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A LINK HOUSE MAGAZINE

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STUDIO SOUND
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BBC Radiophonic Workshop, Maida Vale, UK

STUDIO SOUND
AND BROADCAST ENGINEERING

February 1990
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Compose yourself

Back in 1977 when Bob Marley rallied everybody to join a global jam in his song *We're Jamming*, he can't have realised that a few years later the word jam would most likely imply having a meaningful interaction with your sequencing software in the comfort of your spare bedroom. Of course in the '70s the process of composition would most likely entail hiring a cottage in a remote area of the country and pouring out your soul to a 4-track. Life has certainly become easier with the '80s MIDI revolution, but easier doesn't necessarily mean better music. The most common charge levelled against the current 'pops' seems to be one of sameness, the same sounds used with the same rhythms by the same groups. The justification is that the records are being bought in millions, perhaps by the same people.

So is it MIDI's fault that today's chart seems like one long line of segues? Of course it's not that simple but it does seem to be a little more than coincidence that the growth of the home MIDI studio has mirrored the demise of the singles market. A lot of the music at the moment is self-regenerating, a better than average track is released which spawns half-hearted clones. The whole thing turns into a merry-go-round of demented sampling without an original melody in sight. The line between the titles 'composer' and 'programmer', however, has never been more vague. Everybody is making music through the availability of cheap MIDI equipment whether they're composers, programmers, engineers, tape-ops or A&R personnel. And why not? It's a lot of fun. But where do you start when you sit down at your computer screen, where do you take your inspiration from? For a lot of people that's already taken care of through sampling, and so a clone is born. For others the only remaining source must be the radio, but unfortunately that's where programme controllers are concentrating on giving the public what they think they want not what they need. And so the merry-go-round keeps on turning.

To a certain extent composition originated from a home MIDI studio is slave to the sounds available. That situation is further muddied by trends when the use of a piece of equipment becomes a pre-requisite to the track, but in the music industry that's nothing new. The popular practice of MIDI-retrofitting old analogue synths is just a variation on a theme but does go some way to showing the extent of people's frustration with the same old sounds.

You can't disinvent something, especially something as important in certain areas as MIDI. For example the article in this issue on the BBC's Radiophonic Workshop describes one of the most advanced MIDI studios in the world, advanced in its use of MIDI as a control language, MIDI's use in automation and on stage is also now standard practice and rightly so. But for a lot of people MIDI has become a bandwagon. Computers have taken the place of the old six string as far as writing is concerned, and as the games market can testify MIDI has produced its own fair share of computer widows. Are there really writers who, having invested in a MIDI studio, will close the door on the world and their only source of inspiration and have a quiet night in with the software? Surely there's something missing! What about asking a few musician friends round to swap musical ideas before you step into the studio. Perhaps that's just another form of sampling but it has more chance of being original than reaching for the CD rack.

Although the future for the use of MIDI is unquestionably secure, the novelty will soon wear off, and the record buying public will soon tire of hearing music that is on the fringe of supermarket musak. Indeed there is already evidence of this in the sales of 'World Music' with its wealth of new acoustic sounds and rhythms.

The music you grow up with is as important in keeping memories as your photo albums, the songs can be called the music of your life and frequently are in the golden oldie radio shows. Those songs trigger off memories of what you were like and what you were doing when you heard them. I can't help thinking that in years to come when we're trying to remember what was going on in our lives at this time that we wish we had taken more photographs. **Julian Mitchell**

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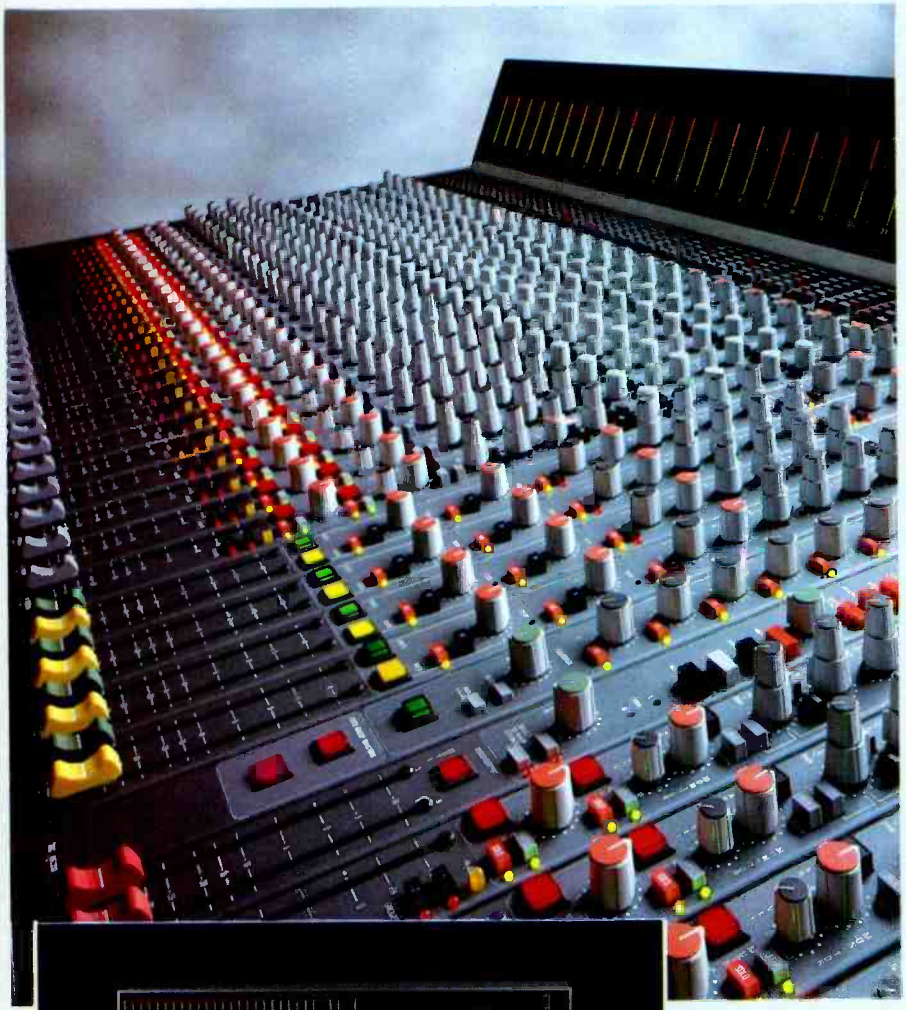

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SOLO/PFL SYSTEM: MAGNUM has a Master Status controlled in-place solo switch on both channel and monitor paths. When de-activated, solo switches provide a full Pfl facility. MAGNUM also has two independent Mute groups.

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Six years ago, Thatched Cottage Audio was created with the idea of specialising in eight and sixteen track recording systems. From the very first, service and value were our primary concerns; for the purchasers of our multitrack systems we provided a standard of back-up previously only associated with large scale professional installations. We were open every day of the year, we opened a recording school to offer free courses within recording packages; we started a service department specialising in the equipment we sold, and we offered free loan items if we encountered problems we couldn't solve speedily.

Not surprisingly, we soon came to dominate our market, becoming . . . the largest pro-audio supplier with nearly all the companies we dealt with (Alesis, Korg, Drawmer, Casio, Fostex, Tascam, Yamaha, Studiomaster and a good few more). This had two knock-on effects; firstly, our customers received (if possible) an even better service – if we were unable to help quickly the manfacturer could; secondly, we were in certain instances, able to purchase 2 items at discount rates giving savings which we passed on to our customers.

This combination of service plus value proved unbeatable, and last year we sold nearly SIX HUNDRED eight and sixteen track systems. During this period though, two significant things happened. Many of our customers began asking if we could supply more advanced systems – budget 24 tracks for producers and artists, and full-blown systems for large installations. Unfortunately there seemed to be a huge price gap between sixteen and twenty-four track (in any case, top end multitrack was already handled very capably by existing companies).

At the same time though, technology came to our assistance! Many of the companies we dealt with who were known for budget multitrack suddenly start launching leading edge technology items at previously unheard of prices. For the small professional studio, or producer/artist, Tascam developed the cheapest 24 track in history, Allen & Heath at the same time designing a medium price, full feature MIDI console with 24 track compatibility. For more advanced systems the Yamaha digital multitrack, the Fostex timecode R-DAT and the Digidesign hard disc recorder were all examples of high tech initiatives from a new breed of companies.

We could not only fulfill the demand for this high quality professional but affordable equipment – in many ways it was a logical extension of what we already did (no other company had the experience we had with regards to the manufacturers producing the new product ranges). We realised however, that to capitalise on this advantage we would have to rival the facilities of the most up-market retailers.

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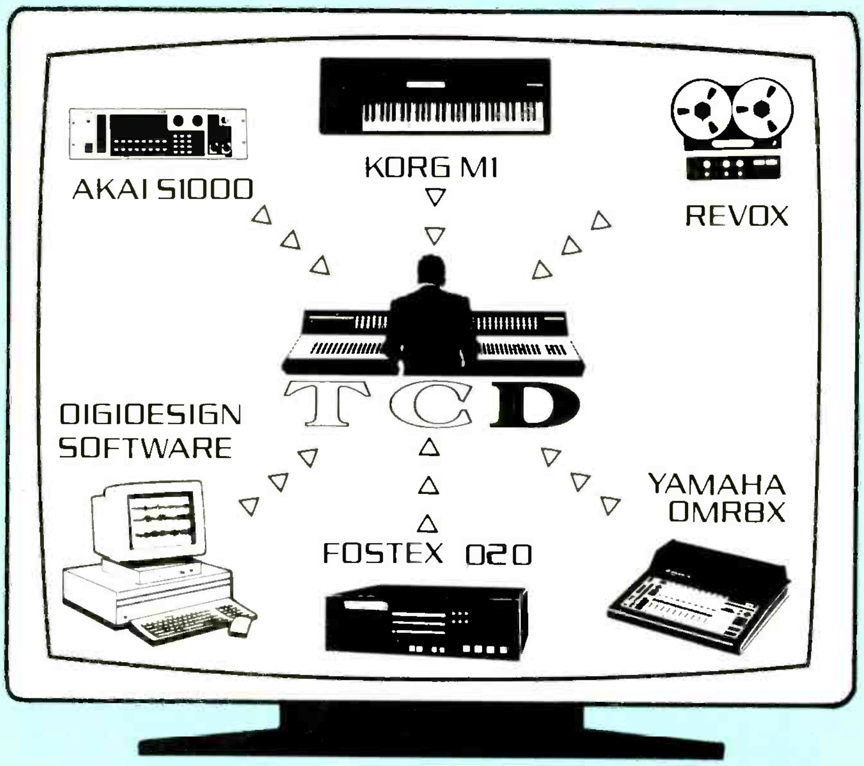

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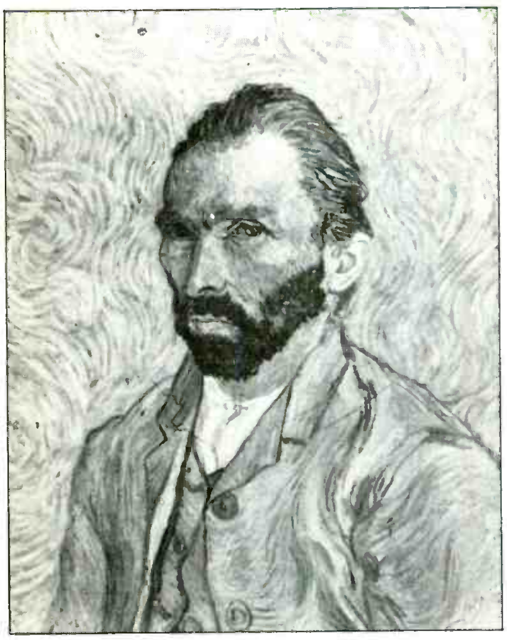

Vincent van Gogh - self portrait

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Sony Broadcast & Communications

New APRS chairman

APRS, the UK professional recording association, has announced that its next chairman is to be Rodger Bain, director of CBS Recording Studios, with Dave Harries, director of Air Studios, serving as deputy chairman. Ken Townsend, general manager of EMI Abbey Road Studios, who retired from the chairmanship after the usual 3-year term, remains a director of the Association.

First elected to the APRS Board in 1985, Bain has served on both the membership committee and the marketing committee. He is also an honorary member of the British Record Producers Guild.

In brief

- Ipswich, UK: Loudspeaker chassis and enclosure manufacturer **Celestion International** have recently announced expansion. The production workforce has grown by 53% during the past year to cope with the increased demand generated by an expanded R&D programme.
- California, USA: Manufacturers **Electro Sound** are expanding their cassette duplication business in line with the boom in the industry. A number of facilities throughout the world have upgraded with 80:1 duplication speed equipment thus increasing capacity by 25% to 30%.
- Hatfield, UK: The **Mitsubishi Pro-Audio Group** have re-introduced Diamond Leasing, a financing scheme designed for recording studios and broadcast facilities.
- London, UK: **Nomis Studios** have signed a special publishing and production contract with Warner Chappell Music. Nomis will now handle four to five recording and production projects a year for new artists signed to Warner Chappell.
- Miami, USA: The Music Engineering Program at the **University of Miami** have announced that again this year its graduating class has achieved 100% placement in the audio industry.
- Milton Keynes, UK: **BSI** have announced the publication of *BS 6288: Part 10: Magnetic tape sound recording and reproducing systems: Part 10: specification for time and address codes*. It specifies requirements for time and address codes for 6.30 mm twin-track magnetic tape and recording/reproducing equipment for professional use. Copies of this standard from BSI Sales, Linford Wood, Milton Keynes MK14 6LE.
- Basingstoke, UK: The Advanced Developments Department of **Sony Broadcast & Communications** and Reading University's Department of Cybernetics are to collaborate on a new undergraduate sponsorship scheme. Under the terms of the agreement, SBC's Advanced Developments Department will each year offer sponsorship over three years to a first year student on the BSc honours degree course in 'Information Technology'.

News from the AES

By the time you read this we will have started a new year and a new decade. We hope this will be good to the audio industry and those who work in it. May we take this opportunity of wishing you all a happy and prosperous time ahead.

In the UK, as in most other countries, we hold regular monthly meetings covering a wide range of subjects, members and visitors are most welcome to attend these events. To help future planning the dates, speakers and titles are listed below (more details will be available on each nearer the time).

1990

February 13th
Digital Audio in the TV Studio
Paul Evans

March 20th
To be announced

April 10th
Amplifier Differences
Paul Miller

May 8th
DAT Timecode
Sony

May 16-17th
Hard Disk Recording
AES Conference

June 12th
Active Acoustics
Philip Newell

This month's lecture is by Paul Evans of Thames Television on **Digital Audio in the Television Centre.** "With the introduction of low cost of high quality Nicam VHS recorders and TV sets, considerable pressure has been put on the broadcaster to ensure the highest quality of audio is originated and transmitted to the viewer/listener. In addition stereo often imposes considerable operational overheads. It is therefore very desirable that means can be found to alleviate some or all of these operational problems and maintain the signal to the highest possible quality.

The cost of digital audio equipment has been falling steadily in recent years. It has now reached a stage where it is cost effective enough to be considered for wide scale use within the TV Studio Centre. It is intended to present an overview of how current and future technology can be applied within the studio complex."

This lecture will be held at the IBA, 70 Brompton Road, London SW3 starting at 7.00 pm with coffee at 6.30pm.

Please also note that the next AES Convention, the first of the '90s, will be held in Montreux from 13th-16th March. In addition to a full series of Papers and Workshops, there will be a large exhibition of international manufacturers with the latest technology on show.

For further details on any of the above or information on joining the AES, please contact: **Heather Lane, AES British Section, Lent Rise Road, Burnham, Slough SL1 7NY. Tel: 0628 663725 Fax: 0628 667002.**

Letter: The Box isolation

Dear Sir, It has come to my notice that several readers have misinterpreted one of the paragraphs in the article 'Modular Acoustics' *Studio Sound*, November 1989.

I would like to clarify this. The specification quoted is the *minimum* isolation for a single walled box placed in a reverberant sound field, ie 36 dB at 63 Hz and 60 dB at 1 kHz. It must be realised that in order to meet the incremental ILR Code of Practice for isolation between two on-air studios or on-air studios and talks studios, a *minimum* isolation of 42 dB at 63 Hz and 70 dB at 1 kHz must be achieved. Our completed installations provide appreciably higher levels than these. NC ratings are frequently NC20-25 and RT is nominally 0.2 secs midband/HF rising to 0.3 secs at 63 Hz.

Should anybody have any further queries do not hesitate to contact me.

Yours faithfully, P D Berg, KFA Associates Ltd, 1/2 Guillemot Place, Clarendon Road, Wood Green, London N22 6XG, UK.

New Virgin venture

The Euromagnetics Group have announced a joint-venture with the Virgin Group and Telfos, the engineering company, which will create a new company called Euromagnetic Holdings Ltd.

Euromagnetic Holdings will produce magnetic computer tapes, floppy disks and blank video and audio tapes. The new company aims for sales of £22 million and some of its products will be sold with the Virgin brand through Virgin retail outlets.

Euromagnetic will be formed from merging three companies, which are already jointly owned by Telfos, Virgin and Peter Mowland, a private businessman who will become chief executive of the new company.

The three companies are Euromagnetic Products, 50% owned by its chief executive and chairman Peter Mowland, 14% by Telfos and 35% by Virgin. Shape Technology, 50% owned by Peter Mowland, 50% owned by Telfos, and Wabash Datatech, 50% owned by Peter Mowland and 50% Telfos.

Telfos, Virgin and Mr Mowland will each own ⅓ of Euromagnetic.

NEWS

30 years of Studio Sound

February 1964...A completely self-contained mobile video recording unit, the Ampex *Minicruiser*, has been placed on the market. The heart of the compact vehicle is an Ampex *VR 660* weighing 97 lb. Designed around a Studebaker *Wagonnaire*, the *Minicruiser* is equipped with a sliding roof, enabling the cameraman to shoot directly from the vehicle. The first mobile video recording unit, the *Videotape Cruiser*, was developed by Ampex in 1959. It incorporated a flexible bus chassis, an Ampex *VR 1000* video recorder and associated camera and power equipment.

February 1969...Recent experiments by Dr K A Mulholland at the University of Liverpool have shown that the application of Polystyrene to a panel, with the aim of increasing insulation, can actually have the opposite effect. The transmission loss of a plasterboard panel with a 50 mm Polystyrene layer was found to be 5 dB poorer at 600 Hz than with plasterboard alone. By contrast, a 50 mm Rockwool layer gave a 22 dB improvement in transmission loss at 4 kHz, relative to plasterboard alone, and 24 dB relative to plasterboard plus Polystyrene. This negates the common assumption that Polystyrene can be used for sound insulation. A standing wave resonance inside the Polystyrene has been suggested as the cause of the deterioriation in transmission loss.

February 1978...The BBC has recently been using an experimental 2-channel digital transmission system to access the feasibility of conveying high quality stereo sound programmes signals from OB sites to London in digital form.

February 1970...A data recorder produced by Lockheed Electronics will be incorporated in the unmanned Mariner spacecraft due to orbit Mars for a 90-day period in 1971.

People

● **Trilion Video**, London, UK, have announced that Dick Allott, one of their directors, has left to pursue his own career in the television sector.

● **SSL** in the USA have appointed a new treasurer/controller. John B Kilcullen joins from Oratronics Inc, where he was vice-president of finance, with over 19 years in financial management.

● Appointments at **Celestion**, UK, include Phil Pell, who will create a UK dealer network for the *SR* series sound reinforcement system, and Linda Brame, sales and marketing co-ordinator.

● Video services company **VSC**, NY, USA, have appointed Shelly Riss vice-president of sales and marketing. He was previously manager of his own production company for 13 years.

● The new studio manager at **Picnic Recording Studios**, Kent, UK, is Dawn Adamson. Adamson comes from other management positions in the USA.

● Phillip Neighbour joins **Tannoy**, UK, in the newly-created post of professional products manager.

● **Neve** UK's new director of sales is Hazel Simpson who joins from Harrison Information Technology.

● Jon Ridel, has been appointed UK sales manager for the complete range of **Soundcraft** products. Ridel was previously in charge of studio sales.

● Martin Kelly of **Harrison Information Technology**, while still responsible for UK sales, will now also be involved in export sales worldwide.

Letter: Neutrik loudspeaker connectors

Dear Sir, With reference to Ken Dibble's very comprehensive article on speaker connectors in the December issue we would like to make a few comments.

1 The 8-pole versions of the Neutrik *NK8FC* for cables and *NL8MPR* receptacles are now in full production—all sockets are airtight.

2 As to the two-component concept, the 'free' connector and the 'receptacle', this constitutes a trade-off between the following considerations:

Plus

● simplicity and unification of cables

● lower cost in manufacturing and for holding in inventory

Minus

● extending a cable needs the in-line adaptor. We have found that only 15% of potential users have to extend cables and we offer the coupler as a service part at a very reasonable price. It can also be permanently attached to a free connector thus forming an extension cord

● the possibility of connecting two amplifier outputs in error. Today's amplifiers are so well protected that accidental paralleling of outputs should not cause any damage. It is a possibility that exists presently anyway with the use of binding posts or ¼ inch jack plugs

3 With regard to the section headed Wiring Convention, Table 1 and Ken Dibble's comments we would like to summarise the history and results of our broad discussions on this subject. Wiring the connector as we originally recommended, as Ken Dibble mentions, has the benefit of using one receptacle and one 4-pole cable for stereo, split somewhere close to the speakers.

Using a special dedicated cable, wired to 1+ and 2+, allows mono bridged application. But apparently only few people are using 4-way cable from the amplifier to the speakers. The next step was providing two receptacles on the amplifier, both wired in parallel, one used for left, one for right with dedicated cables—left wired to ±1, right to ±2, or a mono bridged cable. This would have the benefit of being very flexible and knowing at the cables 'arrival points' what it carries, left or right or mono. The disadvantage apparently is the need of different cables for the two channels and mono.

Recent discussions, however, at the AES in New York showed a clear preference to what Ken Dibble seems to have in mind as a solution for 'Disaster 1':

● A 2-channel low Z amplifier should have three 4-pole *Speakons*, one for left, one for right and one for mono-bridged

● The left and right ones are wired the same way: left channel to 'left' receptacle—LH signal to no 1+, LH ground to no 1−; right channel to 'right' receptacle—RH signal to no 1+, RN ground to no 1−; the mono receptacle shall be wired—LH signal to no 1+, RH signal to no 1−.

Using this system has the benefit that only one cable is needed, that it comes close to the conventional binding post/phone plug/*XLR/3* wiring and that ±2 contacts are reserved for HF-drivers (bi-amp) only. Anyhow, 4-way cables cannot be used as easily anymore.

With regard to 100 V line, 'Disaster area 2', when the connector is wired as follows nothing disastrous should happen.

1−: 0 V
1+: safety ground (if applicable)
2−: NC
2+: 100 V (70 V)

For this case a dedicated 100 V cable is needed but if a normal cable or a biamp cable is inserted nothing will happen.

In case a tapped output transformer is used for, eg a low Z *and* a 100 V output, wiring could be as follows:

1−: 0 V, common
1+: low impedance tap
2−: NC
2+: 100 V (70 V) tap

For low Z a standard, for 100 V the dedicated cable can be used.

4 On the subject of our contact designation with the connector, it originates from best heat dissipation considerations, which we think is a very important point.

There is one last point we would like to mention. The twist-lock design was not only chosen for its mechanical strength (hard to achieve with any other fast latching method) but mainly since we found it to be the only method to achieve both a good multi-area high current contact, and a fast arc-blowout action when disengaged under load. Tests with brand X products showed substantial temperature rise with a 30 A load and started burning when disengaged under load.

I thank you for this chance to comment.

Bernhard Weingartner, Neutrik AG, FL 9494 Schaan, Furstentum, Liechtenstein.

Agencies

● **Focusrite Audio Engineering**, UK, have entered an exclusive dealership agreement with HHB Communications.
● Martin Audio, New York, USA, have been appointed by **WaveFrame** as their representative in that territory including New York City metropolitan area.
● JTM Productions have been appointed as distributor for **Saturn Research** in Holland, Hilversum, Netherlands. Tel: (035) 211920. Fax: (035) 47649.
● Ampsound are now distributors of the **Tascam** *Pro* range of tape recorders. Ampsound, St Albans, Herts, UK. Tel: 0727 50075.
● **New England Digital** have named three new distributors to represent them in Japan, Hong Kong, Taiwan, Singapore and Malaysia. Japan Digital Systems, Tokyo; Linfair Engineering, Taipei, and ECM Research Ltd, Singapore.

Contracts

Audio Kinetics MasterMix II console at The Coach House Studio, Bristol

● Television company TVB, Hong Kong, have selected **Nexo** sound systems for installation throughout their studio complex to be supplied by local distributor Power Source Developments. Other Nexo clients in the area include ATV.
● Full Sail Center for the Recording Arts, FL, USA, have recently purchased the *MSL-3/650 R2* main system from **Meyer Sound** for use in their new Sound Reinforcement class.
● Nightclub chain ClubLand have just opened their newest facility at Detroit's State Theatre using **Electro-Voice** *Manifold Technology* and **DeltaMax** speaker systems.
● **Harris, Grant Associates** have supplied *Boxer 4* monitors to Mike Oldfield, UK, and *Boxer 5*s to Sound Studio 'N', Koln, West Germany.
● **BTS Broadcast Television Systems**, West Germany, have been appointed general contractor to supply TV equipment for two radio and two TV studios at the Kremlin, USSR. The studios will be used for the first time to broadcast nationally and internationally the debates of the Supreme Soviet.
● **Harris, Grant Associates** have been appointed by Granada Television as consultants for the acoustic and isolation treatment of Stage One in Manchester, UK.
● Uduco, **Saturn Research**'s French distributor, have recently installed a *Saturn 824* multitrack at CNIT International Communications Centre.
● Hilton Sound have purchased two **Sony** *PCM-3324A* digital multitracks in addition to their existing fleet of *PCM-3324* and *PCM-3348* recorders.
● The Coach House Studio, Bristol, UK, have become one of the UK's first users of **Audio Kinetics** *MasterMix II* console automation system.
● **MVC Crow**, Newbury, UK, have recently completed a BBC contract for the construction of a radio news vehicle intended for private interviews on locations around the London area.
● Among numerous video installations **Philip Drake's** Broadcast Systems division have recently supplied a new sound dubbing suite to the BBC's South-East regional news department and a

new audio system in Elstree's Studio D based around a Calrec desk.

● Recent **Studer** US sales include a *A820-24* multitrack to Criteria Recording Studios, Miami, and Studio In The Country, Bogalusa, Louisiana. Five Studer *Dyaxis* hard disk audio production systems have been supplied to Music Animals, a Los Angeles-based jingle company.

● Pacific Studios, London, have recently had a 48-channel **Soundtracs In-Line** console fitted in their studio.

● UK dealers Michael Stevens & Partners have recently installed equipment into HMV Records, London, in-store DJ booth, including a 16-channel Soundcraft *SAC 200* desk with **Sonifex** and **Denon** cart machine and **Technics** *SP10* turntables.

Rear view of BBC radio interview vehicle

Exhibitions and conventions

February 20th to 21st Sound 90, Heathrow Penta Hotel. Contact: SCIF. Tel: 0628 667633. Fax: 0628 665882.
March 13th to 16th AES 88th Convention, Centre de Congres, Montreux, Switzerland. Contact: AES Exhibition Director, Herman A O Wilms. Zevenbunderslaan 142/9–B-1190 Brussels, Belgium. Tel: (2) 345 7971. Fax: (2) 345 3419.
March 30th to April 3rd NAB, Atlanta, GA, USA.
April 22nd to 25th Vision and Audio International, Earls Court Exhibition Centre, UK. Tel: 01-776 0709.
June 1st to 6th AV & Broadcast 90, China International Exhibition Centre, Beijing. Contact; Business & Industrial Trade Fairs, 28/F Harbour Centre, 25 Harbour Road, Wanchal, Hong Kong. Tel: 5-756333. Fax: 5-8341171.
June 6th to 8th APRS 90, Olympia 2, London, UK. Contact: APRS Secretariat. Tel: 0923 772907.
August 19th to 22nd Video Expo '90, Palacia Das Convencoes do Anhembi, Sao Paulo, Brazil. Contact: (UK) Ms Alison Carew-Cox. Tel: 021-455 9600. Fax: 021-456 1785. (Brazil) Para maiores informacoes. Tel: 021-220 3386.
September 21st to 25th International Broadcasting Convention, Metropole Conference Centre, Brighton, UK. Contact: IEE Secretariat. Tel: 01-240 1871.
September 21st to 25th AES 89th Convention, Los Angeles Convention Center and Los Angeles Hilton, Los Angeles, CA, USA.

Address change

● **Monster Cable** have relocated to 274 Wattis Way, South San Francisco, CA 94080-6761, USA, Tel: (415) 871-6000. Fax: (415) 871-6555.


THE TECHNICS DAT RECORDER IS HERE.
US
57961
3020
US
30200
GLORY

The Technics portable DAT recorder would have been an admirable choice for the Admiral's Cup.

While the yacht sways and heaves, the last thing a journalist needs is an unwieldy tape machine.

The Technics SV-260 portable DAT recorder's small size is due to a 15mm head drum.

The quality however is heavyweight and on a par with a deck-type DAT recorder mainly due to MASH ADC and XLR balanced Cannon connectors.

Like the U.S. America's Cup team, it goes really fast forward and like them is quick to rewind in the event of an error.

Using the portable recorder is plain sailing in dramatic productions and audio research, where DAT picks up the smallest vibration.

Docked in a studio, the SV-360 DAT deck is equally impressive.

Like the portable, it offers all the flexibility of tape with the sound quality of C.D.

It can be used on its own or as a back up system in a recording studio.

It can also be used alongside our already well-established C.D. players and turntables which have become classics in the studio.

Out of the studio, the DAT deck is equally effective on location.

Where it is often used for outside broadcasts.

Its popularity with the professionals is due to full digital in and out terminals, analogue sampling of 44.1Khz for C.D. mastering and hard wired remote control.

It's 4 DAC 18 bits also delivers higher fidelity.

Something yacht crews could learn a lot about.

Technics

FOR MORE INFORMATION ABOUT THE RANGE OF TECHNICS PROFESSIONAL AUDIO PRODUCTS CONTACT ONE OF THE FOLLOWING ADDRESSES
ASC, 1 COMET HOUSE, CALLEVA PARK, ALDERMASTON, NR. READING, BERKS RG7 4QO. TEL: 07356 79565. HHB, 73-75 SCRUBS LANE, LONDON NW10 6QU. TEL: 01 960 2144
RAPER & WAYMAN LTD, UNIT 3, CRUSADER INDUSTRIAL ESTATE, 167 HERMITAGE ROAD, HARINGEY, LONDON N4 1LZ. TEL: 01-800 8288

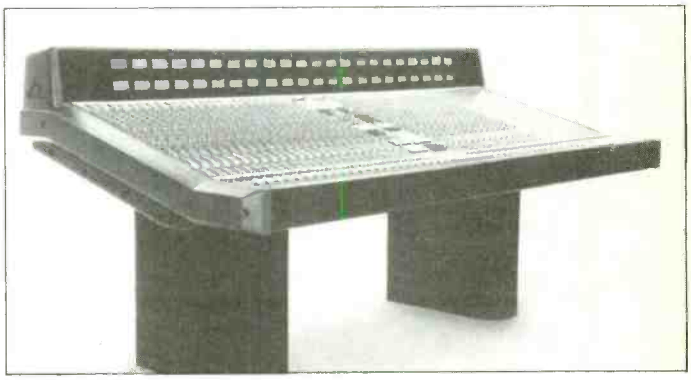
Yamaha digital products

Yamaha launched a number of new digital products at the NY AES. The main introduction was based around an 8-track stationary head digital recording system that was shown in two forms. The first was as the *DMR8X* mixer/recorder comprising the 8-track recorder, digital mixer, timecode locator and automated mixing system all in one unit; the second was the *DRU8X*, which is the digital recorder in a rackmount format that can be added to the *DMR8X* or run separately.

The *DMR8X* performs all functions in the digital domain under microprocessor control with onboard memory for storage of all automation data as well as 32 snapshots of the entire signal path setup that can be stored on memory card. Mixing capability is as an 8-bus stereo format with on-board effects. Channel capabilities include 3-band parametric EQ, independent effects such as comp/lim, high and low filters, delay, echo, flange, emphasis, three effects sends etc. In mixdown 24 channels of audio can be handled in banks of eight. The unit also contains the equivalent of three *SPX900/1000* processors. All mixing data is recorded as sequence data and can be recorded at the beginning of the tape as a table of contents. Digital interface is Yamaha *MEL2* format (24 bit serial) allowing digital interface with other Yamaha digital products such as the *DMP7/DMP11*, *DEQ7* and *SPX* series; as well as two DAT/SPDIF inputs and outputs, AES/EBU stereo inputs and outputs. Analogue inputs are available with the use of the new *AD8X* A/D 8-channel converter. MIDI timecode is included for synchronisation with MIDI devices.

The tape cassette is much larger than an R-DAT cassette and uses a metal particle tape. Playing time is 20 minutes at 48 kHz, 8-track.

Yamaha Corp, PO Box 1, Hamamatsu, Japan.
UK: Yamaha-Kemble Music (UK) Ltd, Mount Avenue, Bletchley, Milton Keynes MK1 1JE. Tel: 0908 71771.
USA: Yamaha Corporation of America, PO Box 6600, Buena Park, CA 90622-6600. Tel: (714) 522-9011.



Trident Vector 432

Trident have launched a completely new console in the form of the *Vector 432*. This is an in-line design that can be operated in such a way that it allows suitability for a wide variety of multichannel types of work. Features include four matrixed stereo buses, 32 group outputs and 16 externally and internally triggerable automute groups that can be channels, monitors and aux mutes in any combination. All inputs, outputs and buses are electronically balanced. There is a stereo bus compressor across the main output; a Broadcast mode that allows simultaneous stereo and multitrack feeds; and two dedicated foldback systems selectable from any combination of console sources. Standard frame sizes vary from 40- to 72-channel supplied with on-board or remote patchbay. The Vector Studio Computer manages machine control and automation functions in a single unit that will lock to SMPTE timecode and other sources. Metering choices include vu or ppm or a vu/ppm bargraph combination. Channel facilities include four mono aux sends each with level, pre/post fader, cut and monitor path sources; four aux sends configured as two stereo pairs with level and pan; continously variable high- and lowpass filters, 4-band EQ that can be split and may be switched separately into the monitor path. The HMF and LMF have switches that alter the frequency ranges over which the bands operate making it possible to configure the EQ section as two identical units. Four dynamics modules can be fitted anywhere within the console offering compression/limiting/gate/expander/duck functions with filter, key listen and link controls.

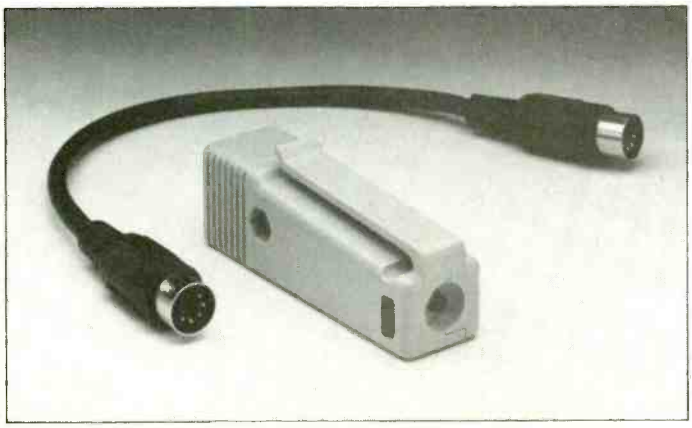
Automation possibilities at present include the model *MCA Studio Computer*, which offers VCA fader automation, two monitors for continuous display of machine and fader status. The basic *Studio Computer* also has the ability to interface with moving fader systems that the *AT* keyboard offers as standard. The computers use hard disk and floppy drives in 3.5 and 5.25 standards.

Trident Audio Developments Ltd, Rodd Estate, Govett Avenue, Shepperton, Middx TW17 8AQ, UK. Tel: 0932 224665.
USA: Trident Audio USA, 2720 Monterey Street, Suite 403, Torrance, CA 90503. Tel: (213) 533-8900.

Musonix MIDI Beacon

A simple but useful new device comes from Musonix in the form of the *MIDI Beacon*. The *Beacon* will detect a MIDI signal in a MIDI cable just by being attached to the cable end. It measures 3½ inches long, is equipped with a standard MIDI In jack and a green LED indicator. Presence of the MIDI signal on the cable causes the LED to flash. Internal circuitry is used to ensure that short MIDI messages such as System Reset (32 μs) will cause a visible flash. The manufacturers also cite applications such as detection of non-standard MIDI outputs and cables without grounded shield; tracing of MIDI wiring; verification of MIDI switchers operation, THRU boxes, signal splitters, etc. The *Beacon* has a clip for convenient carrying and requires no batteries.

Musonix Ltd, 2537 North Ontario Street, Burbank, CA 91504, USA. Tel: (818) 845-9622 (US toll free (800) 888-0848).



Audio Design PCM 2500 mods

Audio Design have announced retrofit modifications on the Sony *PCM 2500* DAT machine to provide for external EBU sync with the facility to provide sync to external word clock to follow. This will enable the *2500* to playback as a slave to a house sync rather than having to act as a master for the complete installation. The unit can further be modified to provide an error status information port as per the Audio Design *PRODAT* protocol. This Error Status port can be linked to the Sony *DTA2000* Digital Tape Analyser unit to produce error reports on DAT recordings. A program is currently being written to work with IBM-compatible *PCs* to providing an alternative report writing output.

Audio Design, Unit 3, Horseshoe Park, Pangbourne RG8 7JW, UK. Tel: 0734 844545.
USA: Gotham Audio Corp, 1790 Broadway, New York, NY 10019-1412. Tel: (212) 765-3410.

NED Synclavier 6400 system

New England Digital have added a third mid-range *Synclavier* to their workstation series. It features 32 stereo voices, 64 Mbytes of RAM, newly enhanced velocity/pressure keyboard, SMPTE and VITC interfaces, 320 Mbytes of hard disk storage, stereo 100 kHz inputs and a *Macintosh IIx* graphics interface. The *6400* fits halfway between the *3200* and *9600* systems in the series but the system can be expanded with hardware options such as optical disk storage system, DSP card, *Direct-to-Disk* recording system, etc, and software.

New England Digital, 49 North Main Street, White River Junction, VT 05001, USA. Tel: (802) 295-5800.
UK: New England Digital (UK) Ltd, 77 Fulham Palace Road, Hammersmith, London W6. Tel: 01-741 9911.

In brief

• **Audio Kinetics** have announced an enhancement for the *Reflex* console automation system in the form of a new 8-channel Mute Interface Board. This enables existing console channel mute buttons to be written to the automation memory. Previously channel mutes could only be written to memory via the automation control pad.

WaveFrame additions

WaveFrame Corporation have announced three new products for the *AudioFrame* digital production system. The Digital Signal Processing Expansion module *DSP-X* for the Disk Recording module provides 12 digital inputs and outputs in PD or SDIF-1 formats and a pair of inputs and outputs in SDIF-2 or AES/EBU format. The interface can be routed directly to the disk recorder or through the *AudioFrame* Digital Bus to any other modules in the system.

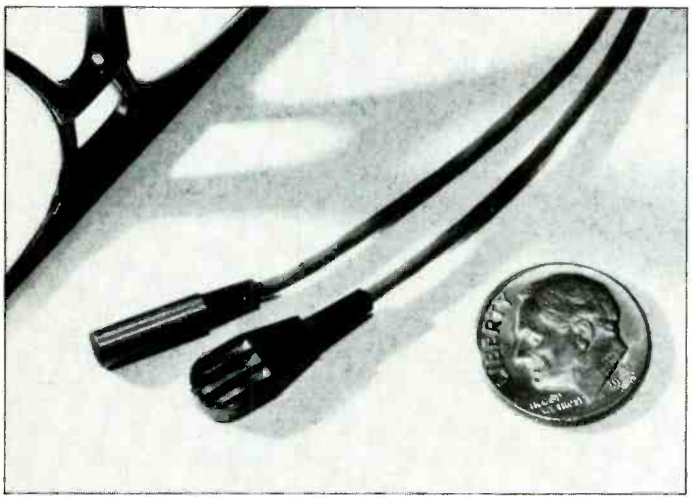
Available from the beginning of this year is an erasable magneto-optical removable drive for session back-up and archiving. It is intended for use as an alternative to the 8 mm tape streaming system currently used in the *AudioFrame* and offers improved access speed.

The third new product is the Storage Expansion Rack (SER), which connects with the *AudioFrame* and accommodates four SCSI devices—disk or tape drives—increasing disk storage capacity from 1 to 8 hours with the possibility of linking multiple units for greater storage.

WaveFrame Corporation, 2511 55th Street, Boulder, CO 80301, USA. Tel: (303) 447-1572.
UK: Syco, Kimberley Road, London NW6 7SF. Tel: 01-625 6070.

Sanken mini mics

Sanken have announced two new miniature microphones that result from a joint engineering effort by Sanken and NHK. The range of mics are designed to maintain high quality audio standards while meeting the concealment needs of high resolution film and video. Known as the *COS* series, the *COS-11* is a tube-type mic and the *COS-12* is a flat lavalier type. Designed for music and voice applications, Sanken claims new levels of transient response, extended frequency response, low sensitivity to mechanical noise and 'unprecedented' omnidirectional response. Sanken have employed a vertical placement of the diaphragm for a greater effective area within a much smaller casing.



The *COS-11* tube lavalier measures 4×11.5 mm providing a frequency response of 40 Hz to 12 kHz and a dynamic range of 93 dB. The *COS-12* flat lavalier measures 2.7×6.8×13.4 mm with a frequency response of 40 Hz to 20 kHz and a dynamic range of 97 dB.

Sanken Microphone Co Ltd, 2-8-8 Ogikubo, Suginami-ku, Tokyo 167, Japan.
UK: Stirling Audio Systems Ltd, Kimberley Road, London NW6 7SF. Tel: 01-624 6000. Fax: 01-372 6370.
USA: Audio Intervisual Design, 1032 North Sycamore, Los Angeles, CA 90038. Tel: (213) 469-4773.

Beyer have added a combined shock mount and wind shield specifically designed for use with the *M160* (cardioid) and *M130* (Fig-of-8) ribbon mics for MS recording techniques. The complete assembly is capable of being hand-held.

A question of culture

British engineer, producer and recording artist Alan Parsons had some words of warning and help for the audio electronics industry in Japan recently. As guest speaker at the International Broadcast Equipment Exhibition (InterBEE) symposium in Japan, Parsons spoke on the history of sound recording technology in the UK, its relationship to the current structure of sound recording in Japan and the future of the field in general.

Parsons compared the recording industry in Japan today with that of Britain some 20 years ago. Record producers in Japan are known as 'house directors', akin to the artists' managers or recording managers in the early days of recording at EMI Abbey Road, Parsons said.

This has tended to stultify the Japanese recording industry, as it has encouraged a very conventional and almost dictatorial approach to recording. Parsons suggested that the recording engineers in his audience would not have the freedom of experimentation that has become the norm overseas, primarily because of the house director system.

"Originality can be destroyed by holding back and going along with conventional ideas," he said.

British culture allows free speech without any jeopardy to one's standing in the company, Parsons remarked and went on to observe cautiously, "I wonder if in Japan today the company standing holds back the creative process in the Japanese recording industry. Possibly Japan is about to have an industry revolution as we had in Britain and in America."

Reminiscing his experiences in the 1960s and '70s at Abbey Road, Parsons outlined the revolution he saw occur with regard to the role of the producer, engineer and recording personnel.

In 1963, when the Beatles had a number of top ten hits in the charts, their producer George Martin received no royalties or commissions for his part in their success. This has changed since then in Britain: "all UK producers...now receive royalties on the records they make," Parsons noted. In Japan, however, producers are still likely to receive very little for their efforts.

Parsons credited George Martin's move to independent production with the support of the Beatles as providing a turning point for the recording industry, both in terms of what studio personnel did and in what manner the studio was used. "The Beatles were responsible for the revolution in the design of studios and reshaped the roles of studio personnel," Parsons said. "They were not afraid to take risks...and used the recording studio as a musical instrument."

By encouraging experimentation, the Beatles helped the development of a whole new breed of creative engineers and producers came to play an active creative part in recording.

The major limitation at that time was in the technology supporting the recording industry.

"The console technology wasn't keeping up with the machine technology and the ever-increasing demand for more tracks," Parsons said.

After the disastrous episode of the 'Magic Alex' console—supposedly offering 72 channels but really a "ghastly console with every conceivable problem of buzzes, hums, clicks"—the Beatles were able to get a new machine for their studio. "*Abbey Road* was recorded on a genuine 8-channel console on a genuine 8-track machine," Parsons recalled.

Even this new technological wonder wasn't enough and Parsons remembered a glum George Harrison complaining about having to wait all the time for the engineer to find the right plug. Tape machines were the main backbone of the studio, providing repeat echo, extra sounds, loops and the other effects that artists and producers wanted to hear.

By the time Parsons got to work on Pink Floyd's *Dark Side of the Moon*, 16-track machines were available but even these were not enough. "The tracks ran out very quickly as there were a large number of overdubs," Parsons said.

In the mid 70s Parsons received a number of producing and engineering offers, and was able to combine both roles: "I was one of the earliest people to do both jobs at once," he said.

"Fifteen years ago not one commercial digital processor was available. It's hard to believe, living in the age of the '80s, the age of digital processing." By the time digital equipment began to be used in the recording industry, Parsons had added 'record artist' to his list of roles, with the production of the Alan Parsons Project album *Tales of Mystery and Imagination*.

" 'Nevermore' (the first track on *Tales*) was the first recording I know of to use a digital vocoder—the digital age had arrived."

Parsons noted that the new age wasn't greeted entirely with open arms. One complaint he had to make was the distortion in musical attitudes, with some artists unable to write material without a full backup of drum machines, sequencers and the like. In addition, the ever-developing market has caused difficulties in remaining familiar with the equipment available.

"Since the dawn of the digital age, the technology has been changing so quickly it's been difficult to keep up with developments. With more delay lines, digital flanges, digital chorus boxes, everyone said we didn't need tape machines any more. Now with drum machines, *Synclavier*, Fairlight and sequencers, everyone said we don't need musicians any more. Then MIDI came along and everyone said help, help, help."

Japan has a great advantage in this new age of music production, according to Parsons. As a major manufacturer of professional audio equipment, the country has a huge impact on industry movements.

"The key to success in the future is to be able to deal with the narrowing gap between the musician, the engineer, the producers and, most importantly, the manufacturer. The barriers need to be broken down and communication put on a better level," he maintained.

Parsons deplored the development of "nightmare machines" with "little tiny buttons" marked with a huge array of arcane instructions and tiny screen displays that are unreadable.

He called for interaction between studio personnel and manufacturers to produce user-friendly machines that are able to do what is required. "The 'ideal' machine in the high-tech world is the one that can anticipate what you want to do," Parsons remarked. To obtain that ideal, audio personnel have to be able to have input at the design level, rather than it being left up to some "electronics engineer straight out of college".

With the huge variety of equipment, particularly among sequencers and editors, it is the manufacturers that dominate with their designs. Parsons called for more emphasis on sensible default settings and functions that have been fully thought out. You end up pressing a button 18 times and then you get a sign lighting up saying 'data entry error'—"I'm not interested," he said.

"With all this new technology, I think we all hoped that everything was going to become easier and easier. Now it seems to take 20 times longer than it ever did," Parsons said. Much of the time is spent reading manuals for overly complicated machinery, with very few people able to master the entire range of synthesisers, keyboards, digital boxes and the plethora of equipment now occupying even the most simple studio.

"The key is to make products that fill needs, not ask us to find a need for a product you come up with," he told his audience, many of whom are involved in the production of hi-tech audio equipment.

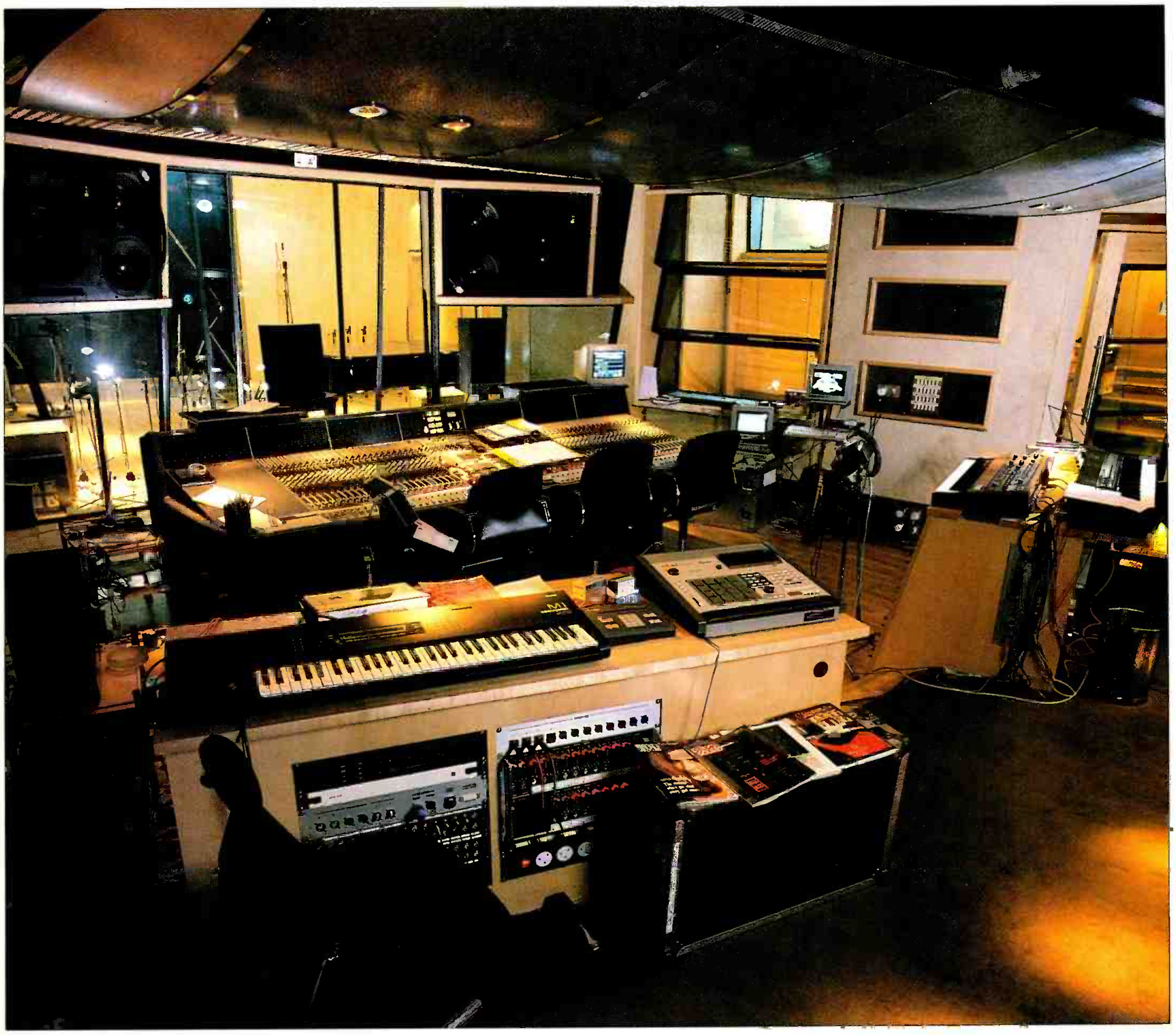
"Manufacturers are always asking us to make critical assessments of their latest devices. It is increasingly difficult to discern quality differences between one machine and another," Parsons noted. "We are pushing our ears to the limit to tell what is very good quality and what is excellent." This was possible in the days of analogue technology, he maintains, but no longer.

The development of so much equipment has meant a further blurring of the roles of studio personnel. "Musicians became engineers, engineers became musicians and producers became computer programmers," he said, not entirely tongue-in-cheek. With this blurring of roles comes a new awareness of the needs and responsibilities of each role, a good thing in an industry that has always tended to narcissism.

The Producer's Guild, formed 3 years ago in the UK, is also seen by Parsons as a good influence on the industry. Its initial aim was to provide interaction between producers, recording companies and manufacturers but has also encouraged producers to talk to each other about the equipment they like, the personnel, the studios and the other facets of recording.

Parsons suggested that such an organisation might benefit the recording industry in Japan. The country is at the forefront of audio technology and, until recently, has tended to concentrate on its own domestic market. Parsons sees that changing in the future, with the "European influx" leading to better developments and better communication between audio personnel throughout the world.

From the reactions of many in the audience, the Japanese appear to welcome such a future. **Vicky Hyde**



Control room Studio A in a working mode

METROPOLIS

After three years of thoughtful planning and design a new London studio complex emerged. Janet Angus reports

It says something about a studio when they send you off with a copy of *The World of Interiors* magazine inside which it is featured. Apart from the obvious fact that it is interesting architecturally and design-wise, it also indicates the emphasis its owners put on image.

"You cannot underestimate the importance of visual impact and environment," producer and co-owner Carey Taylor leaps in defensively. He may well have a point and there is no doubt that he and his partners have gone in for this approach in a big way. We are talking about Metropolis Studios, recently opened in the heart of London, bringing with it a radically fresh approach to studio design.

There may be a clue in one of his quotes from that article: "It was conceived with a new breed of rock musician in mind; people who want more than pinball machines and hamburgers—who live nice lives and don't want to go down (literally) to the grottiest level when they make a record." On talking to Taylor however, the truth appears to have a much more solid and realistic financial foundation. The fact is that, technologically, you would be extremely hard put to do something spectacularly different in building a studio today while still guaranteeing interest on the major recording circuit. The design area certainly provides scope if you are feeling brave enough, which Taylor and his partners (producer) Gary Langhan and Karin Clayton obviously were.

Nevertheless the maxim of more haste less speed was definitely applied here. While agonising over design decisions they watched Virgin acquire, design, build and open their Olympic Studios, which Taylor confesses was very frustrating.

Housed inside a massive Victorian power house in Chiswick the Metropolis creation is a violent tangle of metal and wood modernism, all the more striking for its 19th century setting. It is the final choice of four designs, each of which took 9 months to prepare, so determined were they to hit the right formula, delicately balanced between adventurousness, ergonomics and, naturally, economics.

Comprising two large recording facilities so far, the complex is due to open a further three rooms by the late spring. Representing a staggering investment of £7 million this is a serious business venture with very definite responsibilities to its ▷

investors to succeed. As Taylor says, it's not about making a 'profit'. You can't fail to make some sort of profit with the vast range of equipment available; the kind of profit required to justify the vast capital input is sizable to say the least.

Whatever possessed the three young partners to take on such a heavy responsibility in what must be acknowledged as a struggling industry? Langhan and Taylor both began their careers at Basing Street and followed the tape op/assistant engineer/engineer/producer route from there. Meanwhile Clayton made a name for herself managing Sarm East and going on to form One Management with Taylor. The point is they all have 'large' studio backgrounds. They haven't experienced anything less and so they say it is natural that they should want to compete in that league.

"We saw Advision and Olympic Studios in the 1960s," explains Taylor. "In the '70s we had Trident, Basing Street and Air after that. Obviously, there were the Abbey Road and CBSs but in the pop music context those were the big studios of their day. In the '80s there was Townhouse and Sarm. We wanted to be one of the ones that showed what happened in the 1990s.

"You have to be right in the centre of things, not a satellite. We wanted to set a standard. Naturally we have ambitions for a record company and publishing. If you want to do something you might as well do it well. It is not for us to decide whether we are the best studio in London but that is what we are striving to be.

"Spending a lot of money is the only way to make sense out of running a studio. When the going gets tough (and it is going to) only the big studios will survive. People will always want to make the best record they can. The current market rate is catastrophic. It is very difficult to make sense out of any studio."

It is the same theory that makes it easier to borrow £1 million than £100 in this present stressed economic climate. Metropolis are convinced that the only way to succeed and make money is to spend a great deal of it in the first place. Not that that is their sole motivation: "We all live, sleep and breathe music business. It is unusual to find a studio of this scale set up by people who know what it is like to sit behind the desk."

This is his big selling point. Taylor repeats his belief that although there are plenty of 'good' studios in London, most do not offer all the facilities under one roof to wholly satisfy artist and producer. There are those that have all the technology but leave the artist to hang around in shabby corridors while not actually playing; then there are those that cater for all the comforts of home but are sadly lacking in the equipment department.

"We thought it would be possible to come up with a complex for recording the way people want to record today," Taylor continued. "There has been a swing towards people spending 3 months in a studio. I can show you lots of studios you wouldn't want to spend 3 months in! We have tried to build studios which sound essentially good; rooms which are just straightforward and sound great. We didn't think we needed rotating drums although Real World are also trying to do something different and individual.

"The fact is, people in London are used to working in a specific way. If you try to throw the whole book away it is very hard for people to get used to it. I don't think it could work in the London circuit. You can come in here from any studio and slot in and feel at home. If you want to compete on the major circuit you have to accept the way people work. Gary and I have both spent a lot of time behind the console. I could point you to at least half a dozen good control rooms but very few artists would enjoy being in there. We have tried to make an environment where the guy performing has a great time as well as the guy behind the desk. The relaxation areas are as important as the control room. It all has to be stimulating and fun.

The motivation for dramatic design, then, was in order to create a stir. "If you are really spectacularly different you create a demand. However, in order to do something genuinely different we had to be absolutely certain. So we spent an enormous amount of time working it all out, which meant that when it came to actually implementing it the time on site was relatively fast and efficient."

There are not many studios in London, or come to that in England, which have been built from the ground up. Although Metropolis nestles inside the impressive Power House shell, it doesn't come into contact with the outside walls and is, in effect, a free-standing facility. There were, therefore, no constraints as to shape and size. An architect/studio designer's dream you would think. The fact is it's not as simple as that.

Determining the cost of converting an existing building into a recording facility is a great deal easier than looking at an empty space and costing the whole thing out from scratch. Add to this the fact that you are determined not to use typical 'recording studio' materials and you are stepping into unknown territory.

Metropolis decided to appoint the Acoustic Design Group (more specifically Sam Toyoshima and John Flynn) as acoustic consultants along with architectural group Powell-Tuck Connor and Orefelt.

"We started talking to Sam in 1986. When you are building this many rooms in close proximity your biggest problem is isolation. There are lots of people in England who build studios but they all use the same method of floating floors, etc, which is very expensive. Sam has built many large facilities in Japan and he has a very in-depth understanding of isolation. We also wanted someone who could work with a large team as this was a massive project. Again Sam is used to this method of working

Control room Studio B

and he was very good in that role. He and John Flynn were both very helpful and they never actually said, 'No, you can't do that.' It was slightly daunting to be undertaking this build with our acoustician in Japan but we were helped a great deal by John Flynn (based in the UK) and it worked very well."

When Taylor first set eyes on the 17,000 ft^2 Power House it was being savaged by a wrecking team, filled with burning piles of rubble. Notwithstanding that, Metropolis' first job was to remove 1,000 tons of rubble from what was eventually a 6,000 ft^2 floor area. Perhaps the most remarkable thing about the design is that while it is admittedly visually dramatic it uses fairly simple materials. Langhan and Taylor firmly believe that the best studios are finished in wood. They were keen to preserve the warmth and liveness of feel that wood provides. Many of the other wall surfaces are bare rendering which, strangely enough, doesn't look as though the builders forgot to finish the job.

"We wanted the materials to be natural; we weren't going for terminal superficial trendiness. Although it has been highly designed it doesn't fall into clichés. There is glass but it is in the form of windows, not mirrors, marble or slate."

Another natural material used is steel—perforated with trapping behind it. Natural materials used imaginatively have transformed what could so easily have been traditional recording areas into visually exciting and inspiring spaces.

The main Studio A is a cluster of recording areas arranged in a horseshoe shape round the control room. Floor to ceiling windows make artist/producer contact much stronger and more immediate. A 'stone' room in fact features split blocks but has a similar effect ("quite tight, live and short"). Rather than a plain wood ceiling, the main recording area has suspended pieces of maple cascading from the ceiling down one wall ("armadillo-like scales"); this room is described as live and long. Finally there is a 'deadish' trapped area. Acoustics may be further varied by use of custom-built screens, which are either completely dead or completely live. One surface of perforated steel absorbs sound into hanging traps behind, while the other side is solid wood. Not so much a recording room with associated booths these are three distinct areas, which can be opened up into one large space or shut off completely—each a room in its own right.

The control room design is acoustically fairly conventional in that it is hard at the front (the aforementioned 19 mm thick glass) and becomes progressively deader towards the back of the room. Visually, however, it is again startling. It has to be left to the photographs to tell all as once again the materials used are a combination of shapes of maple, perforated steel, render and glass—impossible to describe.

Integral to the control room are two separate machine rooms—one for tape machines and one for amps. Sony *3324*, and Mitsubishi *X850* digital multitracks are available as well as Otari *MTR100* with Dolby *SR*. Metropolis have taken a bit of a stance here too: there is one rate which includes one tape machine—analogue or digital. So 75% of clients choose the Mitsubishi as it gives the most tracks on one machine and is providing 32-track digital for less than 48-track analogue. Taylor explains that they believe in digital recording and therefore want their clients to be able to avail themselves of it without having to argue with the record company about budgets

Studio A with cascading maple ceiling

in order to do so. Extra machines cost an additional £250 a day. The machine room has a separate patchbay and monitoring thus enabling it to operate independently of the control room, which is useful for copying, etc. Amp, machine and control rooms each have their own air conditioning circuits. Sitting on a raised floor section behind the console are the producers' tables doubling in their role of housing effects racks, which are naturally full of equipment. These also house MIDI matrices, video tie-lines, 8-channel foldback and RS buses linking all the rooms together. Not only that, the wooden tops slide out to form tables on which to set up keyboards.

Having stated that recording equipment design has been virtually stationary for the last 8 years and how it is impossible to do anything adventurous, Taylor had omitted to mention that the console in Studio A is a Neve *VR* with *Flying Fader* automation and monitors are Genelec *1035*s. Although there are many *VR* consoles up and running in America it has not been the obvious first choice in the UK.

Taylor explains: "The Neve console sounds great and has been around for a long time. The *Flying Faders* are fabulous, and incomparable with the SSL system in terms of use. You can get to grips with it literally in 5 mins. Menu driven with a mouse, you don't need to remember anything and can be off and mixing within 5 mins. It is not intimidating to use at all. And because it is moving faders there are no VCAs and it therefore sounds very good. We decided that if we were not going for an SSL or a Focusrite (which we did order before they had their problems) the only other option was the (Neve) *V*. It has proved to be beyond all our expectations. The console itself is very flexible and therefore a bit complicated but it is a great success.

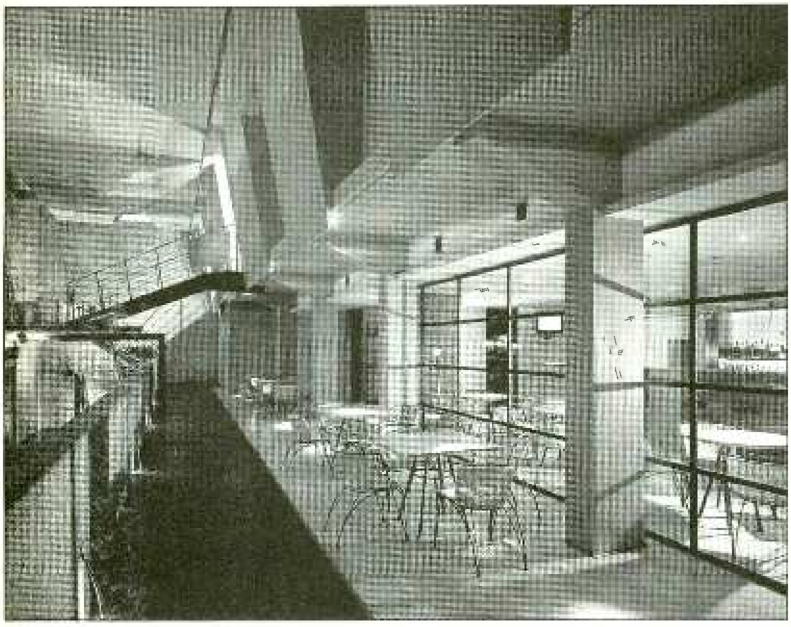
"Working with a new console is a big step but most people are prepared to take it. Because we are producers ourselves we have always felt we could break a console. Obviously the first few months were difficult but a lot of people are very keen. We just needed to unlock the door."

Studio B on the other hand features the more predictable SSL *4000G* with *Total Recall*. "We didn't feel it was possible not to have an SSL. We are not trying to knock Solid State Logic, after all it dominated for 8 years, which is an incredible achievement. But we tried not to dictate the options—we want to make them all available. When the project is finished we aim to have, at the most, two SSLs in five rooms plus a Focusrite or a Neve. There has been much more concerted demand for Studio A than B but you can't gauge how much of that is the console and how much the room. Nevertheless there aren't many Neve studios in the UK."

There aren't many that feature Genelec *1035*s as their main monitor either. "We very much liked the nearfield *NF30*s, which sound fabulous. They are about the only viable choice if you want a lot of volume. They give fantastic mid and high frequency detailing, which to some extent can be a bit of a problem—anything harsh or sibilant gets emphasised—but the fact is that if it sounds good on those it is guaranteed to sound good outside the studio. You don't need to work with 6,000 W pointing at your head; it's fun sometimes but not necessary. The fact is that if the monitors sound good people will use them at low volumes. Genelecs are very expensive but then we have a lot of expensive things here. The power amps and crossovers are built together with the monitors as an integrated system. They look like a softdome design but are in fact hybrids."

Situated above the control room and overlooking the recording areas is a private relaxation area for the exclusive use of Studio A clients. It features black suede furniture (the colour scheme throughout the complex being blacks and greys) and suitably styled TV and hi-fi equipment. There is a kitchen (although you don't need it as there is a very good restaurant and bar within the complex), a dining area and a shower. Plenty of privacy here for conducting business and interviews, simply lounging around or keeping an eye on what is going on downstairs in the studio. There is also plenty of scope for your video, as several clients have already proved.

Studio B is a slightly more compact version of Studio A. Access to the control room is via the rest area, which again features lounge/dining area/kitchen and shower. Large doors in the rear control room wall open up to integrate the lounge area and create one large room if desired. Although designed more with overdubbing than backing tracks in mind, Studio B nevertheless features three recording rooms, which again can be



Restaurant area and walkway to the three new studios

opened up into one large area. The main room has a maple floor and rendered walls. Opening the doors up on either side renders the acoustic medium live as there are absorbent side walls in each of these rooms which keep the reflections down. Closing the doors turns it into a very live space. The ceiling once again features suspended maple, this time likened to a space ship coming in to land. It all sounds a bit freaky but isn't when you see it.

The control room is slightly more conventional than in Studio A but nevertheless features floor-to-ceiling glass windows through to the recording areas, and liberal use of perforated steel and render. The front floor section is maple and once behind the console it is carpeted. The rear wall is bass trapped and finished with foam covered bolsters, which are blue for a change.

Machines and amps are housed in separate areas off the main control room. In addition to the multitrack formats mentioned earlier, there are ATR stereo ½ inch or ¼ inch machines available as well as Mitsubishi *X86*. Taylor and Langhan prefer the instant response and audio quality of ATR machines over the more usual analogue machines and went to a certain amount of trouble to locate these secondhand models. "They are very quick and they sound fantastic. Lots of the tools people use to make great records are not new. We've got things like the Fairchild compressors, valve microphones, Pultec EQs."

The outboard equipment racks are housed in trolleys, and flightcase racks enable artists to set up keyboard equipment with minimum fuss and spaghetti as the MIDI matrices, etc, are all ergnomically placed.

As well as the racked equipment there are additional floating items available at an extra charge (see equipment list).

The Victorian exterior of this extraordinary facility is striking enough with its massive curved windows and ceramic surrounds. Once through the doorway you are confronted with a tangle of metal ramps and walkways, which are strangely not completely out of place in this Grade II listed building. Designed by William Curtis Green (of Dorchester Hotel fame) at the age of 21 it was owned by London Transport and provided the power for the West London Tram System from 1896 to 1911.

Funnily enough it wasn't the 20 flats above the studio that proved a 'change of use' problem but the fact that the power house was formerly subject to a time restriction on site after 7pm. The Metropolis team purchased subject to planning. It was to be 18 months before their battle with the local council was finally won, so although purchased in September 1986, building did not commence until May 1988.

"When the complex was first conceived," said Taylor, "it was thought that there would be a 20% increase in studio rates by now. What has happened in fact is that rates have fallen 20%. It was very difficult to cost and is equally difficult to make sense of it economically. If you keep it full enough it can make some sense but there is not much room for cock-ups. There is something to be said for keeping on a really high plateau—it keeps you on your toes. The Townhouse, with whom we are really competing as we are in the same area both geographically and marketwise, they have got years of client base and respect. We have to try harder than them, which isn't necessarily bad for us.

"I have every reason to believe things will happen in the studio market. We are in a better position to survive and I am afraid we are likely to lose some of the competition. It's tragic for the industry but there is no other end to the ridiculous rate cutting that is going on. Discounting will have to end and I think we may even see a rate increase. The situation is very serious in London. There has to be more resolve among studios or there just won't be a recording industry here."

Metropolis then is a bid by young music business professionals to make the industry sit up and take notice. Coming from a traditional large studio background they value things like training, tape ops, engineers, continuity and incentive—things that all too many studios have been content to see going out onto the freelance circuit.

"We might prove to be the last of the dinosaurs," says Taylor, "but I'd quite like that."□

Metropolis Studios Ltd, The Power House, 70 Chiswick High Road, London W4 1SY, UK. Tel: 01-742 1111.

Equipment

Console Studio A
Neve *VR* 60-channel console with *Flying Faders* mix cue and tape machine automation, *Recall* and 8-way mixable cue system

Console Studio B
SSL *4000G* with *Total Recall*, 52 mono and four stereo channels, 8-way mixable cue system

Tape machines Studios A&B
Mitsubishi *X850* mk 3, Sony *3324*, Otari *MTR100A/90*, ATR stereo ½ or ¼ inch, Mitsubishi *X86* digital, Sony *PCM 2500* DAT

Monitoring Studios A&B
Genelec *1035A*, Yamaha *NS10*, Auratones, AR *18LS*, Electro-Voice *Crystals*, Rogers *LS7*

Reverbs Studios A&B
EMT 140 valve echo plates, Lexicon *224XL*, Yamaha *SPX90* mk 2s, Lexicon *PCM 70s*, AMS *rmx16*

Compressors, EQs & gates Studio A
GML *8200* stereo EQ, Klein+Hummel *UE400* stereo EQ, Tube-Tech *PE 1A* valve EQ, SSL stereo compressor, UREI *1176LN* compressors, dbx *263* de-essers, dbx *160* and *165* compressors, Tube-Tech *CL1A* valve compressor, Drawmer *201* gates

Compressors, EQs & gates Studio B
GML *8200* stereo EQ, Focusrite *ISA 115 HD* stereo EQ, Tube-Tech *PE 1A* valve EQ, SSL, stereo compressor, UREI *1176LN* compressors (2), dbx *263* de-essers (2), dbx *160* and *165* compressors (2), Tube-Tech *CL1A* valve compressor, Neve compressors, Drawmer *201* gates (2 pairs)

DDLs & digital FX Studio A
BEL *BF20* flanger, AMS *SDMX* 6.5/6.5 stereo lock in, Lexicon *Prime Time*, Roland *SDE 3000As*, Roland *Dimension D*

DDLs & digital FX Studio B
BEL *BF20* flanger, AMS *SDMX* 6.5/6.5 stereo lock in, Lexicon *Prime Time*, Roland *SDE 3000A* (20)

Noise reduction & misc Studios A&B
24-channel Dolby *A* or *SR*, four Dolby *361*s with *A* or *SR*, Hinton MIDI matrix, Atari computer with Notator sequencing, Akai *S1000* 32 sec stereo sampler, Fazioli grand piano

Microphones Studios A&B
Neumann *TLM170, U87, U47* FET, *KM84*; AKG *414, 451, D12, D112*; Shure *57, 58*; Beyer *M201, M88*; Calrec *CM1051*; Sony electrets; Sanken *CU41* (with or without transistors); Schoeps *CMC5*; valve mics including pairs of Neumann *U47, U49, U67* and AKG *C12*.

Additional equipment hirable
Lexicon *480L*, Lexicon *224*, Yamaha *REV7*, *EMT 244*, Yamaha *SPX1000*, AMS *dmx 15-80 S* 6.4 sec, Eventide *H3000*, TC *2290* ddl, Roland *SDE 3000A*, Pultec valve EQ, Pultec mic amp, Tube-Tech mic amp, GML mic amp, Fairchild *670*, UREI *1176LNs*, Drawmer *201* gates, Ortofon *STL732* de-esser, Aphex *Aural Exciter Type C*

YAMAHA C1 MUSIC COMPUTER

Mike Collins reviews Yamaha's new music computer, the C1, and takes a look at an accompanying sequencing program, the CSQ1

This article is based upon 4 months use of the Yamaha *C1* computer on a regular basis between other work in my programming room. Jim Corbett (Yamaha UK's MIDI specialist) called in several times to help, and Roger Evan (MIDI computer specialist) from Computer Music Systems in London called in to demonstrate the Voyetra Sequencer and Score. Several other musicians, programmers and engineers all had opportunities to try things out and pass on their comments, all of which were invaluable.

General features

The casing is black plastic and a lid swings open to reveal the LCD display on its under-surface. The keyboard is fairly standard and there is a numeric keypad to the right, with the two data-entry sliders to the left. Overall, the *C1* is stylishly designed and would look very much at home in any hi-tech environment. It is portable but has no carry handle and no 'feet' on the underside so you always have to lay it flat when you put it down. It is also a mains-powered machine, so you won't be able to use it to write your letters on a plane or train.

The *C1* uses an Intel *80286* microprocessor as its central processor unit running at a 10 MHz clock rate. The *80286* is a powerful IBM *PC AT* compatible processor and an excellent choice. The basic operating system, as with all IBM-compatibles, is MS-DOS. This has a small operating system overhead, thus allowing properly designed programs to achieve greater speeds of operation. (An example of such a program is the Voyetra Sequencer, which is now available on the *C1*.) One of the limitations of MS-DOS is the 640 kbyte memory space. The *C1* overcomes this limitation by providing a 512 kbyte RAM extension area, which may be expanded up to 16 Mbytes.

In order to improve the *C1*'s performance when sending or receiving data via the MIDI port, the two in and eight out UARTS are implemented in a special MIDI LSI chip, which makes MIDI data transfer to and from the *C1* potentially much more efficient than with other computers. This is one of the several areas where the *C1* aims to score over its competitors, at least for musical applications.

On the assumption that the sequencing application will be the most popular program, the *C1* has a dedicated timer for most of the music-related application programs to use. An Intel *8254* chip is used to provide two application timers: the A-timer generates an interval of up to 131.07 ms and can be used for any purpose defined in an application program, and the B-timer can be used to generate a basic MIDI clock interrupt of 1/480 quarter notes.

Still with musical applications in mind, Yamaha have anticipated that some data is better manipulated via continuous sliders, eg tempo, volume, general-purpose data entry for synth patches, or whatever. Obviously this feature will be very useful as long as a particular software package takes advantage of it, which not all can. On the *C1*, the slider's positions are scanned and converted into 256 steps by a sub-CPU and fed to the *C1* via a parallel interface chip, the Intel *8255*.

This same sub-CPU is used to generate and read SMPTE timecode. However, Yamaha have decided that the *C1*, as a musician's tool, will not be used for its SMPTE facilities as often as for its MIDI facilities, a fair enough assumption in the case of those many musicians who do not synchronise their MIDI performances with tape machines or video machines. The *C1* is not intended to servo-control tape machines, nor to provide 'gen-lock' or 'jam-sync'. Neither does it have the ability to compensate for round-up errors when reading code originally generated by other timecode readers/generators (as several of the more recently manufactured SMPTE units will do). The assumption here is that the *C1*'s SMPTE reader will be used in a Yamaha-only world, not in a world where other such equipment exists, or has been used previously on a particular project. So only basic read/write SMPTE operations are supported on the *C1*, however, it has all the common timecode formats (24, 25 and 30 frames/sec, and 30 frames/sec with dropframe), which makes the *C1* more than adequate for the most common syncing tasks.

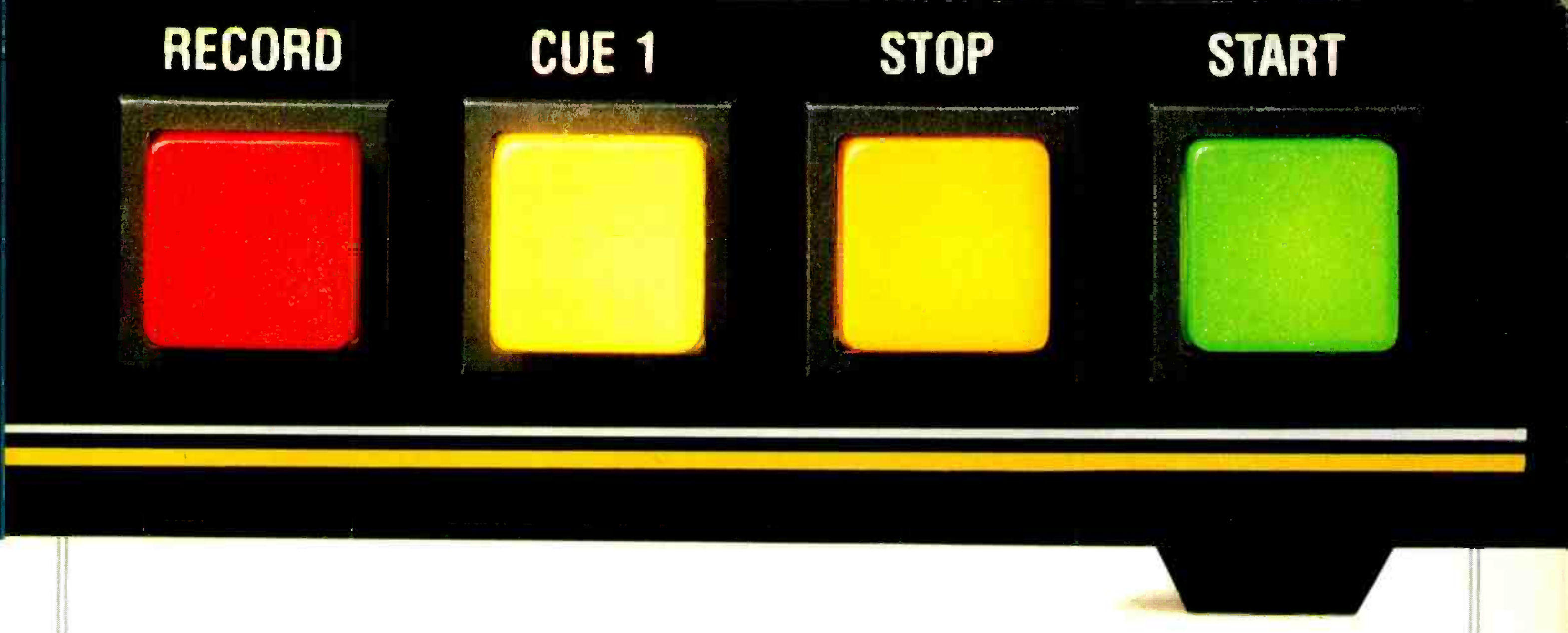
Getting started

With the help of a couple of friends who were conversant with MS-DOS, I quickly put together a set of batch files that drew a menu selection on the screen as soon as the *C1* powered up. Five choices were available to run either the Yamaha Sequencer, or the MIDI Monitor, or Bulk Manager, or Bacchus *DX/TX Voice Editor/Librarian*, or Microsoft *Windows*. Simply by typing a number between 1 and 5 for the menu selection and pressing ENTER, the correct program would be launched—much easier than entering MS-DOS commands! Upon exiting any program the batch file would take you back to the menu display to let you select another program to run.

Unfortunately, the Sequencer program would not exit back to the menu for some reason, even though my expert friends assured me that the batch files were set up correctly. Then I discovered that after exiting the Sequence program, the Bacchus *TX802 Editor/Librarian* would not load, with an error message appearing to warn that there was not enough memory left. I discovered later that the MetaWindows drivers and their corresponding screen and MidiPort drivers remain in memory even after you quit a MetaWindow's program. These can consume up to 130 k of memory, and this is what caused the problems when I tried to run another memory-intensive program—the Bacchus Editor—after the sequencer. The simple solution was to turn the *C1* off, then back on but this does seem to be a rather inelegant way round the problem!

CSQ1 sequence program

Yamaha see the *C1* as a successor to the *QX1* hardware-based sequencer. They anticipate the primary use of the machine will be to run sequencer software, with the added bonus that you will be able to run other IBM-compatible ▷



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◁ software on the *C1* if you wish. Obviously, the *CSQ1* sequencing software supplied with the machine deserves a very close look, as many potential users of the *C1* will want to know if it will serve their needs adequately.

Operational aspects

The program has one main display screen and three types of editing window, the Numeric 'Event Edit' window, the Bar Graph window and the Master Track window.

A *CSQ1* 'track' is a basic song unit—a musical part played by one instrument on one MIDI channel which typically would extend for the whole length of the song. There are 400 tracks, of which you can replay up to 200 simultaneously. There is also a special mastertrack, or 'conductor' track, for tempo and meter changes, available via menu selection.

You can create up to 1,024 patterns in the sequencer. These are 'floating' tracks, like *QX5* 'macros'. Again, these contain one musical part—long or short as you like—that may be played back by specifying the location within any track (ie bar number/beat number) from which you wish playback of your pattern to begin. This feature allows you to put together any musical parts (say, drum patterns, or any other type of repeated sections) by building them up from sections written into the sequencer as patterns. These would then be 'chained together' by putting a message in, say, Track 1 to start the first pattern playing from Bar 1/Beat 1. If the first pattern lasted for say, eight bars, you would then put a message in Track 1 to start playing your second pattern from Bar 9/Beat 1, and so on. To play the complete drum track, you would then just put Track 1 into play mode and hit START. If a single pattern is used more than once in the same or different tracks, you will save on *C1* memory space and your programming time by using these repeated patterns.

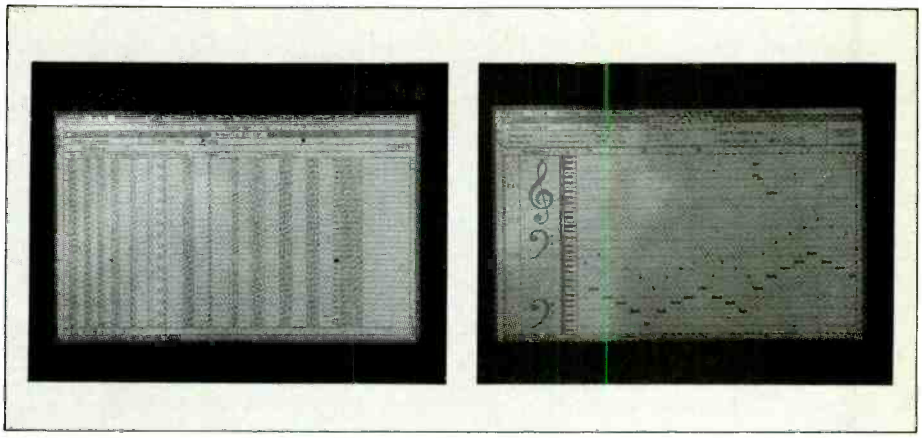
You can create up to 128 MIDI macros. On the *C1* (in potentially confusing contrast to the terminology used with Yamaha's *QX5*) a MIDI macro is a string of MIDI data that can be transmitted as part of a track's data (eg a SysEx message to control some special feature of a synth). You can record SysEx data directly into a track but the macros may easily be re-used in another song or another part of the song.

Step recording

Having spotted the musical symbols printed on various keys, I decided to try the Step Recording function first. I had once experienced problems sequencing a complex piece of music that had various unusual note divisions on the Steinberg *Pro 24* sequencer, so I decided to check out the *C1*'s ability to accept nested tuplets via step entry. The *C1* proved to be perfectly capable of accepting an 11-tuplet with the second note in the group subdivided into a septuplet. The musical symbols on the computer keyboard proved useful here and I was reassured to find that the *C1* out-performed many other sequencers, particularly those on the Atari, in this respect.

Realtime recording

I transferred four tracks from Performer sequencer on the *Macintosh* computer to the *C1* in one pass and was pleased to discover that they transferred perfectly first time, with no apparent delays on the received data. In contrast, I have transferred data into Performer from various other sequencers and have had to requantise afterwards because delays of up to several clocks occurred at random on various parts of the data.



Main display screen showing Numeric 'Event Edit' window and Bar Graph window

Next I recorded in an accelerando section of music from an existing Performer sequence and found that the audible results in the *CSQ1* were entirely accurate—the piece transferred perfectly, complete with the series of tempo changes I had previously entered into Performer's Conductor track. The tempo data in the master window of the *CSQ1* looked different due to the limitations of MIDI clock resolution but the results were perfectly acceptable, giving a graduated rise in tempo from 86 bpm to 160 bpm in the space of two bars of music.

I then tried recording a keyboardist playing a solo Fats Waller stride piano piece, *Alligator Crawl*. The first thing I needed to do was create a MIDI clicktrack because the internal *C1* click could not be heard over the sound of the synthesised piano, so I recorded a track with four bars of ¼-notes playing a high C on one of my expander modules. I then wished to repeat these four bars for the duration of the piece. Unfortunately, the program does not allow you to loop an individual track for this purpose. If you instruct a sequence to loop, it loops all the tracks within that sequence. This is not the best way to implement a loop feature and I cannot understand why Yamaha chose to do it this way. So I looked for some sort of quick way to repeat the four bars of click to create data to fill the track—most sequencers allow you to do this. The only way to do this in the *CSQ1* Sequence program seemed to be either to copy and paste repeatedly until the track was full, or to define the first four bars as a pattern and then insert instructions to call this pattern every four bars throughout the length of the song. Either way a tedious process.

Once this was sorted out, I recorded the keyboard performance into the sequencer. The performance was very good but not absolutely perfect, so the next obvious step was to 'spot' edit any individual notes that were 'wrong'. I am very used to the Numeric editing window in Performer on the *Mac*, so I tried the *CSQ1*'s Numeric Editor first. Editing the Numeric window in Sequence was similar to using Performer, with a few exceptions. For instance, if you wish to correct the timing of a note that was played a few clocks before a beat, you click on the bar location, enter the correct bar number, then press ENTER. At this point the screen gets redrawn, quite slowly, as the information is updated. You then click on the beat location within the bar and correct this. Again, when you press ENTER, the screen gets redrawn. And again after you enter the clock location. Frustratingly slow. Performer is much quicker at this and, for instance, if you were at Bar 10/Beat 4/Clock 456, and wished to correct this to Bar 11/Beat 1/Clock 000, you would only have to increase the clock value of 456 to 480 (the maximum number of clocks per ¼-note) to advance the note into the next bar. Much quicker!

Disappointingly, the *C1* would not let me take this short cut. Also, although there is an Undo function in Sequence, which is available each time you make an edit, there is no Redo function, as on Performer, to allow you to compare quickly between edits. Neither could I find any simple instruction in the Event menu to allow you to insert a note into the Numeric window. Any other type of MIDI event can be inserted here, including bytes of MIDI data, so you could, in theory I suppose, write a note by specifying the actual MIDI bytes in the numeric list. But I don't know of anyone who would prefer to do this.

Standard features

The *CSQ1* allows you to define a sequence as a 'pattern'. You may then insert this pattern into any track at any point. The existing data is shifted forward in the track to accommodate the new data. I tried this and found that the existing data was shifted forward to start from the end of the newly inserted pattern but the data that had previously existed 'under' the new pattern continued to play even though it did not show up in the editing displays. This seemed to be a bug. Talking of bugs, the *C1* and its software did seem to be relatively bug-free. Although one or two mysterious crashes did occur. This does happen occasionally with the very best software on all the computers, so the *C1* did appear to be pretty much OK in this respect.

I recorded several tracks of music data into the sequencer just to see what the overall program control features were like. One of the first things I wanted to do was solo a track and check it timing against a click. I discovered that there is no way to hear the click when a track is soloed. I was very frustrated by this, and surprised as virtually every other sequencer in existence allows you to do this. The next surprise came when I was playing several tracks back and wished to temporarily mute one or two to listen to the others better. There is no way to do this in the *CSQ1* Sequence program without stopping the song from playing first. Again, most other sequencers, including those made by Yamaha, allow you to do this simple thing. What happened? Did they not ask the guys who designed the *QX* sequencers what to do?

Sequence will record, SysEx dumps of up to ▷

◁ 10 kbytes directly into any track. This is useful for recording patch dumps into the actual sequencer tracks rather than using program changes. The benefit here is that when you replay the tracks, you do not have to worry about putting the correct patches into your synthesiser, the sequence does this for you. SysEx messages may also be typed directly into a track via the event editor. Macros containing collections of MIDI SysEx messages may be created via the edit menu and can be used for a variety of useful purposes. MIDI Bulk Data files created via the Bulk Manager, or even via a wordprocessor or other program, may be selected from disk via a file menu option. Once selected, such a file will be transmitted exactly as is, to your MIDI device. These SysEx capabilities are more comprehensive than on most other sequencers and allow for very powerful control over your MIDI setup, especially if you have mastered the use of the Bulk Manager software.

Special features

A feature I particularly liked is that when a controller is listed in the Event Edit Display, the name of the controller is also listed in parentheses—very handy for speedy identification. Also, I found the Bar Graph window to be quite useful. Here it is easy to insert notes, and edit their timings, lengths, pitches or velocities using the mouse (however, that is no defence for not allowing the user to insert notes easily into the Numeric window). I would have preferred to have seen vertical grid-lines as well as horizontal ones on this display, like on the Voyetra Sequencer, as I feel that these would have made the display easier to work with. I also found scrolling the bargraph display very awkward, due presumably to the low screen refresh rate of the C1.

Other unusual features of the *CSQ1* include Trill, Tremolo and Arpeggio functions. A simple arpeggio function used to be provided on many older synthesisers but most newer models do not have one. It may be thought to be redundant on a sequencer as arpeggios can be created in step record or by playing in manually but I feel that this is a very useful additional feature. I have not seen trill or tremolo features on any other sequencer and would personally find these very useful.

Another feature I have not seen elsewhere is the Tempo LFO feature available on every track. The idea here is to simulate the timing discrepancies that would occur between a group of live musicians, in order to better simulate a live performance. I experimented with this feature and found settings of 15 for the depth and 2 for the rate to give fairly usable results when simulating a 'ropey' rhythm section but I was not sure if the control was sufficiently fine to enable you to program the very small timing differences between, say, the members of a string section playing in unison.

A Rhythm Note Assign Table is provided. By changing the settings here, you can change the transmitted notes without changing the track data. Great idea to allow you to quickly change note-mappings for use with drum-machines. I have had this feature on my 'wish-list' for a long time now.

There are some very useful 'advanced' features in the Sequence program. You can set up an initial Program Change message to be sent from any track whenever playback begins. You can also set an initial MIDI volume message (using MIDI controller 07) to premix the balance of any MIDI sound generating device that accepts these messages, or set an initial pan amount (using MIDI controller 10), or set up an initial MIDI macro containing whatever MIDI messages you like to be sent at the beginning of a track. You can do all these things (apart from create macros) in any MIDI sequencer but the advantage of having a special place to enter these also serves to remind you to actually do these things—and it is good programming practice to set things up like this initially.

One of the best of the more advanced features on the *CSQ1* is the 'chase' facility. If you jump to a particular rehearsal mark on most sequencers, you often find that the sequence should be playing a different synth patch at that point. But you have not played the section before so the synth still has a patch selected from an earlier part of the music. Also, if the last part of the sequence you played prior to jumping to this new section finished with a pitch bend, or a control change message, without resetting these, you will find that they still apply to the new section, which you almost certainly don't want them to. Sequence checks the prior parts of the sequence and automatically chases up the correct patch changes and controller changes to make sure that you always hear the right thing when you jump to a new section in this way. Brilliant!

File formats

The *CSQ1* supports Yamaha's proprietary E-SEQ file format for MIDI sequencer files. This is the format used on the *QX3*, so files from a *QX3* will transfer directly. Older *QX1* files may be transferred to the *QX3* first, then saved as E-SEQ files and subsequently transmitted to the *CSQ1*. Thus an easy upgrade path is available from *QX1* to *QX3* to *CSQ1*. Unfortunately, Yamaha have not provided the OpCode Midifile format, which has become virtually standard on Dr T's *KCS* and on most other major sequencers running on Atari and *Macintosh* machines, which would allow easy transfer of *CSQ1* sequences to other sequencers.

Conclusion

The *CSQ1* Sequence software has the makings of a first-rate sequencer program. There are a number of well thought-out and useful advanced features, which I wish were available on other sequencers: the Chase feature, the Rhythm Note Assign Table, the initial setup messages, and so on. There are several unusual features you won't find elsewhere, such as the Tempo LFOs on each track and the Trill and Tremolando functions. The MIDI control features are probably more comprehensive than on any other sequencer available. Although several of the most basic features, such as solos and mutes and individual track looping, did not seem to be properly designed to give the functionality that most sequencer users have come to expect. In practice, comparing it with Performer on the *Macintosh*, I found that the *CSQ1* was frustratingly slow to use in certain areas, such as 'spot' editing individual notes in the Numeric window, although it was 'rock-solid' in its timing. If Yamaha can sort out the problem areas with this program, it could well become a very popular sequencer for the *C1*. In fairness to Yamaha this is the first version of this sequencer software and we can anticipate further revisions in the future, which will surely improve the functionality of the program. ▷

Main display

The main display is divided into sections including:

Position Control block: upper left portion of screen. Here you can:
- Specify locations for recording and playback
- Set rehearsal marks
- Click to jump to mark
- Set Loop On/Off: use for drum machine-style recording onto a track
- Set Punch In/Out locations

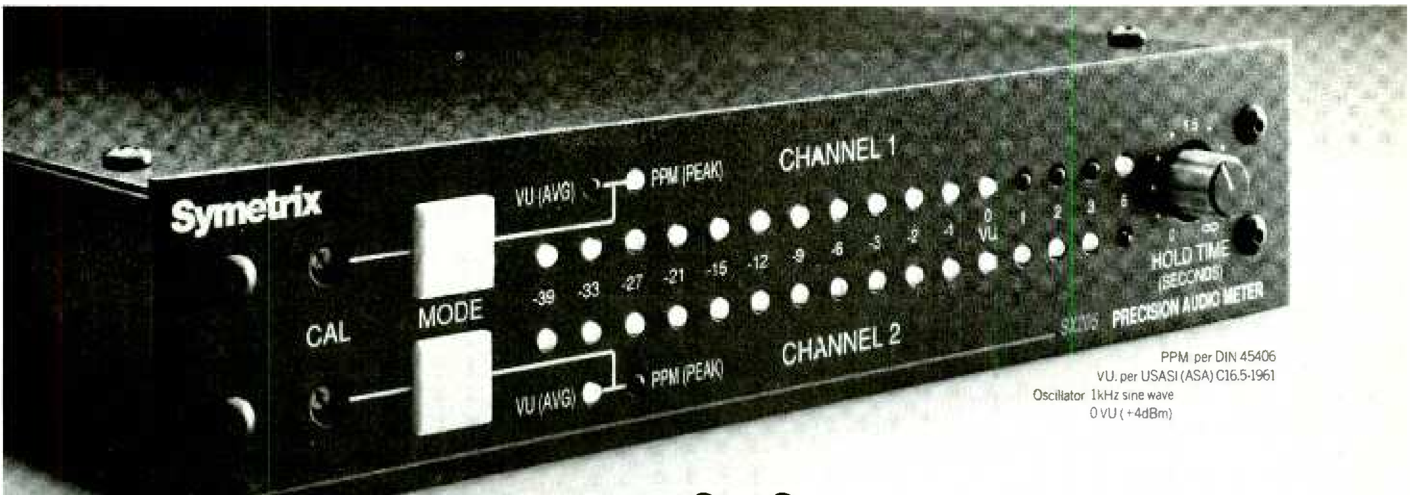
Recorder block: upper middle part of screen. Tape recorder-style controls: REWIND/STOP/PLAY/PAUSE/RECORD. Click On/Off. Function keys may be used as alternatives to the mouse

Recorder Control block: upper right part of screen. Here you set:
- Countdown on/off, 1-9 measures
- Clock source internal/MIDI input/SMPTE/MTC/Tap on space bar to set tempo
- Clock Out: MIDI/MTC
- Repeat: on/off
- Rec Mode: replace/overdub

Status block: along top of screen under menu bar. Three display items:
- Filename: shows name of song you are working on
- Port 1/Port 2: shows if MIDI is coming in on any port
- Tempo Rec: shows if you have selected Tempo Record to record a tempo map

Master Track block: middle display. Mastertrack on/off
Here you can alter tempo data anywhere in a song without opening the mastertrack window. Time signatures and rehearsal marks and measure numbers also displayed here. You can quickly select editing regions covering the Master Track and all the sequence tracks from here in units of measures

Track Control block: bottom half, right two thirds of display screen
The top line of the track data area allows you to select four types of data to display:
- **Track data** shows marks horizontally, displaying notes as dots, chords as stacks of dots—just to indicate that data is in a track. Scroll horizontally with the bottom scroll bar to look along the track. Scroll vertically to look at the next track, or those off screen. Click on ≪**'track data'**≫ to get:
- **Track attribute display** allows you to modify timing offsets for each track, transposition, polyphony and initially transmitted messages for each track
- **Track comment** for 32 character comments
- **Velocity meter** this bargraph display is an unusual feature that meters the strengths of Note On velocities being transmitted during playback from each track. It can sometimes be useful to have such a visual display to let you know that note data is being transmitted from a track at a particular point in time



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- Pull-down menu commands from mouse or keyboard
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- Auto punch-in/out
- Realtime and steptime record
- Both measure/beat/clock and hour/minute/second/frame displays
- Set and jump to 52 rehearsal marks

The philosophy behind the C1's design

The philosophy behind the design of the *C1* was outlined by Tak Nakata and Hiro Kato at the 85th AES Convention, Los Angeles, in November 1988. The hardware was designed by a Mr Miyamori and the software by Messrs Yoshika, Makita, Hirakata and Kawasaki. In their presentation they defined 'musicians' as "composers, writers, performers, 'sound patchers', music students or music hobbyists". They go on to list the typical uses they expect the computer to be put to. Sequencing, display for visual editing of sound patches, sequencing data and scores; integrated control of MIDI equipment, connectivity to printers, modems, MIDI, or SMPTE equipment; with business usage as a secondary goal.

A lap-top configuration was chosen on the assumption that the computer would not normally be a permanent installation in a music studio but would be used by musicians travelling from studio to studio, or going 'on the road'. Thus the size needed to be compact and the MIDI and SMPTE jacks, display screen, keyboard and disk drives all needed to be integrated into one package for simplicity in setting up and taking down. The *C1* succeeds very well in this goal.

The hardware designers thoughtfully anticipated that memory-hungry musical applications would need to use the *C1*'s extended memory as a cache for music data. "One way to access this area under MS-DOS would be to use a RAM disk utility. However, normally, access to the extended memory is done in such a way that the *80286*'s 'real mode' is turned into the 'protected mode' to extend the span of the address space from the limit of 1 Mbyte to 16 Mbytes. To get back to the 'real mode', the CPU resets itself to restore the original state, using information stored in the calendar RAM (which has battery backup) before the mode was switched. During this complicated process, the interrupts are disabled. Obviously, when frequent interrupts are feeding MIDI or SMPTE data into the system, accurate time scheduling of such data could not be guaranteed."

So, another method for accessing the memory area above 1 Mbyte had to be devised and the solution the *C1*'s designers came up with was to provide various Direct Memory Access (DMA) channels. The AES paper informs us: "One DMA channel is dedicated to accessing this extended memory area. In addition to the normal single-transfer mode available on the Intel *8237*, the *C1*'s memory-to-memory DMA provides a CPU priority DMA mode, where a single-word transfer is only requested every 32 μs, which allows the CPU to maintain a good balance of bus use between multiple MIDI ports accessed by the CPU in the foreground and DMA transfers to and from the extended memory area in the background."

This means that, unlike normal *PC/AT*-type RAM cache utilities, access to the extended memory does not compete with interrupts, so interference with musical events is minimised. Obviously, this gives the *C1* distinct advantages over other IBM compatibles (which have extended memory available) when running musical applications.

"Although the 31.25 kbyte/s baud rate of the standard MIDI interface is reasonably fast when using just a few MIDI devices, it is not sufficient for sending time-division-multiplexed data (MIDI data is transmitted on different channels via Time-Division-Multiplexing) to 16 MIDI devices if accurate high-speed concurrent performances are required. So, eight separately-addressable MIDI Out sockets are provided on the *C1*, enabling up to 128 MIDI instruments to be separately controlled if desired." In normal use, however, it is intended that only a few MIDI devices be run from each output, with the advantage that higher timing accuracy can be achieved this way. This is a great idea, which Yamaha originally used on their flagship *QX1* MIDI sequencer. More manufacturers should take note of this.

One MIDI Thru socket is provided, which is entirely adequate, and two MIDI In sockets are provided, which Yamaha say they believe is sufficient for the most demanding MIDI applications. I feel they have perhaps not considered this particular requirement carefully enough, nor looked closely enough at the competition, or they may not have made this particular statement. It would certainly be true to say that two inputs are sufficient for most applications. However, MIDI interfaces for the Atari and the *Macintosh* computers have been available for some time with four mergable MIDI inputs. The *Synclavier MidiNet* has eight mergable inputs. You may ask "Why?" but the reason is simple: many current sequencers allow simultaneous recording onto separate sequencer tracks/MIDI channels so you can record the performances of more than one musician simultaneously, just like a traditional music recording used to be (and sometimes still is) made—with several musicians interacting musically to give a more cohesive performance, which may or may not be improvised. It is quite conceivable that four, or more, musicians may wish to record simultaneously and that MIDI-controllable mixing or effect data may need to be recorded as well. In such cases, the most demanding ones admittedly, more than two mergable MIDI inputs would be required. And where would such instances most probably arise? Why, in a recording studio, of course. But then the *C1* was designed as a personal computer for musicians, who Yamaha obviously anticipate will record their performances into the *C1* individually, or at most, in pairs, and for these applications the *C1* is perfectly adequate.

The AES paper points out that one frustrating aspect of MS-DOS machines is the lack of a solid standard user interface framework. This is no fault of the *C1*, it is simply a fact of life in the IBM-compatible world, and may be contrasted with the philosophy behind the *Macintosh* design, which places the *Macintosh* standard user interface at just about the top of its list of priorities. Various commercially available window packages (GEM, Windows, MetaWindows, etc) are widely used with IBM-compatibles to provide user-friendly graphical operating environments, so Yamaha decided not to develop another such graphics layer in the 'Toolkit', apart from the minimum requirements needed to take into account small differences between the Yamaha PCDC chip and other graphics chips.

For people who wish to write their own 'applications' or programs, the *C1* provides a 'ToolKit' containing low-level software MIDI drivers, accessible from MS-DOS, so that application programmers can incorporate these into their programs, or at least use them as templates.

The AES paper describes an IBM-compatible, excellently equipped as a Musician's Computer, with many more on-board musical features than on any other personal computer made so far. But people don't buy computers for the hardware capabilities *per se*, they only buy computers which will run the software they wish to use. Virtually all the software people who have designed IBM-compatible music programs have started to port these to the *C1*. Even if you cannot get a particular IBM music program to run on the *C1* today, you will be able to soon. Obviously, this high level of music software support for the *C1* should contribute enormously to making it a successful machine. I will be most interested to see any new programs developed especially for the *C1* to use its unique features, such as the sliders and music data entry keys. The first of these is the Yamaha *CSQ1* Sequence program, which comes free with the machine, along with the MIDI Monitor and Bulk Manager programs.

Conclusions

The *C1* represents Yamaha's second attempt to produce a music computer. A number of people will recall the unfortunate *CX5*, a much cheaper computer based on the unsuccessful *MSX* 'standard'. The *C1* is a much more serious offering than the *CX5* but have Yamaha got it right yet? Well I believe they have got quite a lot right but I do have a few reservations. For a start, I think that they should have used an *80386* processor, or provided an easy upgrade path to this, because it won't be all that long before the *80386* becomes the processor no serious IBM-compatible computer use can do without. I put this point to Jim Corbett of Yamaha, and his reply was: "The forthcoming Microsoft Windows 3 operating environment is rumoured to make extensive use of the extended memory of *80286*-type machines. So, as any Microsoft Windows 3 based music software would seem to have a long term future, being able to make good use of the powerful multitasking features of Windows 3, Yamaha felt that the *80286* was the best choice of processor for the *C1*, especially as an *80386* would have added significantly to the cost."

The choice of an IBM-compatible machine was a reasonable one for Yamaha to make, especially as there is plenty of well-established software in the marketplace already. The *CSQ1* sequencer includes several exciting new features but the screen refresh-rate is frustratingly slow, and a few very important basic features are missing—like the ability to loop tracks individually, or to turn tracks on or off while a sequence is playing. Yamaha may well sort these problems out in the future via software revisions but this could take some time. Consequently, the Voyetra Sequencer is a much better choice if you want a reliable and 'mature' sequencer software package right now.

If Yamaha intend to see their computers in use in studios, I feel that they ought to consider the practical needs of recording engineers more carefully. The *C1* would obviously be a good choice 'on the road' but for studio use, a more powerful multitasking central processor and operating system that would allow several programs to be used concurrently, and a large screen would be much more desirable. In a studio situation it would be very handy to have sequencer, patch editors, sample editors, MIDI monitor, and possibly other programs more readily accessible at all times. Having said all this, the *C1* is a powerful machine with plenty of the right features built in, such as the SMPTE facilities, the MIDI interface with its eight separately addressable outputs and the extra processors, which handle various low-level tasks leaving the main processor free to handle the more important tasks.

I would have no hesitation in recommending the *C1* to any musician who would like an IBM-compatible, portable, music computer. The *C1* is in a class of its own here, with plenty of first-rate popular music software available right now. With the right combination of hardware, operating system and software, future generations of the *C1* (or its offspring) could become the 'Number 1' music computer in the world.□

Specifications

General: Laptop 10 MHz IBM *PC-AT* clone. Based on Intel *80286* microprocessor with special additional circuitry; slot available for *80287* maths co-processor (speeds up some software such as Sample Vision sample editor, and some maths-intensive business software)
Memory and storage: 640 k main and 512 k extended memory. 64 k ROM. One 720 k 3.5 inch 2DD FDD floppy disk plus one 20 M hard disk. 64 k of VRAM. High-capacity external hard disks available from many other manufacturers
Display: Backlit supertwist LCD with 640×400 resolution. External video CRT output for IBM CGA (Colour Graphic Display) colour and IBM *Hercules*-compatible monochrome display monitors
Extra features: Two data control sliders, music symbols available on keyboard for music notation and sequencing programs
Interfaces: Two MIDI Ins, one MIDI Thru and eight separately addressable MIDI Outs. SMPTE In and Out jacks. External video signal and power jacks. One Centronics compatible parallel printer interface. Two RS232C serial ports (allowing you to connect a mouse and still have a terminal available for a modem or other peripheral). Toshiba-compatible expansion bus. An extension Memory Board is available which can be installed in the expansion slot to increase memory capacity.
Items included: MS-DOS 3.3, RAM disk, MIDI Monitor and System-Exclusive Bulk Manager, Yamaha *CSQ1* sequencer, Yamaha mouse
Size: 15½×3¼×15 inches (whd)
Weight: 18 lb 12 oz

MIDI'S FRONTLINE

The BBC Radiophonic Workshop has now added two digital music studios of a radically different design. Mike Collins and David Mellor report on all aspects of the new studios from MIDI routing to working there

The theme from TV programme *Doctor Who*, space age music, sound effects and unusual electronic music of all types. That is probably what is conjured up in most people's minds here in the UK whenever the BBC Radiophonic Workshop is mentioned. Of course, if you are the sort of person who checks all the credits attached to a BBC radio or TV programme, you will already be aware of at least a part of the vast range of music and sound effects created at the Workshop over the last 30 years or so.

Originally, the techniques used for music composition and realisation at the Workshop borrowed heavily from so-called 'Musique Concrète', popular among electronic composers in the 1950s, which involved the use of tape splicing and looping to manipulate sounds for musical purposes. These days, digital synthesisers and samplers are used to achieve similar ends. The interesting thing is that the Radiophonic Workshop has stayed in the forefront of the latest developments in the musical and sound world since its inception.

Throughout the last year or so, an exciting new development has been taking place at the Workshop leading to the design and installation of a virtually all-digital studio, with centralised computer control over much of the equipment. One of the Workshop's composers, Peter Howell, has spent this time working closely with development co-ordinator Mark Wilson to plan, design and build what they see as the ideal electronic studio in which to compose and record music for BBC programmes.

A composer's studio

Over the years, a fairly standard layout has developed for the equipment in the Workshop's six studios. Usually, a large mixing console is positioned in front of a pair of loudspeakers and a video monitor, with keyboards and racks of ancillary equipment to the sides. The problem with this layout is that it is awkward to operate the equipment while watching the video and listening to the sound balance. Also, the various items of equipment, which usually have both audio and MIDI connections, need to be reconnected in different configurations quite frequently. This can be time consuming and often interrupts the 'creative flow'.

As well as problems with the layout and interconnection of the equipment, it often turns out that there is a need to go back to pieces of music other than those currently being worked on. As anyone involved in this type of composition and recording will be aware, it can be difficult to say definitively that a piece of music is 'finished', even as the transmission date approaches.

Although MIDI composition makes altering a piece of music easier than it would be if musicians had to be called back for re-recording, it is still necessary to reprogram all the sounds that were used, recreate all the original effects and duplicate the mix. In addition to improving the ease of operation of the equipment, the new design had to make it as straightforward as possible to rework pieces of music according to the changing requirements of programmes in production.

Brian Hodgson, head of the Radiophonic Workshop, explained the sequence of events leading to the opening of the new studio. (Then) newly available equipment such as the Yamaha *DMP7* digital mixing console (a compact 8/2 console with motorised faders, every parameter controllable via MIDI), the *Akai DP3200* electronic switching matrix and the Apple *Macintosh II* computer were making new ventures possible.

"In October 1987, Peter Howell returned from a period of leave from the Workshop with fresh ideas about new ways to arrange studio equipment and wished to try his ideas out in the design of a new studio for Elizabeth Parker (one of the Workshop's composers). We had been looking at software-based 'virtual' mixers and had started to evaluate the Yamaha *DMP7* for use as a submixer. Peter suggested that the new studio could incorporate *DMP7*s if the noise levels proved to be acceptable and the idea of using the Akai routers also emerged at this time. We originally planned to use one *Mac Plus* to control the routers and one for the music programs but when the *Mac II* became available, we realised that it could not only do the job of two *Mac Plus*es but it could do it faster and provide a larger screen as well. We also realised that the computer could be used for virtually all the control operations necessary in the new studio. By April 1988 we had set up an experimental studio layout using pieces of wood cut out and laid on top of filing cabinets, and using *DMP7*s on loan from Yamaha. The *DMP7*s proved to be adequate for our needs, so we decided to proceed in this new direction and at the end of November 1988 the new studio became operational with Peter Howell composing and recording in there so that he could make sure that everything was working OK before handing over to Elizabeth in January 1989. From my point of view, one of the most important achievements with the new studio design was that we had built a fully automated studio for the same amount of money which it would have taken us to build an old-style room around a conventional mixing console."

Although the first studio to use the new design went to Elizabeth Parker, Peter Howell now has a similar studio that incorporates all his previous ideas, plus several improvements. The development period of the design is now close to an end as the consolidation period starts. ▷

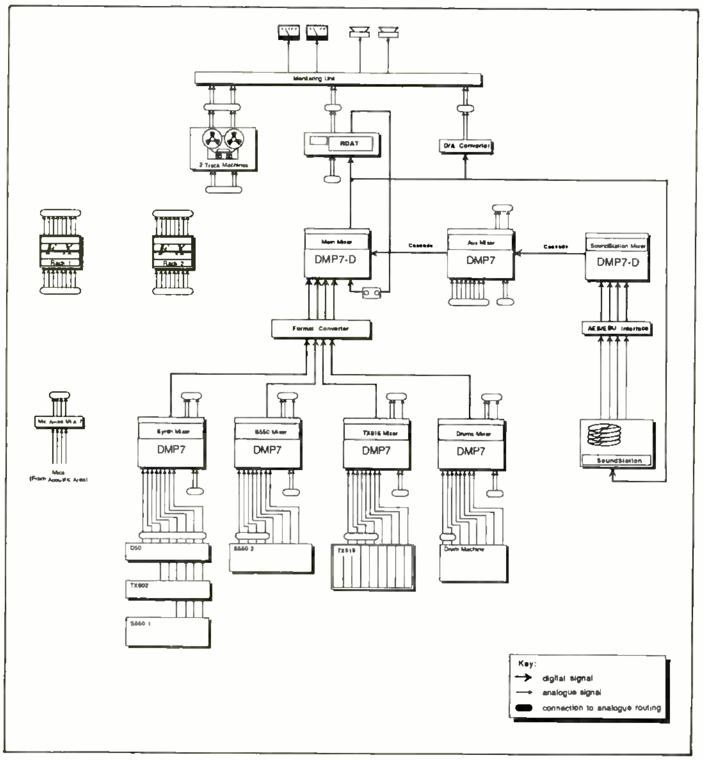
Studio layout

The Radiophonic Workshop's new studio design uses a wraparound layout, with woodwork by cabinet maker Jeremy Quinn, which allows the composer to sit in a central position facing the video and audio monitors. A Yamaha *KX88* MIDI master keyboard is placed in front of the composer. On a shelf above it is the *Macintosh II* keyboard plus mouse. The *Mac II* monitor is slightly to the left, while two *DMP7*s (one *DMP7-D* with digital inputs and outputs) are also close to hand a little to the right. Five tracks of synthesisers and effects are mounted either side of this central position within easy reach. On the lower level are four *DMP7*s and a *DMP7-D*, which act as submixers for the MIDI equipment, effects and the DAR *SoundStation II*.

A custom-built monitoring system was designed and built by in-house engineer Ray White, to provide monitoring facilities for the tape machines as well as a talkback facility for a live recording area. These facilities are not available on the *DMP7*s. Video synchronisation is provided via a computer system based around the BBC microcomputer called *Syncwriter*, which was developed by Jonathan Gibbs several years ago, and is still in use today (see *Studio Sound* April 1985 for an in-depth review of *Syncwriter*).

Underneath the working surfaces are cupboards containing three Akai audio routing matrices, providing 72×72 switching crosspoints, *SycoLogic* MIDI routing matrices and other associated items. Easy access for maintenance is provided via large panels, which may easily be detached. All the wiring, apart from connections to the multitrack, monitors and tie lines to the live area, travels around the circumference of the horseshoe-shaped table, with slots cut into the rear of the working surfaces for cable access to the equipment.

As a result of the modular design approach, the whole of the main studio installation could (at a pinch) be lifted in sections and moved out of the room without having to remove cabling from under the floors or in the walls.



Studio B: Audio distribution (main signal paths only)

Digital connection

Although the studio is not entirely digital, it probably comes closer than most other studios to the ideal of having all the devices talking to each other in the digital domain. In the early stages, an analogue connection was used but this proved slightly less than satisfactory due to noise problems.

Even in the days of digital synths and samplers, their outputs are still in analogue form. Each rack of synths, drum machines and effects has its own *DMP7* submixer with links to and from the Akai routers for extra versatility. The *DMP7-D* on the lower level is used to mix digitally the outputs of the DAR *SoundStation II*. The digital cascade outputs on the four *DMP7*s on the lower level feed via a Yamaha format converter to the digital inputs of the master *DMP7-D*. The DAR's *DMP7-D* cascades digitally through the auxiliary *DMP7*, mixed if necessary with signals from the router, into the master *DMP7-D*. The digital outputs of the master *DMP7-D* go directly to a Sony *PCM 2500* R-DAT, which is the master stereo recorder, and to a D/A converter feeding the monitor.

Since, basically, this system has only two buses, recording on to the multitrack, the DAR, takes a circuitous but logical route. It is often necessary to transfer material from the master R-DAT to the DAR, hence the connection of the R-DAT's digital output back into the master *DMP7-D*.

Although the digital and analogue audio paths are working well, synchronisation of the various types of digital equipment is more of a problem. Ideally, the main source of machine sync would be the timecode track on the video, translated to MIDI Timecode to run the sequencer and DAR *SoundStation II*. Unfortunately, the DAR only accepts SMPTE/EBU code as yet, which means that there has to be a video running all the time to sync sequencer and DAR, even if the composer does not want to see it (as he or she might not if working on a radio programme!). An alternative is to record timecode on track 8 of the DAR and run the sequencer from that.

Digital sync is also a problem as the digital sync source may be either the R-DAT or the DAR, signals travelling in either direction. These are, of course, problems that exist at the moment and will not prove insoluble. A few problems is the price you pay for being first in the field.

Cue card

The controlling software for the equipment operates, on the *Mac II*, under an 'umbrella' program called 'Cue Card' written by Mark Wilson and Peter Howell. Cue Card is a Hypercard program and is run under 'Multifinder', the *Macintosh*'s application switcher program, so the sequencing and patch librarian software may easily be switched to and from while Cue Card is in use. (Hypercard is a *Macintosh* application that contains a simplified *Mac* programming language, Hypertalk.)

Why had Mark Wilson chosen to use Hyperecard rather than any other programming language to develop this system? "Hypercard is the sole reason we were able to put the new studio together. Had we had to develop the program from scratch using C or Pascal, the development time would have been far to long. Instead we used Hypercard to its limits, using many external commands written in Pascal or C. I also used the MacDraw, MacDraft and Diagram Maker software to plan and draw up the studio furniture and equipment layout, once I had settled on the way it should be done. I worked very closely with Peter Howell to establish exactly what this layout should be, as well as to establish exactly how the software control of the studio should be handled. Having a composer and a software writer develop the system together in this way really helped us to achieve a result which we believe works extremely well in practice.

"With this new system, the main control of the studio is achieved using the *Macintosh II* from a central position. All the audio and MIDI routing, the loading of required sounds into the synthesisers, the control of the seven *DMP7* digital mixers, as well as the music sequencing and notating and the synthesiser editing can easily be achieved using the *Mac*. One of the beauties of this system is that all the data relevant to a piece of music can be stored and retrieved from one folder containing just one Performer and one Cue Card file."

Howell provided a guided tour of the new studio, pointing out all the new features.

"The first thing you will notice here is that there is not a cable in sight and the studio is incredibly tidy compared to most of the others. This is because we use electronic routing for all the audio and MIDI connections."

He demonstrated how quick and simple it was to re-route some of these connections using the Cue Card software; just a couple of mouse clicks on screen and new connections were made.

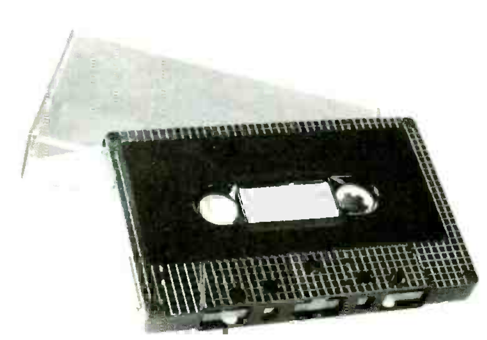
"Before, there was much more to learn and remember, as well as more actual moves to make in many cases. And now we can easily write macros containing several repatching instructions, which will then be carried out at the click of a mouse or press of a key, and this speeds the process up even more."

Howell explained that originally the program was thought of as an electronic *Filofax* into which you could enter details about each piece of music being worked on. Hypercard seemed ideal for this type of work, and besides the title of the programme, starting and completion dates, and so on, there would usually be a list of particular timings at which music would be required. These are called music cues and this led to the name Cue Card. The program has been further developed so these cue entries on Cue Card may be clicked on with the mouse to reveal further data cards from which the synthesiser sounds, mixer setting and routing setups can be stored and retrieved via MIDI for each Cue.

"In this way, the music for any cue may be completely rebuilt at some later date. This was not really possible to anywhere near this extent previously. So we now have a virtually total recall system in operation."

PRODUCT PROFILE

PRODUCT PROFILE



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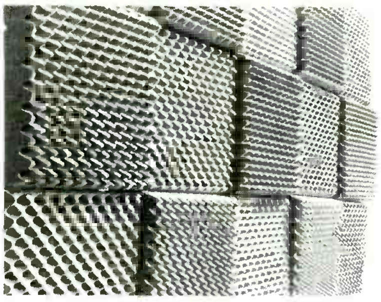

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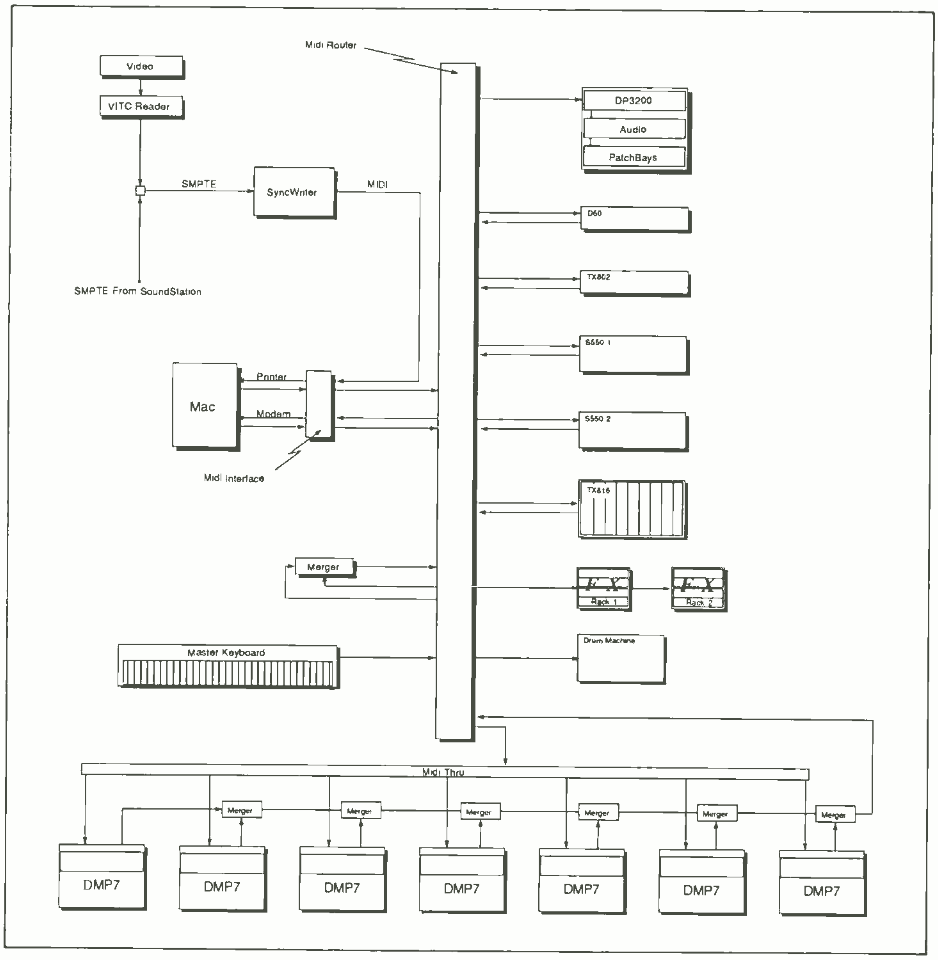
Although the new studio may seem complex, the complexity is very much underneath the surface, allowing the composer greater freedom and flexibility, while having to pay less attention to the needs of the equipment.

Hodgson: "This new system gives the composer more time to try out ideas, more time to think about the composition. Cue Card removes the technology barrier to a great extent, to allow the composer to spend more time with the music, while the *Macintosh* copes with the technicalities."

It is clear that the Radiophonic Workshop are paving the way for the future in their wholehearted adoption of computerised tools to aid their work. They have opted for a modular system based round 'off-the-shelf' components currently available from Yamaha, Akai, Apple and others. They have added customised hardware and software, using their in-house expertise, to achieve an extremely user-friendly and flexible studio system with an ideal ergonomic layout to suit their particular needs.□



Studio B: MIDI distribution

Cue Card software

There are three main Hypercard 'stacks' (as individual Hypercard programs are called), which comprise the Cue Card system of studio control programs. These include the 'Control' (Home) stack, the 'Router' stack and the 'Cue Card' stack. Ancillary stacks are provided to give Phonebook and Diary utilities within the Hypercard 'environment'.

Access to the applications environment is extremely easy: either a click on the appropriate icon on the Control Strip, or a menu selection.

This control strip is an excellent feature, and contains icons which launch Performer; Composer; the Op-Code Editor/Librarians; the Cue Card stacks; the Hypercard Router stack, which allows the composer to re-configure the audio connections; a diary for session notes; or a phonebook containing a list of useful telephone numbers. Every Hypercard program has to have a 'Home stack', which may be thought of as a sort of master control section from which you may access all the other stacks. The Cue Card 'Home' stack is known as the Cue Control stack and contains the global information about the studio as well as the Hyperscript MIDI in/out routines for all the MIDI functions.

When you click on the Performer Icon, or one of the other non-Hypercard applications, the Hypercard window drops below the bottom of the *Macintosh* screen, just leaving the control strip visible. The application windows then have a large screen area available but you can still move quickly to anywhere else, such as to the Cue Cards, using the control strip, which is now at the bottom of the screen.

To launch the 'Cue Control' stack, you point to the appropriate icon on the screen using the mouse to move the screen-cursor and 'click' the mouse button. This gives the Cue Card title page containing general information on the current project.

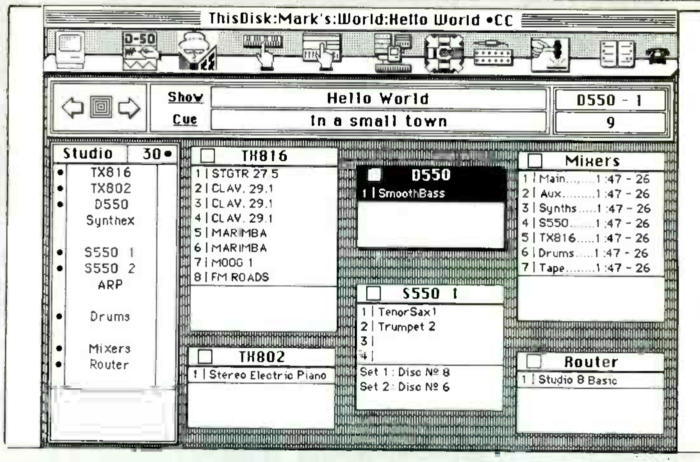
At the start of a project you would click on the Cue List button to access the Cue List card and enter details of the music cues supplied by ▷

the producer of the TV show.

If you click on one of the numbered buttons to the left of the cues, you will be presented with a window, which allows you to store all the data relating to that particular piece of music. This data would typically include the synthesiser sounds used, mixer settings, and so on.

To the left of the screen is a list of all the MIDI-controllable devices in the studio. A click on the name of any of these devices will open up a window to the right of this selection area. A menu is available with various selections to allow you to save or load the memory states of any or all the devices within the studio setup. Typically, you might edit your *TX816* sounds using the OpCode editor and save these sounds as Cue Data here. Similarly, you would set your audio patches up using the Cue Card Router stack and, again, save the settings for your project here. For a project with several music cues you would repeat this process for each cue.

Various sets of information relating to each project will be needed by the different departments within the BBC, so a stack is provided to allow assembly of the different sets before printing them out. These would then be supplied to the production office, the Radiophonic Workshop office, the video dubbing mixer, etc. The necessary information is automatically gathered from the project stack and inserted into the various categories, which need to be printed out. Even the length of time worked on each project is automatically logged by the computer and recorded where this information is required. This saves a greal deal of time that would otherwise be spent in compiling the administrative documentation.



Cue Card data: Typical display

Computer control of audio routing in the studio is one of the most innovative aspects of this system. The Router stack may be opened in the usual way by clicking on its icon on the Control Strip. This icon (on the right hand side of the control strip, just before the diary icon) looks like a traditional audio patchbay, to help the user to identify its function at a glance.

There are various icons in the main Router display for the different MIDI devices in the studio. To make connections via the Router, you click on the icon for the device whose outputs you wish to route, and then on the icon for the device to whose inputs you wish to route the outputs of the first device. For instance, if you want to connect a synth to a *DMP7*, you click on the synth's icon and a small window containing a list of all the synths pops up. Similarly, you would then open up a window with a list of the mixer inputs. If you then select one of the synth outputs and one of the mixer inputs by clicking on their names in these lists, this will automatically insert the names into a strip above the display area. The first named device's output may then be connected to the second named device's input by clicking on the 'Intercity'-like icon displayed between these names.

Peter Howell of the BBC Radiophonic Workshop is currently developing a Sound Index stack to provide an overview of all the sounds he has available in the different synths and samplers. This is a database program that allows the user to attach descriptive 'tags' to the name of each sound to allow them to be sorted appropriately when looking for, say, all the marimba sounds. The location of each sound, whether in ROM in a synthesiser, or on a cartridge or floppy disk, is listed also.

For instance, you could find all the sounds you have described as useful 'lead' sounds, and once the search and sort process was complete, only this selection would be displayed on screen.

Finally, a special Studio manual was written using Hypercard to provide on-line help for new composers using the studio. It becomes easy to see why Hypercard has proven to be so useful, when one considers that it can be used for all these different applications, once it is customised appropriately. And, with the simplified programming facilities available, you do not have to be a professional computer programmer to do this work yourself.

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Elizabeth Parker interview

You have been using the new studio since the beginning of last year. Would you tell us about your initial impressions?

Terrific. Once I got used to the new ways of working! It was very hard work for the first few weeks and I had to take manuals home with me every night. Although it seems relatively easy now, at the time I remember saying that it was just about the hardest thing I had ever had to do. One major difference in the way I find myself working in the new studio is that instead of recording to multitrack as I am building the piece up, I can make much greater use of samplers, thus retaining the ability to make quite drastic changes, if they are required, right up until the final stages of mixing-down to R-DAT. I find that I'm making very heavy use of sampled sounds at present. For instance, I'm part way through assembling a new library of suitable samples for a BBC programme about the life of Christ and I am taking these from the BBC sound effects library, as well as from the Roland Factory samples. I sometimes end up sampling sounds which I have constructed using different synthesisers, such as the PPG, just for the convenience of having all my sounds available from one source in the sampler.

How long did it take before you felt comfortable with the new system?

Well, when I took over the studio in January, Peter and Mark were still putting the finishing touches to the software, and there were still one or two minor bugs which needed to be sorted out; but I was able to start recording my first serious piece of work after about 3 weeks. I had to learn how to use the *DMP7* mixers as well as the new Cue Card system, the routers, and so on, as well as getting used to the new studio layout. I found Cue Card to be the simplest part of the system to learn how to use—it's just so easy!

Are you recording mix data from the DMP7s in realtime into your Performer sequencer or do you mainly use the DMP7's stored 'snapshot' mixes and recall these via MIDI Program changes?

What's a 'snapshot' mix anyway? I always record the *DMP7* moves into Performer so that I can get the mix to 'play' back immediately. It is only with the advent of this new studio system that I have been able to do everything using just the samplers, synthesisers and the *DMP7*s and mixing straight to R-DAT, without necessarily using the multitrack, and not feel that I am losing anything. I now have total access to the sounds I am using at all times. Previously, once things were on multitrack my options for further manipulating these sounds were limited. For instance, I could apply EQ and rebalance the levels but to reshape the sound more drastically meant re-recording.

Could you make some brief comments about the 'Mark of the Unicorn' and OpCode software which you use?

Performer works just fine for me as a tool to aid me in my work. From that point of view I simply could not live without it now. I find the numerical event-list editing totally acceptable as a way of working, now that I am used to it and I don't personally feel that there would be any greater advantage to me in using a sequencer with a graphical or conventional music notation editing system. I particularly like one of the new features of the latest Performer upgrade (2.41) which lets me hear any note I select in the editing display.

As far as Composer is concerned, I only use it very occasionally. This is partly because I find that the user-interface is not too friendly and partly because it often won't cope with the more awkward time signatures or key signatures which I use. I still like to write out parts of my compositions by hand anyway, because I somehow feel that this lets me put more of my own personality into the music.

I don't actually edit my *DX* and *D50* sounds very extensively, so I don't make very heavy use of these editor programs. However, when I do need to edit these synths, these programs are a real boon compared to having to edit from the synthesiser front panels with their tiny displays.

I would like to point out that I regard all these things as tools and I don't find myself getting terribly excited about the latest software updates, or whatever—if the technology helps, that's great but it's the creative music-making which turns me on.

Do you use live instruments much in your recording work?

This depends on the individual piece. As it happens, I have just recorded a signature tune for the *Everyman* programme featuring a viola. I recorded the viola player in a corner of the new studio, using screens to provide acoustic separation from the audio monitors and the recording was totally successful. Obviously, for something more ambitious than one viola this would not be practical, so I would use one of the main BBC recording studios here at Maida Vale and bring this recording into my studio on R-DAT.

What would you like to be able to do in your studio which you can't do at the moment?

I would like a VITC system which works 100% reliably. We do have a VITC system at present but there are some problems with it and I do feel that I need the ability to move slowly up to particular video frames and stop on the exact frame where I need to place a particular effect. It would be nice if this worked properly all the time.

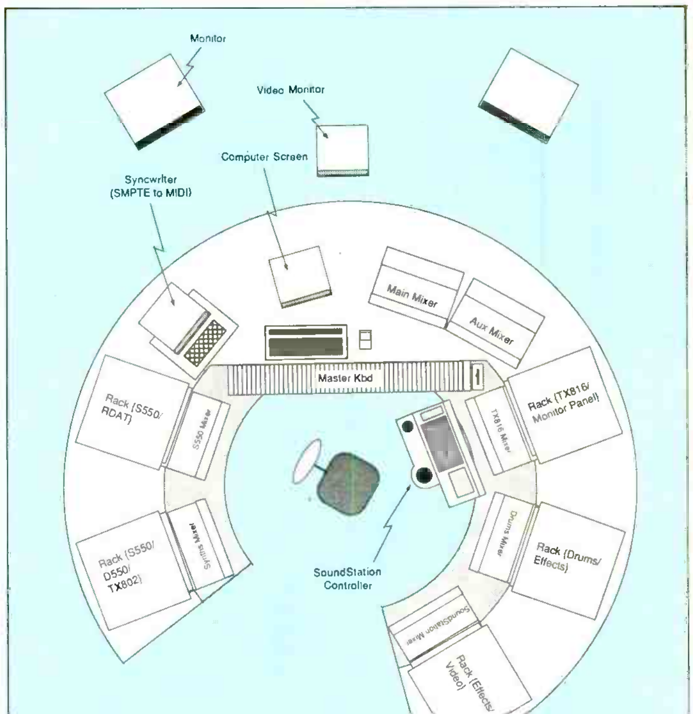
I would like to have access to a large central sampler library. I realise that it would be a mammoth task putting such a library together for all the composers here to share but it would be great. I would also like to have longer sample times available on the Roland *S550*s which I am using, say 30 seconds or so. With enough sampling time available, I could dispense with the multitrack altogether.

I would very much like to have improved sequencing facilities and I am currently looking at Vision sequencing software from OpCode, which offers the ability to use different time signatures on different staves, for instance.

Also, I wish that the panning on the *DMP7*s worked more effectively and reliably. Maybe I am asking too much of my *DMP7*s by sending them too much data to process but I have had some very unsatisfactory results when using panning effects on these. But apart from these 'niggles', I am very satisfied with the facilities in general.

Do you find that the new system has made things easier, or faster, or is the main benefit the virtually total recall which you now have?

It is more complicated now but the advantages of total recall far outweigh the extra work involved. Yes, I would say that the total recall is the main benefit—it is a much neater, tidier, way of working. I do find myself doing more things with the sound, now that I have three effects units available in each *DMP7*, as well as my outboard effects units. Obviously it takes longer if you make use of the many more possible ways to control the sound but in the end, this helps me to achieve better results. On the other hand, I have lots of preset routings for the various common operations which I have to carry out, and these are now available at the touch of a button or two. This makes re-routing much faster than it was using manual methods. As a result, I can afford to take more time doing more specialised tasks which would not have even been possible previously, and still finish my work within similar time-scales.



Studio B: Physical layout

Barry Fox

More publicity for Nicam stereo, 20 bit debate, Sleeveprint go into CD production

Responding to criticism over failure to publicise the milestone launch of Nicam digital stereo on September 11th, the IBA waffled: "...feeling confident that the IBA has done everything within its resources" to publicise the fact that the IBA transmitters in London and Yorkshire covering 30% of the population became "fully operational for Nicam digital stereo" on September 11th and that Channel 4's entire sound-in-sync distribution network has been modified for stereo. The IBA also passed the buck to the TV companies, saying "it is up to them to promote the programmes that are available in stereo".

The next day Michael Grade, chief executive of Channel 4, spoke at the 'Images of the Future' conference in London.

A questioner from the floor asked about Nicam and why the IBA, Channel 4 and the ITV stations had done so little to publicise the milestone launch.

"What is Nicam?" said Grade. "It's not a phrase I have ever heard."

Other panel members then whispered in Grade's ear telling him that Nicam was the stereo service his Channel had supposedly been offering for the last 2 months. Grade then dug himself deeper by publicly blaming the IBA for not providing the necessary equipment.

The man from the IBA—who had just posted a letter saying he felt confident that the IBA had done everything in its resources, and that it was up to the TV companies like Channel 4—then had to stand up and tell the tittering audience that Grade was wrong, because 100% of the Channel 4 distribution network has been converted to Nicam operation, transmitters covering 30% of the population were already operational and there would be 75% coverage by the end of 1990.

Trying to cover for Grade, but making things even worse, Geoffrey Owen, chairman of the Television and Radio Industries Club, said he didn't want people to get the wrong impression; of course Michael Grade knew what "Neecam" was, he just hadn't understood the question.

Nothing I could ever say or write says better than this how badly the IBA has fallen down on the job of educating not just the public but even the Heads of Channels about Nicam.

What was needed, just prior to the September 11th launch, was a press conference at which all the issues surrounding Nicam and stereo broadcasting were explained, and the system demonstrated. Even now it would not be too late for the IBA to hold one.

Remember all the publicity for Virgin's first-in-the-world in-store CD pressing plant, at the Megastore in London's Oxford Street?

I wrote about it a few times, noting that the monoline system seldom seemed to be working. Often it was throwing half-finished discs into a scrap bin. As Nimbus pointed out, when they installed monolines, you have to have a side chain or buffer to hold half-finished CDs if a later stage of the line develops a fault.

Well the Virgin line has now gone, needless to say without any publicity. It has been replaced by a travel sales counter for Virgin Atlantic airlines.

After 2 days of phone calls I finally found out what happened.

After a lengthy shut down, caused by unidentified cracks in the metallisation chamber, Virgin jumped at an approach from Fleming Mercantile. This is an investment trust, also known as London Wall Investment and part of the Robert Fleming Merchant Bank group.

Fleming did not actually buy the CD line. They "financially engineered" (their words) the purchase of the CD line from Virgin on behalf of a company called Sleeveprint in Bedford. When the line left Virgin it went to Cornwall to be refurbished and was delivered to Sleeveprint in early September.

Sleeveprint has previously been in the business of printing record sleeves for the music industry. Recently they decided to go into the manufacture of vinyl LPs, cassettes and compact discs. Sleeveprint's philosophy, which makes sense, is that some of the major record companies have moved out of LP and vinyl production too early. Managing director Roger Masterson says that by the end of September Sleeveprint had started pilot LP and cassette production, with "a few teething problems".

The Virgin CD line is being run by a separate company inside Sleeveprint called Music Manufacturers. The men behind this company are Roy Matthews and Nick Flower. Those with long memories will recall how Matthews, an ex-EMI tape man, was talking about setting up a CD plant right at the start of the CD era.

When I spoke to Roger Masterson he was clearly unaware of the kind of problems Virgin had encountered with the in-store line. But, said Masterson, the Virgin line has been modified and should be up and running sometime towards the end of 1989. It will offer short run services for the smaller record companies.

Masterson is obviously not used to dealing with the audio press. He would not talk any more unless I promised to let him vet what I wrote prior to publication.

"I have no problems with print magazines over this," said Masterson, refusing to say how much he had paid for the line and how it had been modified. "They do a nice presentation story, then fax me a copy before publishing it."

The prospect of Sleeveprint getting the Virgin line up and running and producing saleable discs within 6 weeks of installation has since produced chortles from Brits and polite bewilderment from Japanese. More to the point, where will Sleeveprint get the masters cut and stampers made for pressing the short runs? And what will the turnround time and cost be if Sleeveprint employs Telemedia Sonapress in Germany or Digipress in Paris to do the job?

As I was not prepared to let MD Masterson edit my text I never got the chance to ask this vital question.

The next hot topic looks like being 20 bit recording. There have been some muddled and misleading press releases. These I believe and hope are the facts.

In March and April 1989, Sony was secretly experimenting with a 20 bit DASH format (*Studio Sound*, September 1989, p. 82) using a modified *3402* in the EMI mobile. The system had big box external 20 bit DACs working at close to 20 bit accuracy, and 20 bit code went onto tape.

Before CBS had a chance to issue the sessions (Berlin Philharmonic with Dietrich Fischer-Dieskau singing Mahler and the London Philharmonic doing Mozart's *Requiem* at Walthamstow) on the 'Classical Sony' label, Hyperion had got in first with a CD recorded by Tony Faulkner with 20 bit DACs made by Cambridge company Data Conversion Systems. DCS had developed 24 bit converters for radar and digital radio systems and then used their experience for 20 bit audio use.

The Hyperion CD, of the new London Orchestra conducted by Ronald Corp, playing Poulenc and Hahn (CDA 66347), is probably the first commercially available disc made with a 20 bit system. Deliberately the record sleeve note makes no mention of this fact. The idea was to see if the public could hear the difference, without being told that there was a difference to hear.

But the Hyperion CD is really only a half-way house. The 20 bit signal from the DCS DACs was crunched down to 16 bit code, at the session, for recording onto a Sony *1610*.

Probably the first all 20 bit session with commercially available equipment (but *not* the first ever 20 bit recording as is claimed by DCS) was on July 3rd, at Walthamstow Town Hall when Faulkner used DCS DACs and a Mitsubishi *X-86* to put 20 bits on tape while recording the Royal Philharmonic under André Previn playing a string of Beethoven Symphonies including the *Ninth*. The *X-86* is the only recorder available 'off the shelf' with the ability to put 20 bits on tape, although the DACs that come with the *X-86* are only 16 bit capable. Hence the use of DCS DACs.

The 20 bit signals on the *X-86* tape will of course have to be crunched to 16 bit for CD release on the BMG Classics label.

I suspect we will now see a lively debate over whether it is best to put 20 bits on tape and then crunch to 16 bits at the mastering stage, or convert from 20 bits before taping, or even if there is any real benefit in going to 20 bits at any stage of the game.

One point to bear in mind is that some 16 bit DACs may in fact be working at only 15, 14 or even 13 bit accuracy. This makes quiet music sound grainy. More quality is lost when the studio tape is transferred to a master disc from which CDs are pressed for sale to the public. With 20 bit DACs there is a better chance of genuine 16 bit operation.

MONITOR SYSTEMS-

ROOMS AND PHASE RESPONSES PART SEVEN

Phil Newell considers the role of loudspeaker design in relation to different rooms and phase responses

From the point of view of accurate studio monitoring, loudspeakers and the rooms in which they will be installed cannot be designed in isolation. They are part of one and the same system. This is true from the point of view of phase and directivity responses, drive unit positions, sensitivity and many other parameters.

It is frequently overlooked that the listening room has an enormous effect on the loading of a loudspeaker. This loading in turn, affects electro-mechanical impedances, back EMFs, resonances and many other things. If one climbed into a large loudspeaker cabinet, reversed the drivers, used the box as a small listening room and the room as a cabinet, the loudspeakers themselves would perform in their usual manner, only the listening environment would change. Subsequently changing the room would probably affect the perception from within the cabinet. Accepting this, it is much easier to believe that the room itself. 'the big loudspeaker cabinet' must now be taken fully into account when considering loudspeaker loading and performance. Even changes in air humidity, and hence density, will affect this loading. Except in a climatically controlled environment, we will always see some minor day-to-day performance shifts.

Very few people make a habit of listening to loudspeakers in anechoic rooms, or true freefield conditions, why then do we measure them in such environments? With modern gating techniques, all measurement can be accomplished in real circumstances. Anechoic chambers and freefields do not represent realistic environments for the loudspeakers. Remember the room is a cabinet and if that cabinet is not typical, the loading, transient and resonance responses will not be typical. Obviously. the big problem is in defining just what does constitute the average room. be it control room, or domestic lounge. The truth is that loudspeakers are still dependent on so many compromises that no absolutely definitive set yet exists.

It has been suggested by Martin Colloms, among many others, that the compromise points in loudspeaker design would differ depending upon whether the source material were digital or analogue, classical or rock, or even upon the use of near or far microphone positioning. For example. for digital programme, extended bass response could override small colouration compromises, whereas on analogue material, one may not require the extended bass, but would not wish for any additional colouration as the analogue signal would already contain more colouration than the digital signal. Remember that no loudspeakers are anywhere near perfect, so compromise is a matter of fact. Furthermore, loudspeakers designed for radiation into a nominally 2π space such as when mounted flush in a wall. will have inherently different low frequency characteristics than those intended for nominally 4π or free-standing use. Let's look at the contents of a 'natural' sound.

As a beater strikes a tympano, a whole range of frequencies is produced. Some are built up as resonances but most are present to some degree from the moment of impact. The low frequencies radiate from a large surface area with wave motions rippling through the entire diaphragm. High frequencies travel at the appropriate speed of sound through the skin material, intermodulate with other frequencies, and radiate from many points on the skin's surface. If we stood 20 ft to the left of the tympano while it was being played, the frequency content of the instrument could be heard and measured. Should we then be lifted by a crane, travel in a semi-circular arc through a point 20 ft above the tympano and finally land 20 ft to the right, the phase relationships of the transients, throughout the journey, would be chaotic. Nevertheless, the subject would be very well aware of the nature of the instrument, the overall tympano sound undoubtedly being preserved. There may be a slight change in timbre but unmistakably, the instrument would be heard 'undistorted'. It is probably this fact that for over a 100 years led to statements, originally by Ohm himself, that it was the overall amplitude response of a sound, which was of overriding importance for the accuracy of reproduction of that sound. Until recently, it was considered that 'phase' was not of prime consideration, in many contexts it was believed to be inaudible!

Nonetheless, it would probably be the changing phase relationship that would be responsible for any change in timbre when travelling over the instrument. While the amplitude/frequency response or the power spectrum of the instrument would remain largely the same, the waveform shape would change with position. This waveform change with position can never be accurately recorded then reproduced via a loudspeaker.

Once we record the tympano, we're in the mire! The physical requirements of drivers for low and high frequencies are almost diametrically opposed. Large propagation area and high mass for the bass drivers; point source and lightness for efficient response of the high frequency units. For this reason, the practical difficulties in obtaining 20 Hz to 20 kHz from one loudspeaker is, to this day, all but impossible. Inevitably we resort to multi-element systems. We now have a situation where the point of origin of any part of the sound is entirely dependent upon its frequency; 10 kHz will come from a very small source, the high frequency driver, and nowhere else. This bears no relation to the origin of the sound from the tympano. A person passing over the face of the loudspeaker, will no longer receive a recognisably uniform wavefront. High frequency arrival time will vary depending on the relative location of the drivers; frequencies will no longer be jumbled in their origin as in the case of the instrument itself. Even in a dual concentric loudspeaker system, the centre of propagation of the sound will be somewhat further forward or further back for the high frequencies, relative to the low frequencies. This is because the HF diaphragm is not co-located with the centre of propagation of the low frequency driver. Indeed, it could not be, as the precise centre of propagation of almost any driver moves forwards and back with frequency. True, by using so-called aligned in time crossovers, they could be co-located at the crossover frequency but only at that frequency. At all other frequencies, there would be a time shift. Even if digital delays were incorporated, the frequency dependent propagation shifts would still render less than perfect results.

As we see from travelling over the tympano, the absolute shape of the waveform is not a mandatory requirement for our recognition of the sound as 'natural'. We have already flown in the face of nature, by distributing our sound sources in the loudspeaker, with respect to frequency. This is not to say that we can ignore phase integrity in recombining the signals, but the importance of various aspects of this should be investigated. Where is the phase response really relevant?

The time of arrival at the ear of the different frequencies of a transient wavefront has a great bearing on the perceived timbre of the sound. With the low frequency components of a wavefront reaching us before the high frequency components, despite the system having a 'flat' overall frequency response, the sound will be perceived by the brain as a bass-heavy signal. The inverse is also true, this is irrespective of the fact that the steady state pressure amplitude response is 'flat'. On signals with low transients content, the effect will not be so noticeable. Here then, we have loudspeakers with flat frequency responses, sounding similar on steady state signals, but with significantly differing characters on transient signals. The forward or backward positioning of the HF units, can give a top or bottom heavy transient characteristic, depending purely on driver locations.

Certain designs that are aligned in time go to great lengths to ensure in-phase, on-axis signals at the crossover points but quite often, chaos reigns at either side. This is one effect of phase shifting/time delaying, either by electronic means or by physical positioning, where the prime criterion is the effect of phase on the steady state amplitude response of the entire system. Phase cancellation at the crossover points of a 24 dB/octave, or 18 dB/octave system, is restricted to a relatively sharp dip at the crossover frequency. You can measure it but the inability of the ear to detect high Q, moderate amplitude ▷

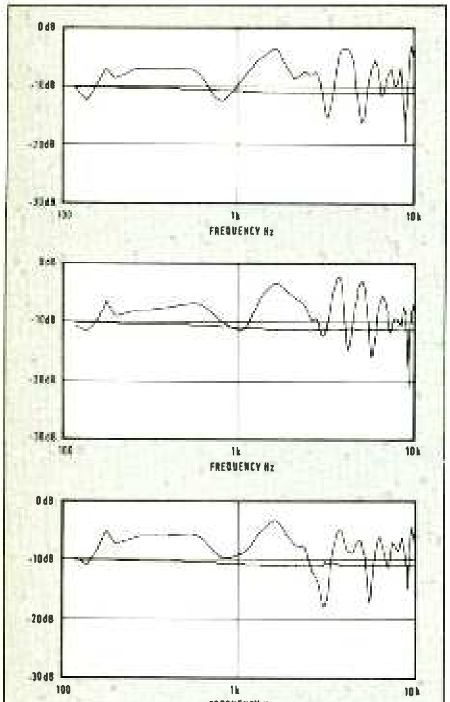


FIG 1a: One loudspeaker measured in three rooms, on top of three different mixing consoles
Note different high frequency responses due to varying reflexion patterns

troughs, is considerable. Rule one—don't compromise your overall phase response for a dead flat amplitude/frequency readout! I strongly feel that an adherence to a policy of minimum overall time displacement, is of greater importance than absolute phase integrity at the crossover points. Given that the impulse response is dependent upon amplitude and phase, I believe that maintaining the impulse waveform is the prime consideration.

Here, some horns can produce a stumbling block. In many horn designs, there is 18 inches from the driver mounting flange to the mouth of the horn and a 6 inch throat tube in the driver itself. In round figures, we have a sound velocity of around 1,000 ft/s in air at sea level, around 1 ms for every foot! Given that the diaphragm in

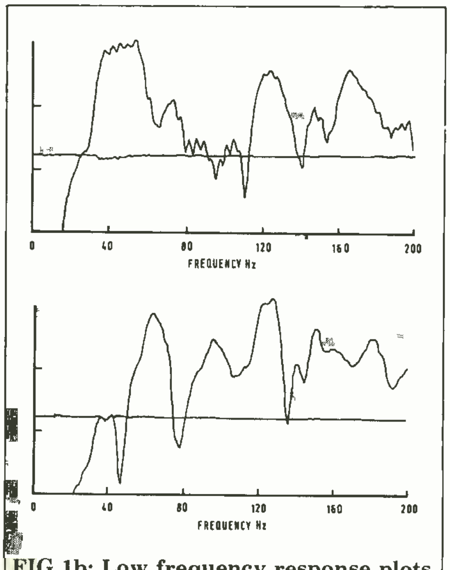


FIG 1b: Low frequency response plots of two identical loudspeakers in two different control rooms. Measurements are taken at engineer's position

the above horn/driver combination is 2 ft from the mouth, that is a 2 ms time delay if the horn mouth is mounted on the same baffle as the other drivers. What can one do? Stick the horn 2 ft out from the baffle? If you did, you'd have lost your integrated plane of the baffle for minimising reflexions, while giving everybody something to bang their heads on as they walk past. As yet, we have no means of moving time forward, so you can't 'time advance' the signal to the horn diaphragm. You can insert a phase shift in the crossover, giving the horn a 2 ms equivalent phase lead at the crossover point only but this, largely, only straightens out the steady state frequency/amplitude graph. Two octaves up, you're 2 ft out of sync again. You could put a 2 ms time delay, analogue or digital, in the crossover outputs to the amplifiers for the other drivers but then you are introducing more electronics, more filters and potentially more problems.

I believe this leaves us with a big problem on our hands if we do use a horn/driver combination with 2 ft between the diaphragm and the mouth. With well chosen driver positioning, coupled with judicious choice of crossover parameters and loudspeakers positioning, we can achieve minimum overall phase loudspeaker systems sounding even more natural than designs where undue attention has been paid to crossover point phase integrity. It is the time integrity of the leading edge that is more relevant than precise amplitude response.

All too frequently, too much attention is paid to one aspect only. A constant directivity is often the main goal of phase aligning. Time discrepancies in the crossover region, cause a tilting in the axial response of vertically aligned drivers. With 18 and 24 dB/octave, the region of mutual propagation of any two drivers is much reduced. In these higher order crossovers, the lobing and tilting effects are far less apparent. The lobing is a function of phase shifts in the crossovers at the crossover points only and should be dealt with there. The overall time/frequency response of the entire system should be considered a genuine priority, in order that a system may be as neutral on transients as it is on steady state programme material.

These seemingly confusing and sometimes contradictory sets of circumstances are usually born of attempts to build loudspeakers for use in average rooms. The compromises are therefore chosen by the individual designers based on their own sets of important criteria. Room reverberation times and the presence of early reflexions will have an enormous bearing on the priorities for any loudspeaker design. If the room is reverberant and is troubled by early reflexion problems, then overall power response and an even directivity/frequency performance will be very important. Possibly much more important than axial integrity. If the room has a very short reverberation time and is free of early reflexions, then the axial impulse response would be a prime target.

Given the complexities of the widely different directivity characteristics of the different instruments, together with the fixed directivity/frequency pattern of any single monitoring system, we can currently only hope to approximate the original sound field. To some degree, however, all loudspeakers suffer from the same drawbacks. They are, therefore, at least reasonably representative of each other. What can really have an enormous bearing upon the performance of a loudspeaker, is the interface with the room. It is totally unjust to criticise any

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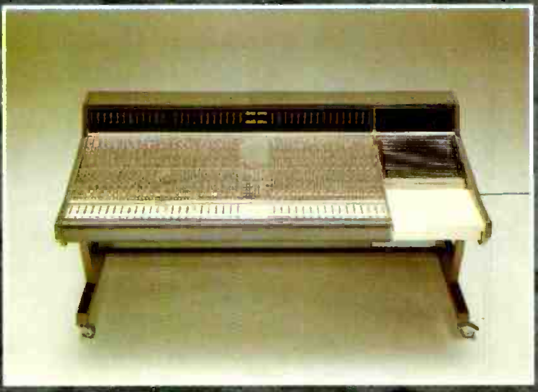

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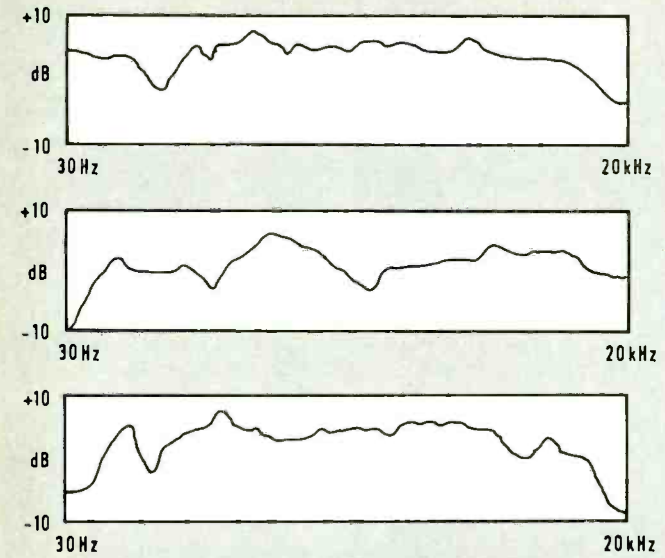

FIG 2: Response of one loudspeaker at three different locations in the same room. Differences are entirely due to room position

loudspeaker system after evaluation in just one room! **Figs 1, 2, 3** and **4** clearly show the effects of rooms and room positions on the responses of various loudspeakers. Note the relative similarity of **Fig 4**, the responses of the two *different* loudspeakers in the same location in the same room. **Fig 1(a)** shows the high frequency response variation of one set of near-field monitors placed on the top of three different mixing consoles, each in a different control room. **Fig 1(b)** shows the low frequency response variation of two identical monitor systems in two different control rooms. **Fig 2** shows the full range responses of three identical monitor systems in three different rooms. **Fig 3(a)** shows the response differences of one loudspeaker in one room but measured at two different distances. **Fig 3(b)** shows the low frequency response differences of another monitor system in another control room, but measured in two different positions; 1 metre and the listening position. **Fig 4** shows the relative *similarity* in low frequency responses of two entirely different loudspeaker units when mounted in the same position in the same control room and measured from the engineer's listening position. This puts a very strong case for considering the room and the monitor system, as being part and parcel of one and the same entity.

Due to the directivity characteristics of the loudspeakers, the effective space into which they radiate, varies as a function of frequency. This in turn affects the on-axis power response, and the subjective frequency balance in the room. The distance to the adjacent room boundaries also has a time function, in that the closer the room boundaries are to the loudspeakers, the faster will be the arrival of the early reflexions from those boundaries to the listening position (see **Fig 5**). The faster the arrival of the early reflexions, the less distinct the imaging becomes, so the overall definition suffers. It is for this reason that larger rooms frequently sound less coloured than their smaller counterparts.

At low frequencies, where the wavelengths are long, the loudspeakers will radiate in an almost omnidirectional pattern. If the loudspeaker is a free-standing box and has been optimised for 'freefield' conditions, the low frequencies will increase when the loudspeaker is placed in, or against, a wall. Frequencies below 200 Hz can no longer radiate in all directions and so will be concentrated back into the room. Side walls, ceilings and floors, all have a similar effect, depending on their proximity to the loudspeakers. A 'freefield' aligned system can experience anything from 2 dB to 10 dB of boost, below 200 Hz, depending on its location in a room. As far as the loudspeaker is concerned, it is the self same loudspeaker but, from the listen pointers of view, there has been a drastic change in the perceived frequency balance and general sound character. As we have already seen in **Fig 1** and **Fig 2**, two dissimilar loudspeakers may sound more alike in one position in one room, than one loudspeaker in the two different locations. This may go some way towards explaining why many different loudspeakers may appear substantially the same when measured in controlled conditions on the widely used swept sine, or on a spectrum analyser, while having little in common on the reproduction of music in real life.

In a room, the spectrum analyser only tells us of the overall, direct and reverberant average of the frequency distribution of the sound power in the room; and what is more, only at that microphone position. The room will tend to dominate! Moreover, up to 100 different microphone positions would need to be computed and averaged, to ever have any hope whatsoever of gaining a representative display below 100 Hz. Standing waves and resonances of the room itself may swamp the actual loudspeaker output. Such a measurement system tells very little about the loudspeakers themselves.

It is only with the reconstitution of the impulse response that reasonably accurate assessments of loudspeaker performance can be made. Accuracy is still worth pursuing, despite the fact that we are not in control of loudspeaker placements in people's homes. We must aim for some sort of standard of accuracy in the studios, otherwise all points of reference will eventually be lost. The problem of standards of reference must be a function of what is actually heard in the control room. From **Fig 1** and **Fig 2** it can clearly be seen that no 'reference' standard can be given, neither to any particular set of loudspeakers, nor to any individual room. The overall performance of any monitoring system, must be a function of the loudspeakers, crossovers, amplifiers and the rooms. It is futile to declare one amplifier the best or a certain crossover correct without full reference to the systems of which they form component parts.

There seems to be some innate, human urge to pigeon-hole things; to find security in having one particular loudspeaker or amplifier to cling to. Unfortunately, of course, it just does not work that way. No loudspeakers are going to give good stereo imaging and punchy bass while situated in a ceramic tiled bathroom. I realise that this is an extreme case but it does show how the reverberant field can dominate. Multipoint, digital inverse filtered, active control systems that will tie together the room and all other components of the monitor system are already being developed. These should take the hit and miss out of the interface but such computerised, digital technology is unlikely to become universal in the near future. In the absence of such digital reconstitution, the balance of impulse response to reverberant response can only be achieved by a very careful balance of the characteristics of the monitor system components, and the room.

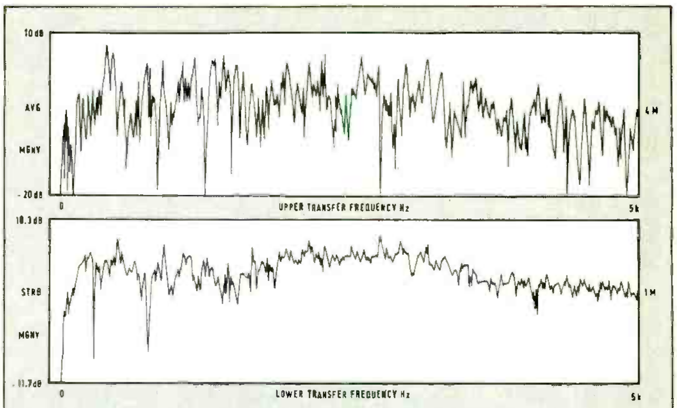

FIG 3a: One loudspeaker at one location in one room
Lower trace measured at a distance of 5 ft
Upper trace measured at a distance of 15 ft
Note the predominance of the room at low frequencies and the comb filtering in the upper trace caused by room reflexions

The typical domestic living room, while varying from house to house, can, reasonably, be averaged. Unfortunately, from the point of view of the worldwide distribution of music, the domestic average varies considerably from country to country. An international average from igloo to palace, would probably be representative of very little. Even if it were established, the vast majority of the record buying public would still live in 'non-average' conditions and hence would be no better off than now. Nobody has yet succeeded in achieving any generally accepted frequency/reverberation time industry standard ▷

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for control room acoustics. Without any standard of control room acoustics, it is difficult to achieve any standard for monitoring in general. As so much recording is now passing directly into the hands of the musicians themselves, the possibility of any sort of consensus could well be slipping out of reach. All too often, those who seek to conduct do-it-yourself recording, bite off far more than they can chew. The entire recording process takes a great deal of subjective and objective thinking. The requirements for a crew of producer, engineer, assistant engineer and musicians is as necessary now as ever it was.

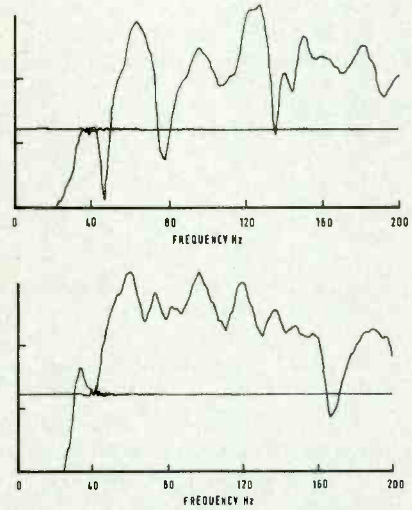


FIG 3b: Low frequency response plots of one loudspeaker in one control room but in two different positions

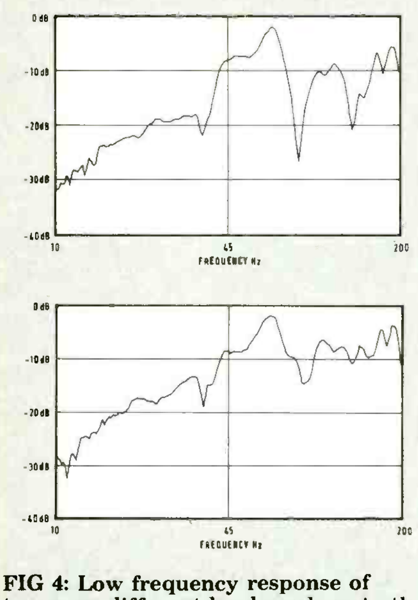


FIG 4: Low frequency response of two very different loudspeakers in the same location in the same room Note greater similarity than in Figs 1, 2 or 3

With such disparity in the performance of studios, coincident with the advancement of home facilities, the record companies have in many cases seen little difference in the sound quality of home or studio produced recording. When A&R people have frequently been brought up themselves in a era of good home facilities, they have often favoured the provision of recording equipment directly to the artists, as opposed to providing the time-honoured 'recording advance' in studio time.

The whole face of the recording industry is

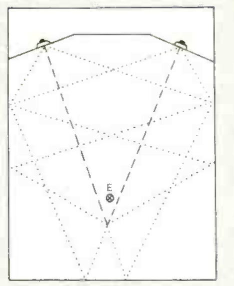


FIG 5a: Relatively 'live' room, with loudspeakers mounted at the narrow 'end'. Reflexions can reach the engineer after travelling relatively short, additional distances compared to the direct waves

KEY
E ⊗ LOCATION OF ENGINEER
--- DIRECT WAVE PATH
..... REFLECTIVE WAVEPATH
~~~ ABSORBENT SURFACE

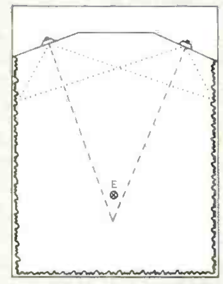


FIG 5b: Relatively 'dead' room, absorbing reflexions at room boundaries. This may produce an unnatural internal acoustic and below a few hundred Hz, the room's absorbent properties will revert to those of the 'live' room (a). This can render an uneven overall response

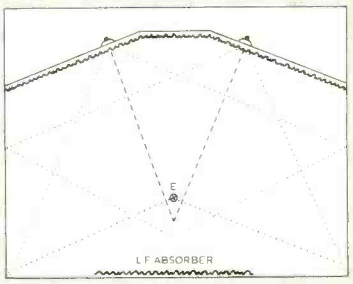


FIG 5c: Moving side walls further from the loudspeakers, and placing an LF absorber behind the listening position, will prevent early reflexions. If the first reflexions can be delayed 15 ms – 20 ms after the direct wave arrives—the brain will accept them as separate signals, and not as blurring of the initial signal

changing in an unpredictable manner. Sooner or later it will become apparent that in many circumstances, a more orderly approach will be required. When this time comes, it will be the studios that produce the most consistent, reliable results, and are staffed by the most objective and knowledgeable personnel, who will be best suited to survive. A wider understanding and more open discussion of the truly relevant aspects of monitoring *must* soon be instigated. With modern technology the use of conventional ⅓-octave spectrum analysers and graphic equalisers are totally outmoded. Furthermore, in the wrong hands, the end results can be about as accurate as a control room setup by the use of a secondhand lavatory brush!

Fast Fourier analysis of the impulse reponse gives reliable, easily interpretable representation of both the amplitude and phase characteristics of the monitor system. The Maximum Length Sequence System Analysis units (MLSSA) are also capable of excellent results in this area. Pink noise analysis will provide reverberation analysis of the rooms themselves. Whilst the two things must be considered in parallel, they must be measured in isolation. Once again, so much of the confusion in monitoring has been brought about by people making measurements who only think they know what they are doing; usually measuring the wrong things or interpreting incorrectly. Temperature and humidity have more effect on monitor response than much of the day to day electro-mechanical 'drift'. There is no reason why endless twiddlings and re-alignments should be required under normal circumstances. It is always a strong indication of the phase/impulse accuracy of a system, when the desire to 'adjust' that system is a rare occurrence. Personnel and equipment brought into a control room, will in all probability, significantly affect the reverberation characteristics of the room. Unless placed between the monitors and the listeners, they will not affect the impulse response. It is the impulse/phase response that carries the stereo spacial imaging, which can only be effectively blurred either by room reflexions arriving within the first 15 ms or so of the direct sound or, alternatively, by being swamped by high levels of reverberation; within around 10 dB of the direct signal level.

The importance of the integrity of this impulse response cannot be overemphasised but it is difficult to achieve in practice. A single, full range loudspeaker, would be a very desirable piece of equipment, 10 octaves, however, is too tall an order in practice as the laws of physics demand widely differing properties for the generation of 20 Hz as opposed to 20 kHz. Crossovers and physical separation of the drive units are something of a fact of life. Before I hear somebody say 'time aligned dual concentrics', they still have similar problems in practice. They are not necessarily the answer as in many instances, it has been a case of one step forward, two steps backwards! Crossovers suitable for monitoring purposes are complex devices, almost impossible to build with an entirely minimum-phase response but with very careful dedication to the specific loudspeaker system for which they are designed, they can produce acceptably phase accurate systems. They must be designed in the full light of understanding of the physical layout and phase/frequency characteristics of the loudspeaker drive units. Nothing in a monitor chain, from the desk output to the listeners' ears can be fully
~~~

developed in isolation. The overall system must really be considered one single entity and that includes the space within the room!

There are four major parameters affecting tonality in any monitor system:

Pressure amplitude (frequency) response
Phase response
Non-linear distortions
Reflexions and reverberation
NB Delayed resonances can be considered in the phase/amplitude domain.

While non-linear distortion is primarily a function of the loudspeaker/amplifier system, the other three are primarily a function of the rooms/loudspeaker/amplifier/crossover combination. If the room has an uneven reverberation time, the peak in its response will appear to hang over and colour or 'muddy' the sound at and around those frequencies. This cannot be controlled by graphic equalisers as any reduction in the energy from the loudspeakers in an attempt to drive the room less at its reverberant peaks, will distort the impulse response and hence the direct transient signal which carries so much of the definition to the ears. Room reverberation problems must be dealt with in the room. Any room reflexions will add to and subtract from the direct signal, depending on their amplitude and phase relative to frequency. Here we have two possible options, control the loudspeaker directivity, or control the room. One advantage of controlling the room is that loudspeaker drivers can be chosen for tonality, power handling and other parameters, without as much consideration being given to directivity as would need to be the case in a room with more reverberation and early reflexion problems. Nonetheless, abrupt directivity changes within the loudspeakers should be avoided as other colouration problems may arise.

Reflexions will also cause phase confusion, which, in many instances, will mask many subtle perceptions of spaciousness and position, not to mention the overall tonality. If a designer has control of a room, it becomes a much more manageable task to optimise the amplifier/crossover/loudspeaker system for 'accuracy'. Manufacturers who design and market loudspeaker systems for general use without any control of the listening environment are facing an uphill struggle, and the performance data from the loudspeakers alone are virtually meaningless. It's not that such manufacturers should pack up and go home but rather that users should pay more attention to the intended mounting regimes those manufacturers may specify. They just cannot sound great in any room, so without close co-operation with the manufacturers, the users cannot aspire to any degree of accuracy, unless they are fortunate in their environmental circumstances.

In larger rooms in particular, more directional loudspeakers may well be desirable on two counts. Firstly, if the room tends to be live a directional monitor would minimise the reflexion problems by concentrating the sound on-axis. Secondly, if the room were relatively dead, a wide directivity monitor would require a much greater acoustic output as so much of its radiated power would be off-axis and absorbed. A more directional loudspeaker would concentrate the available power on-axis and hence would be more efficient in those circumstances. Once again though, individual circumstances must be dealt with individually.

To sum up see 'Newell's Law', which is a letter I sent, some time ago, to another studio designer I had known for many years and his reply. All I can add is that it assumes the rooms and loudspeakers would be designed for use together, and that the title was tongue in cheek—I was not claiming absolute originality!□

Newell's Law

Monitor problems must be fixed in the monitors, room problems must be fixed in the room!—and never the twain shall meet. The job of a monitor system is to drive the room with the desired signal. That desired signal is the sum of its instantaneous parts. I maintain that the 'transparency' and 'neutrality' of the perceived sound is almost entirely due to the integrity of the leading edges, which means an absolute minimum of amplitude or phase distortions. Room problems are mainly resonance problems, reverb time problems. The monitor system therefore needs to be accurate to the impulse; the room needs to be accurate in steady state conditions. Should a room have a peak or dip, the option to drive the room to a lesser or greater degree by means of monitor equalisation is not available. True, a peak in the room will be less objectionable if it is driven to a lesser degree by a correspondingly equalised monitor system but nonetheless, to a slightly lesser extent, the resonant overhang or underhang will still exist. As the correct reconstitution of an impulse requires absolute fidelity to both amplitude and phase, then by definition it can no longer be accurately reproduced if any form of monitor equalisation is used to correct a non-monitor problem.

Monitor equalisation can have an application when, for certain reasons, the monitor system itself displays small peaks or dips. If these deviations are of a minimum phase nature, then an equaliser exhibiting minimum phase properties may under certain circumstances, restore both the amplitude and phase accuracy of the entire system. Large deviations, however, are rarely of a mimimum phase nature and, therefore, whilst amplitude accuracy can be restored, phase accuracy most certainly cannot. Monitor equalisation can only be used for the compensation of monitor deficiencies of a minimum-phase nature; they must never be used to correct for a room reverberation time problem. We have, in past years, been fools. If the impulse integrity is accurate upon arrival at the listening position, then once again, by definition, the amplitude and phase responses must be correct. Get this right and the rest will fall into place—neutrality, transparency, stereo imaging, fatigue-free listening—we are then left with one major problem. Due to the inherent group delay properties of conventional filters, no crossovers, except for the 6 dB/octave on-axis, can exhibit minimum-phase and correct amplitude properties at the crossover point. These group delays time-smear the composite output at the crossover points in such a way that the physical realignment of the drivers cannot adequately compensate. It was for this reason that I pursued the only path which I knew to be open—the use of a 2 ms modelling delay and an adaptive digital filter for phase and amplitude reconstitution in the general mixing area by means of a least mean squares approach to response over the desired listening area. For the purposes of impulse reconstitution, there will be a feed-forward system into the monitor drive circuits via the modelling delay.

If the room microphones are to be used in any way for room control then the required data for this will be fed only into the circuits of the room control drivers; never back into the monitors. These will then operate in conjunction with the velocity sensing microphones at the rear of the room to absorb the acoustic energy striking the wall by means of the active control drivers. We have come full circle back to Newell's Law, deal with monitor problems in the monitors, and room problems in the room! Never try to use one to 'fix' the other; you can't!

Notwithstanding this, I do realise that if the room is acoustically of low reverberation time and is essentially ⅓-octave flat to within ±2½ dB, then recent work by Elliott, Nelson, Hamada and others does indicate that overall impulse control may well be achievable by digital inverse filtering of the monitors alone. Solve the non-linearities after that and we should be home and dry!

And another studio designer replies

Philip, I hope that with the above approach we will, in the near future, be working with a new degree of repeatable definition in terms of control room monitoring. Maybe then, there will be a return of overall confidence in consistency. By no means, however, is it all in the hands of the studio designers nor was it ever the sole fault of loudspeaker manufacturers. Demands upon control room monitor systems become increasingly complex. They must be capable of horribly accurate, nit-picking truthfulness at prodigious sound pressure levels. They need to be representative of reasonably 'average domestic conditions', which, even if that could be quantified, would be subject to fashionable change. They must also, ideally, be capable of inspiring the musicians to greater creativity; which sometimes means creating 'on stage' environments. This really calls for three or more monitor systems, yet in inexperienced hands, the more systems available, the more confusion is possible. The requirements are not practically realisable in any one system, hence the dual, near and far field systems evident in most present day control rooms.

Art, by definition, is beyond science. The recording business is expanding rapidly with more widely based points of view, and in many cases, proportionately less in-depth knowledge. The days of tape operators waiting 5 years before getting any chance of sitting behind a mixing desk, are long gone. This wider breadth of knowledge with less depth of understanding, makes some pretty heavy demands on the available science. No room, and no monitor system, can be expected to be all things to all people, but we can work towards more fixed references. Accepting that the end result, music in people's homes, is an artistic interpretation, we can still aim for more consistency in the practical realisation of that goal. It would, however, appear that for the moment (1990) entropy rules!

EUROSOUND MOBILE 3

Andrew von Gamm visits Eurosound's Mobile 3 to check out the claim that it is Europe's busiest

The heart of the mobile is a specially adapted *MXP 3000* from Sony. Because the *3000* was at the time only available with a maximum of 36 channels, Eurosound developed their own mother-board for channels 37 to 48 and extended the vu meter bridge and the patchbay. This development was later sold back to Sony who now offer it as an option. The I/O modules are the *MXBK 3001* transformerless and EQ is with the *EQ31* module, which is a Wien fixed Q type giving ±15 dB for four frequencies and the *EQ 32* whose Q is switchable from 0.5 to 1.5. Because the desk is very simply and logically laid out and every button seems self explanatory, Mobile 3 can be used by many different engineers.

A look at Mobile 3's schedule for the last year would seem to verify the claim that it is Europe's busiest. It has averaged one international act per month and has been on the road for numerous smaller acts and events.

Apart from having a conventional 24-track recording studio, Jan de Groot, owner of Mobile 3 and its brothers 1 and 2, has been specialising in mobile recording for over a decade, but it has been Mobile 3 that has taken his company, Eurosound of Herveld in the Netherlands, into the big league.

Mobile 3 can provide the customer with recording formats from the usual 24-track analogue to all three digital formats. Although it has only been in existence for the past year, the truck has already recorded such acts as Elton John, Duranduran, David Bowie, Bryan Adams, Bon Jovi, Prince, George Michael and many other European acts who are superstars in their own countries. It was on one such occasion that I was able to visit de Groot and Mobile 3 as they went to Bonn in West Germany to record the band Bap, an act that is hardly known outside Germany but can draw crowds of tens of thousands in their home country.

De Groot sees part of the reason for the success of his latest mobile in its configuration and the flexibility that this allows. For example, they recorded Bon Jovi on a Friday in London using two 24-track machines and 2 days later they were over 800 miles away in Milan to record Duranduran on 48-track digital.

Jan de Groot did not begin his working life in recording. Although music and recording were his hobbies, he started out as a proof-reader for a Netherlands daily publication. Recording had to happen in the evenings and at weekends. In 1971 he opened up his own studio working with local bands and choirs and just 3 years later he was able to leave publishing and run the studio full-time. His first mixing console was a Sennheiser but this was replaced by a Midas and, when he went full-time, he invested in an Ampex *MM1000* 16-track recorder which had been used for the 1974 Olympic Games in Munich.

Once he was a full-time recording engineer, de Groot got off to a good start with customers such as Golden Earring, Ultravox and Joe Jackson, as well as TV productions. But the Ampex recorder proved heavy and cumbersome so, as the customers came and turnover increased, he re-equipped with the new MCI 24-track in 1976.

Today, most of the machines used are Sony, including the desk on Mobile 3. But why Sony? De Groot says that there was no specific decision to install one of their machines but each piece of equipment has to stand or fall on its own merits and Mobile 3 can use any recording format.

Eurosound does not advertise or engage in any price-cutting, so why is Mobile 3 successful in a field where others have failed? When asked what was better about his mobiles, compared to others, de Groot flatly refused to answer. He did point out, however, that he went to great pains to get Mobile 3 just right. It may only be 1 year old but it took 3 years to plan and build. Three journeys were made to the US to study the market and talk to potential customers. It is no good thinking one knows what the customer wants: one has to go and find out exactly, says de Groot.

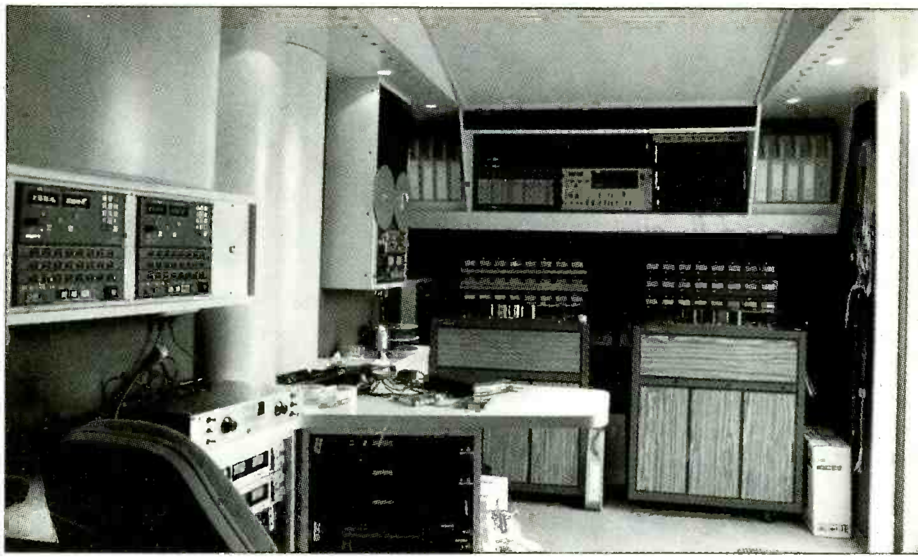
De Groot sees every one of his three mobiles as being a self contained unit. Mobile 1 is mostly used for classical recordings and uses a 32/24 desk with Eela inputs and specially developed dynamic sections for each input to cut out any noise from open spot mics. Recording is done on two Studer ▷

◁ *B67*s as well as a Sony DAT. The mobile is prewired for 24-track. Mobile 2 is almost the same, except that it uses a 30/6 Midas desk and is used more for rock/pop music.

Eurosound have over 100 microphones and can equip their mobiles with almost any mic imaginable. If the customer does not specify a particular choice of microphones, there is a standard collection that goes out with each and every mobile consisting of six Shure *SM 58*s and two 57s, three Sennheiser *MD 441*s and eight *MKH 416*s and four Neumann *KM 84*s, as well as six active DI boxes.

Other microphones are from Schoeps, Beyer, B&K and E-V. Mobiles 1 and 2 carry an *EMT 245* reverb, a K-T *DN360* EQ, a *1176* UREI stereo limiter as well as extensive intercom and communications equipment.

Mobile 3 comes complete with two Yamaha *REV7*s, a *REV5*, an *SPX90* and an *SPX90II*, a Lexicon *PCM 70*, a large collection of UREI and Burnie limiters together with 10 Drawmer noise



Sony *JH2424*s at the rear of Mobile 3

gates. EQ and spectrum analysers come from Klark-Teknik and there are three timecode generators/readers and a *Zeta 3* synchroniser from Adams-Smith, as well as an Aphex *Studio Dominator*.

Recording equipment for Mobile 3 is usually two Sony *JH 2424*s but these can be substituted by two *PCM 3324*s, a *PCM 3348*, two Mitsubishi *X 850*s or Otari *DTR 900*s. Stereo equipment is Sony U-matic with *PCM 601*, Sony DAT recorders, Studer *A810* with SMPTE timecode and two Nakamichi cassette decks. There is also a collection of submixers available, though usually an Eela 12/4 or a Soundcraft 24/6 is used.

Mobile 3 comes with a crew of three or four: one sound engineer, one man on stage and one or two helpers. Usually, de Groot is one of those helpers. He also seems to do most of the driving. All three of his mobiles are on Mercedes chassis. Mobile 3's cab sports such luxuries as a bed and electric windows. They found Mercedes trucks relaxing to drive and as they run a 24 hr breakdown service throughout western Europe, they were the obvious choice.

Selling the services of Mobile 3 has so far been mainly by word of mouth. Peter Brant is the first foreign representative (in Germany) for Eurosound and is seen by de Groot as being a model for further development. In order to do successful business in another country, he says, it is very important to have proper representation. Each European country has its own market structure, laws and language. The German customer, for example, expects to be able to speak in German to a German company and do business under German laws. Knowing who to sue when things go wrong, says de Groot, is an important business consideration! He does not plan to build any more mobiles. Three, he states emphatically, is quite enough. Expansion will take place by having more representatives in other countries to service their individual markets.

On the road, de Groot, when he is not driving, sees his role as trouble-shooting and talking to customers, leaving the important work, as he puts it, to his crew. In the office he answers all his own correspondence and does his own book-keeping. In all he does, de Groot seems to show the attention to detail that one would expect from a former proof-reader and this is probably one of the reasons for his success.

Eurosound, Dijkstraat 5, 6674 AG Herveld, The Netherlands. Tel: 31 8880 1048. Fax: 31 8880 1042.

MONITOR SYSTEMS-

COMMENT ON THE SERIES

Our series on Monitor System design, which began in August 1989, has provoked a number of people into writing to us with views for and against.

Dear Sir, I started reading Mr P Newell's two articles recently published in *Studio Sound* with great interest as he said "I draw on a great deal of practical and academic experience". Having an MSc in Acoustics, and having been a loudspeaker designer for various major loudspeaker manufacturing companies almost continuously for 18 years, I looked forward to the articles that were promised. I was so disappointed that I have taken the unprecedented action of preparing for your consideration some sound reasoning of my own. As the articles invited comment I look forward in anticipation to receiving the hospitality of your columns.

Mr Newell rails at the lack of understanding concerning horns and says "much of the criticism seems based on unfounded or ill-informed prejudices". This in effect pays the average recording engineer quite a compliment, because I find most engineers are not worried about understanding the loudspeaker but simply complain, "they sound hard and horrible and give me bad mixes". In fact I find Mr Newell's arguments full of prejudice and lacking in understanding, let me explain why.

First, a discussion in the defence of the direct radiating loudspeaker. A sensible statement needs to be made about the sound pressure levels required in studios. The often mentioned 120 dB at the desk, repeated in Mr Newell's articles, is basically a joke. At that level every person with normal hearing will be experiencing great discomfort. Sensible monitoring is clearly not possible at these levels. The inner ear itself at these levels adds significant levels of distortion to the signal sent from the ear to the brain. In experience even listening for several hours at 100 dB is very tiring. I am confident in suggesting that most engineers will not monitor sounds much louder than 96 dB. This level enables the optimum satisfaction of the following objectives.

- To make the best artistic decisions
- To discern undesirable features clearly
- Relaxed, stable decision making a fatigue-free environment

Most of those engineers reading this letter, who have actually measured the level they prefer to work at, making artistic and quality decisions, will be in general agreement.

If you have some customers with an insatiable desire for levels that make most people's ears bleed, we all know what has to be done. But let us be honest, nobody can listen to quality at that level, it is difficult to even tell if the balance is satisfactory.

Consider the following situation, you are listening at 96 dB at 2 metres from the speaker, half the energy is going into the midrange. That is fairly realistic for the average direct radiating loudspeakers, it is used as a so-called 'nearfield' monitor. If we decide that a 10 dB margin is sufficient for the continuous sound level, this adds up to the midrange driver being required to produce a continuous SPL of 106 dB at 1 metre.

The ATC 3 inch soft dome has a sensitivity of 94 dB/W/m so to produce 106 dB requires an extra 12 dB of level over 1 W which is supplied by 16 W. The voice coil of this speaker reaches 100°C with a continuous 100 W input. The mechanical integrity of the units surpasses its thermal capability. From this you can see not all direct radiating loudspeakers are at the limits of their ability at commercially used listening levels. In fact some have considerable reserves, at least another 8 dB for the 3 inch dome, limiters have only been included in ATC systems to prevent the amplifiers clipping when they are driven at insane levels.

I have shown above that all the negative comments about thermal power handling and sensitivity are not of prime importance. A system using a cone driver with a 1 or 1½ inch voice coil will reach problems earlier than the ATC 3 inch dome but the picture is still not as bad as painted in Mr Newell's articles.

Despite Mr Newell's obvious optimism, there are two major reasons why horn speakers will never replace direct radiating loudspeakers as the premier monitor design, perhaps the most important is distortion.

Mr Newell complains that for horns "many charges against them are simply not true". I am afraid I have to complain that many of Mr Newell's charges for them are untrue and cannot be substantiated. Especially his claim that they have less distortion. Level for level I do not know of any horn cleaner than the ATC 3 inch dome by a significant margin. I expect this is true for other direct radiating midranges too.

A loudspeaker should reproduce a sound with its various frequency components at the same relative levels as the original sound. To do this, the loudspeaker should have a flat pressure amplitude response, or to most people a flat frequency response. However this is not, within limits, of paramount importance as most engineers will know it doesn't matter how much you EQ a sound there are other ingredients that define its quality. Incidentally horns usually have far worse irregularities in their frequency response than direct radiators.

Loudspeakers typically have the poorest harmonic distortion figures out of most audio equipment, so manufacturers are quite happy not to shout about them. This means comparable distortion measurements between various drive units are difficult to come by. I must categorically claim here though that the harmonic distortion characteristics have a most profound correlation with perceived sound quality.

Perhaps Mr Newell thinks horns have less distortion because he has not undertaken their measurement yet. In his university level investigation of various horns, he shows surprise that his listening impressions for a horn did not correlate to the frequency response measurements.

Instead of measuring harmonic distortion at this point he appears to have pondered what I call time domain distortion quite seriously for he says, "I believe that the truly audible and idiosyncratic signature of any given loudspeaker system exists in the domain of phase and the closely related impulse characteristics."

It is plain common sense that a loudspeaker's 'signature' is defined by how all the various distortion types combine and affect our subjective judgement. The relative offensiveness of the various types will depend on the listener, but to single out one distortion type is something of an oversimplification.

In the 1970s several loudspeaker manufacturers, with the advent of digital measuring systems, began to investigate the time domain aspects of a loudspeaker's design. The Audio Engineering Society's Journal at the time flowed with papers discussing this new audio topic. Even the popular magazines discussed the subject to death, so the facts are far from hidden.

Resonances to me are the number one nasty time domain distortion, they are clearly audible, sounding muddy, removing clarity and detail. A major resonance in the cone ▷

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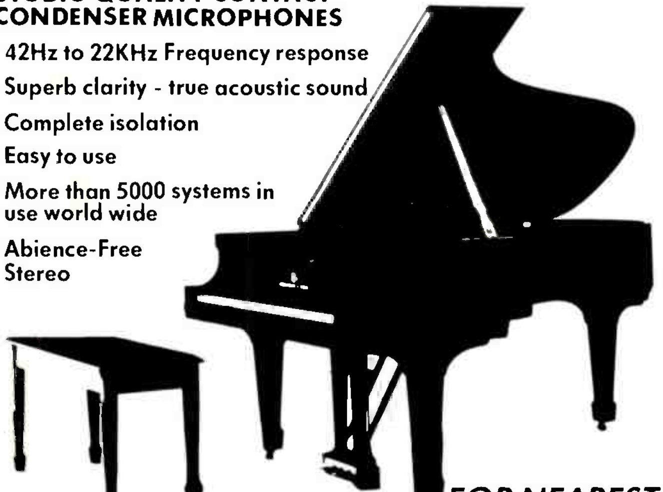

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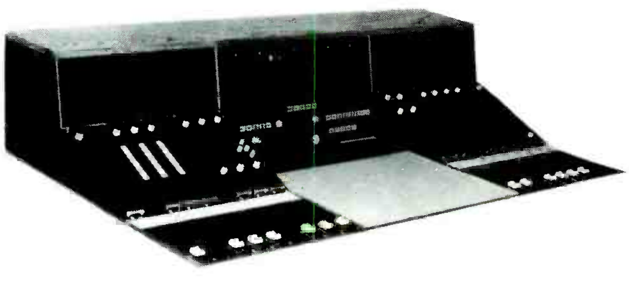

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Unit A8, Pinfold Industrial Estate, Rhyl, Clwyd LL18 2YR, U.K. Tel. (0745) 338864 Fax. (0745) 343693 loudspeaker is generated where the surround resonates in sympathy with the cone usually around 100 to 500 Hz, often causing what is called a 'surround bump' or 'dig'. Higher order resonances causing cone break are usually only a problem in very cheap drivers.

For horn loaded compression drivers, Mr Newell kindly went to great lengths to discuss their time domain problems caused by what he calls phase dispersion, high Q phase shifts at cut off, and reflexions from mouth causing horn resonances, he insists a design is around the corner where all these things will be made negligible. (Note that a direct radiating loudspeaker does not have any of these distortion types at all.) He's a dreamer, companies with budgets greater than his have earnestly sought to improve the horn, they know the problems and Mr Newell soon will too, by all accounts.

An emphasis was placed on the superior transient behaviour of horns due to the very light diaphragm in the compression driver. This is akin to saying motor bikes accelerate faster than cars because they are lighter. In fact many cars accelerate faster than motor bikes, the important factor is not the weight alone but the power/weight ratio. If you put a larger engine in a car it not only accelerates quicker, its top speed is increased too. Similarly if a speaker can operate to a sufficiently high frequency it will have a creditable transient response, other distortions permitting.

The weight of a compression driver diaphragm by itself is of no consequence, concerning transient response, in fact such a diaphragm is usually made from a very thin sheet of aluminium alloy that resonates away quite merrily causing very high levels of audible distortion. When combined with all the other time domain distortions and harmonic distortion produced in horns, that characteristic hard sound is soon explained. It was inferred by Mr Newell that horns sound hard at high listening levels. I could contend and say horns always sound hard and progressively more so as the drive level is increased.

An interesting way to describe a crossover point in a loudspeaker system is to call it a resonance, for indeed that is what it is and they sound similar to other modes of resonance too.

The ear discerns with greater detail distortions of all types in the midrange, so most hi-fi manufacturers strive to keep crossovers out of the middle frequency range. A quality speaker having crossover points around 3/400 Hz and 3/4 kHz, there are no secrets here among accomplished speaker designers. However, with horns, the mouth size required to go down to 400 Hz is a bit large for a studio control room. In practice, the useful range of a horn in a studio can only go down to 1 kHz. This is where the ear's ability to detect distortion is near maximum.

Mr Newell says more research is required on the audibility of phase shifts and slopes but if he had investigated open literature more thoroughly he would have found a mine of information.

A full discussion of this matter is beyond the scope of this letter, but I wish to make the comment that Mr Newell's comments about polarity of drivers relative to crossovers, while interesting, can be further explained. Audible phase effects occur in what has been called a critical bandwidth, even quite large changes in phase outside of this bandwidth have been demonstrated to be inaudible. Therefore Mr Newell's preferred inphase crossover, with discontinuities in amplitude and phase, cause audible effects and the smooth responses of the reversed phase crossover cause inaudible effects. I assume his listeners simply preferred the sound of the audible phase distortion in his listening tests.

There is yet another major reason why horns will never yield to the prowess of the small diameter direct radiator.

When we listen to a sound we do not just hear the direct sound, but early reflexions and the quasi-reverberant field too. The non-direct sounds need just as much attention as the direct for a natural reproduced sound quality. The frequency response of the non-direct sounds will be influenced by the off-axis frequency response or directivity of the speaker and of course the acoustic geography of the room.

To reproduce the original sound field is it not obvious the early reflexions should have similar response to the original sound? This is achieved by a loudspeaker that has a very wide dispersion. Mr Newell agrees with this at one point but later says horns have an advantage over direct radiators in that they can direct the sound over certain areas. The fact is that the most natural sound field is set up by a very wide dispersion loudspeaker, beaming of the sound is a PA trick for large highly reverberant halls. If this trick is needed the control room is too big for ultimate monitoring. A wide dispersion loudspeaker requires much greater care in its positioning to ensure the arrival pattern of the early reflexions is compatible with a high perceived sound quality.

While reading this letter not just the light from the word you are reading at a given time enters your eye, but from a much wider area. It is possible to read by viewing through a card with a couple of holes but it would not be your normal relaxed reading experience. Similarly with sound it is important not only to get the direct sound information to the listener accurately but by careful engineering of non-direct sounds a natural relaxed listening environment is produced.

Some people will contend with much of this implying it is just my opinion, these same people claim acoustics is an art. Well, for some acoustics is a science and I claim what I say is fact.

Yours faithfully, Ray Ellaby (formerly of ATC and Turbosound).

Dear Sir, I read with great interest the opening of the series by Philip Newell in *Studio Sound* August 1989 ('Monitor Systems').

My highest congratulations to Philip for caring enough and taking the time to compile the words of wisdom for our audio industry to contemplate.

Philip presents some of the fundamentals required for acoustic 'sonic neutrality' that all in our industry should be aware of, and as familiar with as morning coffee or tea.

However, if Philip continues with this line of thinking and expands on these principles in further editions (and I hope he does), rest assured, the bleeding hearts in our industry with prized fat cat bounties to preserve will crawl out of the cracks and yell foul!

These 'fat ones' so out of touch with reality, with pride and antiquated ideas to preserve, will never concede their fundamental approach has come to the end and is found wanting.

Take courage Philip, I'm told a person in quicksand shouts and thrashes about before going under (and there will be a lot of shouting if you expand on these topics). Unfortunately, in our industry, plain truth is often considered synonymous with fanaticism. *Sonic neutrality is all that counts.* Keep pushing: all leaders and innovators in this industry have to take the shouting in their stride. You are on the right track. The truth and sonic neutrality will win with the professional. The mob will eventually follow as truth prevails.

Tom Hidley, Route de Ghon 24, 'Residence Panorama' 6-B, CH-1820 Montreux, Switzerland.

Philip Newell replies: Dear Sir, In reply to Ray Ellaby's letter, let me begin by commenting on his final statement of "for some, acoustics is a science, and I claim what I say is fact". Far too little is known about the human audiological processes for loudspeaker manufacturing to become a pure science in the foreseeable future. Loudspeaker design problems are fractal, they continue to reveal more detailed problems as each larger problem is solved. From the most quiet of perceptible sounds to the threshold of pain, the ear covers a power ratio of around 1,000,000,000,000:1. If the noise in recording is 60 dB (pressure) down on the music, then that noise contains only one millionth of the power of the music peaks, yet such a noise level would not even be deemed adequate for domestic hi-fi. When we can begin producing loudspeakers capable of producing an output faithful to the input to the degree of only having one part in ten million of spurious products, then we may, possibly, be starting to look at *science* of loudspeaker manufacturing.

So many of the imperfections audible in the output of a loudspeaker, are masked by greater imperfections. When a greater imperfection is removed, the lesser ones become audibly apparent. The problem continues virtually *ad infinitum*. Obviously, large differences in harmonic distortion figures between two comparable drivers, would no doubt lead to the lower distortion unit being deemed preferable when subjected to listening tests, but even when distortion comparisons show ratios of 3 or 4:1, other characteristics can still override and mask the distortion differentials. Every loudspeaker component and accessory has a natural period of resonance. Every item is different due to individual molecular structures, fibre lengths, adhesive thicknesses, resin densities, and a myriad of other functions. Mechanical shocks or electro-mechanical impulses will excite all the component resonances. I believe that this has a greater bearing on the perceived tonal character of a loudspeaker to a far greater extent than the steady state production of lower order harmonic distortions; the figures usually quoted. This is why I have been considering impulse time ▷

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UNIT 3 CRUSADER ESTATE 167 HERMITAGE ROAD LONDON N4 1LZ FAX: 01-809 1515 histories as being so important. We are currently getting good correlation between the impulse envelope shapes of loudspeakers that sound similar, even when harmonic distortion patterns are quite noticeably different.

Of course, harmonic distortion is important but when reduced from gross levels, it is no more important to perceived sonic character than a whole host of other properties. We do, now, have compression driver/horn combinations that are being perceived as sonically similar to direct radiators. Traditionally, compression drivers and horns have been aimed at the sound reinforcement market, where sensitivity, extended range and directivity control have been major design parameters. In studio use, where the full output potential is not required; and especially in mid-range applications where extension of high frequency performance is not a pre-requisite, much lower distortion figures are now achievable. By reducing the compression ratio in the phasing plug which reduces sensitivity, together with larger diaphragm to phasing plug spacing, which is possible where very high frequency performance is not a consideration, harmonic distortion can be reduced greatly. The reduced sensitivity still allows a 10 dB advantage on typical direct radiators.

While on the subject of compression drivers, Ray Ellaby made the statement that recording engineers had rejected them as 'horrible'. This is exactly the point I was trying to make. Horns seem to have all been tarred with the same brush as only a very few designs have been specifically and thoroughly designed and manufactured solely for studio use. Engineers have judged PA horns and drivers shoehorned into monitor cabinets, often by companies who should have known better, then adjudged them 'hard and horrible'. No wonder, they were never designed for that application. It is like judging all cars by the performance of a bulldozer.

Mr Ellaby goes on to say that my comment about 120 dB levels at the mixing consoles are a joke. Roger Hayler at Nomis, UK, with a degree in acoustics from Salford University, has measured much higher levels (four times) on many occasions, in their very large control room, that even taxes their Kinoshita monitors, with their mid-range horns 16 ft from the listening position.

Inner ear distortions cannot safely be cited as a consideration because at those levels, that is what we hear no matter whether the sound is coming from a monitor system or 3 ft from the bass guitar amplifier and drum kit that originally produced the sound. Inner ear non-linearities are not the concern of the monitor system, whose purpose is to compare like with like.

I have recently been approached about control room designs of 50 ft front to back. The rooms are becoming huge. I am not developing loudspeaker systems for general sale to the public, I am investigating the technologies I need to supply clients' design needs. I have to optimise harmonic distortion, as only one of many other parameters, over the whole performance envelope of a particular room/monitor combination. If my clients wanted domes, they would ask for domes. Having said this, however, I sincerely believe that the ATC dome is a very fine driver—in its place.

Mr Ellaby states: "Incidentally, horns usually have far worse irregularities in their frequency response than direct radiators." Usually? I was discussing highly specialised horn designs for static monitoring purposes. This once again highlights the point I made earlier that all horns have been lumped together. As with the comment implying that I have not been measuring horn distortion, once again, we are looking at highly specific designs and I do not believe that harmonic distortion is the be all and end all.

By suggesting that impulse time histories are of little significance as no conclusive result has fallen out of the magazine's "discussing the subject to death", Ray Ellaby implies that all is known about the subject. All is *not* known about the subject, and far from it! Hence much of our current work with the audiologists.

He goes on to say that direct radiators do not have any of the problems peculiar to horns. But we were discussing problems peculiar to horns—direct radiators have other problems. I *am* not and *have* not been claiming that horns are superior in all respects to direct radiators. I am in no way putting down direct radiators, I use them more often than I use horns. The whole thrust of this research is to achieve sonic performance *compatibility* between horns and direct radiators but at a level, a sensitivity and a ruggedness that direct radiators are not currently capable of achieving.

Ray Ellaby states that companies with bigger budgets than my own have sought to improve horns and have failed, and so will I. Firstly, I do not have, nor am I associated with, any company. Secondly, the larger budgets of certain companies *still* tend to look for cost effective answers for their largest market, studio applications have in their terms, not been worth the effort. I decided to approach the Institute of Sound and Vibration Research; with 300 brains and being one of the world's largest acoustics research institutes, who could be better equipped to attack this problem? Maybe I am a dreamer, but many people have already heard my dreams coming true. Let time be our judge!

I was indeed somewhat ambiguous in my reference to superior transient response of horns. The point I was trying to make was that lighter diaphragm tends towards greater sensitivity, which tends towards greater headroom, which tends towards cleaner transients. Remember, we were talking about high level monitors. As for horns sounding hard at all levels, I say again, we now have designs which do not; horns which have never been chosen for comparative listening tests even *at low levels*! At high levels, the harshness is being reduced by lower compression ratios in newly designed phasing plugs, which are in themselves much easier to design when one is not looking for responses out to 20 kHz and beyond.

Once again, I assume that Ray Ellaby is primarily considering harmonic distortion perception at 1 kHz when discussing crossovers, he clearly cannot mean phase distortions as he claims they are inaudible. It has been my personal preference to maintain the 1 to 6 kHz bandwidth from one driver, and in general I have avoided the 3 kHz region as a crossover point. From this point of view, a 1 kHz lower limit for a horn is of no disadvantage. On the 1 kHz vs 6 kHz crossover point subject, I think that Ray Ellaby and I will just have to agree to differ. I stand by what I believe, though I can by no means claim it to be absolute fact.

The same applies to the subject of the audibility of phase slopes. The open literature is just that—discussion documents. There are as yet no known absolutes.

The field of direct and non-direct sound is one in which Ray Ellaby and myself are working in very different areas: 90% of the monitor systems I produce are installed in rooms where I have great control over the acoustics. I have always maintained that the loudspeaker systems and the room cannot be separated, they are part and parcel of one and the same installation. If anybody thinks they can achieve accurate monitoring by the hit and miss approach of hoping that the precisely desired amount of frequency conscious reflexions will bounce off the walls is taking an enormous risk. On this point, we must in all seriousness, have some degree of understanding for the ridiculous task loudspeaker manufacturers face. People buy their products, and from that point on, the manufacturer usually has no control whatsoever over their use. It is grossly unfair to judge the efforts of a loudspeaker manufacturer by virtue of having listened to them in an unsuitable room. Ray Ellaby's claims that the most natural sound field is set up by loudspeakers of wide directivity angle is nullified when one has control over the room environment. Remember, the series of articles is entitled 'Monitor systems' and in later months will begin to deal with the rooms themselves. The control room is part of the monitor system. However, Ray Ellaby says that if such large monitor systems and directivity control are required, then the room is too large to be an accurate control room. Well, achieving what the client needs is my whole function. If people now need 50 ft long control rooms, as a consultant, it is my job to achieve the best results I can from their specification. There are many, real world, practical applications, which are not ideal. Ideal in one area can mean a worse overall end result. My function is to achieve the best overall *working* compromise. Hence the *art* of acoustic design; the number of variables is far too great and frequently, the equations of the 'science' are non-linear—they cannot yet be resolved without the intuitive leap.

Martin Polon

Random notes from here, there and everywhere. Comment from our US columnist

Random notes from here, there and everywhere. Items of interest that do not make up a full column but which bear passing on from time to time. Where shall we start? Yes, let's look at the extreme high regard the world in general and the news media in particular hold for audio.

Summer 1989, USA, a major band tour on the road. Articles appear in a spread of major newspapers and regional magazines, focused especially on the band's hi-tech state-of-the-art persona. TV stations do in-depth reports on the tour and its equipment. Article after article, programme after programme, stresses fact after fact. This particular group travel with over 100 people, more than a dozen large trucks, numerous buses and 'makes an airline's day' by keeping them busy. Daily expenses run into tens of thousands of dollars. Numerous television cameramen, directors and technicians work on the event to capture it for the audience at hand and the audience at home. Video equipment and tasks are identified. The diversity of lighting is illuminated with a small army of spotlight operators in the rigging and several on the floor highlighted. The special lighting instruments including the high intensity Xenon lights and computerised lights are carefully profiled by the journalists. What the band and the entourage eat, how they and the show are kept secure, how they live on the road are all carefully delineated in various features. The daily movements of their 100 or so pieces of luggage and a full time luggage co-ordinator are analysed. And oh yes—by the way, the group are using a customised sound system.

Now, there is no one in our industry more concerned with the provision of superb entertainment and a superb experience for the audience than the band in question. Everything they do on tour is interesting precisely because they are such consumate showmen. Their concern with sound quality is legendary. The company that supply the sound reinforcement are similarly revered. They have achieved status as being at the top of their profession in concert work. No question that the other elements of the tour are awe inspiring. The tour has been variously described as travelling rock'n'roll amusement park. There must be 'one heck of a sound system' on the tour but you wouldn't know about it from the news media.

The media more or less tends to overlook the aural portion of the package. In our society audio is best heard and not seen. You will almost never read a review of a concert or a group performing and find out that the sound was ideal or perfect or even just good. You will usually learn nothing about the sound, unless of course there is a problem with it, in which case the operator will be tried, convicted and sentenced as a mass murderer—of music. Check out movie reviews.

Never, and I mean never, do you find anything out about the sound in the theatre or the overall quality of a movie's soundtrack/mix or the special audio effects, etc. No easy answers here but isn't it time the media recognised the fact that the public can hear as well as see.

October 1989. Beware the ides of October. Now boys and girls, let's look in our old friend Mr Calendar. Here is what we find.

- ITU-COM '89: First World Electronic Media Symposium and Exhibition, October 3rd to 8th, Geneva, Switzerland.
- SBE: Society of Broadcast Engineers Annual Broadcast Engineering Conference, October 4th to 8th, Kansas City, Kansas, USA.
- MITAS: International Broadcasting and Telecommunications Exhibition, October 12th to 14th, Milan, Italy.
- MIPCOM: Motion Picture and Television Media, October 12th to 16th, Cannes, France.
- UNIATEC: Organisations devoted to the technology of film exhibition and production, 17th International Congress, October 14th to 18th, Montreal, Canada.
- AES: Audio Engineering Society 87th technical meeting and exhibits, October 18th to 21st (revised from Oct 19th to 22nd), New York City, USA.
- SMPTE: Society of Motion Picture and Television Engineers 131st technical conference and equipment exhibition, October 21st to 25th, Los Angeles, California, USA.
- BROADCAST '89: World broadcasting exhibition, October 25th to 28th, Frankfurt, West Germany.

There were eight important events going on in the western world during October 1989, which all more or less focused on the wonderful world of electronic entertainment, its production and distribution. All eight of these events directly or indirectly overlapped with another event making it very difficult to attend more than one or two of the eight during the month. Which event was the most important to attend?

In the so-called good old days it would have been easier to decide. Either AES or SMPTE would have to be considered premier events but these days, the judgement call is a lot harder. If one is working on digital sound for broadcast, for example, and focusing on the international marketplace, then giving a paper at ITU or MITAS or even at Broadcast '89 might be as, or more relevant than, AES or SMPTE. If advances in film sound are your bag for the world market, then UNIATEC or MIPCOM might be more important than even SMPTE or AES. For the manufacturer, the judgement call becomes even more difficult. How do you maintain a viable presence if you must exhibit at AES, Broadcast '89, ITU, MITAS, SBE, and SMPTE all during the same month?

This is all getting to be so confusing that what was once an orderly industry has now become a less than functional conundrum. Granted October, like mid-April to mid-May, is the peak meeting time worldwide due to relatively desirable weather conditions. Nevertheless, it is unfortunate that this multiplicity of events is fragmenting the audio and visual industries and preventing greater co-operation as the issues of the next century loom ahead. Curiously, in the case of several of these events, planners were not even aware of all the conflicting programmes that were in juxtaposition with their evolution. It looks a lot like back to the drawing board time.

The question is, is there trouble in River City? Today it is not Professor Harold Hill of *Seventy-Six Trombones* fame asking the question but some of the leading lights of the audio industry. The trouble that has belatedly caught the attention of the audio pros is the process of legal precedent being set in the regulation of the audio business. Whether it is a commission setting sound pressure level limits for all workspaces in Europa 1992 or the City of New York trying to force outdoor concert performers to use city sound systems and city operators, the trend is clear. The audio business has become a target of the courts. Wrongful death suits are circulating in the US based, in one case, on a woman's miscarriage after attending a painfully loud concert and in another case on the suicide of a teenager after a concert. In all cases, significant legal action could forever clamp limits on sound pressure levels found in performance and in recording. Whether the constriction is via codified laws or the result of established legal and financial liability, loud sound is in trouble.

> "The trouble that has belatedly caught the attention of the audio pros is the process of legal precedent being set in the regulation of the audio business. The audio business has become a target of the courts. Wrongful death suits are circulating in the US"

There were those in the audio community who were upset by the most prominent case, that which reached the US Supreme Court, involving concert activity in New York's Central Park. The ▷

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feeling was that the audio industry should have been allowed to go to Washington and testify as to the absurdity of the legal positions *vis-á-vis* technology. What seems absurd to the audio community makes good sense to the legal community as a case reaches the benches of the nation's highest tribunal. The opportunity for such comment occurred back when the case was at the New York State appellate level; not when it reached the Supreme Court. The audio people had missed the boat on the earlier hearing, the lawyers tell us. But in fact, the time to have really dealt with the sound level problem was 10 years ago, at the close of the '70s. The answer was self-regulation and the industry worked arduously to avoid any kind of controls—even those that were internal. Despite clearly documented medical, communal and social impacts of high level sound, nothing was done that might impede the technical and creative freedom to be loud—as loud as technology would allow, if that was what was needed.

Curiously, at the same time a wave of conservative politics swept in—Reagan and Thatcher—and swept out the previous era of big government. The plans of the US Occupational Health and Safety Administration and of the British Home Office were thrown out the windows as the winds of political downsizing withdrew funding of regulatory functions. In both cases, moves were afoot to control the levels allowable for sound systems. Ten years later, with a furtherance of the maximum SPL threshold as the only artifact of progress, the legal world has acted. Now it seems terribly wrong for the audio industry to be forced by government or by liability settlements to turn down the volume. But in today's sharply focused climate of responsibility for the environment, it is interesting to note that civilians do not call loud sound 'high level' or refer to it as 'SPL;' it is known simply as 'noise pollution.' Without some internal setting of limits by self-regulation the inevitability of the court process and its penalty system seems irreversible.

Of course, no look at the entire spectrum of audio recording and reproduction technology could be complete without another review of the progress being made excrementally in audio studies and development. In the studies department, we find a crisis brewing in the continued predictability roll for roll of the acoustic performance of toilet paper. It seems that major manufacturers of toilet paper no longer keep a predictable formula for a given brand over time. The makers of toilet paper (TP) today have tremendous pressures involving conservation, Japanese consumption of lumber at above market rates and an overall problem of availability of raw material. Advertising and marketing costs are also much higher than the cost of producing the product itself. Despite the product's status as a vital commodity, there is a constant tinkering with the formulas so as to free even more dollars for the purpose of convincing the public of the increased viability (or is that pliability) of a specific brand of TP. Wet strength and overall robustness have also suffered of late, as reprocessed paper and lumber production by-products have been used in greater amounts in the so-called TP formula. In short, so are the fibres in toilet paper.

Now, what in the name of all that is logical does this have to do with professional audio, you ask? Well, in many recording studios, post-production houses, broadcast facilities, reinforcement sound mixing booths, home studios, A/V facilities, etc, there are monitor speakers. In many cases, toilet paper is used to absorb some of the high frequency energy produced by modern high frequency drivers. This approach is felt by most observers of the monitoring TP scene to be significantly better than using some kind of in-circuit attenuation to reduce energy at certain frequencies. The TP modifies the driver's acoustic output without affecting the frequency response of the monitor system the way electronic attenuation would. Honest folks, that is the way it is in many facilities. And quite rationally, those who use TP state that there is some good sense in using a kind of 'scrim' to change the relationship of high frequency output without modifying the response. In the same way, a photographer would modify a light source without electrically changing its colour temperature by inserting a 'scrim'. In fact, several manufacturers now make toilet paper 'scrim' kits with *Velcro* holders and mounts to be used on monitor systems.

"It is strongly rumoured that one of the more successful audio supply houses is stocking specific brands of toilet paper in large quantities and at high prices"

The problem with all this is that the audio practitioner, like the photographer with film, can no longer depend on a reliable source of a proven TP 'emulsion'. Also, like the photographer, audio specialists are beginning to stockpile desirable 'vintages' of toilet paper. Since a good deal of effort goes into evaluating certain papers for monitoring use, the only logical response is now to stock-up on the most desirable papers as soon as testing indicates a desirable set of characteristics. It is strongly rumoured that one of the more successful audio supply houses is stocking specific brands of toilet paper in large quantities and at high prices. Woe will befall the assistant who mistakenly grabs the wrong roll for the company loo!

And last but surely not least, a return to developments in the area of computerised toilets in Japan. As we discussed in a previous column, the concept currently being developed in Japan is of a luxurious toilet facility with warm water sprays, scented cleansing solutions and biomedical analysis of the passed effluent. In addition to the provision of pregnancy testing enhancing the range of medical analysis offered by the so-called 'smart john', published reports indicate a new direction in what is called 'audio assist'. It is felt by those developing the super toilet that the environment the toilet operates in is as important as the actual facility. Current theory holds that stress reduction and a state of relaxation will enhance the experience offered by the 'hi-tech' toilet. The use of stereophonic music to create an ambience will assist the user in successfully culminating his or her needs for ablution.

Several technologies for adding audio have been considered. The use of a pair of small speakers and an amplifier are obvious. Source material remains a quandary. There is some interest in using DAT but the current shortfall in software availability is a problem. Analogue cassettes would also serve but there is some concern that music not be repetitive and the scenario of the user trying to flip through a collection of cassettes while trying to resist the 'urge' is not a pleasant one. The interactive modem capability of the new range of toilets could allow the digital delivery of appropriate music. Now it seems likely if the concept of digital delivery of music to the home via telephone line is successful and if the concept of the 'hi-tech' toilet washes (or is that 'flushes'), then there will be digital toilet channels—with appropriate music therein. There could well be a digital toilet channel for 'number one' and a different channel for 'number two'. These channels, known as DT1 and DT2 would carry music appropriate to the moment. An example for programming capable of assisting the occupant of the hi-tech toilet with number two activities might be excerpts from the *March of the Toreadors* and the finalé from the *1812 Overture*. Of course personal taste would differ but the point is to provide music that would be 'inspiring'.

Although much of this particular report is being written slightly tongue-in-cheek, there is no truth to the reports that the first site in the US for manufacturing the 'hi-tech' toilet will be Flushing, New York, or that *Billboard* magazine is thinking about running a weekly 'Hot Seat 100'. In fact, the Japanese are deadly serious about this whole business. They see no humour here at all. The view held in Japan is that this is a growth opportunity in consumer products similar to that offered by television 30 years ago and automobiles 20 years ago. It is felt by some important people in Japan that along with high definition TV, the 'smart' toilet could be the high technology 'hot button' of the '90s. The need for reduced use of precious water supplies due to increased population worldwide will spur the mass replacement of water-wasting conventional toilets in the next decade and the Japanese calculate that there could be 500 million old toilets suitable for replacement. What with the current slowdown in the audio business, it may be time to work the audio for the 'Tidy Bowl' rather than for the 'Rose Bowl'.□

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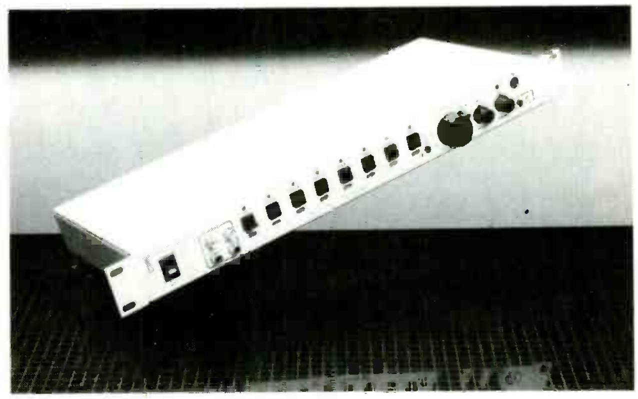

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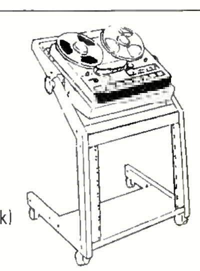

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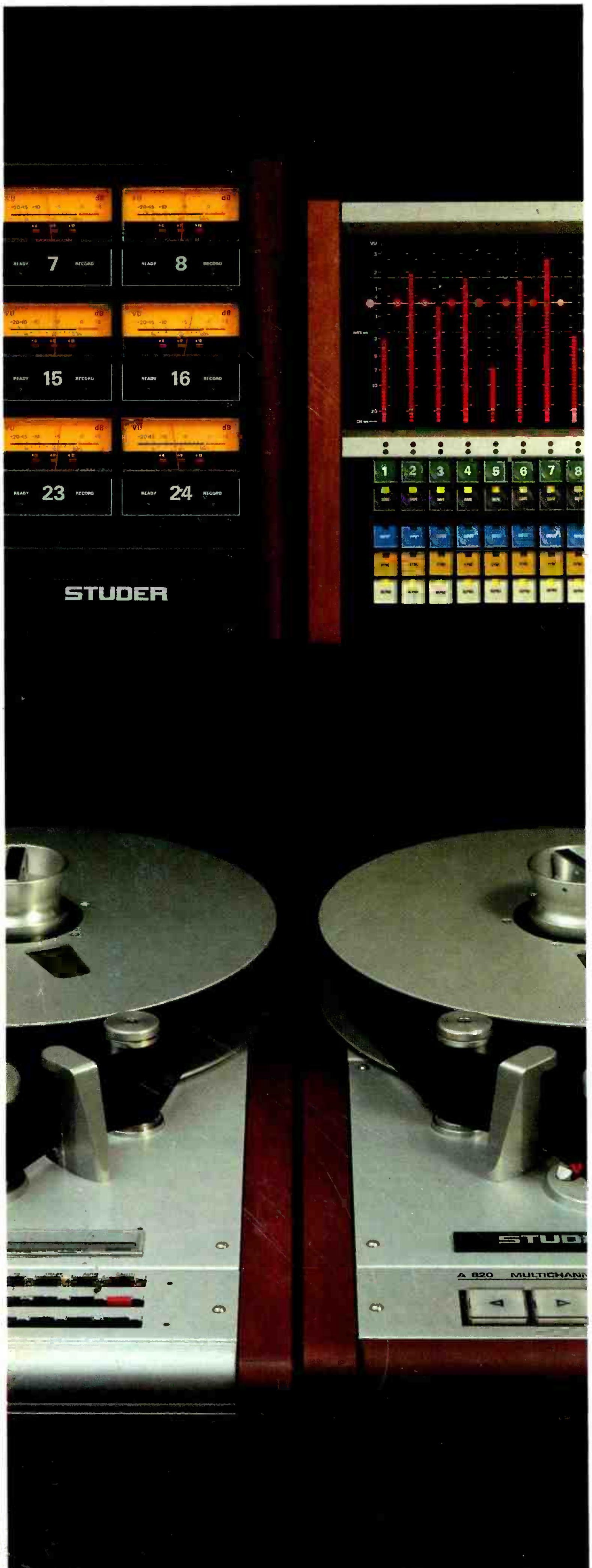

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