bowed area and the f-hole are a good starting point in miking-for-economics. If you are generating a stereo track or several tracks, the signals from each instrument should be positioned slightly apart from other to emphasise any differences between them.

In a brass section it is not quite so easy. For most brass instruments there is really only the sound eminating from the horn that is usable. The use of the clip on miniature mics on the bell of the horn is probably the only hope of a very close sound.

Next comes the mic in the nearfield. Being at a distance of three to five feet there is a more natural tonality and an increasing amount of room tone. We also have the blending of sound from the other instruments; this mic would most likely be aimed at covering a desk in a string section or at least an identifiable musical part.

There then come the distant mics. There is the tendency to want to make these sound impressive in their own right but in this mic technique this is not ideal. They have to be heard as part of the composite as they offer the most difference in tonality and time to the closer signals. The studio that this technique was developed in had a lowish ceiling of nine to ten feet quite absorbent surfaces. So a mic around 20 feet away had a harmonically integrated sound, a slight delay and in this studio, a quite dry character. It was essential to have at least two distant mics so they could be hard panned or the stereo picture would be drawn together.

The key to success was in the balancing. Judiscious use of EQ, level and panning could lead to quite large-sounding results. EQ is used to create slightly different shades on closely-related signals. This has to be subtle — just a few dB is quite sufficent. It is the intention to gently emphasise what little differences there might be between mics. Panning is used to create a positional spread between mics on the same instrument with other instruments overlapping in their positioning.

There may be a need to compromise mixing in this way to keep balance between the parts and the original orchestration relevant.

If you built these mics up into stereo pictures it was best to try and keep as many of them on separate tracks as possible. Although the combined effect worked well behind an existing track, there was no way of predicting the result if it had to become too exposed. The ability to revert to only the nearfield mics was essential.

Such techniques are solely for a specific economic requirement and are not recommended for their excellence. More their necessity .

You must listen carefully for significant phase problems between the close mics. Again, as I mentioned earlier, their effectiveness also varies from room to room and your ability to talk the musicians into co-operation.

At the other extreme I also came across notes from a lecture that Bruce Swedien gave during the 87th AES Convention, New York in '89. Swedien has had a very long and distinguished engineering career but perhaps he is better known these days as engineer for Quincy Jones and Michael Jackson.

Swedien has a quite unique approach to many recording situations and we may return to his techniques in the future. An essential part of his philosophy is to capture aspects of the sound in the room to give it a sonic character. With this in mind it is worth looking at his thoughts on backing vocal tracks.

If the track is going to be doubled, he will move the singers close in on a coincident pair of mics. For the second track the mics will remain in position and the singers will move. The left and right will remain in the same place — not flipped. He ends up with a stereo signal that has depth. By not flipping the double he retains the room character which is reinforced by the mics remaining static and adds to his concept of sonic personality.

He took this further on a Quincy Jones track, *Places You Find Love,* where aside from the main choral parts there are answering choirs on the left and right. The technique was to use a spaced pair in the room but place the choir just on one side of the room for the first track. The mics stayed and the choir moved around the room to the other side for the answer part.

These stereo pairs are then spread hard left and right in the mix giving the choirs a real location in the stereo picture — not just a panned image. And it works. You use a lot a tracks though — something that does not worry Mr Swedien.

In utter contrast to my opening points I recently came across the term 'Straight Orchestra' meaning an orchestra 'so constituted and so orchestrated that it produces a balanced sound without the aid of microphones'. Anyone worked with one of those lately?

And finally, Mr Swedien extends his thoughts on microphones far wider than just where to stick them. He has a totally integrated approach to sound recording and the role of the microphone within that.

'A microphone is a modern symbol of the human urge to capture a bit of the living world and examine it later at our leisure.' Bruce Swedien, New York, 1989. ■

Illustration: Carl Flint

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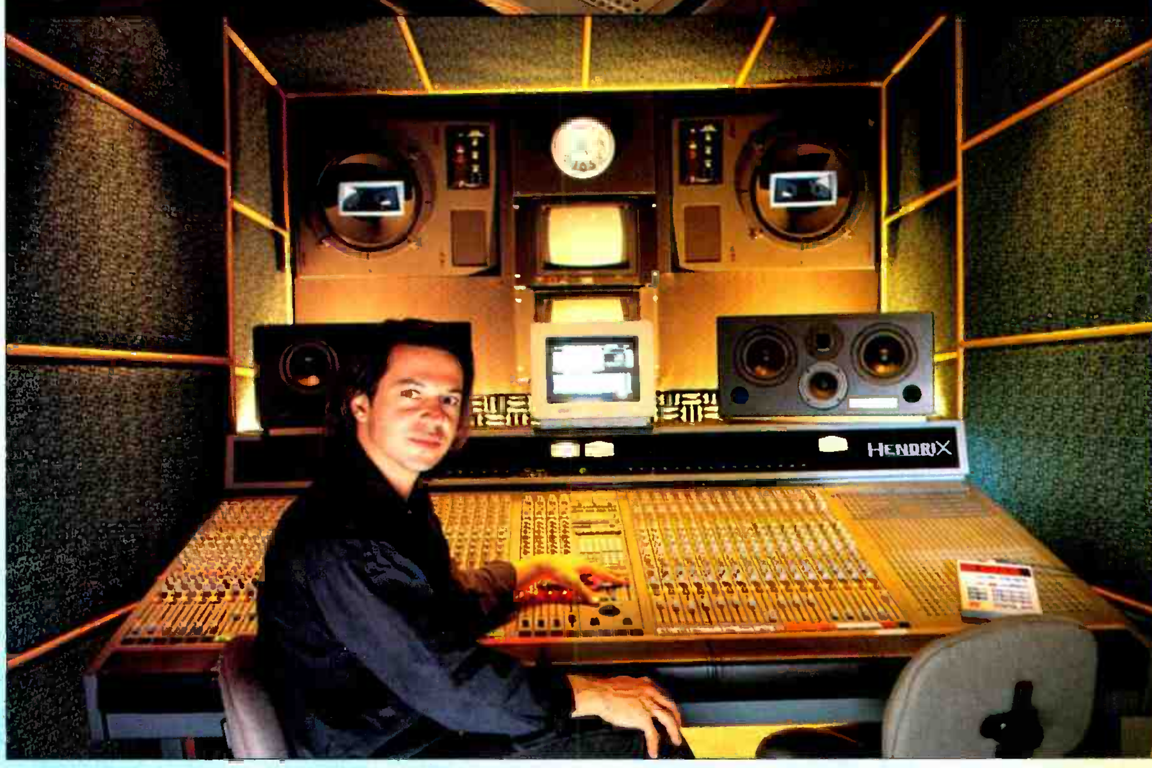

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