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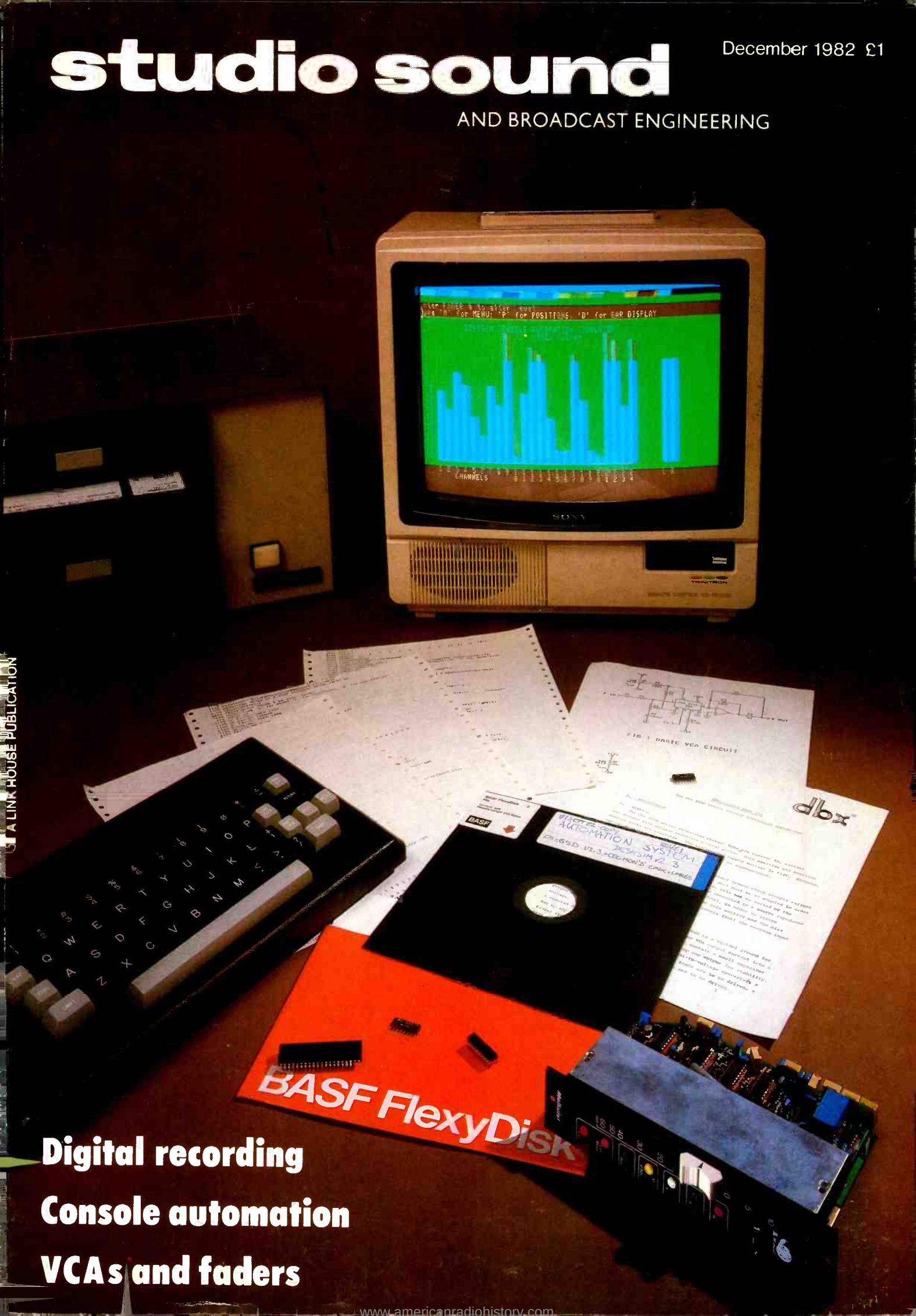
AND BROADCAST ENGINEERING

December 1982 £1

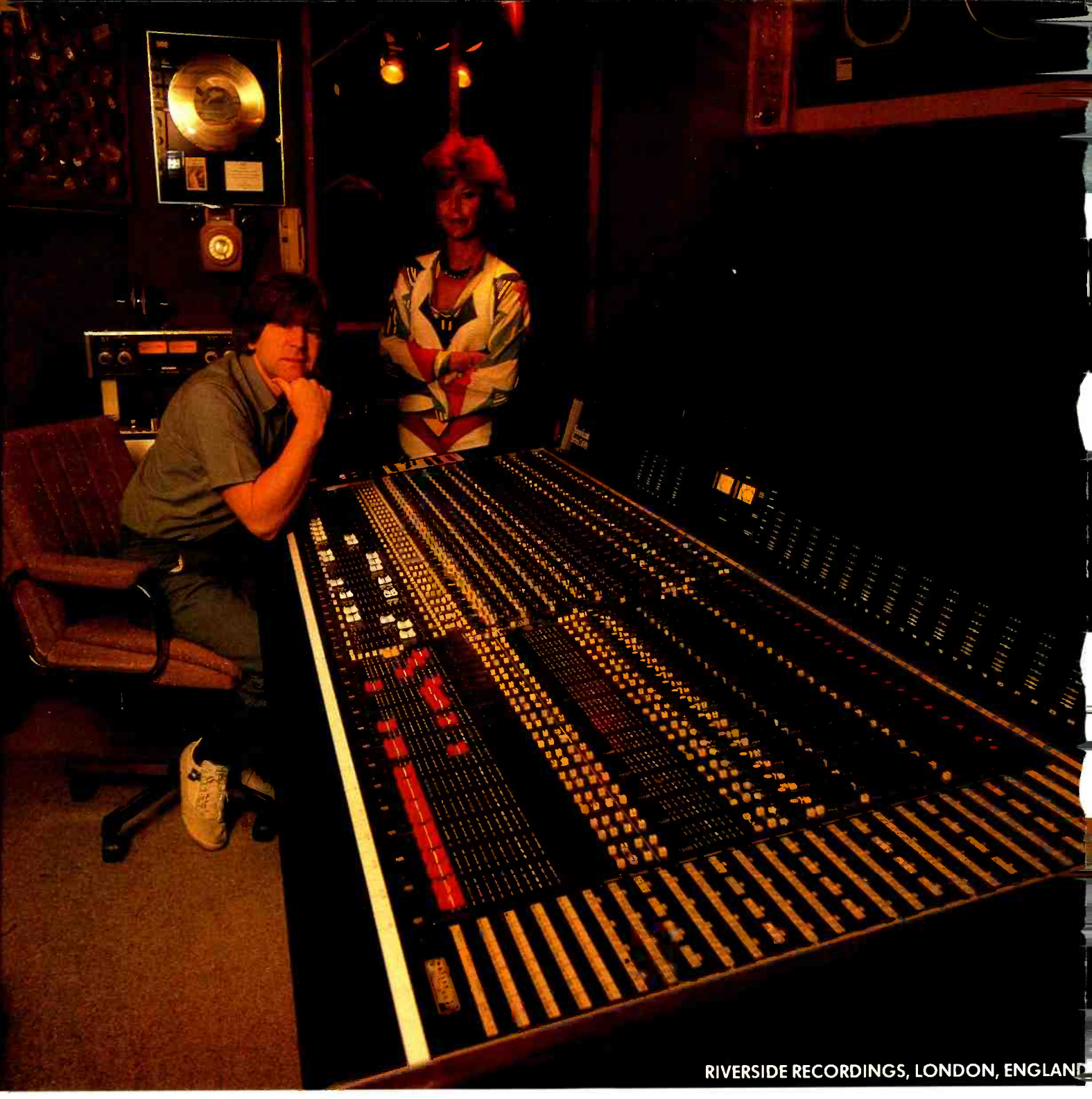
A LINK HOUSE PUBLICATION



Digital recording

Console automation

VCAs and faders



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AND BROADCAST ENGINEERING

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A LINK HOUSE PUBLICATION

ABC MEMBER OF THE AUDIT BUREAU OF CIRCULATIONS

Cover photograph by Roger Phillips

Contemplating digits

Now there is a whole gamut of digital recorders, both multitrack and stereo, on the market—most of which still won't talk to each other in terms of taking a tape from one machine and playing it on another, although there is a fair degree of compatibility in stereo standards—many studio owners must again be wondering whether or not now is the time to invest in the benefits of this way of working. It's a difficult decision.

Leaving aside the question of compatibility for a moment, there is a number of possible approaches. You can go all out for digital multitrack: an expensive business. You can go for an all-singing, all-dancing digital stereo mastering system and editor. Or you can invest in a PCM adaptor (for example, the Sony *PCM-F1* reviewed in this issue), knowing that there are enough digital editing suites around for clients to put masters together. Let's look at the options one by one.

Digital multitrack has capital investment as its main hangup. The digital multitrack machines around today definitely offer exceptional quality—it is interesting to note how much quality has improved even in the past year—but the money question is a big one, despite rumours of much cheaper machines 'in the pipeline' from unsuspected sources. If you buy a digital multitrack today, you will have to charge a higher rate to clients to use it (and if you have only one studio, you'll no doubt have to charge a higher rate whether clients use it or not). Will you attract enough clients who want the new technology to justify the expense? It is similar to the problem of investment in automation, except that it is on the one hand more expensive than many console automation systems, and on the other it will probably be more impressive to potential users. Is it a good idea? In the United States, the answer is 'possibly'. In Britain, I doubt if it's a good idea right now at all. I don't know that there's the work around at the moment. You'd probably make more money offering video-audio post-production instead. Wait until the economy improves a little.

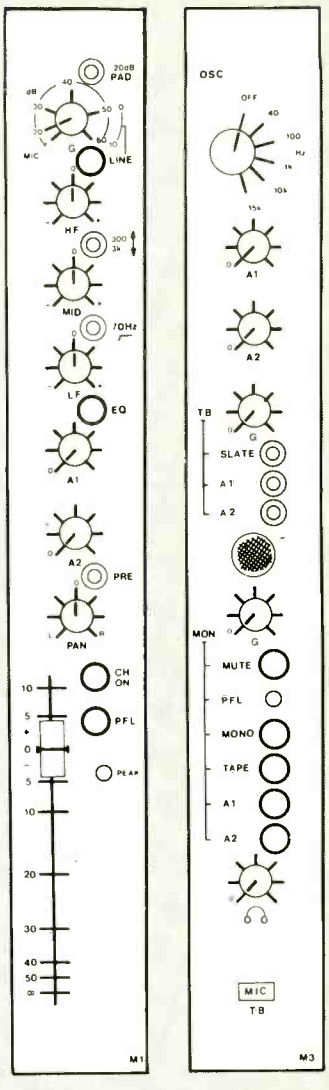
How about digital stereo, then? This is a far more cost-effective investment. And while cutting rooms may well want to invest in digital replay gear for mastering, they should also consider the possibilities of setting up digital audio editing suites where masters can be assembled. As with video, people are likely to be relatively easily convinced that it is not a serious problem to book time at a separate location for editing. This would be even more cost-effective, as an installation, for video post-production facilities, of course, where the requisite video recorders may already be available (but take care with PAL/NTSC standards where applicable). And with increasing numbers of studios investing in video gear for sound-to-picture, bringing in a few PCM adaptors for mixing down to and an editor to go with them is not such a bad idea.

For those who merely want to dangle a foot gingerly in the digital tide, devices like the *PCM-F1* offer a marvellous incentive. If you already have a VCR in the building, stereo digital will cost you *less* than even a 'reasonable' analogue machine. Tape's cheaper too! But here again, check very thoroughly with regard to standards to make sure that someone, somewhere, will be able to edit, and cut from, your tapes. Analogue tape-recorder manufacturers, especially those making portable machines, should begin to have thoughts about new product lines just about now, by the way.

And when the tape finally gets to the cutting room, be careful there too. Some early efforts have given digital a very bad name in audiophile circles. Some of it may have been due to quantisation and dither circuitry (unnatural reverb die-aways, for example) which is now much improved: current systems sound excellent off-tape. A bigger problem appears to be HF cutter-head resonance, which is alleged to cause a harshness at the top end, and we would appreciate comments from cutting engineers on the problems of cutting from digital and their solutions. How does one do it properly?

Overall, it would seem that there is a great deal to be gained from investing in digital stereo systems, but the present high cost of multitrack heavily offsets its audible benefits—this month. **Richard Elen**

A Sound Investment for Video and Broadcasting



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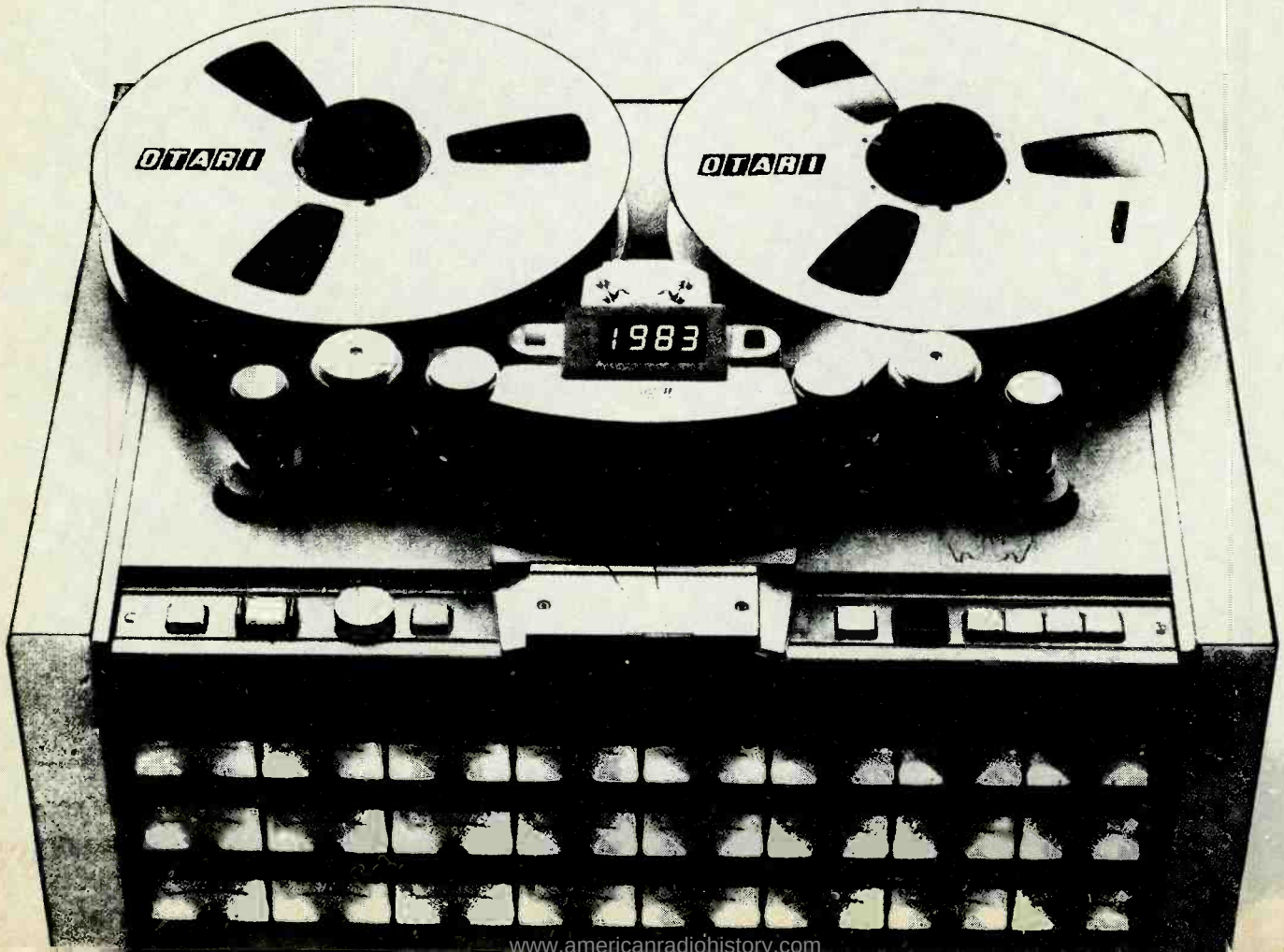

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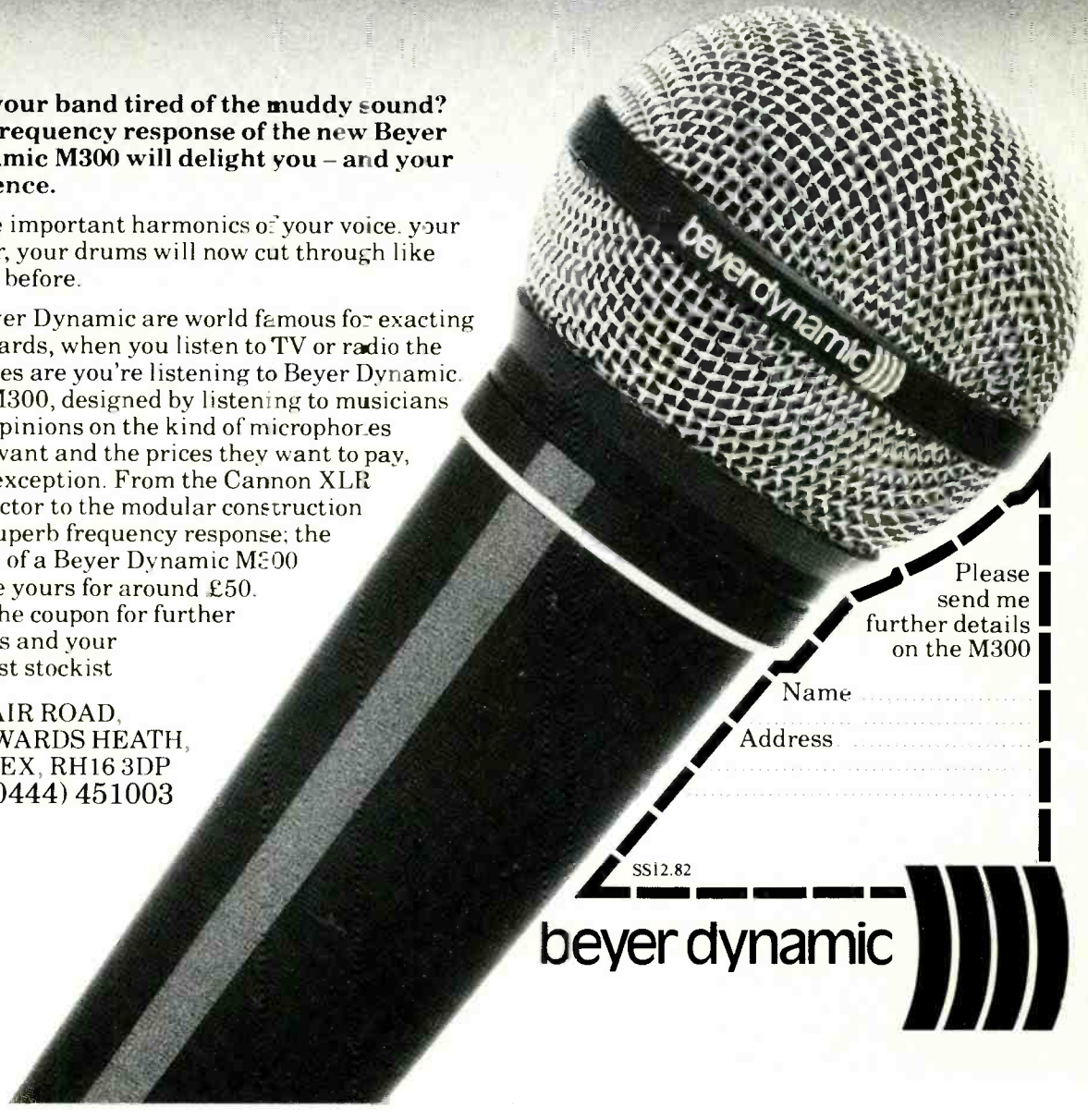

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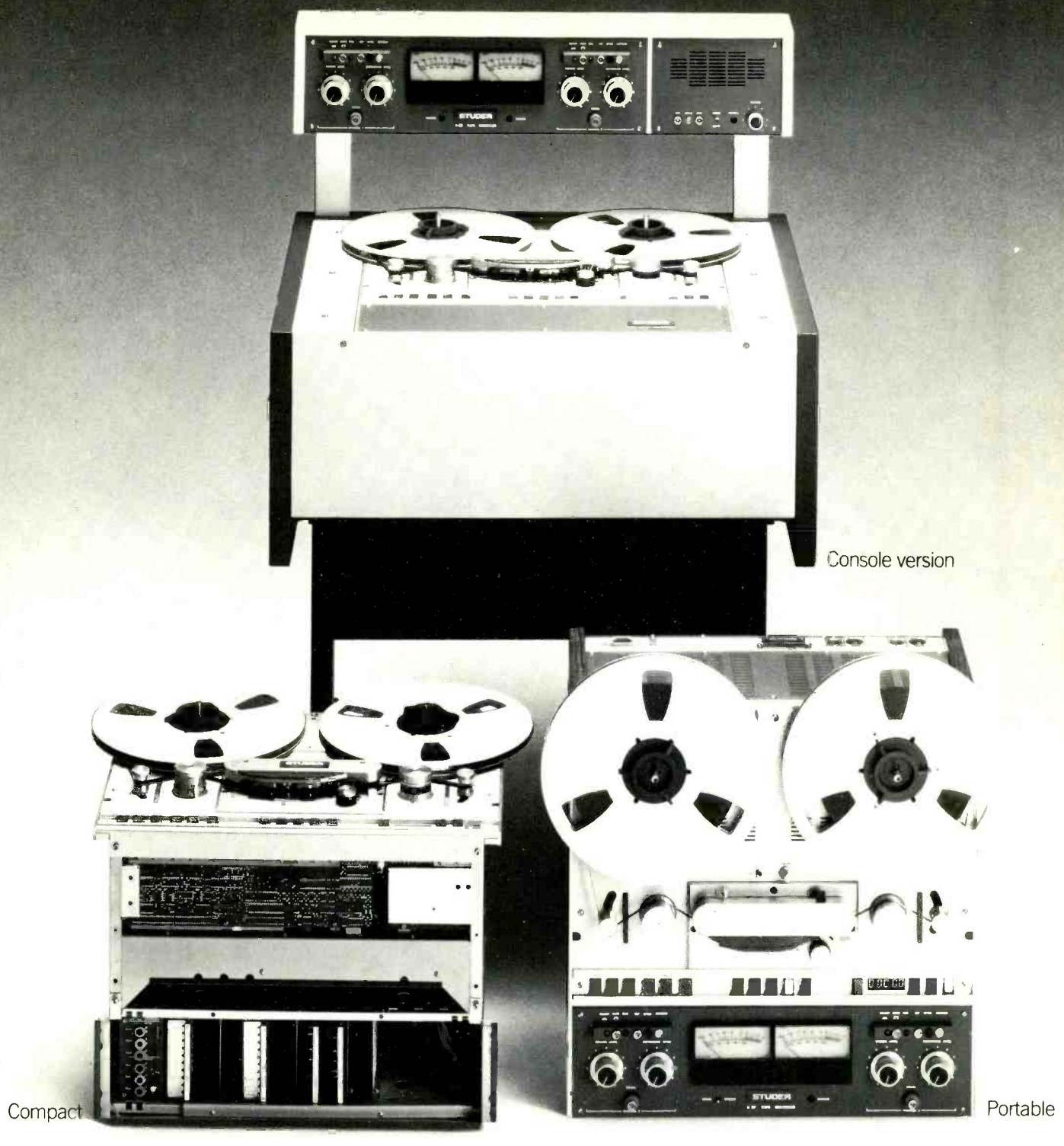

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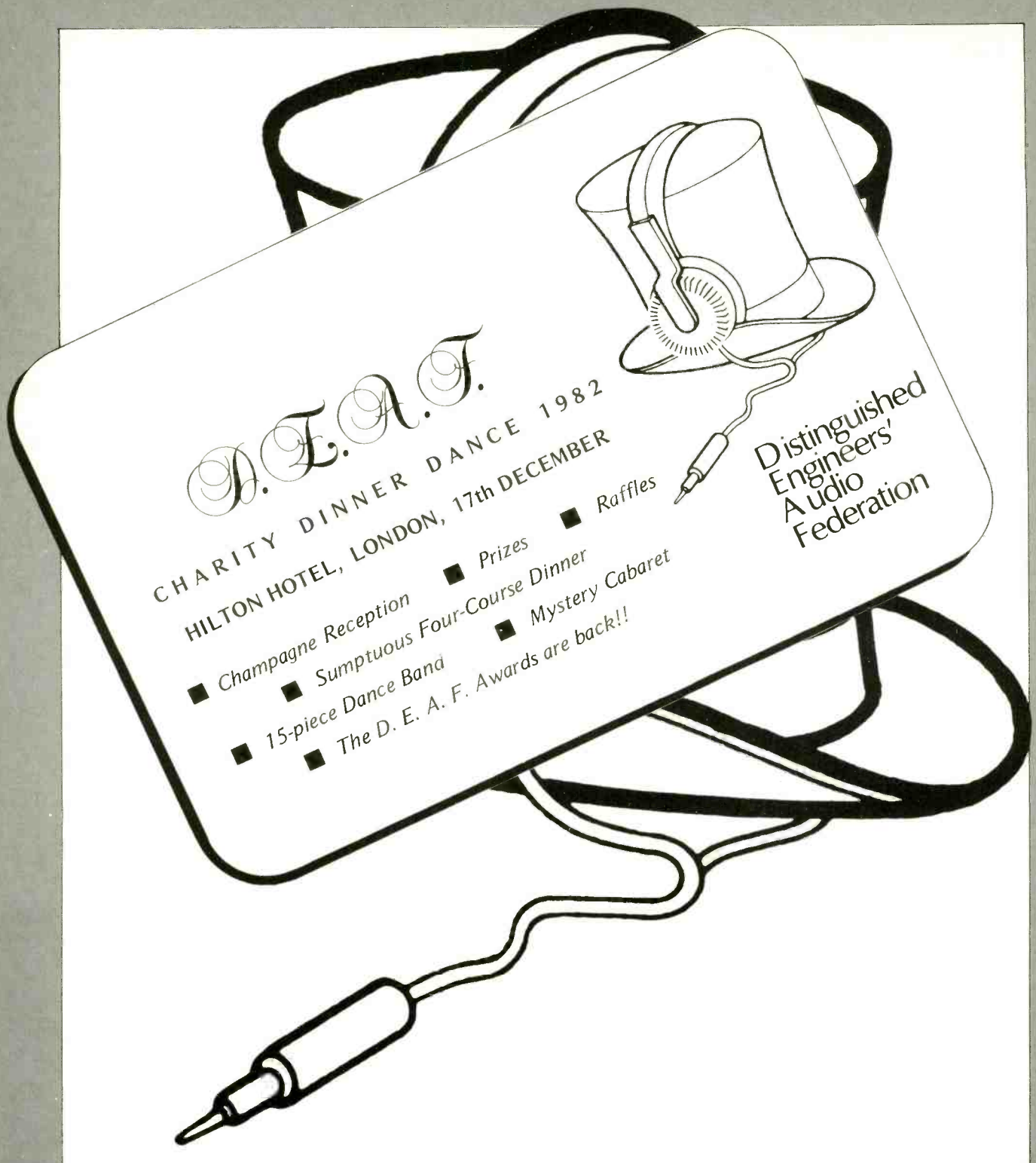

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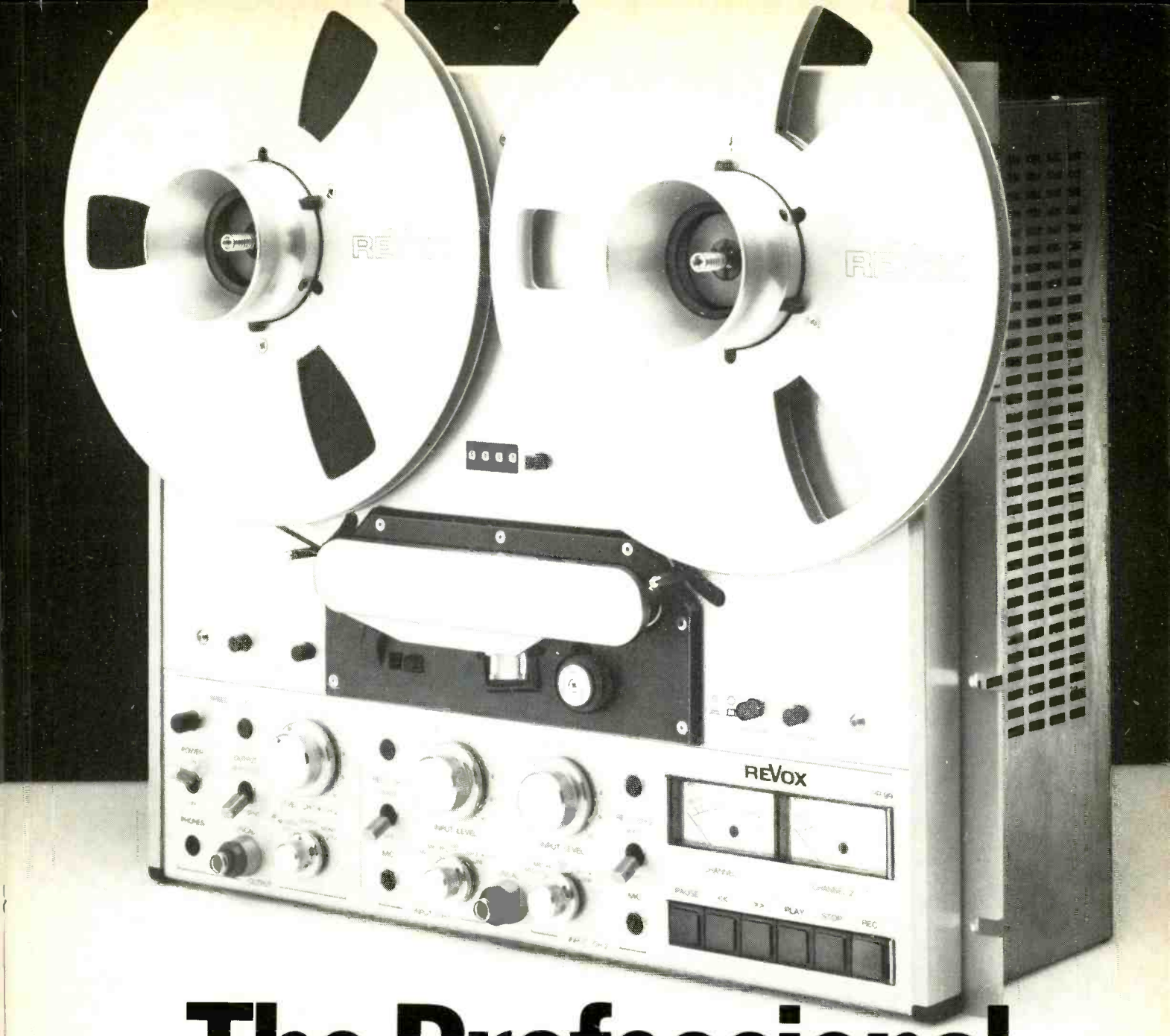

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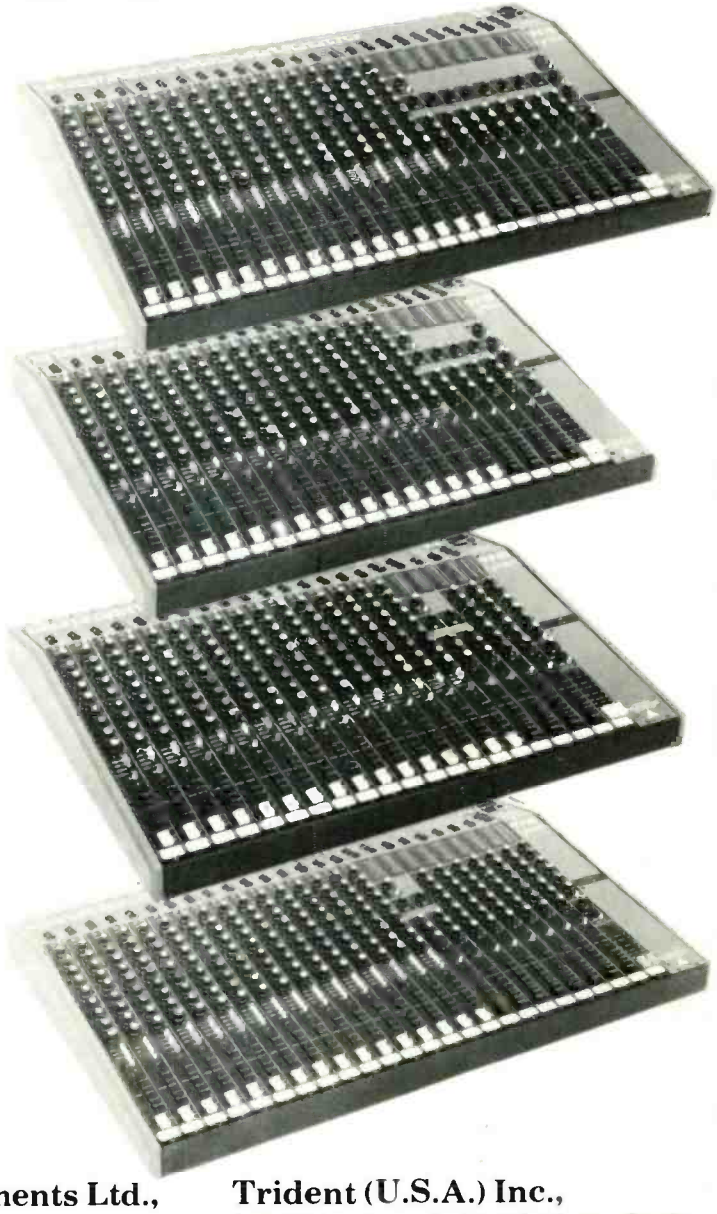

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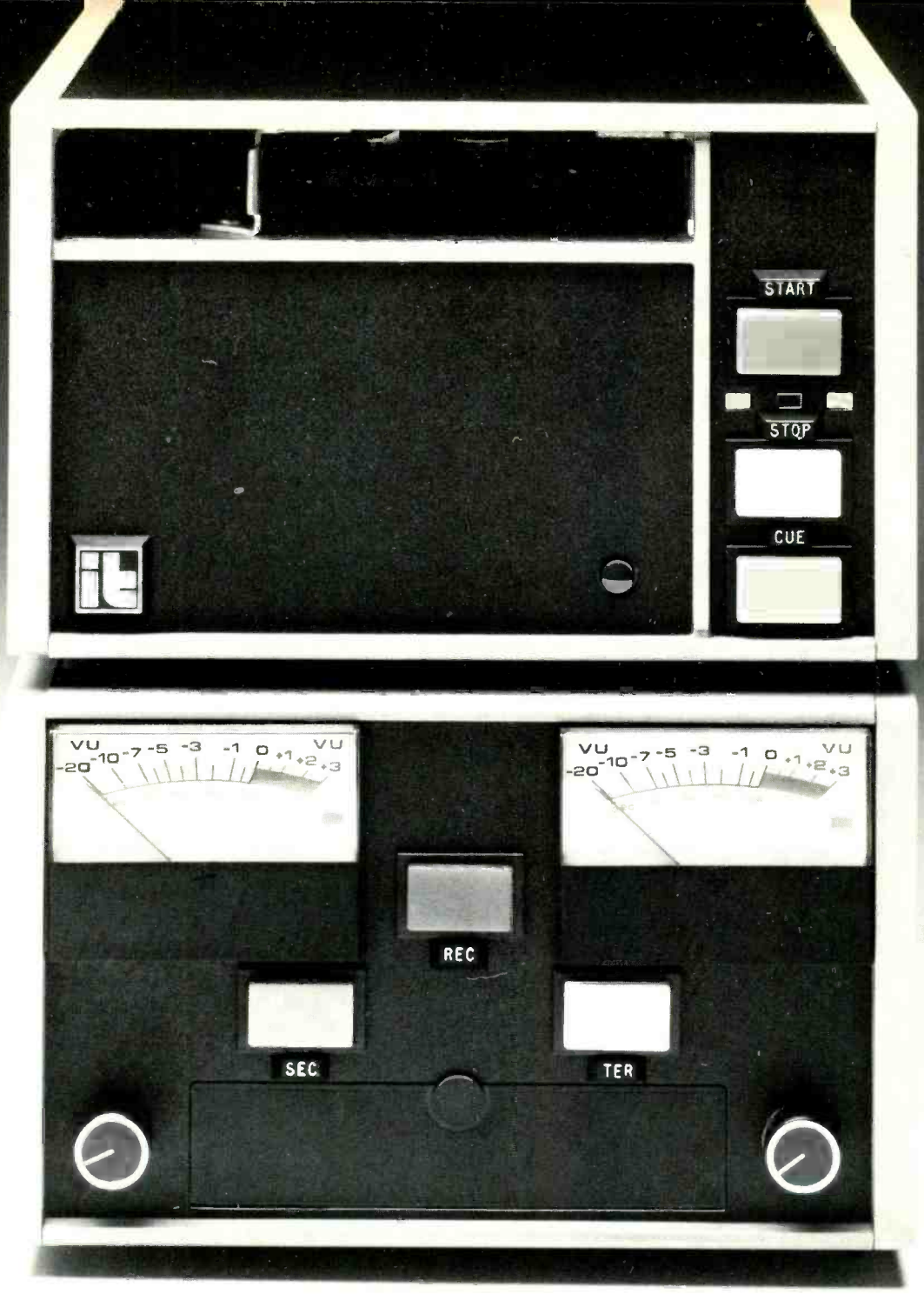

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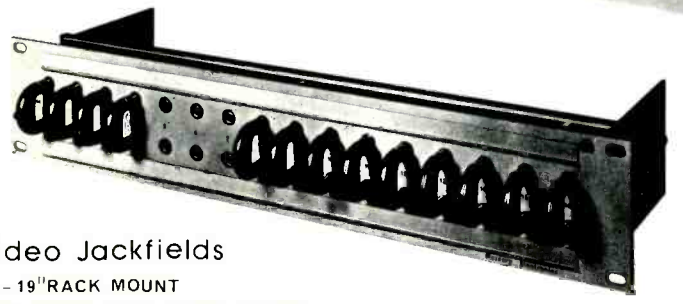

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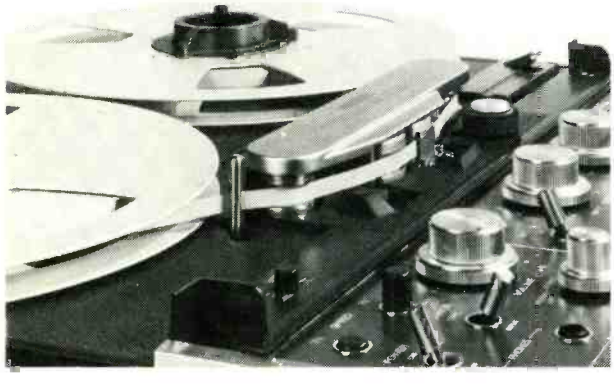

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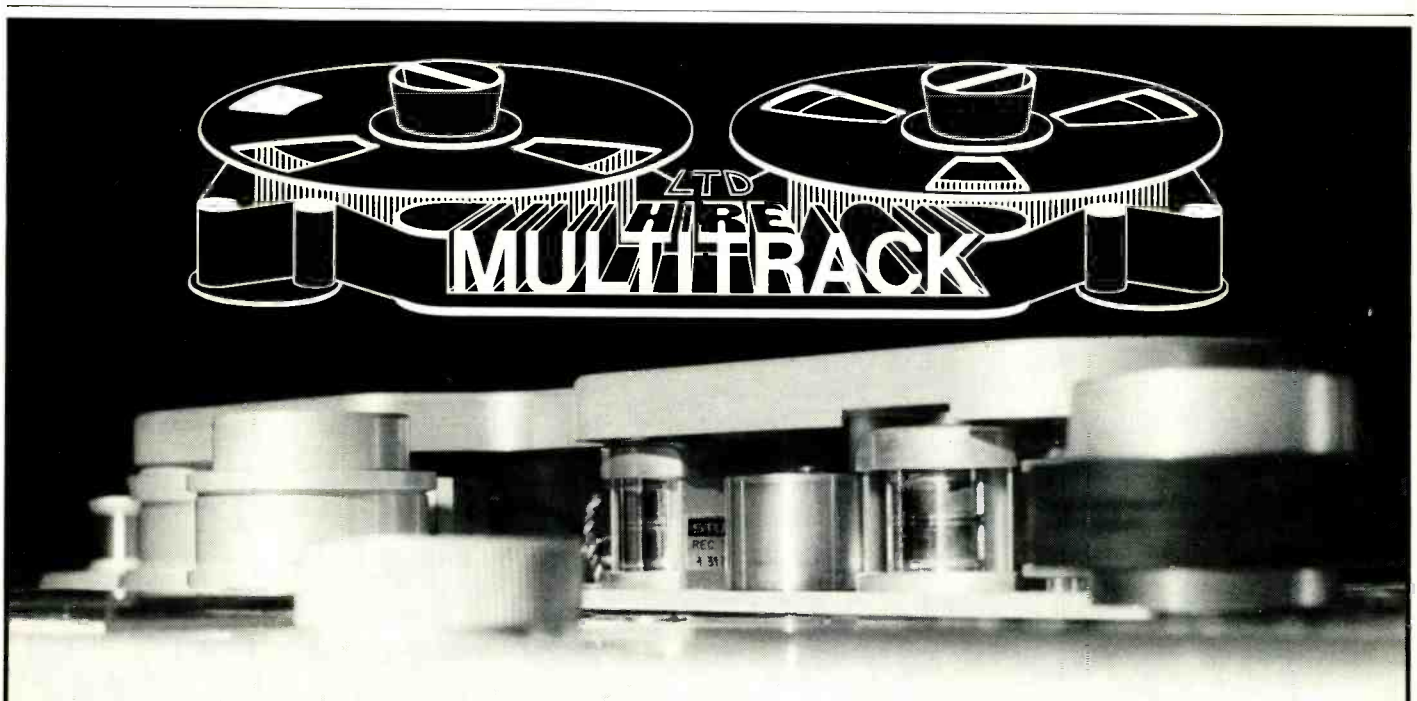

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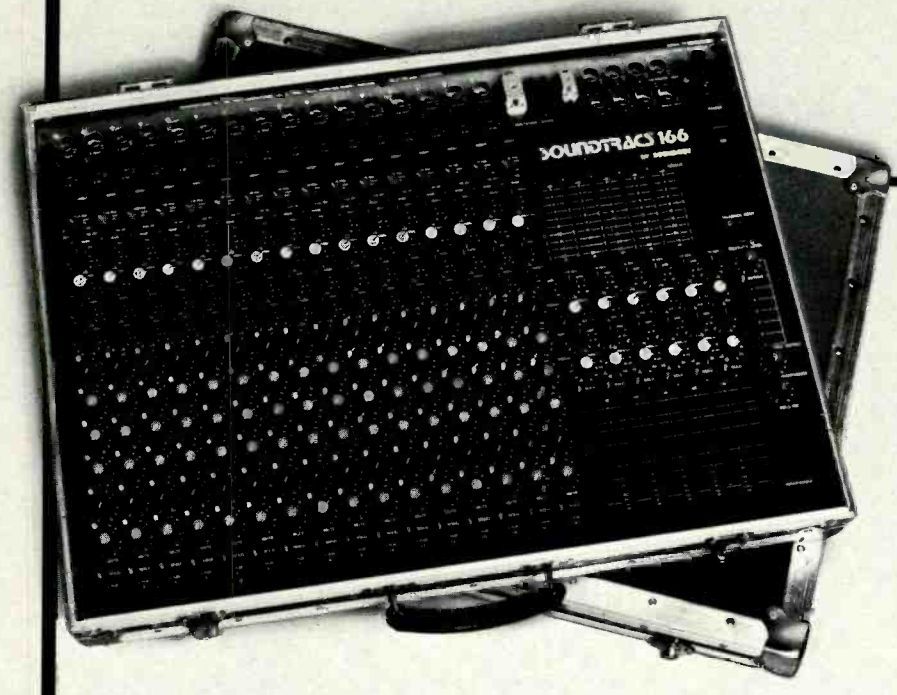

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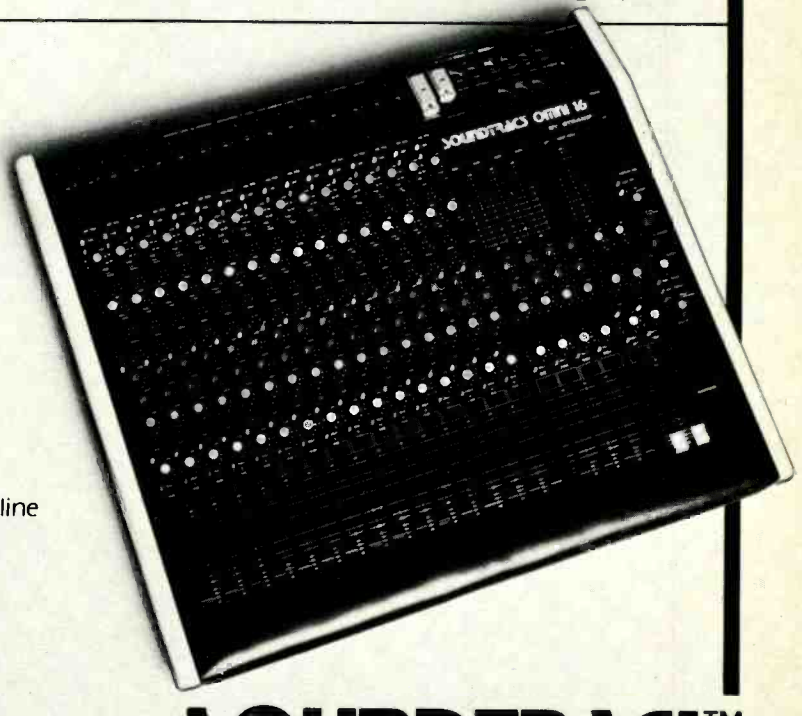

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Just grab it and go. **THE LOCATION MIXER** *was designed to travel. If there's no AC at your location, no problem. You've got the battery pack right in the lockable cover. No need to bother with batteries or power supplies for your mics.* **THE LOCATION MIXER** *powers them all – 12-48V. phantom, and even T-power is provided.*

You can set up everything from a single mono mix, to a stereo mix to an eight channel direct feed for your multitrack. You can even set up all three mix feeds at once!

You've got the flexibility to insert effects to individual channels. And you'll appreciate the convenience of the tape return to the monitor section.

An instant, easy way to check your tape, or the output of an outboard compressor.

With Mic/Line switches on all channels, audio and video sound people can monitor multitrack playbacks without patching – just press a button. Multitrack mixdown in the field? Absolutely. And with no compromise in quality because you feed line inputs directly – NOT padded down through the mic. preamp. You'll love the ability to edit electronically through the flexible Solo system and channel mute... It can be a great time saver.

Doing a live remote? **THE LOCATION MIXER** *is perfect. All four mix busses (2 main, 2 AUX) drive a 600ohm line to 20dBm (with balanced Xfmr option) giving both mono/stereo and back up feeds at the same time. And the ability to solo pre or post fader with the channel on or off makes set-up and trouble shooting a bad cable or mic. easy.*

What's in **THE LOCATION MIXER** *for film people? Plenty! No more battery or power supply hassles for your T-power mics. Just flip a switch. And there's a low-cut filter ahead of the mic. transformer to keep wind and boom movement from saturating your tape. This filter is independent of the EQ section. The EQ itself has sweepable mid, detented pots for resetability and can be totally bypassed with the EQ in/out switch. Need a quick talk-back system for your boom man? Just use the pre fade Aux bus on an unused mic. input, and he can hear his mic., cues, and your closed-circuit directions – his own custom mix!*

For reliability and quick positive switch indication, we've used self-indicating colour-coded push button switches that – along with colour-coded knobs – keep you informed of control status at a glance.

For your next location job, get a **LOCATION MIXER** *and you're ready for anything.*

***Just unlock it, open the cover and you'll find the most versatile 8- input mixer you ever uncovered. Compare features, compare performance, compare price. You'll choose* THE LOCATION MIXER.**

AT £975.00 CAN YOU AFFORD NOT TO?

For your **free** colour brochure contact sole UK distributor,

PROGRESSIVE ELECTRONIC PRODUCTS LIMITED, 83 Leonard Street, London EC2 01-729 5411 or see your nearest dealer – London: **ITA,** 1-7 Harewood Avenue, London NW1 01-724 2497 Kent: **MICHAEL STEVENS & PARTNERS,** The Homesdales Centre, 216-218 Homesdale Road, Bromley, Kent 01-464 4157 Manchester: **WIGWAM ACOUSTICS,** St. Annes House, Rycroft Avenue, Heywood, Greater Manchester, 0706 68766. **Manufactured in the UK by OXGLADE LIMITED.**

diary

Anniversary

On December 19, 1982, the BBC is celebrating the 50th Anniversary of the official start of the Empire Service (now renamed the External Service). To commemorate this, Ariel Radio Group has obtained special call-signs and will be using them during the period December 1 to 31.

The amateur radio stations will be GB2BBC, GB3BBC and GB8BBC in Central London, G3BBC in West London and GB4BBC at Caversham near Reading. In addition several other BBC Club Stations around Britain will participate. The bands in use will be 80 m, 40 m, 20 m, 15 m, 10 m and 2 m, and maximum activity will be centred around December 19. SSB will be the main operating mode on HF.

A special QSL Card will be issued for contacts made with these stations.

Solid State for Editel

Editel in New York, a division of Columbia Pictures, opened a state-of-the-art sound mixing facility in September. Mixing capabilities consist of all forms of video programming plus regular album recording. Virtually every tape format will be available, from 2in to ¼in plus cart machines, 35/16mm film, and VTRs. Studer recorders will be used in conjunction with a custom-designed SSL *4000E* automated console.

The Editel mix room has been designed by Vin Gizzi to match any major recording studio control room, the three basic specifications being top acoustic performance, technical capabilities, and working atmosphere.

All machines will be capable of operation with or without SMPTE timecode and can be locked to picture at all times. SSL's events and effects controllers will also be installed along with the console.

Editel, 222 East 44 Street, New York, NY 10017. Phone (212) 867-4030.

APRS digital recording survey

The Association of Professional Recording Studios has published the first impartial study of digital recording techniques and available equipment, the Association having decided that a full-scale handbook, along the lines of *Sound Recording Practice*, would be impractical at the present time due to rapid developments in the field. Instead, this review takes the form of a technical booklet, edited by APRS committee member David A. Pickett, lecturer in recording techniques at the University of Surrey. Other contributors from the industry include fellow committee member Roger Cameron of Advision studios.

The booklet covers the history of digital recording, a description of its workings, digital recorders, monitoring and mixing digital tapes and editing them, an appraisal of digital standards, and a recommended reading list. There is also a comparison table of available equipment culled from manufacturers' responses to a questionnaire.

The booklet is available free to APRS members, but will also be available generally for £3.

APRS, 23 Chestnut Avenue, Chorleywood, Herts WD3 4HA.

Low-cost digital from dbx

Unveiled at the recent AES Anaheim Convention was a new, low-cost digital audio processor from dbx Inc. Called the dbx *700*, the unit utilises delta modulation (in which changes in levels are recorded rather than absolute values as in PCM) plus precision companding and linear prediction technology. Thus the system is termed Companded Predictive Delta Modulation (CPDM). The inherent cost-effectiveness of delta mod plus the other aspects of the system—which allow DM to be used to professional standards—combine to produce a system of high quality yet low cost (under $5,000), thus making digital recording an affordable proposition for the majority of studios.

The inclusion of companding circuitry eliminates a major problem with DM, namely that there is a limitation to the available dynamic range. Adding high-precision companding offers a range of over 110 dB. Linear prediction is used to 'predict' the future of a signal by monitoring its recent past. This is done 700,000 times per second and further enhances the dynamic range capability. This circuitry combination additionally eliminates the need for costly anti-aliasing filters as used in PCM systems. Specifications are given as follows.

Two-channel system utilising VCR storage medium; frequency response 10 Hz to 20 kHz ± 0.5 dB; dynamic range (unweighted, maximum RMS signal to noise floor, input shorted, noise bandwidth 20 kHz) >110 dB; wow and flutter <0.01% unweighted, <0.006% RMS; THD (1 kHz, 1 V RMS input) less than 0.03%; sampling rate 700 Kbits/s; error correction complete up to eight times per video frame ($\frac{1}{30}$ s) for a 1024 bit burst error; precision-companded, linear-predictive delta modulation for A/D conversion; two columns of LED metering (switchable between record level, pre-emphasised), 60 dB range; wide range signal-level meter, unweighted RMS, dBV reading, 120 dB range; and loudness meter, 120 dB range, matching Stevens curves to within 2 dB); optional mike preamp adding less than 1 dB to mike noise for all mike impedances from 100 Ω to 1 kΩ, balanced *XLR* in; headphone jack; balanced line-in at 10 kΩ, electronically balanced line-out driving up to +24 dBm into 600 Ω; NTSC 75 Ω, 1 V p-p, BNC connectors for video in/out; audio in for editing, balanced 10 kΩ impedance; audio out (also for editing), unbalanced, drives 2 kΩ or greater; dimensions 19 × 5¼ × 11½ in (whd); weight 20 lb approx.

Corrections

In the Tape Duplication product guide, August issue, there are two corrections to be made under the entry for Recortec Inc. The address should read Recortec Inc, Mountain View, California 94043, USA. Phone: (415) 962-0220. Telex: 910-379-5022. Additionally the last sentence should read 'Duplicating ratio 50 and 100:1; speedmaster 375 in/s, slaveloader 188 or 94 in/s (for 7½ or 3¾ in/s master tapes).'

It has also been brought to our attention that there was an error in Control Room Acoustics Part 1 published in July. The equation on page 107 second column should read

$$10\log\frac{1}{100} = -20 \text{ dB}$$

and not as printed This was a printing error that crept through and Andy Munro's original manuscript was in fact correct.

Finally, in the October issue Intercoms and talkback product guide, we gave the incorrect telephone number for Millbank Electronics, Uckfield, Sussex. It should read 0825-4811. Apologies for any inconvenience.

Free Ampex with Studer

For one year, from September 1, 1982, purchasers of Studer professional recorders in the United States will receive a free reel of the appropriate width of Ampex *456 Grand Master* audio tape. In addition, the machines will be aligned for the Ampex product.

Agencies

● Electro-Voice has changed its distribution in the UK and Ireland and in future it will be handled by Shuttlesound Ltd, 200 New Kings Road, London SW6. Phone: 01-736 0907. Shuttlesound will have responsibility for all Electro-Voice and EV-Tapco audio products in these countries in addition to assuming existing guarantee obligations.

● The UK agent for the Enertec range of tape machines is now Crow of Reading Ltd, PO Box 36, Reading RG1 2NB. Phone: 0734 595025. Telex: 847056.

People

● Peter Thompson has been appointed Field Sales Engineer covering Eastern England for Bourns Electronics Ltd.

● The National Council of Acoustical Consultant, Springfield, New Jersey, USA has elected Douglas Muster, a consulting engineer from Texas, as it's new President.

● Jon Raper and Rodney Wayman, ex-employees of the now defunct Studio Equipment Services have set up their service, sales and hire company for professional audio equipment. They can be contacted at 34 Danbury Street, London N1 8JU. Phone 01-359 9342.

● Harrison Systems Inc have appointed Ken Fay as a factory marketing representative. He will be responsible for the administration of sales and support of all Harrison products for broadcast, recording and live performance on the West coast of the USA and Canada.

Contracts

● Scenic Sounds Equipment has just completed the installation of a customised 36 input/24 output *M2500* console for the new Molinaire video complex. This is the first *M2500* post-production console to feature the new stereo input module and was installed in record time to meet Molinaire's tight schedule.

● ITA have recently installed Otari equipment in Windmill Lane Studios, Dublin, an *MTR90-II* 24-track and *MTR10* 2-track; Music Factory, Cardiff, *MTR90* 24-track and Soundcraft *2400*; Rediffusion Music, six Otari 2- and 4-track *MTR10*s; Castle Sound Studios, East Lothian, an *MTR10* 2-track with ¼ and ½ in headblocks to add to their new *MTR90* 24-track.

Steve Graham Audio name change

With effect from 26th July, 1982, Steve Graham Audio Ltd, 20 Victoria Road, New Barnet, Hertfordshire EN4 9PF, UK, changed its name and will now be known as Connectronics Ltd. All other details remain unchanged. ■

You may think that at less than a pound a watt, you are in for a bargain.

The specifications are impeccable, and every feature on the Fostex Lab series amplifiers, suggests that they are built for professionals.

SPECIFICATION	MODEL 300	MODEL 600
RMS (Min.) Continuous Power Output: per channel 20Hz to 20KHz (both channels driven) Power Output Mono	8Ω 100W (THD 0.02%) 4Ω 150W (THD 0.05%) 2.5Ω 200W (THD 0.1%) 8Ω 300W (THD 0.1%)	8Ω 200W (THD 0.02%) 4Ω 300W (THD 0.05%) 2.5Ω 350W (THD 0.1%) 8Ω 600W (THD 0.1%)
Total Harmonic Distortion (THD):	0.05% Max	0.05% Max
Intermodulation Distortion (IM): 250mW to rated power at 8Ω with any 2 mixed frequencies between 10Hz and 30KHz at a 4:1 ratio	0.05% Max	0.05% Max
Frequency Response at Rated Power	+0,−0.5dB 10Hz-50KHz	+0,−0.5dB 10Hz-50KHz
Signal to Noise Ratio	Greater than 97dB	Greater than 95dB
Transient Response (any square wave)	1.5μs rise/fall time	1.5μs rise/fall time
Damping Factor	150 Min (8Ω 20Hz-20KHz)	150 Min (8Ω 20Hz-20KHz)
Input Sensitivity	0.8VRMS (for rated output)	0.8VRMS (for rated output)
Input Impedance	10KΩ	10KΩ
Stability	Unconditionally stable with any type of load including full range electrostatic loudspeakers	
Semiconductor Complement	67 Transistors 58 Diodes 1 IC	75 Transistors, 58 Diodes 1 IC
Heat Sink Surface Area	3700cm²	6100cm²
Power Transformer VA	700VA	1.3KVA
Overload Protection	1. Low impedance electronic sensing circuit limits with output current below 2Ω without limiting with 2.6Ω or higher 2. Thermal sensing of inadequate ventilation	
Loudspeaker Protection	Relay circuit protects loudspeaker from low frequency oscillations and D.C. offset voltages. Five second turn on/off delay eliminates on/off transients	
Power Consumption	700W	1800W
Shipping Weight	12.5Kg	16.5Kg
Dimensions: (W x H x D)	482 x 88 x 378	482 x 132 x 378

Like the accurately stepped input attenuators and LED display. Or the array of input connectors for looping and linking, and the massive torroidal and huge fins helping keep things cool.

Built around a sturdy chassis, they are meant to withstand normal handling. However, and this is the only catch, they do need proper protection if you intend to throw them around.

After all, a precision instrument that's priced as if it fell off a lorry, shouldn't need to.

For more information, call Chris Collings or Ivor Taylor now on 441 1133.

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letters

Expert's errors

Dear Sir, I refer to Don Davis' efforts to confuse in his article Expert's Errors in the October 1982 edition of Studio Sound.

I'll start with his second example – the Bel is a relative logarithmic unit of power, ie it refers one power level into a given load to another power level into the same or another load.

$$Bel = log_{10} \frac{Wa}{Wb}$$

This turns out to be an inconvenient formula in practice as the Bel is too large a unit for convenience and also measuring power is not a common measurement. To resolve the first matter the decibel – one tenth of a Bel came into common use.

$$decibel = 10 \times log_{10} \frac{Wa}{Wb}$$

Rather than measuring power it is far easier to measure voltage across a load or current in the load. Remembering that power in watts = voltage squared or current squared multiplied by the load the decibel may be defined as follows provided that the load remains unchanged:

$$decibel = 10 \times log_{10} \left(\frac{Va}{Vb}\right)^2 = 10 \times log_{10} \left(\frac{Ia}{Ib}\right)^2$$

To ease calculation this formula is commonly written as:

$$dB = 20 \times log_{10} \frac{Va}{Vb}$$

Don Davis' statement "the actual fact is that a two to one change in voltage is 6 dB" is confusing. The real fact is that the decibel is a power ratio and a two to one change in power is 3 dB.

The dBm came about from the common use of 600Ω lines, 0 dBm being defined as a power of 1 mW in a load of 600Ω which happens to correspond to 0.7746 V RMS.

Strictly the dBm should only be used with reference to 600Ω loads and it has been proposed that dB.7 V should be used to refer voltage to 0.7746 V across other loads or dBV to refer to 1 V (with dBu sometimes being used, but I've never seen this defined). These units require the load to be constant and in such circumstances strictly remain units of power.

Having got this off my chest I will return to Don Davis' example No 1 which really inspired this letter. He criticises an 'expert' who stated that 0 VU = +4 dBm and goes on to quote "Zero volume units (ie 0 VU) is indicated when the instrument is connected across a 600Ω resistor in which there is dissipated a power of 1 mW at 1 kHz (ie 0 VU = 0 dBm)."

This is highly confusing as anyone who has tried applying 0.7746 V to a standard volume indicator will find that it indicates −4 VU. It would seem appropriate to clarify this confusion and part of the secret lies in American Standard C16.5-1954 Volume Measurements of Electrical Speech and Program Waves *which states in the last paragraph "For example, the meter sensitivity might be such that 0 VU on the meter scale corresponds to a volume of +4 VU."*

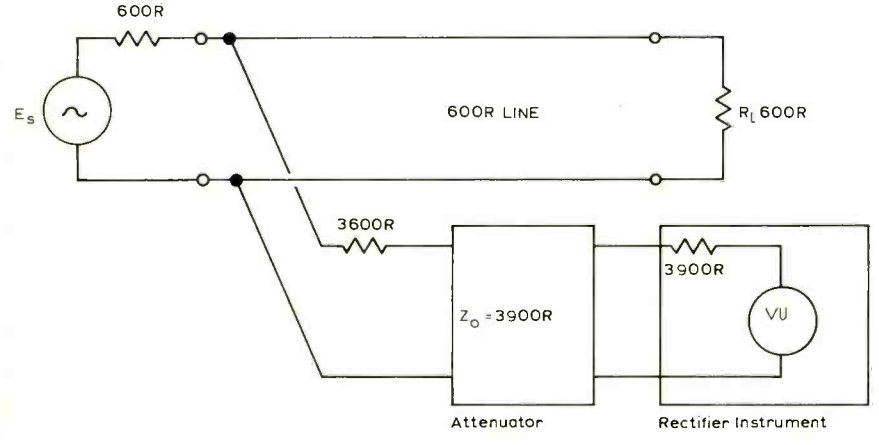
The original VU (correctly a Standard Volume Indicator) was described in the Bell Systems Technical Journal of January 1940. This instrument was designed to indicate the volume of speech or programme waves in terminated 600Ω lines. Being an instrument with defined ballistics and using a full wave rectifier both loading of the lines and the introduction of harmonic distortion due to the rectifier being placed across the line were design criteria. The result was an instrument with an impedance of 3900Ω when measured at the 100 mark (0 VU) on the scale using a sinewave. Intended for use with a 3600Ω series resistor the impedance becomes 7500Ω which is the recommended 12.5 × the line impedance.

Considering that a terminated 600Ω line has an impedance of 600/2 = 300Ω the loading presented by the meter affects level by less than 0.4 dB. As the scale of the instrument is cramped at low indications the instrument was intended to be used with an attenuator calibrated in VU with the correct arrangement being shown in Fig 1.

Nowadays VU meters in audio equipment are commonly fed by a variable gain buffer amplifier which eliminates both loading and distortion problems, but unfortunately, many instruments bearing the letters VU bear no relation to the genuine ballistics and rectifier requirements.

Yours faithfully, Hugh Ford, 24A Trinity Road, Richmond TW9 2LD.

Fig 1



Roosevelt tapes

Dear Sir, In the May Business column, Barry Fox seems to have been caught up in the overly dramatic promotion and hype American Heritage *magazine gave to its 'disclosure' of the 1940 FDR White House recordings. The recordings were not all that secret, and Robert Butow was not the first person outside the Hyde Park FDR Library staff to know of the existence of these recordings. A full explanation and listing was published in 1973 in Volume V, Number 2/3 of the* Association for Recorded Sound Collections Journal. *When the Watergate tapes were disclosed there was public speculation about the recordings of earlier presidents and I am so sure that I heard news reports about the FDR recordings that I never felt it necessary to be the one to make them public. I, and other archivists of my acquaintance, all assumed the FDR recordings were common knowledge. Butow himself admits that the recordings were long available to any researcher who had cared to listen to them.*

There is one other problem with Barry Fox's account. He states that the machine that was originally used was an optical sound recorder manufactured by RCA Photophone engineers in Camden, New Jersey. This is not borne out by the American Heritage *article which never asserts that the machine used the optical process nor was made by the Photophone branch of RCA. All they said was that it was a film recorder. That is not synonymous with optical film. Illness prevented me from fully researching this when I was recently in Washington, but from evidence thus far available it appears that the machine probably embossed a groove on a continuous loop of film. This concept was first seen in 1934 in the* Filmon *developed in Japan by Shozo Konishi, Koichi Tsubota and Ginjiro Sato, and may have been similar to the* Cellaphilm *said to have been used in California by Freeman Lang in 1935. By 1940 an American inventor, Jay C Fonda, was developing a machine which was actually produced in 1944. And while I was visiting the National Archives in May I had a chance to inspect a working model of a continuous loop grooved film machine widely used by the government and armed forces in World War II. It is the Amertype Recordgraph Corporation's* ARC Commando Model A *which was featured on the cover of the October 1944 issue of* Radio-Craft. *The concept of a continuous loop film with numerous parallel embossed grooves fits all of the descriptions of the quality of the FDR recordings, while an optical system fits none.*

Yours faithfully, Dr Michael Biel, Route 5 Box 174, Quail Hollow Road, Morehead, Kentucky 40351, USA.

Barry Fox comments: I am very grateful to Dr Biel for his valuable comments, as will be any reader with an interest in audio history. I would only point out that I did not see the *American Heritage* article until long after I had written my piece, which was researched by telephoning numerous people in the USA, including Robert Butow and the FDR Library. ■

Expert's errors

We have had a great deal of correspondence as a result of Don Davis' *Expert's errors* and will be publishing further comments in forthcoming issues.

F 462

RANGE OF MASTER TAPE RECORDERS

A range of tape recorders designed for demanding users (typical wow and flutter .03% CCIR at 15 i.p.s.), who can upgrade their equipment as their needs evolve through a wide range of accessories and options such as: remote control panel, varispeed unit with ± 13 semi-tone range, tape marker, scissors, NAB hub adaptors, monitoring loudspeaker...

When ordering your recorder, you may select: 2 of the 4 capstan speeds, the tape deck height, your usual equalization standard (CCIR/NAB), the head assembly corresponding to your type of operation (mono / stereo / two-track, with full or separate overlapping track erasure), etc.

...And you can change your mind later by upgrading your configuration at your convenience for a minimum cost.

F 462: an outstanding and flexible range of master recorders.

Give us a call. We'll let you know how and where you can try this product (Why not in your own studio?).

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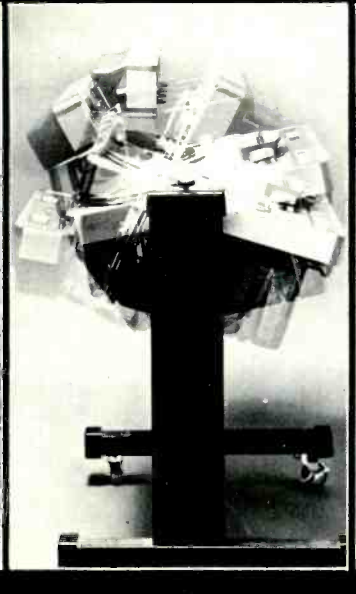
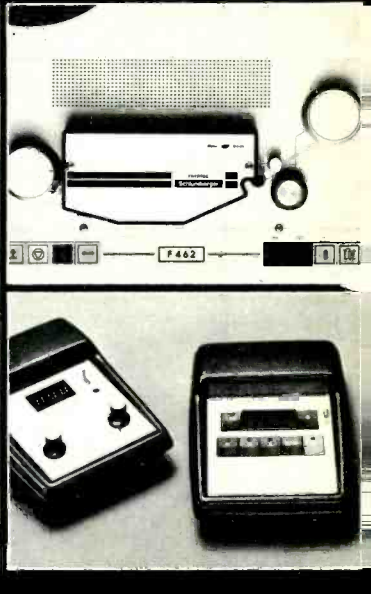
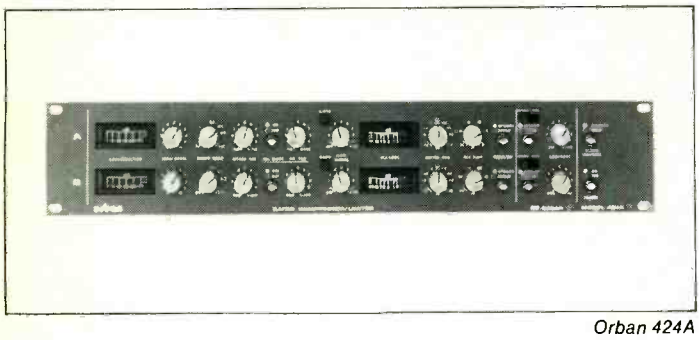




new products



Orban 424A

Orban 424A available

The new Orban gated compressor/limiter/de-esser, previewed at the Montreux AES in prototype form is now available for delivery. Two units are available, the *422A* (mono) and the *424A* (stereo). Described by Orban as 'the studio *Optimod*', the units offer several unusual features which ensure unobtrusive operation even at very high compression ratios. Both linear and exponential release slopes are available; a noise gate is built in; and 'idling gain' adjustments ensure that the compression level is maintained when there is no signal, or when the unit is gated off, thus avoiding noise surges and other unwanted effects. A dual-intensity de-essing circuit is also fitted.

Orban Associates Inc, 645 Bryant Street, San Francisco, California 94107. Phone: (415) 957-1067.

UK: Scenic Sounds Equipment Ltd, 97-99 Dean Street, London W1V 5RA. Phone: 01-734 2812/3/4/5.

New Citec faders

Citec is a new name in faders, but they may be better known under their previous title, Plessey Resistors. The company's new *100 SF* series is the result of an 18-month development programme originally instigated to produce custom fader units for Neve Electronics. Particular attention has been paid in the design to consistent operating force, and in addition, the wiper is claimed to provide superior mechanical and electrical performance.

The new faders will be widely available from early 1983, samples being available at present for evaluation. The range is to be expanded in the coming year.

Citec Ltd, Cheney Manor, Swindon, Wiltshire, UK. Phone: 0793 487301. Telex: 897971 CITEC.

C-Tape LP2

C-Tape Developments have produced a version of their C-Ducer contact microphone system specially tailored for use with Latin American drums, e.g. congas, bongos, timbales etc. The *LP2* comprises two tape contact microphones and a small preamplifier unit with a balance control to adjust the level between the two mikes. The tapes follow the standard C-Ducer system and fix to the drums with the use of double-sided tape and are simple to attach. The output from the preamp mixes the two contact mikes down to a single unbalanced jack output.

C-Tape Developments Ltd, Transducer Laboratories, 73 High Street, Aldershot, Hampshire GU11 1BY. Phone: 0252 319171.

USA: Griffith Jones & Associates, 3074 Greenwood Trail SE, Maryetta, Georgia 30067. Phone: (404) 953 0697.

USA: Hammond Industries Inc, 8000 Madison Pike, Madison, Alabama 35758.

Denon digital editing

Nippon Columbia Co Ltd, or Denon as they are more widely known, distribute a newsletter entitled *Denon News* which details new products etc usually aimed at the hi-fi retailer. Issue No. 39—July 82 however gives very brief details of a digital editing system with the full title of *Random Access PCM Editing System Model DN-036ED*. Denon have apparently been active in PCM development for the last decade and have opted for a disk based system with the *DN-036ED* in the form of two disk drives with a music programme storage capacity of 23 minutes of 4-channel and about 46 minutes of 2-channel. Access to required sections is almost instantaneous and the floor-standing editing console can achieve swift editing with full compensation for reproduction levels and fade rates. Denon close by saying that the *DN-036ED* will soon make its appearance in studios around the world.

Nippon Columbia Co Ltd, No 14-14 Akasaka 4 Chome, Minatoku, Tokyo 107, Japan.

New HH amp

HH Electronic has announced the availability of a new amp in its range of MOSFET power amplifiers. The *M-900* features 400 W RMS per channel (800 W in bridged mode), and has a very high capacity power supply unit. The amp is 19 in rack mounting and optional balanced input transformers are available. Locks are fitted to the gain setting front panel pots. *XLR*-type connectors are used on ins and outs. Full specifications are not yet available.

HH Electronics, Viking Way, Bar Hill, Cambridge CB3 8EL. UK. Phone: 0954 81140.

USA: HH Electronic Inc, 2500 E Fender Avenue, Suite 1, Fullerton, California 92631. Phone: (714) 680 4293.

JBL 4401 compact monitor

JBL have introduced a compact studio monitor to their Professional Series to be known as the *Model 4401*. It is a 2-way system with a newly developed 6½ in LF driver which has been designed for optimum small enclosure performance using JBL's Symmetrical Field Geometry magnetic design giving reduced second harmonic distortion. It also features a rigid lightweight laminated cone with a coating to give proper damping characteristics, in addition to an unusually long voice coil to improve excursion linearity.

JBL monitor

The HF is handled by a JBL one-inch hard dome radiator which maintains a linear response over its operating range.

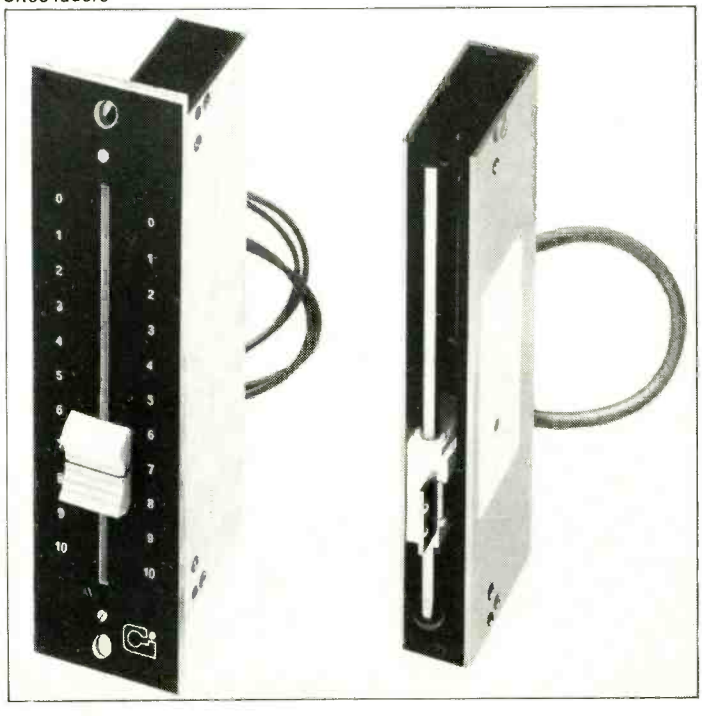
Intended applications for the *4401* include main monitors where space is restricted or as a secondary console mounting system. It may be ordered with optional steel cradle mounts to simplify wall or console mounting. Finish is American black walnut veneer with dark blue grille.

Specification. Frequency response: 70 Hz to 18 kHz ±3 dB; power handling: 60 W continuous programme; sensitivity: 88 dB SPL, for 1 W at 1 m; nominal impedance: 8Ω; crossover frequency: 2.5 kHz. Dimensions: 14¾ in × 9⅜ in × 7⅜ in with a weight of 35 lb.

James B. Lansing Sound Inc, 8500 Balboa Blvd, Northridge, California 91329, USA. Phone: (213) 893-8411. Telex: 674993.

UK: Harman (Audio) UK Ltd, Mill Street, Slough SL2 5DD. Phone: 0753 76911. Telex: 849069. ■

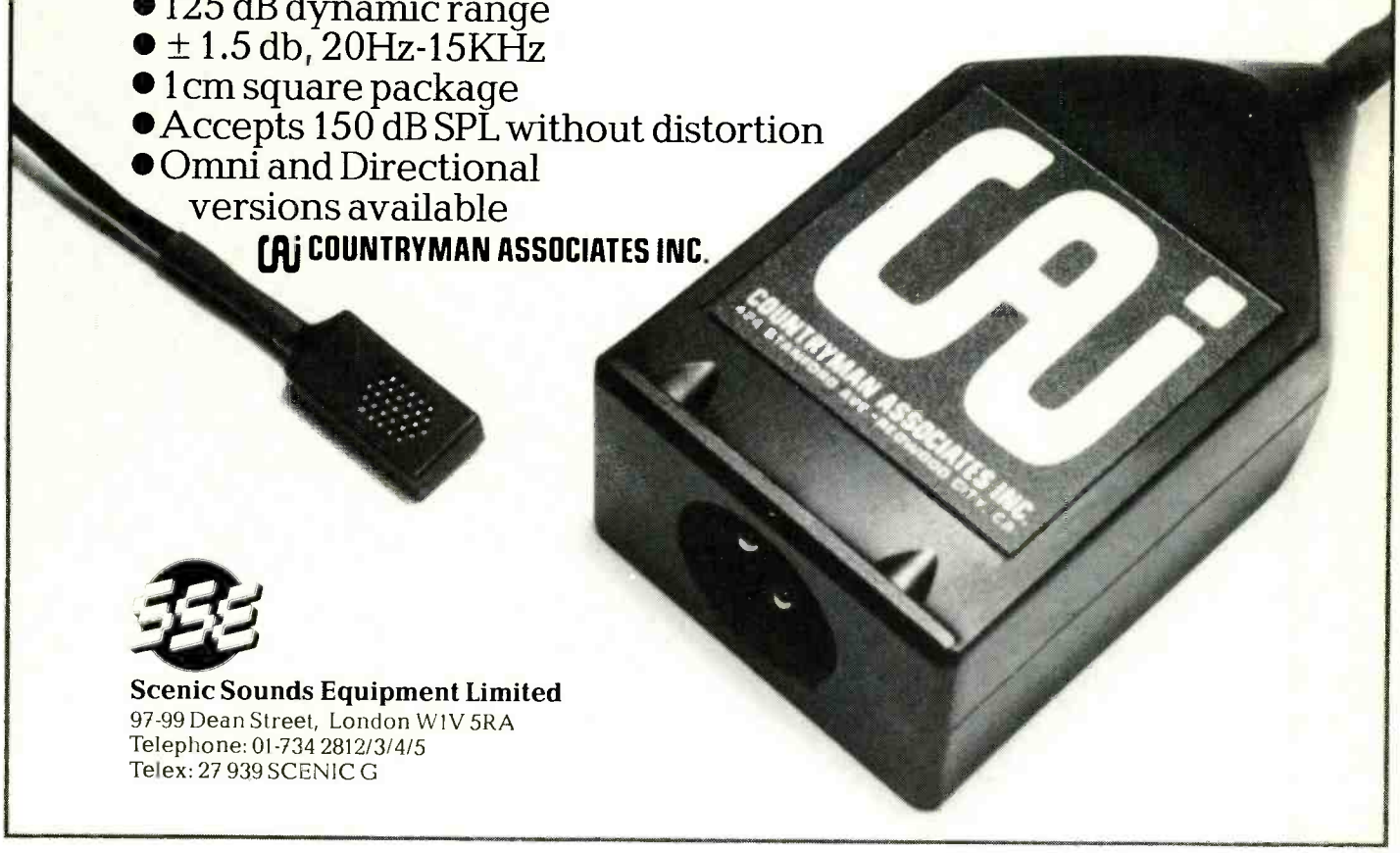
Citec faders



Sub-Miniature Precision Pressure Microphones

- 125 dB dynamic range
- ± 1.5 db, 20Hz-15KHz
- 1cm square package
- Accepts 150 dB SPL without distortion
- Omni and Directional versions available

COUNTRYMAN ASSOCIATES INC.



Scenic Sounds Equipment Limited
97-99 Dean Street, London W1V 5RA
Telephone: 01-734 2812/3/4/5
Telex: 27 939 SCENIC G

Looking for a Distortion Measurement System?

The Amber model 3501 is quite simply the highest performance, most featured, yet lowest cost audio distortion and noise measurement system available.

It offers state-of-the-art performance with THD measurements to below 0.0008% (−102dB), maximum output level to +30dBm and noise measurements to below −120dBm.

It has features like automatic operation, optional balanced input/output and powerful IMD measurement capability. It includes comprehensive noise weighting with four user changeable filters. Unique features like manual spectrum analysis and selectable bandwidth signal-to-noise measurements.

The 3501 is fast, easy to use and its light weight and small size make it very portable. It can even be battery powered.

amber Amber Electro Design Inc. 4810 Jean Talon West. Montreal. Canada H4P 2N5. Telephone (514) 735 4105

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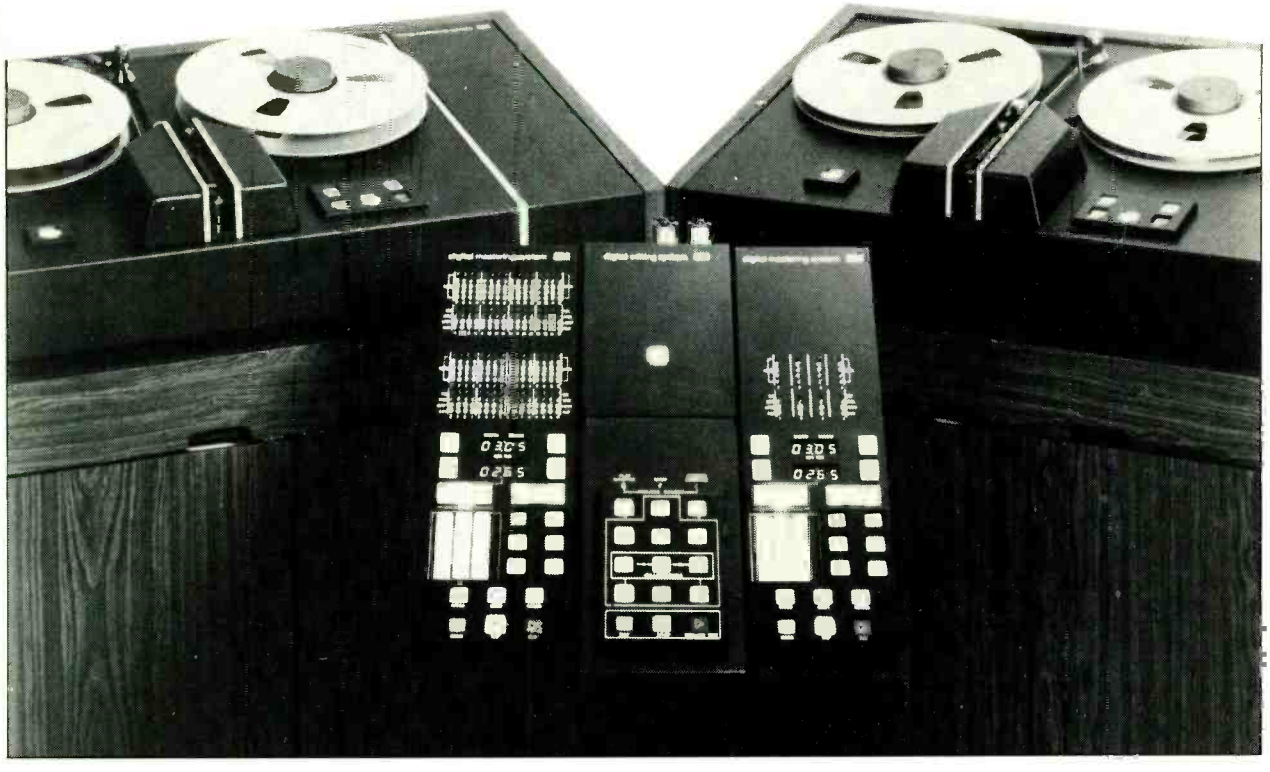
Digital recording the 3M system

Clark Duffey (3M)



Manufacturer's Specifications

Frequency response	20 Hz to 18 kHz — ±0.3 dB; 10 Hz to 20 kHz — +0.5, —3.0 dB
Signal to noise	> 90 dB
Harmonic distortion	< 0.03% 100 Hz to 20 kHz at maximum input/output level (+28 dBm) < 0.03% 20 Hz to 20 kHz with input/output level of +15 dBm
Intermodulation distortion	< 0.03% for any two frequencies 100 Hz to 20 kHz at maximum input/output level (+28 dBm)
Erasure	No measurable residual signal
Print through	Not measurable
Input	Maximum +28 dBm Normal (12 dB headroom) range —6 to +15 dBm adjustable 20 kΩ impedance for balanced or unbalanced line input
Output	May be prewired for —6 to +16 dBm for normal (12 dB headroom) output with 600Ω load. Floating transformer output
Wow and flutter	Not measurable
Timing accuracy	Controlled by crystal oscillator
Tape speed	45 in/s with ±10% vernier
Play time	> 30 min with 12½ in reels wound with 7,200 ft of recommended tape (45 min with 14 in max reel size)
Rewind/fast forward time	< 2½ min with 12½ in reels
Dimensions	37½ × 22½ × 45 in
Number of channels	32 on 1 in tape; 16 on 1 in tape; 4 on ½ in tape
Recommended tape	Scotch Brand 265 Digital Audio Mastering Tape

MULTITRACK digital mastering was first achieved in late 1978 and the first 3M 32-track and 4-track recorders were installed in the US studios in February 1979. Editing other than by physically cutting the tape, however, was not available and the first recordings made were, in effect, 'direct-to-digital'—complete performances with no editing—although in some cases, they were transferred to analogue for editing.

However, 3M engineers in mid-1979 made available a working prototype of an electronic editor that was put to use immediately. It was the forerunner of the present simplified but sophisticated electronic editing system, which allows precision and creativity beyond that achievable with the traditional cut-and-splice editing. Electronic editing also provides these benefits without risk to the integrity of the initial recording or the new master. Physical handling and splicing—which could destroy the extremely high density information (over 890,000 bits/in² or approximately 138,000 bits/cm²) directly or create debris that could later interfere with signal—is eliminated.

Editing capability expands the use of digital recording equipment in both classical and popular recordings. With the 3M digital editor, the equipment operator can audition and preview a potential edit then refine it by as little as a millisecond. The complete edit can also be auditioned and further refined prior to committing to the edit. Musical timing between two takes is typically accurate to one millisecond as well. Because editing is achieved within the digital domain, the signal does not experience the degradation inherent in analogue copying.

Because crossfade capability guarantees smooth transitions for inaudible splices, there is no need to try to match an aesthetically acceptable splicing point with a technically acceptable location as was necessary prior to cross-fade. The flexibility added by this capability therefore translates to a better and faster editing session.

Typical of the reactions to crossfade was that of Andrew Kasdin who used 3M's equipment in the production of several classical CBS Masterworks digital albums:

> "I have made daring splices where I would have been prepared for failure in analogue, and not one of them gave any problem at all. Once I became familiar with this total editing capability, I was making splices at least as fast as with analogue. Crossfade overcomes all my earlier reservations about editing multitrack digital tapes."

3M basic technology

While other digital recording systems have been or are being developed resulting in and from the growing interest in distortion-free, noiseless, high-dynamic-range recording, 3M's system was the first commercially available system to achieve 30 channels of audio with a one-channel-per-track configuration. The recorders were designed to be readily accepted by recording audio personnel, because they looked like and were operated in a similar manner to the analogue systems which had been in use for many years. Essentially, recordists were not required to learn new procedures; the 3M digital recorders could be dropped into an otherwise analogue system. The digital masters thus produced have been used to create higher quality hybrid analogue disc recordings and it is likely that some masters will be used again in the future to produce digital discs or tapes for public consumption.

Typical systems comprise one or more 30-channel recorders, a 4-channel mixdown recorder and an editing system. More than 60 recorders and 15 editing systems are currently in use by American, British, German, French, Swedish and Japanese recording studios.

The one-track-per-channel design is the result of intensive research and development over the years, by 3M and the BBC. Working together, they were ultimately able to design an error correction scheme even more effective than an earlier four-tracks-per-channel format, combining high-quality, relatively blemish-free recording tape (also from 3M) and an error-detection-and-correction scheme that ultimately required only 50% additional recording redundancy on the same track. It is also more cost-efficient as expenditure on recording heads and electronics is reduced.

Versatile recorder controls permit conventional studio techniques, such as punch-ins, overdubbing and ping-ponging. Each track may be controlled separately to achieve the recordist's final product. Because virtually no signal degradation occurs with digital dubbing, each copy is like a new, full-fidelity original so that, theoretically, the 20th-generation dub sounds like the original input. A digital preview unit is also available to optimise pitch-control for disc-cutting.

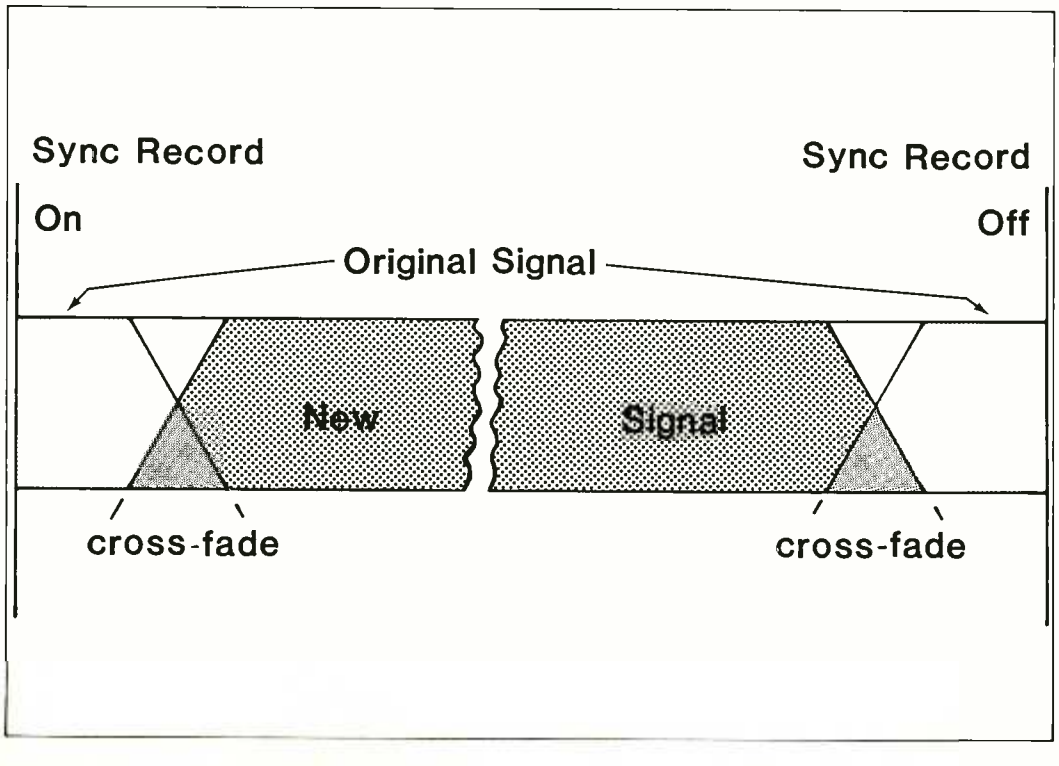
Error-correction scheme

Digital recording uses technologies and procedures much more akin to instrumentation and data-processing than to the traditional analogue technologies heretofore used in audio and video recording. All that is recorded is a series of binary numbers (composed of groups of 0's and 1's). Since binary numbers take more space than the equivalent analogue signal, it is necessary to pack data densely on digital recording tape so every square inch (6.5 cm²) of digital recording tape contains approximately 890,000 separate magnetic bits of information. At a speed of 45 in/s (50 kHz sampling frequency), the number of separate magnetic write-ins and read-outs per second could be as high as 40 million. Even though the tape used in digital recording has a high degree of quality, it can never be completely blemish-free. Further, airborne specks of dust or other contaminants can interfere with magnetic reading and writing so at such densities it is obvious that a very small interfering object would cause a drop-out of vital data. Thus, an error-correcting scheme is necessary.

In ordinary analogue recording, the density is orders of magnitude different. At the extreme of the audio range—20 kHz—the number of magnetic excursions necessary per inch would be only in the order of 40,000 per second spread out over the 15 in or 30 in of tape typically used per second. Even if an occasional drop-out occurs in analogue recording, it is hardly noticeable because the drop and recovery in amplitude is smooth and tends to mask the event. In digital recording, on the other hand, a drop-out does not cause an omission; instead it causes a wrong number to be generated which can create momentary undesirable noise or distortion. Any noise is much more obvious because of the lack of the other noises in digital tapes. Thus, the error-correcting scheme becomes vital.

Having the benefit of years of experience in manufacturing tape for audio/video, instrumentation and data-processing applications, 3M development experts determined the statistical probability of error-causing drop-outs per unit of tape length. Armed with that knowledge and using a computer-like memory to store (momentarily) the musical 'data' read from tape, they were able to devise an error-correcting scheme that allows one audio channel per

32 ▶



3M system

track as contrasted to other systems which must use two or more tape tracks per channel.

In the very rare instances where an error is so great that complete reconstruction is not possible, the system will (by virtue of the data word/parity/frame interrelationships) still be able to reconstruct every other data word or sample. During this period, the good word is repeated for the missing word. This form of reconstruction is not audible through short periods of time.

The recording system basically works by sampling the analogue waveform produced by the microphone/preamplifier system 50,000 times per second. Each 1/50,000 s point in the waveform is then assigned a numeric amplitude value using a 16-bit binary word. (A 'data processing-to-digital recording concept conversion kit' accompanies this article for the benefit of those not familiar with data processing, or who may need some memory-refreshing.)

During recording, two such binary words, each representing one sample of the analogue musical waveform, are moved into an electronic memory and figuratively laid side by side for comparison. The words compared are from adjacent input samples but are spatially separated on tape. Corresponding bits of the two words are compared, and from those comparisons a parity word is generated and recorded at still another location on the tape deck.

This separation of positions on tape is essential to error correction; a blemish or dust particle may interfere with more than one of the associated frames which allows complete reconstruction of the interrupted data. If there is an indication of error, reconstruction circuitry compares the corresponding error-free words of the two 'good' frames, determines what the correct words should be, and then inserts them in the 'bad' frame. When that has been accomplished, the data words are routed in correct order to the D/A converter which recreates the analogue waveform.

Temporary electronic storage is also the key to absence of wow and flutter. A time-base corrector with crystal-controlled clock gates the digital words out of a playback electronic memory providing precise, even timing. Any wow or flutter that might exist in playback is eliminated because the tape speed has no bearing on the speed at which the time-base corrector clock gates the digital words out to the loudspeaker. Minor variations in tape speed are servo-controlled by the electronics of the read-out circuitry so there are neither significant lags of data input or data pile-ups caused by tape speed variations.

Digital audio standards

There has been a lot of talk about digital audio standards. In fact, discussions for an AES standard began as early as December 1977 after 3M and Soundstream showed their digital audio recorders at the New York AES. Today there are more than half a dozen technical and user organisations tackling this complex subject including AES, APRS, EBU (European Broadcast Union), IEC (International Electrotechnical Commission), SMPTE and SPARS. These organisations are discovering what other standards committees have learned: standards are not easily set—and for good reasons. And, standards always involve compromises or trade-offs.

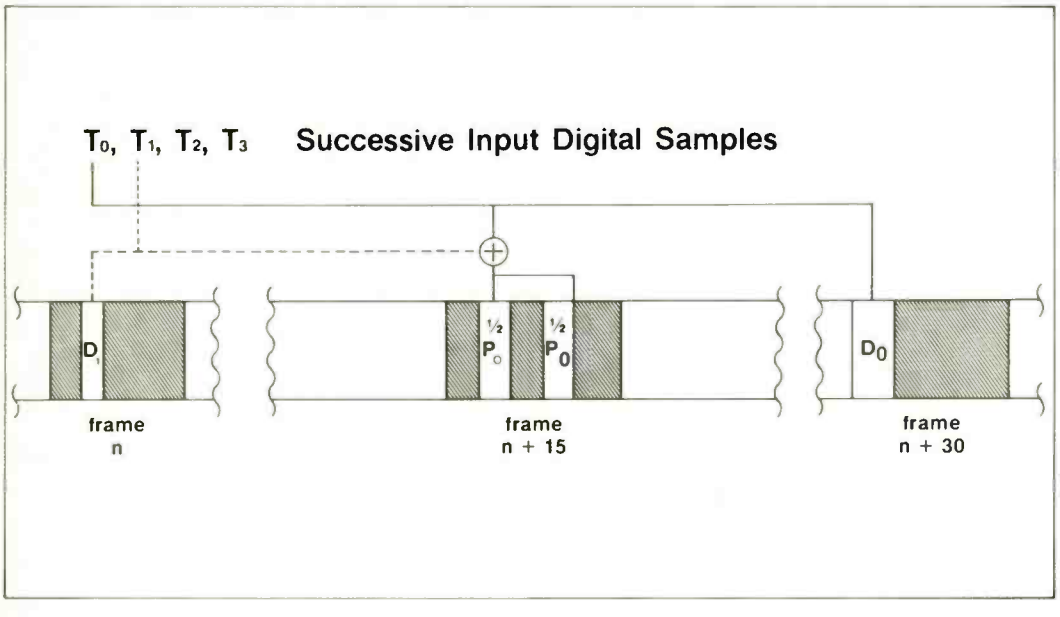
From the manufacturers' standpoint, the equipment finally marketed reflects considerable research into the optimum design to meet the prospective customer's expressed needs. Often extensive field evaluation precedes freezing a final design. Design differences obviously occur because manufacturers have different experiences, may receive different input from the industry and discover methods of achieving the same result.

User reactions to any new technology generally range from feelings that it falls short of perfection to believing that it represents overkill. As this relates to digital audio, there are those who would like to see the sampling frequency extended to 100 kHz. Others feel that 32 kHz or 44.1 kHz is sufficient. The higher extreme is prohibited today by technology and practical cost considerations, while the lowest sampling frequencies tend to preclude offering top quality sound during variable speed operation.

3M, Soundstream and others chose 50 kHz early because it appeared to provide full sound resolution and the best overall compromise to achieve compatibility of digital audio with established TV, film and other digital audio transmission systems. Some manufacturers preferred other frequencies for various reasons, and there are many lesser important but necessary parameters to be chosen if complete interchangeability is to be achieved.

What may be most needed is a single interface so that one brand of recorder can 'talk to' a competitive brand. This will prove increasingly important as more equipment emerges from the initial design phase. On this belief, in August 1980 3M made available formerly proprietary information to potential manufacturers of digital audio equipment. This included the technical interfacing necessary so that these manufacturers, while still finalising their equipment, could provide signal compatibility. Even achievement of signal interface compatibility would be far from automatic, but perhaps this is a realistic step towards standards.

In Europe an interim answer to the standards problem has been found. Now that 48 kHz has been accepted in the UK, Europe and the USA, all future 3M machines will be fitted with a switchable bit sampling frequency facility as standard. This will also include bit sampling at 44.1 kHz which is being used for audio disc masters. ■



Data processing-to-digital recording concept conversion kit

Digital technology, including error correction, is the backbone of the data-processing industry. To readers with a digital background, 'parity', 'cyclic redundancy check' and some other concepts may be no problem. But to help those without such a background, this obviously over-simplified 'kit' is offered.

Binary Numbers: A binary number system uses the lowest range of numbers: two. Only 0 and 1 are used to make up the numbers in even the most complex digital system. This is because the numbers are easily represented by the binary state of electronic circuits—on or off, positive or negative, magnetically 'north' or 'south', etc. Larger numbers are simply made up (as are our 'normal' decimal numbers) by putting the binary digits ('bits') in orders of magnitude after all possibilities in one order have been used, for instance:

Decimal	Binary
0	0
1	1
2	10
3	11
4	100
5	101

A value of any size can be represented by binary digits, if enough are put into place. Each successive bit has a magnitude twice that of the previous one (in decimal notation, each order of magnitude is 10 times the adjacent one). Thus, a 16-bit binary word as used in the 3M digital audio mastering system can hold any value up to 65,536.

Parity: This is a technique to ensure correct storage or transmission of bits in electronic systems. It is something like the old game of 'odd man out'. The corresponding bits in a pair of binary words are compared, and a third—parity—word is created electronically. Its corresponding bit is determined by the bits in the corresponding positions of the two original words. In the 3M system, if both 'data' bits are the same, the parity bit as generated is a '0'; if they are dissimilar, the parity bit generated is a '1'. Then, later on—at playback or at the other end of a transmission—the corresponding bits are compared. Following the ground rules by which they were originally compared and with the parity bit as generated, it is possible to re-create and replace a bit missing from a 'damaged' word.

Data placement on tape: Since digital notations are simply groupings of numbers (and not actual musical waveforms) related ones can be on non-adjacent locations on a tape. As long as any digital 'word'—a specifically defined digital grouping—can be labelled in some manner, it can (within limits) be at any place on a tape and yet routed to the proper electronic 'stall' for parity comparison or other treatment. Conversely, words can be routed from temporary storage circuitry to specified positions on tape.

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Audio production for EPCOT

Bob Anthony

WHEN Walt Disney first conceived Walt Disney World near Orlando, Florida, in the United States, he originally purchased approximately 27,000 acres of land, of which only a few thousand have thus far been developed. But scheduled to open soon is an $800 million addition to the 'entertainment world' called the EPCOT Center, an acronym for the Experimental Prototype Community of Tomorrow. In an effort to fulfil the commitment that this name implies, Walt Disney Studios in Burbank, California, is planning for the future with state-of-the-art production techniques today.

While building Disneyland, Walt Disney formed WED (derived from the founder's name) Enterprises, a diversified group of designers, engineers, craftsmen and mechanics responsible for conceiving and constructing the outdoor recreation parks. Their major projects to date include not only Disneyland, but Walt Disney World, EPCOT Center and a Tokyo Disneyland, which will be ready to welcome the public during the first quarter of 1983. The studio supports many of these projects by providing the skills, equipment and facilities necessary to complete any film production required by the parks.

"When the EPCOT Center project was committed, one of the goals was to achieve the very best sound quality possible for a theme park," remembers Nelson Meacham, assistant to the studio sound director. "That meant doing the film tracks digitally wherever it was feasible. Two of WED's engineers, Dave Spencer and Steve Boze, designed all of the necessary interfacing equipment to allow us to do film sound in sync with digital. We can shuttle the video and audio tapes at least as fast, if not faster, than we can shuttle our film dummies."

The process has since matured to the point where the studio is doing 9-panel, 360° shows, which entail as many as four digital tape recorders and nine video tape machines running in sync in the dubbing theatre. The 9-camera show currently in production is intended for viewing in the China pavilion, and is appropriately entitled *The Wonders of China*. The footage was all shot on location in that Asian country by a process called *Circle-Vision 360*.

Because the project was started some time ago, the only digital multitrack audio recorders available to the studio were manufactured by 3M, offering four and 32-track capability. Almost all of the work is done on three 32-track machines using 1 in tape, and a 4-track that needs ½ in tape. The company also owns a Sony *PCM 1600* digital processor, which principally holds the sound effects library. The Sony utilises a ¾ in *U-matic* cassette for storage and affords two recording channels. This machine is usually circulated around the lot for various recording tasks.

Preparation

During the study phase, the company was debating whether or not to dub to film. Studio personnel had two alternatives open to them in terms of methods for dubbing multiple panel shows. One course of action was to convert Stage Two, a normal sound stage at the studio, into a dubbing stage. This, in itself, meant solving two problems. The film projectors, many of which were 70 mm, would have to be quietened down in some way, probably by building separate isolation booths for the cameras. The planners decided that to make the stage sufficiently soundproof for dubbing would be time and cost prohibitive.

The second problem stemmed from the number of screens available. There were only five in the *Circle-Vision 360* test setup, four short of the required nine. Video projection was considered to supply the additional four rear screens, but at that point, the answer became obvious—make the entire system video projection. "We had already been doing research here at WED on synchronising video tape machines to a timecode source, and we felt rather confident that we could do that," says Dave Spencer, supervisor of audio/video technical services at WED. "And of course, video projection is significantly less expensive than film for this type of work, when you consider the price of film and interlock systems today."

The primary disadvantage is that the pictures are a smaller scale. According to Spender, the Stage 2 screen formats are about 80% of the full size, whereas the video screens on Stage 1 measure 12 ft. According to the audio engineers on the dubbing stage, the video dupes are not the greatest, but certainly adequate for this application. However, if any cues are needed on the pictures, they have to be added before the tape goes to video duping. Any attempt to add cues on the sound stage would entail the use of a fairly complicated, and expensive, character generator.

The rest of the video system is comprised of nine Sony cinemascope video projectors (model *FPH-670W*) arranged in a circular array, yielding a 360° field of view. Front projection is employed, which required a unique support ring that the studio machine shop built to specifications. The projectors are arranged round the top of, and supported by, the ring, which is suspended above the mixing consoles. Feeding them are nine VTRs all of which are synchronised to a master timecode source.

Thirteen standard BTX *Shadow* units were used as synchronisers, but they demanded a custom interface to marry the *Shadows*, the nine video tape recorders and the four digital audio recorders. All the pieces are controlled via a Hewlett Packard *9826* computer with a 16-bit processor. Lem Davis of the studio's scientific systems department wrote the software for the system in BASIC, and worked out all of the protocol. Fortunately, in a distributed system like this, the computer doesn't have to do any real-time manipulating. The *HP-9826* simply controls the BTX machines, which in turn have to carry out the commands in real time.

"We had already done it here on a smaller scale with an *HP-85*, using the RS-232 port to the *Shadows*. On the full system, we basically run two RS-232 feeds out of the Hewlett Packard, and multiplex them to the 13 *Shadows*. The computer has the ability to talk to any combination of machines at any one time. Theoretically, the system is limitless in terms of combinations that can be accessed."

However, Spencer has found that ". . . because we're running RS-232 multiplexer out, the rate of 1200 baud is the limiting factor. We're currently looking at converting the system so we can talk *HP-IB* (abbreviation for Hewlett Packard's implementation of IEEE standard 488-1975 for digital interfacing of programmable instrumentation) out of the Hewlett Packard, and do a conversion to RS-232 at the *Shadows*."

All the equipment chases a SMPTE timecode, which can be derived from a timecode generator, another digital machine, or a 35 mm mag running in a theatre a couple of blocks away. For this project, the timecode is the only signal stored on the video tape's audio tracks. In addition, one track of every digital

36 ▶

Current equipment

1	Eventide *H949 Harmonizer*
1	Sound Effects highpass, lowpass filter
1	Aphex *Aural Exciter*
11	Aphex *EQ F-2* equalisers
7	*CX-1* compressor/expanders
4	UREI *LA-4*
1	Marshall *5402 Time Modulator*
2	Twin Lexicon *224X* digital reverb
3	Nagras
2	UREI *539* ⅓-octave graphic EQ
4	Dolby *Cat 43* noise reduction units
2	UREI *565T* notch filter sets
4	dbx *De-essers*
4½	dbx *Overeasy* compressors
1	Publison *DA-10 89B2*
1	Stereo delay line
1	Lexicon *Super Prime Time*
2	UREI *1178* graphic EQ
5	Yamaha *P-2200* monitor amps
10	Monitors—new JBL Bi-radial Horns (2-way system) (Monitors are behind the screen up on racks) White passive ⅓-octave graphics on all the monitors
3	3M model *M81* digital recorders (32 tracks)
1	3M model *M81* digital recorder (4 tracks)
1	Harrison model *TV-3* mixing console

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A mix control is provided, enabling the unit to be used in one input of a mixing console, or with musical instrument amplifiers. A regeneration control provides for the recirculation of processed signals, creating more and more notes, depending upon the selected interval. This results in multitudes of voices or instrumental chords. An entire new range of sound effects and musical textures, unattainable with any other type of signal processor, is suddenly at your fingertips.

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We have been able to maintain a high level of sonic integrity in this most versatile signal processor. The frequency response of the processed signal is beyond 10 kHz, with a dynamic range exceeding 80 dB.

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MXR Professional Products Group

audio machine is dedicated to SMPTE. That gives the engineers over 90 digital audio tracks on-line and in-sync at all times.

"The people who have been working on Stage 1, have made improvements and streamlined the system as the project has progressed," Spencer points out, "but I think the important point to consider is that the design time on this entire system was about a month, and fabrication was about a month and a half. That's very quick."

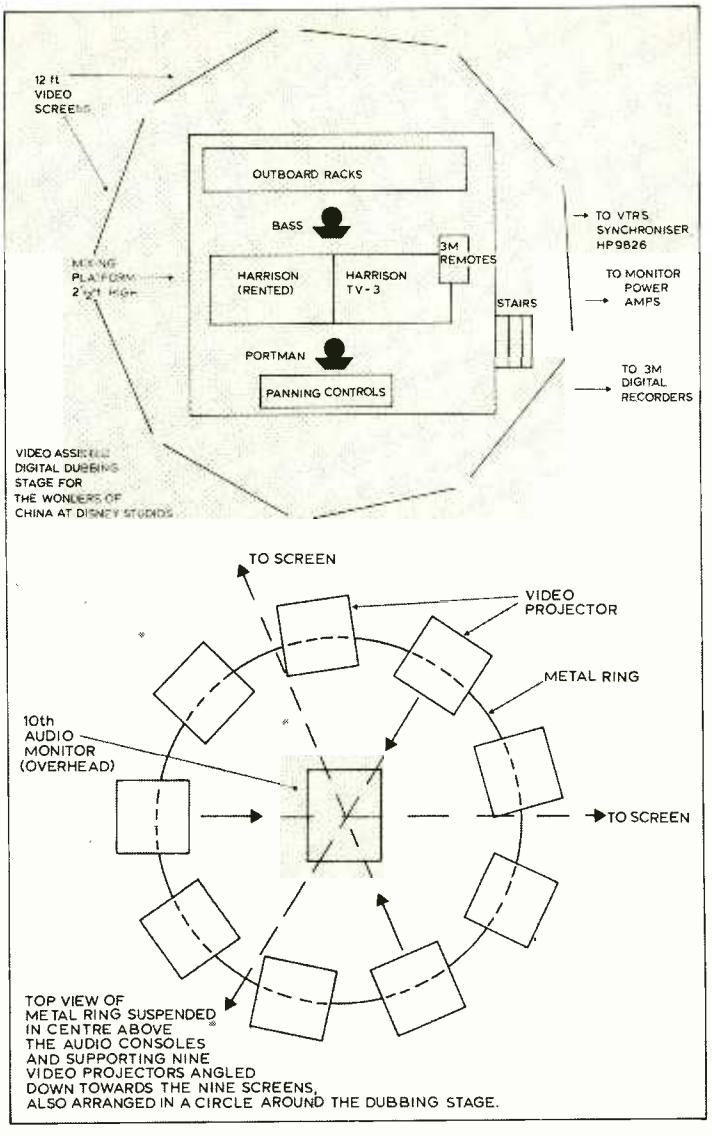
Production

On the dubbing stage, production engineers Andy Bass and Richard Portman are mixing down to ten discrete tracks—one for each of nine screens and one overhead. A typical 20 min show requires two reels of 200 or so tracks—the exact number per channel contingent upon how many sounds are required per panel. "We assign the sounds by the picture," says Andy Bass. "If we have horses galloping around in opposing 360° circles, we may have 50 tracks for that 15 or 20 seconds. A picture with just crickets in the background would mean one track spread across all the panels.

"Every sound has a track," adds Richard Portman. "If there are eight sounds per panel, that's eight times nine panels which means 72 tracks that need to be cut. Those, of course, would be reduced down to nine separate tracks, each one containing the submixes that make up the composite track for a given panel. We generally break the sounds up into backgrounds and foregrounds, and then mix it down."

On this particular show, *The Wonders of China*, there are two tracks of dialogue that need to be turned into 10 tracks of dialogue. Although the off-screen narrator is run through the overhead monitor, the on-screen narrator tends to walk around the theatre. Because the production crew was supplied with only one track of him, that one track has to be expanded to nine tracks, or one for each screen.

Portman has four stereo panpots that sweep through the entire 360°, and explains the situation this way: "Quite often, the editor on this show was not able to record a continuous track. Instead of someone walking through all the panels on one track, the Foley engineer cut one track per panel. We have to dub one panel at a time, which is quite arduous. We're experiencing difficulties in joining between panels when it's continuous motion. It's like doing nine movies at once. Then there's also the problem that some of the tracks are analogue and some are digital; some are library effects and some are new effects. In addition to just matching levels, we have to do a lot of EQ-ing and signal processing to keep everything clean and try to make it match. The original elements are all Dolby-ed, but once we get to digital, there's no more concern about noise reduction."



TOP VIEW OF METAL RING SUSPENDED IN CENTRE ABOVE THE AUDIO CONSOLES AND SUPPORTING NINE VIDEO PROJECTORS ANGLED DOWN TOWARDS THE NINE SCREENS, ALSO ARRANGED IN A CIRCLE AROUND THE DUBBING STAGE.

The sound effects were recorded at Goldwyn studios, because the Disney Foley stage was being rebuilt at the time. The tracks were cut on analogue with Dolby and then transferred to a 3M 32-track digital recorder. All the music, with the exception of the authentic Chinese instruments recorded on analogue tape in Asia, was mastered directly to digital by Shaun Murphy, the EPCOT supervising mixer on the lot. Murphy also handled the dialogue recording at Goldwyn using the 3M 4-track digital machine. The digital machines were even shipped as far away as London for specific music requirements.

The three 32-tracks are used all the time on the dubbing stage—two for playback and one for recording. The fourth machine, a 4-track, is the 'bail-out' machine used to slide sync, or move one track independently of the others on a given machine. The track is moved to the 4-track, where its sync is re-addresssed, and then transferred back to the 32-track machine in its new position. There's no generation loss, because both machines are digital.

"The piece we're working on now has 36 tracks of Foley," says Andy Bass. "Because we have only 28 tracks on a reel, we have to split the tracks up between two reels. And with four multitrack machines, we just can't get it all into one board."

Disney Studios owns a new Harrison model *TV-3* console with 32 channels, and rents a second, a five-year-old Harrison console that provides another 40 inputs. "There's 32 faders in the *TV-3* but actually 64 inputs," mentions Bass. "We use the side that's not going into the fader to drive the monitors. We bring the 'record' machine back into the multitrack return side of the board without running it through the faders, and hard-assign outputs of the recorders to the monitors. That way we're always listening through our master machine."

The board is assigned from left to right, from panel one to panel nine. "In this case there seem to be about four tracks per panel," says Portman. "As we go around from screen to screen, we just slide down the board. Generally, Andy does most of the mechanical mixing. I do the panning and any mixing for the rear panels by leaning over the meter side of the console and mixing upside-down. I have four stereo panpots—basically eight in and nine out—that gives us the versatility of simultaneously panning several elements in either direction at different speeds. But the pots are not as quiet as the total system; they're putting noise in against our floor."

"The biggest concern is that everything is so quiet," adds Bass. "Equipment and elements that would normally be acceptable, aren't any more. I think the overall S/N is about 80 or 90 dB."

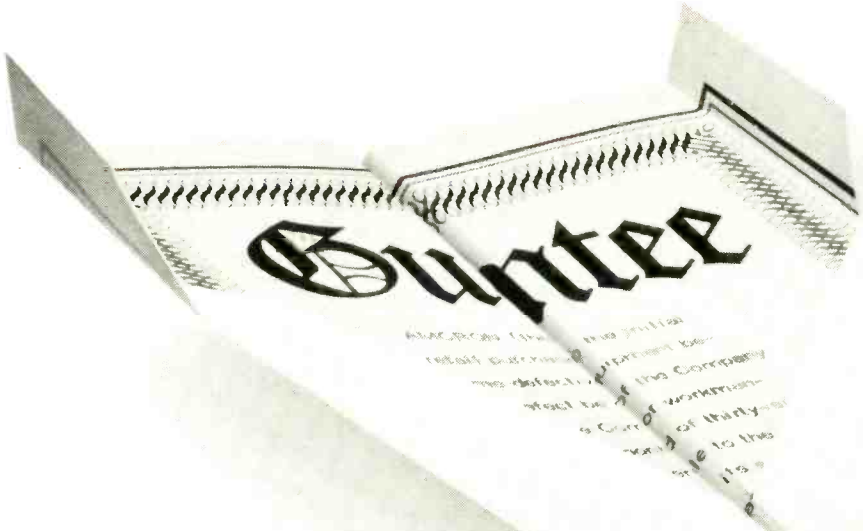
In addition to a master mix, the engineers record separate music, effects and narration tracks just in case the director should want to do a foreign language mix, or make some changes in ratio based on how the tracks play in a particular theatre. The inner elements can be independently called up to be replaced or rebalanced, as opposed to going all the way back to the original elements and starting the mix from scratch.

Andy Bass considers this particular show to be a lot like travelogue, and ". . . doesn't really require many special effects. But we'll use the pitch-changer for Doppler effect. On one occasion, a sound effect clashed with the pitch of the music, so we changed the pitch of the effect. We also do quite a bit of flanging and double and triple-tracking. But what we use most of all is the Marshall *Time Modulator* to give sounds depth."

Presentation

The China pavilion at the EPCOT Center in Florida will show this production on nine custom-built 35 mm projectors that are interlocked to SMPTE timecode. In the past, WED used a modified IRIG timecode (a dual amplitude code similar to the codes used in data instrumentation in the '60s) to synchronise an entire Disney theme show, such as the *Audio-Animatronic* (3-dimensional, animated) figures, curtain controls, lighting controls, audio tape machines, etc, to the projectors. Unfortunately, the code was difficult to restore. For the new project, they wanted to convert to a standard type of timecode. SMPTE is a bi-phase code that's easy to restore, and will be located on a 24-track, 2 in, analogue master audio tape, which WED figures should provide good fidelity for at least 5,000 passes.

According to Dave Spencer: "The decision to use analogue as a theatre playback sound source was based upon the fact that at the present time, there's no feasible or cost effective way to play back 24 tracks of digital audio for 20 hours a day, 365 days a year. It is possible, but it would not be cost effective for us. However, we did want printing masters that were digitally recorded so that when either memory density, or maybe a new medium, was developed where we could play back digitally, we would already have the necessary quality to do a direct digital dub to the new format. We feel the secret of success is always planning for the future." It looks like they're off to a good start. ■



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Rest assured. If anything goes wrong with your Amcron power amplifier within three years of purchase, we'll fix it or replace it. But the odds are it'll continue to function sweetly and smoothly into the twenty-first century without so much as a hitch or a hiccup. And our warranty will be ancient history.

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The D75, for instance, sets an impeccable standard at the lower output end of the range, with its 50 watts per channel into eight ohms, balanced and unbalanced inputs, front panel controls and patent Amcron IOC indicators.

The high-powered PSA2 is the acknowledged first choice for critical applications in studio monitoring and sound reinforcement, with its high-power output, fast slewing rate and sophisticated design.

Together, the Series II D150A and DC300A represent the standard against which all general-purpose power amplifiers must be measured in terms of sound quality, versatility and sheer cost-effectiveness.

And new developments like the PS200 and PS400 with their Multi-Mode Circuitry™ are dramatic testimony to Amcron's continuing ability to turn radical new technology into rugged and practical hardware.

For further information on the world-renowned range of Amcron power amplifiers, just give Ian Jones a call on 01-961 3295.

Amcron

FROM HHB

HHB Hire and Sales, Unit F, New Crescent Works, Nicoll Road, London NW10 9AX. Tel: 01-961 3295. Telex: 923393.

AUSTRIA: HI-FI STEREO CENTER KAIN SALZBURG 37701. BELGIUM/FRANCE: GENERAL TRADING LUXEMBOURG 471548. DENMARK: TEAM SOUND APS GRAESTED 02-292522. GERMANY: AUDIO VERTRIEB PETER STRUVEN GmbH HAMBURG 5245151. FINLAND: STUDIOTEC KY HELSINKI 80-556252. HOLLAND: IEMKE ROOS IMPORT BV AMSTERDAM 972121. ITALY: AUDIST SRL MILANO 8394728. SPAIN: MABEL SDAD. ANMA. BARCELONA 3517011. SWEDEN: ELA-LJUD AB SUNDBYBERG 08-984422. SWITZERLAND: MUSICA AG ZURICH 2524952.

product guide
Digital recording equipment

DENON (Japan)
Nippon Columbia Co Ltd, No 14-14, Akasaka 4-Chome, Minatoku, Tokyo 107.

Random access PCM Editing System DN-036ED: Information on this system is somewhat scarce although we understand that it is or shortly will be commercially available. The storage medium is hard computer disk with a capacity of 4-channels for 23 minutes or about 46 minutes with 2-channel. As the system has been specifically designed for editing work as against a recording medium, the facilities available on the editing desk part of the system allows the engineer full access control over all material on the disk.

EMT (West Germany)
EMT-Franz GmbH, Postfach 1520, D-7630 Lahr. Phone: 07825 1011. Telex; 754319.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ. Phone: 01-953 0091. Telex; 27502.
USA: Gotham Audio Corp, 741 Washington Street, New York, NY 10014. Phone: (212) 741-7411. Telex: 129269.

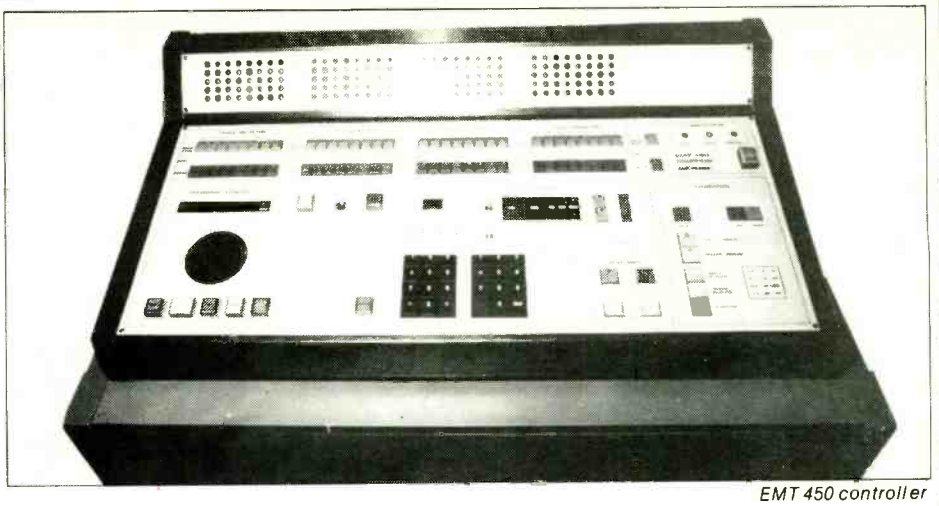
EMT 450 Digiphon: digital recording system using hard computer disk to store digital audio. Disk drive used is the Control data Corp BK6 XX with a 300 megabyte capacity giving a maximum recording time of 70 minutes for a mono signal and a maximum number of 32 tracks may be recorded at the same time although the recording time becomes progressively less with the more tracks used. The Digiphon also incorporates a microprocessor based formatter and controller interfaced with standard tape recorder controls and autolocate. Location time of any point on the disk is 50 ms and sections may be looped for repeated effect. Disks may be removed for off-line storage.
Sampling frequency: 32 kHz.
Quantisation: 16-bit linear.
Frequency response: 30 Hz to 15 kHz + ½ dB, − 3 dB.
Dynamic range: 80 dB below clipping level.

HITACHI (Japan)
Hitachi Denshi Ltd, 1-23-2 Kanda Suda-Cho, Chiyoda-Ku, Tokyo 101. Phone: 03 255-8411. Telex: 24178.
UK: Hitachi Sales (UK) Ltd, Hitachi House, Station Road, Hayes, UB3 4DR. Phone: 01-848 8787. Telex: 933611.
USA: Hitachi Sales Corp of America, 401 West Artesia Boulevard, Compton, California 90220. Phone: (213) 537-8363.

PCM-V100: combined audio PCM digital recorder and VHS format video recorder. Uses LSIs for digital circuitry, compact size, error correction before D/A conversion, available for use with NTSC, PAL or SECAM colour systems, digital electronic editing.
Sampling rate: 44.056 kHz.
Quantisation: 14-bit linear, 16-bit CRCC.
Dynamic range: 85 dB.
Frequency response: ± ½ dB 20 Hz to 20 kHz.

JVC (Japan)
JVC, The Victor Company of Japan Ltd, Tokyo.
UK: JVC (UK) Ltd, Eldonwall Trading Estate, Staples Corner, London NW2. Phone: 01-450 2621. Telex: 923320.
USA: JVC Cutting Center Inc, RCA Building, Suite 500, 6363 Sunset Boulevard, Hollywood, California 90028.

Series 90 Digital Audio Mastering System: Complete digital mastering system comprising PCM Processor, Digital Audio Editor, JVC *U-Matic* video cassette recorder and Digital Audio Delay Unit.
BP-90 Digital Audio Recording Processor: 2-channel digital audio processor. Designed for use with lo-band *U-Matic* video cassette recorders. Full error correction circuitry. Possibility of running other units in synchronisation for multi-channel recording. Dubbing facilities mode. Analogue, digital and video input and output facilities.
Modulation system: conforming to NTSC TV signal,
Sampling rate: 44.056 kHz.
Transmission rate: 3.084M bit/s.
Quantisation: 16-bit linear.
Dynamic range: 90 dB.
Distortion: 0.02%.
Frequency response: DC to 20 kHz 0.5 dB.
Dropout compensation: error detection and correction, 38-bit check, triple error correction.



EMT 450 controller

AE-90 Digital Audio Editor: Enables editing of digitally recorded tapes with two *U-Matic* VCR's and a single *BP-90* processor. Recordings can be monitored audibly by playing back at desired speed in either direction. Search function for automatic and manual scanning of precise editing point within and out points stored and accessed independently in the memory. Cross fading between selected edit points at four different rates. Digital rehearsal facility to 'run' edits by recalling music signal from memory without running the tapes. Level adjustment facility. Editing point can be shifted forward or back in 2 ms steps. Final editing operation performed automatically.
Channels: 2.
Digital input/outputs: 16-bit.
Editing accuracy: 45 μs.
Rehearsal memory: 5.92 s.
Cross fade rate: 0, 10, 17, 40 ms.
VP-1000 PCM Audio Processor: 2-channel PCM audio processor. Not part of the *Series 90* system. Designed for use with *U-Matic, VHS* or *Beta* video cassette recorder. Industrial or home-types. Contains full A/D and D/A conversion, error correction capability, digital dubbing facility, peak level metering with peak hold, calibration signal generator, pre-emphasis circuit, automatic adjustment to match VCR type.
Channels: 2.
Modulation system: PCM system using NTSC standard TV signals.
Sampling frequency: 44.056 kHz.
Quantisation: 14-bit linear.
Dynamic range: 85 dB.
Distortion: 0.03%.
Frequency response: DC to 20 kHz + .05 dB, − 1.0 dB.
Digital Preview Unit: Designed for use with 3M 4-channel digital recorder and provides two channel preview signal for disc cutting. Delay time is selectable in 5 ms increments 0 to 1.3 s or to 1.96 s with extended memory option. The delay is measured and displayed on led display. Sampling rate is 50 kHz but controlled by the speed of the replay machine ie if varispeed being used.

MITSUBISHI (Japan)
Mitsubishi Electric Corp, Mitsubishi Denki Building, Marunouchi, Tokyo 100. Telex: 24532.
USA: Mitsubishi Electric Sales America Inc, 7045 N Ridgeway Avenue, Lincolnwood, Illinois 60645. Phone: (312) 982-9282.

X-80/MS-80A Series Digital Audio Recorder: reel to reel digital recorder with similar appearance to analogue machines. Transport uses servo controlled capstan with speed errors detected by difference in phase between PCM playback signal and internal reference signal in recorder. In addition to the PCM recording on tape, there is also a channel of analogue sound and another of SMPTE address code. The analogue track enables the manual splicing of tapes as analogue tapes with extensive error correction circuitry to eliminate errors occurring at edit point. Can also be used with an electronic editing system. Off tape monitoring.
Channels: 2 PCM, 1 analogue, 1 SMPTE.
Tape speed: 15 in/s.
Tape width: ¼ in.
Sampling frequency: 50.35 kHz.
Quantisation: 16-bit linear.
Dynamic range: 90 dB.
Distortion: 0.05%.
Frequency response: ±0.3 dB 20 Hz to 20 kHz.
Weight: 140 lb.
XE-1 Electronic Editor: Automatic editing using SMPTE code, adjustable recording level, fade in/out functions, microprocessor control, memory for 99 editing points with accuracy of ±3.3 ms.
Disc Cutting Adaptor: Preview delay unit with main signal delay variable between 0.8 and 1.8 s in 0.05 s steps. Full analogue outputs.
X-800: digital multitrack recorder, 32-channels on 1 in tape running at 30 in/s, designed to be used with the *S-80* mastering machine. Digital delay to synchronise replay signal with record for sync recording, punch-in and punch-out, variable pitch. 37 tracks on tape (32 channels, one SMPTE code. two auxiliary digital and two analogue). The auxiliary tracks may be used for SMPTE timecode mix-down, or automation. Internal SMPTE generator and reader included, internal/external master clock, 100 tape position memories, comprehensive search functions, full remote control facilities.
Reel capacity: 14 in.
Quantisation: 16-bit linear.
Sampling frequency: 50.4 kHz.
Frequency response: 20 Hz to 20 kHz + 0.5 dB − 1.0 dB.
Dynamic range: 90 dB.
THD: 0.05%.

SONY (Japan)
UK: Sony (UK) Ltd, Pyrene House, Sunbury-on-Thames, Middlesex TW16 7AT. Phone: 09327 81211. Telex: 266371.
USA: Sony Corporation of America, 9 W 57th Street, New York NY10019. Phone: (212) 371-5800. Telex: 424595.

PCM-3324: 24-channel digital audio multitrack recorder with stationary head recording format. Reel to reel transport using ½ inch tape with a maximum reel size of 14 ins. Transport uses no capstan pinch wheel. Twenty eight track format with 24 PCM channels, 2 analogue audio tracks, 1 control track and one external data track. Switchable sampling rates. Full drop-in facilities etc with electronic editing capability.
Tape speed: 72.38 cm/s with 48 kHz sampling rate (66.5 cm/s for 44.1 kHz).
Sampling rate: 44.1 kHz and 48 kHz switchable.
Quantisation: 16-bit linear.
Dynamic range: 90 dB.
Frequency response: + ½ dB, − 1 dB 20 Hz to 20 kHz.
Distortion: 0.05% THD.
PCM-3204: 4-channel portable digital audio recorder with stationary head format on ¼ in tape. Transport and electronics similar to *PCM-3324* including switchable sampling rate and 16-bit linear quantisation integral A-D and D-A converters. Error correction circuitry. Illuminated meters.
Recording format: two tracks per channel with additional SMPTE code and two analogue reference tracks.
Tape speed: 15 in/s at 50.4 kHz sampling rate.
Quantisation: 16-bit linear/channel.
Density: 46,080 bpi. 30,720 flux reverse/inch.
Channel code: 3PM.
Dynamic range: 90 dB.
THD: 0.05% 20 Hz to 20 kHz.
Crosstalk: 90 dB.
Frequency response: + 0.5 dB, − 1 dB 20 Hz to 20 kHz.
PCM-1610: 2-channel digital audio processor/recorder. Designed to be used with standard Sony *BVU-200A* hi-band *U-Matic* video recorder with *U-Matic* video cassettes as the recording medium. Full error correction circuitry. Analogue audio inputs and outputs in synchronisation with other PCM recorders or video recorders. Relatively compact dimensions allow degree of portability.
Channels: 2.
Modulation system: PCM using NTSC standard TV signals.
Sampling rate: 44.056 kHz.
Transmission rate: 3.5795 Mbit/s.
Code: approx 6 words in 1 TVH.
Quantisation: 16-bit linear.

40 ▶

"No noise, nor silence, but one equal music."

John Donne, 1571-1631.

The new Klark-Teknik high-performance DN30/30 graphic equaliser offers much more than just a quiet ability to balance channels right across the audio spectrum. Thoughtful ergonomics are backed by a new circuit design breakthrough using ultra-stable microelectronic filter networks to set performance standards comparable with Klark-Teknik's 'golden oldie' the DN27A. The DN30/30 is the equaliser to boost a studio's reputation, meet broadcasting specs in less rackspace, cut costs and equipment failures on the road — because ...

It fits two matched high specification graphic channels into a single unit, each providing ⅓ octave equalisation over a full 30 ISO centre frequencies.

It gives fine fingertip low-frequency control covering the subwoofer range down to 25Hz — with touch-sensed centre detents, selectable cut boost level range and fail-safe design giving extra certainty during live events.

Its advanced design, tough construction, stringent testing and long burn-in exceed even Klark-Teknik's previously high standards for reliability and consistent performance on the road.

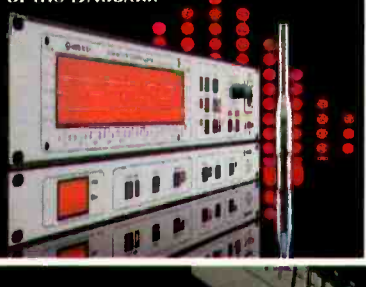
For technical information ask for:
Our DN60/RT60 Data Sheet.
Our DN30/30 Data Sheet.
Our Application Notes on equalisation.

KLARK TEKNIK
sound science

Klark-Teknik Research Limited
Coppice Trading Estate, Kidderminster, DY11 7HJ, England.
Telephone: (0562) 741515 Telex: 339821

Klark-Teknik Electronics Inc.
262a Eastern Parkway, Farmingdale, NY 11735, USA.
Telephone: (516) 249-3660

SOME MORE EQUAL THAN OTHERS
The microprocessor-based DN60 Spectrum Analyser and RT60 Reverberation Analysers provide LED matrix display of system, performer and studio response at 30 frequencies identical with those of the DN30/30.



product Digital recording equipment guide

Dynamic range: 90 dB.
Distortion: 0.05%.
Frequency response: +5 dB, −1 dB, 20 Hz to 20 kHz.
Dimensions: 17 in width × 10½ in height × 18½ in depth.
Weight: 77 lb.
PCM-100: 2-channel digital audio processor. Used in conjunction with Sony *Betamax* or lo-band *U-Matic* video recorders. Digital analogue and video outputs and inputs allowing connection to other digital equipment. Error correction circuitry. Can be synchronised by external input to clock with other equipment.
Channels: 2.
Modulation system: PCM system using NTSC standard TV signals.
Sampling frequency: 44.056 kHz.
Recording density: 2,643 Mbit/s.
Code: 128 bit/1 TVH (including 16 bits for CRCC and 28 bits for error correction).
Quantisation: 14-bits linear.
Dynamic range: 85 dB.
Distortion: 0.03%.
Frequency response: ± 1 dB DC to 20 kHz.
Dimensions: 19 in width × 8 in height × 15¾ in depth.
Weight: 33 lb.
PCM-10: 2-channel audio processor. Intended for less demanding applications and for use with domestic type video cassette recorders, 14-bit quantisation with 85 dB dynamic range but lacks digital dubbing facilities. Meets EIAJ standards.
PCM-P10: 2-channel audio processor. Playback only version of *PCM-10*.
PCM F1: compact digital audio processor designed to be used in conjunction with a VCR. NTSC and PAL versions. Has internal battery facility and can be used as a portable system. Separate mike and line inputs. (For further information see review in this issue.)
Quantisation: 16-bit or 14-bit linear switchable.
Frequency response: 10 Hz to 20 kHz ± ½ dB.
Dynamic range: 90 dB (16-bit).
Distortion: 0.005% THD 16-bit.
DAE-1100: digital audio editor. Can be used with *PCM-1610* or *PCM-100* for full electronic editing between pair of Sony *BVU-200B U-Matic* video cassette recorders. Large search dial gives control over playback speed in both directions to select and locate edit point. When selected, the memory stores 5.95 s of signal around it and by rehearsal editing the precise edit point is determined. There are seven selectable cross-fade speeds between sources, and the programme level may be controlled giving a fade in or out capability. The editor has digital tape counters for both recorders with a maximum reading of 23 h, 59 m and 29 frames. Integral SMPTE timecode generator/reader.
Digital connection: 16-bit, 2-channel.
Video connection: NTSC composite video.
Timecode input: 0 dB, 600Ω balanced or 10 kΩ unbalanced.
Search memory time: 5.95 s.
Editing resolution: 363 μs.
Crossfade time: 1 to 99 ms in 10 steps.
Preroll time: 5, 10 or 30 s.
DQP-6040: digital quantisation processor. For conversion of 16-bit digital signals to 14-bit or vice versa. Completely digital in operation. Sampling rate 44.056 kHz.
DSX-87: digital sampling rate converter. Fully digital operation for converting 16-bit signal of 44.056 kHz sampling rate to 50.35 kHz sampling or vice versa. Clock frequency may be synchronised to external signal.
DMX-800: digital mixer. Digitally mixes 4 stereo inputs to 1 stereo output. Each input has stereo fader and master metering of bargraph PPM type. Switchable sampling rate of 44.056 or 50.35 kHz or under external control. 16-bit operation.
DDU-1500 series: digital preview/delay unit for disc cutting. Comprises the *DDU-1510* preview unit with 12-bit quantisation, the *DDU-1520* unit with 16-bit quantisation, and the *DDU-1530* delay unit, delay variable in 1 ms steps from 1 ms to 2.23 s max. normal maximum being 0.74 s.

SOUNDSTREAM (USA)

Soundstream Inc, 2505 East Parley's Way, Salt Lake City, Utah 84109. Phone: (801) 486-4701. Telex: 388900.
UK: Soundstream UK, 20 – 24 Beaumont Road, Chiswick, London W4 5AP. Phone: 01-747 1391.

Soundstream offer a complete recording service for location recording, either in a studio or remote, mastering and editing as well disc mastering digital replay. Editing requires the use of a computer based system of which there are two in the US and one in West Germany and a possible system in the UK in the near future. Recording is by Soundstream developed digital recorders which they simply refer to as the Digital Recorder. It is available in 2, 4 and 8 track formats using 1 inch tape and a longitudinal fixed head format. Recently the recording hardware has also been made available for sale.
Quantisation: 16-bit linear.
Sampling rate: 50 kHz.
Frequency response: +0.2 dB, −0.5 dB DC to 20 kHz.
Signal to noise: 90 dB unweighted.
Crosstalk: − 80 dB.
Tape speed: 35 in/s.
Dimensions: 20 × 18 × 9 ins wdh, 175 lbs.

STUDER (Switzerland)

Studer International AG, Althardstrasse 150, CH-8105 Regensdorf. Phone: 01 480.29.60. Telex; 58489.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Studer Revox America Inc, 1819 Broadway, Nashville, Tennessee 37203. Phone: (615) 329-9576. Telex: 554453.

Studer announced its intention to follow a common format to Sony in 1980 and will shortly be unveiling its digital product range, prototypes of which have been shown at recent US shows. Advance information details a multitrack—the *A808* PCM multitrack tape machine, a sampling rate frequency convertor *SFC16* and the *DAD16* digital delay line. They will presumably be compatible with the Sony multitrack *3324* but more information is not available at present. (See the article *Digital Audio—the Studer view* in *Studio Sound* March 82.)

TECHNICS (Japan)

UK: National Panasonic Ltd, 308 – 318 Bath Road, Slough SL1 6JB. Phone: 0753 34522. Telex: 847652.
USA: Panasonic Co, 1 Panasonic Way, Secaucus, New Jersey 07094. Phone: (201) 348-7000. Telex; 710-992 8996.

SH-P1: digital audio PCM processor. Designed for use with any low-band rotary head video recorder such as a VHS system. CRC error correction circuitry.
Channels: 2.
Quantisation: 14-bits linear.
Sampling rate: 44.056 kHz.
Signal format: standard NTSC TV signal.
Frequency response: ± 1 dB 20 Hz to 20 kHz.
Dynamic range: 85 dB.
THD: 0.1%.
Digital Audio Tape Recorder: digital reel to reel tape recorder. This is the first model in a range using the same format, 24 or 32 channel recorders on 1 in with this model being 4-channel on ¼ in. Uses thin film heads to achieve greater track density. Transport features isolated loop transport with one large diameter capstan and two pinch rollers. CRCC error correction. Both tape splicing and electronic editing is short and there is an analogue guide track for tape splicing while a separate track is used for SMPTE code and electronic editing. Tape speed of 15 in/s.
Channels: 4 audio, 4 aux, 4 tracks per audio channel.
Quantisation: 16-bits linear.
Sampling rate: 50.4 kHz.
Dynamic range: 90 dB.
Frequency response: ±0.5 dB.
Distortion: 0.05%.
Editor: Electronic editor for handling two digital recorders. Large search dial for locating edit points by controlling speed and direction of transports. Crossfading between edit sections at 5 selectable rates. Separate digital readouts for tape position on both recorders with max 99 h, 59 m 59 s and 999 ms.
Entry cross fade times: 1 ms, 10 ms, 50 ms, 100 ms, 300 ms.
Exit fade out time: 100 ms, 300 ms, 500 ms, 700 ms, 1 s.
Editing accuracy: 119 μs.
Digital Audio Mixer: Completely digital mixer except for analogue monitoring output. Inputs are 4 line input and 4 auxiliary input each with separate level adjustment fader, pan, echo send and peak level LED meter, two channel output with master fader and meters.
Frequency response: ±0.5 dB 20 Hz to 20 kHz analogue output.
Quantisation: 16-bit linear.
Sampling rate: 50.4 kHz with internal clock or externally controllable.
Digital preview unit: fully digital preview unit for disc cutting with D/A on output. Delay time may be set between 0.1 to 1.6 s using keyboard on front panel.
Channels: 2.
Quantisation: 16-bit linear.
Sampling rate: 50.4 kHz by external clock.
Dynamic range: 90 dB.
Frequency response: ±1 dB 20 Hz to 20 kHz.
THD: 0.05%.

TELEFUNKEN (West Germany)

AEG-Telefunken, Postfach 2154, D-7750 Konstanz. Phone: 07531 862460. Telex: 733233.
UK: Hayden Laboratories Ltd, Hayden House, Chiltern Hill, Chalfont St Peter, Buckinghamshire SL9 9UG. Phone: 02813 89221. Telex: 849469.

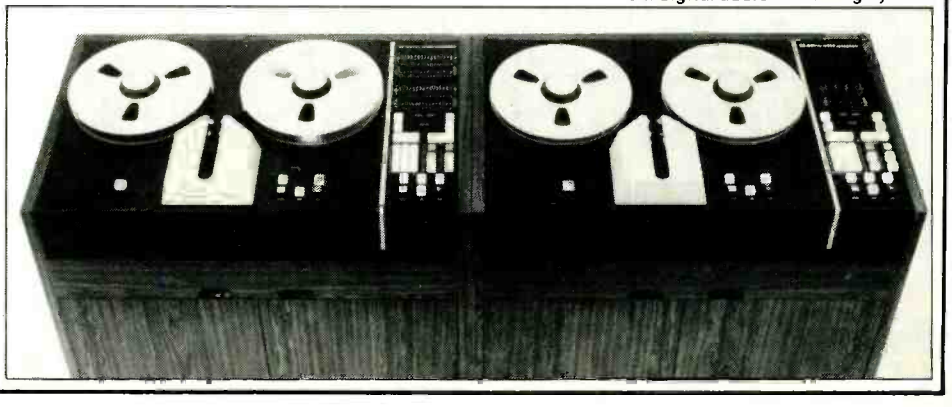
Telefunken are marketing the Mitsubishi range of digital audio equipment under the Telefunken name in Europe and the UK. Specifications are as for the Mitsubishi models.

3M (USA)

3M Mincom Division, 3M Centre, St Paul, Minnesota 55101. Phone: (612) 736-9567. Telex: 297434.
UK: 3M UK PLC, PO Box 1, Bracknell, Berkshire RG12 1JU. Phone: 0344 26726. Telex: 849371.

Digital Audio Mastering System: digital multitrack recorder and mastering machine. The multitrack is 32 channel on 1 in tape with tape speed of 45 in/s. The transport is similar to the traditional 3M *Isoloop*. The mastering machine is 2 or 4 channels on ¼ in. Full remote control facilities. Error correction circuitry. Autolocator with 10 position memory. Variable speed. 14 in reel capacity. Single track/digital channel.
Quantisation: 16-bit.
Sampling rate: 50 kHz. 48 kHz may be available as a future option.
Frequency response: ±3 dB 20 Hz to 18 kHz.
S/N: 90 dB.
THD: 0.03%.
Input: +28 dBm max, 20 kΩ balanced or unbalanced.
Output: prewired −6 to +16 dBm with 600Ω load, floating transformer output.
Tape speed: 45 in/s.
Digital Editor: control module for two *Digital Mastering Systems*. Master tape to be edited requires Edit timecode recorded from beginning to end on one track. The edit point botton selects the edit points on both machines. The pre-audition button used in conjuction with the shuttle button will rewind the selected machine to two seconds before the edit and go into play until the edit, where it mutes the audio and the machine then rewinds to repeat the action. The post-audition button works in the opposite way—2 seconds before the edit and mute until the edit point, then signal for two seconds, then rewind. The edit points can be moved in both directions in various increments from 1 ms. The edit button activates both machines from 10 s before the edit point, places both machines in play and at the edit point the second machine is placed in record and the edit completed. The Preview Edit button will allow the edit to be demonstrated without recording. ■

3M digital audio mastering system



The funny-looking mike that's taken very seriously.

The PZM with its flat back plate, is as unconventional as it looks. Its revolutionary design eliminates phase-induced interference and provides a significant improvement in signal quality.

Indeed the PZM represents the most important advance in microphone technology of the last fifty years.

But why take our word for it? The PZM has won the utmost respect from sound engineers all over the world.

In a recent issue, Studio Sound examined the applications of the PZM in contemporary recording, and its conclusions are a glowing endorsement of all we've claimed for this remarkable microphone.

Top engineers were interviewed and confirmed that the PZM provided an undistorted output, free from comb-filtering.

They found that it gave a 180° pick-up with no off-axis problems and that it was ideal for reproducing anything from ambience to a grand piano.

They spoke of its extraordinary reach and clarity, of the way in which it simplified the business of miking-up and how its low profile made it ideal in hidden applications such as theatre and television.

All in all, the experts are deeply impressed by the PZM and already regard it as an indispensable tool in the creation of a transparently natural sound, free from non-linear characteristics, both on stage and in the studio.

For details of available models, including the new 3LV tie clip microphone, prices and suggestions for further applications of the PZM microphone, just telephone Mike Silverston on 01-961 3295.

PZM FROM HHB

HHB Hire and Sales, Unit F, New Crescent Works, Nicoll Rd, London NW10 9AX. Tel: 01-961 3295. Telex: 923393.

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BELGIUM/FRANCE: GENERAL TRADING LUXEMBOURG 471548.
DENMARK: TEAM SOUND APS GRAESTED 02-292522.
GERMANY: AUDIO VERTRIEB PETER STRÜVEN GmbH HAMBURG 5245151.
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ITALY: AUDIST SRL MILANO 8394728.
SPAIN: MABEL SDAD. ANMA BARCELONA 3517011.
SWEDEN: ELA-LJUD AB SUNDBYBERG 08-984422.
SWITZERLAND: MUSICA AG ZURICH 2524952.

product Console automation guide

B & B AUDIO (USA)

Aphex Systems Ltd, 7801 Melrose Avenue, Los Angeles, California 90046. Phone: (213) 655-1411. Telex: 910-321 5762.
UK: AKG Acoustics Ltd, 191 The Vale, London W3 7QS. Phone: 01-749 2042. Telex: 28938.

Model OAS-24 Grouping System: VCA sub-grouping system in portable or semi-permanent form. Consists of control console with four or more DC group control modules each with 24 group assign switches, group mute and fader, with one master control module with 24 channel master group selection switches, mute sequence switch and master fader. Also the VCA/connector case housing the VCA cards and *XLR* connectors for inputs and outputs. The basic mainframe holds 9 sub-groups and one master module but the system can be expanded to 48 channels.
Frequency response: ±0.5 dB 20 Hz to 100 kHz.
THD: 0.02% at +10 dBV input.
Ouput noise: −84 dBm.
Max gain: unity.
Max attenuation: 94 dB.
Tracking accuracy: 1.5 dB between channels in any combination.
Inputs: 10 kΩ unbalanced.
Outputs: 600 Ω transformer balanced +24 dBm max.

HARRISON (USA)

Harrison Systems Inc, PO Box 22964, Nashville, Tennessee 37202. Phone: (615) 834-1184. Telex: 555133.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ. Phone: 01-953 0091. Telex: 27502.

Auto-Set: originally designed for use with Harrison *24 Series, 32 Series* and *32B Series* consoles but can be used with any console operating with DC control voltages as the automation interface including those designed to work with the Allison *65K* Programmer. Uses include control of video, lighting and special effects control. Input is via a full ASCII keyboard with 5 in CRT. Total system enclosed in single unit excepting power supply. Has ability to store up to four independent mixes on one track of an audio tape recorder with each of these mixes consisting of up to 63 console positions. Information may be extracted from the store as required for a particular channel without regard for the source selected for another channel. Another method of operation is for use when there is no storage medium for information and uses the internal memory of the *Auto-Set* which can record 10 static 'pictures' of the console positions and these are manually selected with a choice of an immediate jump between the two settings or a fade, variable from 0.1 to 9.9 s. These settings may also be recorded on cartridge with up to 630 pictures on each one and recall is initiated by the keyboard controlling the integral cartridge interface of the *Auto-Set*. Further hardware will be made available for the *Auto-Set* including SMPTE timecode generator/reader and floppy disk storage system.

Auto-Set II: hard disk based automation system for *MR-1* and *PP-1* hybrid audio mixing consoles which use DCI (distributed control intelligence). The system comprises a standard Remex Data Warehouse *RDW-3200* disk unit providing a fixed formatted 18Mbyte *Winchester* disk with twin double-sided double-density floppy diskettes for off-line back-up, providing 1.2 Mbyte each, and the main *Auto-Set II* processor based on *Z8000* and *Z80* processors. Required peripherals are an *MR-1* or *PP-1* mixing desk with level I virtual console, and an external-timecode reader or digital feet and frame counter. Optional additions are a CRT data terminal for level II and beyond, a colour CRT graphics generator and various automated peripherals. The main processor is supplied with either high speed (800 kbaud) serial ports for interface to the console, a CRT serial port and a synchroniser interface.
There are currently two command levels, level I allowing the operator to generate data, use data, update data, store data on a diskette, and clear the data space, all controlled from a small switch and indicator array located in the mixing desk, while level II allows data to be stored in labelled files, to merge data files, generate windowed record and write commands, enter off-line data from external sources, and assign labels to timecodes, mixing desk channels and data files. Level II requires a data terminal for the greater complexity of inputing.

Autograph: automation ready graphic equaliser comprising a control head and the analogue processor cards. The control head contains the Harrison *5004 CPU* computer card to interface between the digital signals from the automation system and the analogue processing. Panel controls include 7 EQ bands of ±8 dB with the top band switchable to 6.9 kHz or 8 kHz.

LEUNIG (West Germany)

UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900. Telex: 554494.

Param: Computer assisted equalisation system comprising a minimum of five components including central control panel, computer system. EQ power supply and monitor. The computer, EQ and power supply are standard 19 in rack mounting units of 3 U high. The central control panel is 9 × 5⅞ × 3⅛ in with a soft leather cushioned back enabling it to be placed on top the console while being used. It consists of a calculator type keyboard with additional system control switching and a joy stick. The joy stick can be used to 'draw' an equalisation curve on the video monitor while the computer adjusts the internal filters to achieve it. The alternative method is to shape an already existing response. The EQ may be heard whilst adjusting the curve.
Entire console equalisation can be stored in one of the 64 console memories which may be copied from one to the other to assemble them into a chronological order and during the session they can be recalled by one key-stroke or by an external signal such as from a tape linked automation system or pulse on the tape. 32 standard equalisations may be recalled from any channel with a single button allowing comparison between different channel or complete console settings. All equalisers may be 'zeroed' by a single operation. The monitor can display instantly any of the console settings content of the memory, channel status, channel failure and number of overloads, optional 10 band spectrum analyser display on equaliser channel with display of difference between total equalisation and input signal. All stored information may be transferred to cassette or floppy disc systems for permanent storage. The video monitor may be any model capable of accepting a 75 Ω NTSC composite video signal.
The equaliser consists of low and high shelving EQ. 2 mid peaking EQ and high and low cut filters. Options include programmable gain, equaliser grouping and provisions for automation of send level and pan control in quad.
Noise: −96 dBm 20 Hz to 20 kHz bandwidth all equalisers level.
THD: 0.05% at +20 dBm into 600 Ω.
Max input: +24 dBm into 600 Ω unbalanced (balanced versions available as option).

MCI/SONY (USA)

MCI, a Division of Sony Corporation of America, 1400W Commercial Boulevard, Fort Lauderdale, Florida 33309. Phone: (305) 491-0825. Telex: 51436.
UK: MCI (Professional Studio Equipment) Ltd, 54–56 Stanhope Street, London NW1 3EX. Phone: 01-388 7867. Telex: 261116.

JH-50 Console Automation: originally designed for MCI consoles, this system is now available for retro-fitting in other consoles. Each channel to be automated requires a VCA fader package with the rest of the system comprising a 19 inch rack cabinet containing the digitaliser and micro-processor boards and a power supply. This system will provide automation control for level, mute and sub-grouping (8) and solo-in-place. Information storage is on one track of the multitrack tape machine.

MELKUIST (UK)

Melkuist Ltd, 35a Guildford Street, Luton LU1 2NQ. Phone: 0582 416028. Telex: 825828.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Studio Maintenance Services Inc, 12458/60 Magnolia Boulevard, North Hollywood, California 91607. Phone: (213) 877-3311. Telex: 674901.
USA: Trident (USA) Inc, 652 Glenbrook Road, Stamford, Connecticut 06906. Phone: (203) 357-8337. Telex: 643678.
USA: Valley Audio, PO Box 40743, 2821 Erica Place, Nashville, Tennessee 37204. Phone: (615) 383-4732.

GT800: console automation system designed to operate with mixing consoles capable of producing and accepting up to 64 voltage replicas of control functions. Uses BASF dual drive single density IBM format floppy diskette drives, high speed scanning alogorithm running at twice SMPTE/EBU timecode scan rate, special loose lock timecode reader for

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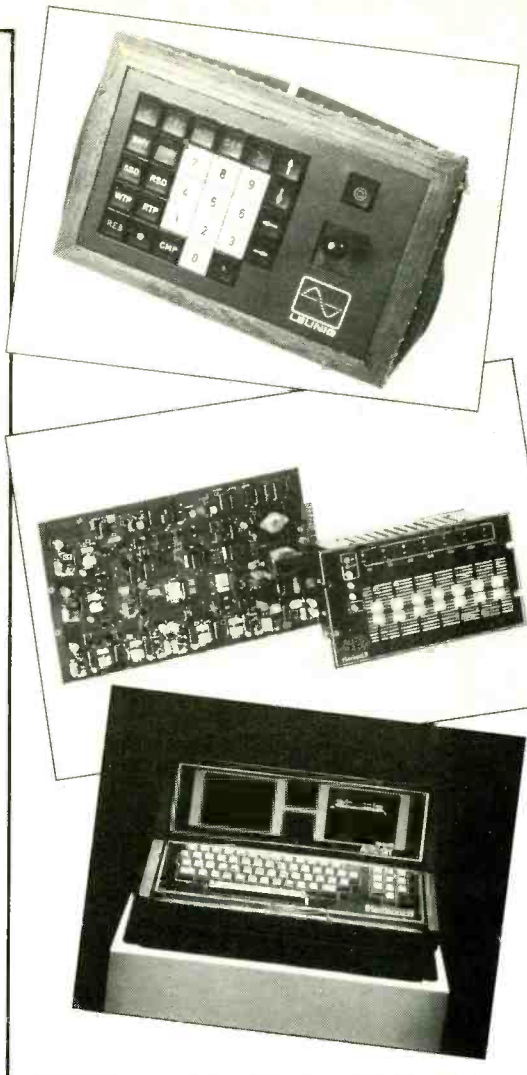

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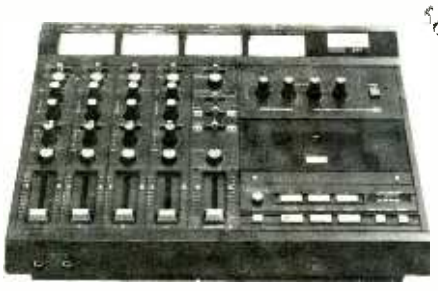
TASCAM 38 MULTITRACK

Everything the 88 had – and more! This 8 track on half-inch recorder boasts on-board varispeed, foot-switch controlled punch in/out, digital tape counter including Zero Search, and an improved tape transport handling. DBX unit also available.



TASCAM 244

Teac's new PORTASTUDIO with improved facilities. Includes simultaneous recording on 4 tracks plus parametric EQ, digital tape counter and full stereo monitoring.

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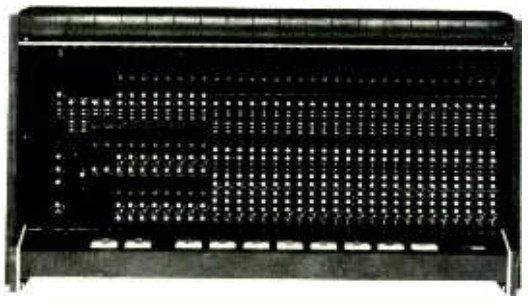
The Who
Depeche Mode
Theatre Projects
Matrix Studios
Royal Opera House
Greater London Council
Blackwing Studios
and many, many more...



TASCAM M-16

The perfect complement of the 85-16B, this 24 channel console is the ultimate, offering 16 track monitoring, stereo solo-in-place, auto-mix switching, remix solo enable, dual-concentric parametric EQ, flexible monitoring, stereo echo, 100mm conductive plastic faders.

TASCAM 85-16B

Tascam's greatest machine! An outstanding performer 16-track/1" plus integral dbx. 15 ips +10% varispeed. 67dB S/N. Quartz controlled DC motors. OPTIONAL MUSIC LAB FEATURES: 30ips capability, 25' remote control with 6-postition autolocator, SMPTE sync system for video.

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good low level synchronisation. The system uses standard EBU timecode recorded on the multitrack. The central processor keeps track of system operation displaying suitable error messages when problems occur, while comprehensive memory management techniques allow most damaged diskettes to be suitably recovered.
Interface to the console is by the digital filter subassembly, which contains all hardware for analogue to digital conversion, data processing, communication, and digital to analogue conversion, with 64 separate input/output ports. Fast digital filtering algorithms process the derived movement coefficient matrix to create a new matrix of interpolation coefficients for transmission down the serial data link to the CPU. The *GT800* system may be used either with Melkulst digital transfer faders, or with conventional analogue transfer faders such as *Fadex*. Each fader module has a linear slider fader, subgroup selection, isolate/read/write/update switch, and cut switch, with ready light. In isolate mode, the fader controls normal audio with no assistance, in write all movements are stored, in read the processor controls the audio and the fader is disabled, while in update the fader may generate manual offset, but not affecting the stored information. A null indicator shows when the current fader position and date returning are within a nominal 2 dB. Various fader central operations are also provided.
VCA Grouping Fader: there is a version of the *GT800* available without the mixdown automation but retains the VCA bypass facility and non VCA group mixdown. In the VCA mode it has both a mute switch and solo function.
Events Unit: a 32-channel event unit stand-alone system with battery supported memory of up to 250 combinations of the 32 events programmed relative to SMPTE/EBU timecode. Can be interfaced with *GT800* automation system for storage of events long term or data may be dumped to tape.

NEVE (UK)

Neve Electronics International Ltd, Cambridge House, Melbourn, Royston SG8 6AU. Phone: 0763 60776. Telex: 81381.
USA: Rupert Neve Inc, Berkshire Industrial Park, Bethel, Connecticut 06801. Phone: (203) 744-6230. Telex: 969638.

NECAM: Computer controlled automated mixdown system. System comprises console fader package type 1785, console peripheral unit, control unit, tape machine peripheral unit, computer rack comprising CAI Alpha *LSI-2/20* computer and dual floppy disk drive with power supply. A SMPTE time code generator is also required. The peripheral items may be built into equipment as may the control panel but is normally free standing on the console. The computer rack is standard 19 in and 4 ft high. The principal feature of the Necam system is the servo-controlled faders which actually move following the original action and making a very easy process with the possibility of manual override in mixing. Full remote and locate facilities are provided for the multitrack tape machine. One track of the tape machine has SMPTE code recorded on it and a special wide band head amp to enable the reading of the code at high winding speeds. Up to 999 points of the tape may be 'labelled' for use in automated runs. The control unit has 16 instruction keys, a numeric keypad and a 32 character alphanumeric display provides full instruction for the computer. The system incorporates a sophisticated merge capability with the ability to merge certain tracks from different mixes. Version of Necam for post-production work known as Necam 'D'.

NTP (Denmark)

NTP Elektronik A/S, 44 Theklavej, DK-2400, Copenhagen NV. Phone: 01 10.12.22. Telex: 16378.

Programmable Equaliser 582-100: remote controlled 14-band graphic equaliser with floppy disk interface. The equaliser section band centres are ⅔ octave spaced and all ±14 dB in 2 dB steps with all bands switchable to a shelving characteristic. Associated microprocessor controls all functions. Up to a maximum of 64 channels may be handled at one time and equalisation may be copied from one channel to another. The internal electronics are capable of storing up to 16 console settings but the addition of the floppy disk will give a maximum of approximately 200. The equalisation curve under the controls is displayed on an LED display.

PLUS 30 (France)

Plus 30, 37 rue des Annelets, F-75019, Paris. Phone: (1) 202.21.02.

Plus 30 have an automation system based around the Allison *65K* Programmer using their own design fader system known as *ADS*. Each fader has a 10 position thumbwheel allowing 9 DC subgroups, group master function, write and update modes, null LEDs, conductive plastic fader. VCA element is the B & B *1537*.

QUAD/EIGHT (UK)

Quad/Eight Electronics Inc, 11929 Vose Street, North Hollywood, California 91605. Phone: (213) 764-1516. Telex: 662446.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. PHone: 01-580 4314. Telex: 28668.

Compumix II: Automated mixdown system with dual drive floppy disk storage. Capacity for 32 takes on each disk. Requires the use of one tape track for SMPTE timecode. Internal SMPTE generator and reader. Editing between mixes by entering time codes into edlting table on controller. Colour CRT to display VCA attenuation and mutes, operating instructions, automatic disk directory table, disk preset editing, timecode, prompting instructions, error instruction. Designed for use with *Coronado* console, and a new version for use with the *Ventura II* console.

ROLAND (Japan)

UK: Roland (UK), Great West Trading Estate, 938 Great West Road, Brentford, Middlesex TW8 9DN. Phone: 01-568 4578. Telex: 888941.
USA: Roland Corp US, 2401 Saybrook Avenue, Los Angeles, California 90040. Phone: (213) 685-5141.

CPE-800 Compu-editor: console level automation system. Self contained unit capable of handling 15 channels of signal. The signal from each channel to be automated is patched through the *CPE-800* and in all other functions the console remains as normal. It will operate in three separate modes – with SMPTE code recorded on multitrack tape generated from an internal unit, from an internal clock or in a one-shot mode where complete sets of levels are stored for instant recall. Level information is digitalised and stored in the internal *CPE-800* memory. For storage this memory may be dumped on to tape or cassette. Provision for graphic display outputs.

SOLID STATE LOGIC (UK)

Solid State Logic Ltd, Churchfields, Stonesfield, Oxford OX7 2PQ. Phone: 0993 898282. Telex: 837400.
USA: Musicworks International, 2352 Wisconsin Avenue, Washington, DC 20007. Phone: (202) 333-1500. Telex: 440519.

Total Recall: automation system as an integral part of the *SL-4000 Series* console. The computer controlling the system can scan the complete console and store the entire console status including input selection, routing, monitor and foldback levels, panning, equalisation, echo sends and dynamics modifications. This information may be recalled instantly and the computer can display on the monitor which controls do not match the stored memory. Uses network of microprocessor controlled data busses which carry low-voltage analogue information to each I/O module and enables use of continuously variable controls rather multiple VCA or stepped controls. Floppy disc storage. Also acts as tape locator.
Real Time System: automation system that can operate with or without SMPTE time code off tape. Consists of presets that are a record of static fader levels and mutes as well as information from the *Effects Controller*. Any number of presets can be stored and recalled or edited. Used with SMPTE time code, times and durations of presets can be entered.
Events Controller: allows the programming of up to 32 timecode triggered relays which may be used to switch outboard equipment. Up to two units can be linked to one SSL computer.
Effects controller: up to 40 devices can have their control voltages stored and recalled referenced to SMPTE time code.

SOUNDCRAFT (UK)

Soundcraft Electronics Ltd, 5–8 Great Sutton Street, London EC1V 0BX. Phone: 01-251 3631. Telex: 21198.
USA: Soundcraft Inc, 20610 Manhattan Place, Suite 120, Torrance, California 90501. Phone: (213) 328 2595. Telex: 182499.

Automation system for the 2400 Series console: comprises *MC6809* microprocessor and VCA modules with full level control capability. The *MC6809* is an 8-bit system with 16-bit architecture with a priority encoding routine claimed to be faster than other comparable systems. A single channel mode button controls entry to read, update and rewrite modes and the software is programmed to determine the next most logical mode although this can be overridden. Mute control is independent to the fader. Any channel can be designated a VCA sub-group and any number may be used together with one master. There is an auto-nulling facility where the point of fader entry becomes the zero with no change in level. Data storage is on one of two tape tracks although future expansion will include a disk based memory.

SOUND WORKSHOP (USA)

Sound Workshop Inc, 1324 Motor Parkway, Hauppauge, NY 11788. Phone: (516) 582-6210. Telex: 649230.

Diskmix: add on computer that interfaces with Sound Workshop, MCI, and Valley People (Allison) automation systems. It is a SMPTE timecode based system that stores multiple mix data on floppy disks. It allows merging and editing of mix data, control of automated console parameters, and storage of session documentation. *Diskmix* consists of the main computer unit (which also houses dual disk drives) a controlling keyboard computer, and a colour monitor (user provided). The ultimate dedication of the *Diskmix* computer will be determined by market input. In its fullest implementation. *Diskmix* could control and store operating parameters for the studio's full compliment of peripheral equipment.

SPHERE (USA)

Sphere Electronics, 20201 A Prairie Avenue, Chatsworth, California 91311. Phone: (213) 349-4747.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.

Datalog: multi-level automation system. Level I consists of a *6800* series computer, command panel, a keyboard and a set of master input faders. It offers facilities including fader grouping, 4 master group faders, 80 preset fader 'snapshots', 2 mute group presets and 2 fader presents. Can be interfaced with Allison *65K* programmer and there are a number of options available. Level II has all the facilities of Level I but with the addition of SMPTE generator/reader, floppy disk drive, 8 digit readouts above each fader where track assignments can be displayed on dB levels, and when used with new Sphere consoles EQ and inserts can be controlled. Level III is a conceptual framework for a Sphere custom designed automation system incorporating computer interface of the console, tape machine operations and possibly session record keeping and client invoicing.

VALLEY PEOPLE (USA)

Valley People Inc, PO Box 40306, 2821 Erica Place, Nashville, Tennessee 37204. Phone: (615) 383-4737.
UK: Scenic Sounds Equipment Ltd, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Allison 65K Programmer: programmer accepting and producing control voltages from 0 to +5.6 V DC. Basic programmer frame contains four *MUX-65K* cards with each one acting as 16 channel input multiplexer and 16 channel output de-multiplex/sample and hold. Extender frames may be used to bring the number of analogue functions to 4096. Each control voltage is divided into 128 increments and when used with VCA having 20 dB/V, a 112 dB dynamic range is processed. The system requires the use of a track on the tape recorder. No special adjustments are required other than normal. Can be used to programme equalisers, pans, echo sends etc and any other programmable elements. Now available, an update kit including digital display section, digital test code section and analogue test and calibrate section. Also available is an Anti-Dither I/O card.
Fadex: programmable fader system designed for use in new equipment or as a retro-fit into non-programmable consoles to interface with the *65K* Programmer. Designed to be 'standard' fader size and are available with differing dimension top plates. The fader element is a P & G conductive plastic with a *EGC-205M* VCA giving a control range of 145 dB. Fadex provides in addition to programmable level control, up to 9 VCA sub-groups, channel mutes and solos, group mutes and solos, VCA master, nulling indication, remote/master control logic. Complete Fadex system includes Fadex module with VCA per channel, one VCA master and one stereo master, master logic card, power supply, console interconnection package and programmer to console interconnection and one *5K* Programmer. ■

Sony have just made the world's most advanced 24-track analogue recorder...

product guide
VCAs, faders & panpots

AUDIOFAD (UK)

Audiofad, Unit 16, 31 Poole Road, Wimborne, Dorset BH21 1QB. Phone: 0202 886322. Telex: 47674.
USA: Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900.

Model 1040P: conductive plastic linear motion fader, mono or stereo, front panel mounting with choice six black aluminium fascia panels with white lettering using different scales in fade up or fade down formats. Panels can also be custom sizes.
Resistances: log 600 Ω, 5 kΩ or 10 kΩ; linear 10 kΩ or 25 kΩ; also VCA law functions. Tracking error ±3% linear, 0 – 20 dB ±1 dB log standard grade and ± ½ dB premium grade. Stereo matching 1 dB max 0 – 40 dB.
Maximum attenuation: 90 dB
Travel: 104 mm
Options: low profile knob in choice of six colours, maximum of two micro-switches, and choice of electrical connections.
Dimensions: without fascia 136 x 18 x 43½ mm lwd.
Model 1240P: similar to Model 1040P but designed to fit 142.24 mm mountings and choice of lever offsets.
Dimensions: (without fascia) 148.6 x 18 x 44 mm lwd.

B & B AUDIO (USA)

Aphex Systems Ltd, 7801 Melrose Avenue, Los Angeles, Cal 90046. Phone: (213) 655-1411. Telex: 910-321 5762.
UK: AKG Acoustics Ltd, 191 The Vale, London W3 7QS. Phone: 01-749 2042. Telex: 28938.

1537A: voltage controlled attenuator in 14-pin DIP package requiring several external IC op-amps and other components for full operation. Claimed to operate as a true class A device to eliminate crossover distortion.
Distortion: THD 0.004% (+ 10 dBm input, 10 dB atten) 20 – 20 kHz, IMD (SMPTE) 0.03% (– 14 dBm input, 10 dB atten).
Noise: – 90 dBV ±1 dB worst case, unity gain.
Modulation noise: 6.5 dB.
Maximum attenuation: 94 dB, 15 dB gain available only to order.
Dc shift v attenuation: 5 mV fader grade, 2 mV select grade.
VCA505: universal VCA card using *1537A* IC, but requiring no external circuitry, all op-amps on sockets so they may be easily replaced when better ICs become available. PC mounted with 15-pin card edge mount. Will retrofit many Allison VCAs if gain is not required.
Distortion: THD 0.01%. IMD 0.03%.
Noise: – 90 dB.
Max attenuation: 100 dB.
Dc shift v attenuation: 5 mV.
VCA500A: retrofit card for the MCI *500 Series* console providing VCA facilities.
2521: operational module 'the unpotted one', high speed, high output, short circuit proof buffer amplifier module. Provides 1 W (+ 30 dBm) into 62 Ω, 50 Ω driver version available to order.

CITEC (UK)

Citec Ltd, Cheney Manor, Swindon, Wiltshire SN2 2PZ. Phone: 0793 487301. Telex: 897971.

Type 100SF Series: conductive plastic linear motion faders, mono or stereo, linear or audio laws, wide choice of output functions and up to 4 built-in micro-switch facilities. Metal housing.
Resistances: 1 kΩ, 10 kΩ, 100 kΩ linear (100SF and 102SF); 600 Ω, 5 kΩ, 10 kΩ, 100 kΩ audio law (120SF and 122SF). Essentially infinite resolution. Stereo matching ±2% linear and within 0.5 dB 0 – 20 dB, within 1 dB 20 – 40 dB and within 2 dB 40 – 70 dB.
Travel: 104 mm.
Options: alternative attenuation laws, two knob sizes in black or white, additional micro-switch options with overpress facility; flying lead or 15-way socket terminations, three fascia sizes, natural or matt fascia with characters or unscaled.
Dimensions: without fascia 149 x 19 x 46 mm lwd.

DANNER (West Germany)

Konstantin Danner, Zienstrasse 28, D-1000 Berlin 62. Phone: 0311 781-1822.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Gotham Audio Corp, 741 Washington Street, New York NY 10014. Phone: (212) 741-7411. Telex: 129269.

A selection of faders from Penny & Giles

Danner manufactures a wide range of high quality (and very expensive) linear motion faders. Further details upon applications.

dbx (USA)

dbx Inc, 71 Chapel Street, Box 100C, Newton, Massachussets 02195. Phone: (617) 964-3210. Telex: 922522.
UK: Scenic Sounds Equipment, Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

2150A: VCA current in, current out integrated circuit housed in a plastic 8-pin single-in-line casing.
Distortion: 0.02% THD maximum at 0 dB gain; 0.01% IMD at + 15 dB gain.
Noise: – 95 dB typical at 0 dB gain.
Gain: – 60 dB to + 40 dB.
Output noise: – 95 dBV at 0 dB gain.
Frequency response: 20 Hz to 20 kHz.
Supply voltage: ± 12 V.
2151: Similar to the 2150A but with lower THD at higher gains.
2001: all discrete VCA module with the same physical dimensions and connections as the 202, a current in/current out device which needs the maximum of external components.
Gain control range: – 100 to + 60 dB.
THD: 0.01%, typically 0.003%.
IMD: 0.01%.
Output noise: – 87 dB 20 Hz to 20 kHz, 0 dB gain.
Frequency response: to 50 kHz ±0.02%, lower frequency limit determined by input coupling capacitor.
Control voltage feedthrough: 10 mV.
202: voltage controlled amplifier with pin connections in an encapsulated package. Requires a minimum of external components. There is also an upgraded version known as the 202C.
Gain control range: – 100 to + 40 dB.
Distortion: 0.03% 2nd harmonic, 0.04% 3rd.
Frequency response: to 20 kHz to ±0.3 dB. Lower frequency limit determined by input coupling capacitor.

DUNCAN/CGS (USA)

Duncan Electronics, 2865 Fairview Road, Costa Mesa, Cal 92626. Phone: (714) 545-8261. Telex: 910-595 1128.
UK: The CGS Resistance Co Ltd, Marsh Lane, Gosport Street, Lymington, Hants SO4 9YQ. Phone: 0590 75255. Telex: 47691.
Note: both Duncan and CGS are subsidiaries of Thorn EMI.

Metric series M400 Sideline: conductive plastic linear motion faders, mono or stereo (and two wipers on a single track), injection moulded plastic construction, solder tag or pin connections, optional fascias calibrated 0 to 50 dB in 5 dB steps, 60 dB and infinity, or 0 to 100 in 10 unit steps.
Resistances: linear 1 kΩ to 250 kΩ (optional centre tap) log 600 Ω to 25 kΩ and balance with opposite log tracks 1 kΩ to 25 kΩ.
Maximum attenuation: 90 dB.
Effective travel: two versions, 70 mm or 108 mm.
Stereo: ganged tracks in same package as mono.
Switching: optional twin microswitches at infinity end of travel.
Knobs: available as round, low profile, square or pointer, in white, black, red, blue, yellow and green.
Dimensions: 70 mm travel version 108 mm x 10 mm x 20 mm high, 108 mm travel version is 152 mm long.

MCB (France)

MCB, 11 Rue Pierre Lhomme, 92404 Courbevoie. Phone: (1) 788.51.20. Telex: 620284.
UK: Cetronic Ltd, Hoddesdon Road, Stanstead, Abbots Ware, Herts SG12 8EJ. Phone: 0920 871077. Telex; 817293.

AT104/AT2104: conductive plastic linear faders with infinite resolution, precious metal multi-contact cursor, outputs on screen cable and metal casing. *AT104* is a 2-track model.
Resistances: 600 Ω, 5 kΩ, 10 kΩ. Track matching on *AT2104* ±1 dB, 0 – 40 dB.
Maximum attenuation: 85 dB.
Travel: 104 mm.
Dimensions: 142 x 18.5 x 47 mm lwd.
RC80/RC120: slim line linear conductive plastic faders, can be fitted with max and min cursors, solder pin connections, screw fixing, no casing.
Resistance: 10 kΩ.
Travel: 80 mm RC80; 120 mm RC1200.
Dimensions: RC80, 124 x 16 x 18½ mm lwd; RC120, 160 x 16 x 18½ mm lwd.

PENNY & GILES (UK)

Penny & Giles Conductive Plastics Ltd, Newbridge Road Industrial Estate, Pontllanfraith, Blackwood, Gwent, South Wales NP2 2YD. Phone: 0495 228000. Telex: 498135.
USA: Penny & Giles Conductive Plastics, 1640 Fifth Street, Santa Monica, California 90401. Phone: (213) 393 0014. Telex: 652337.

3000 series: conductive plastic linear motion faders, available in single or dual channels, full environmental shielding, integral switch facilities, new type of construction so that knob runs along a semi-circular rod providing maximum difficulty to dirt and liquid ingress.
Availability: single or two channel, linear or audio taper.
Resistances: 10 kΩ linear, 5 k and 10 kΩ audio taper on different models.
Maximum attenuation: – 85 dB, – 95 dB, – 95 dB and – 100 dB for four types.
Effective travel: four models, 65 mm, 70 mm, 83 mm and 104.2 mm strokes.
Switching: one per fader closed at infinity/0 end of travel.
Knobs: 11 mm wide red, yellow, blue, gree, black and grey, 16 mm wide red, yellow, blue green, black and orange.
Dimensions: four models, 93 x 31 x 13 mm, 115 x 31 x 13 mm, 115 x 31 x 13 mm, 133 x 31 x 13 mm.
1500 series: conductive plastic linear motion faders, available in mono, stereo, four or eight gangs, metal construction, 15-way socket connector (mating connector supplied) range of four top plates and fascias of different sizes, or rear mounting top plate (no fascia). Fascia calibrated 0 to 70 dB and infinity.
Availability: unbal linear 1, 2, or 4 channels, unbal log fader audio taper 1, 2, 4 or 8 (to order) channels (includes infinity gap), unbal ladder ader 1, 2, or 4

48 ▶

Even the finest 24-track analogue recorder money can buy can't compete with the Sony PCM-3324 Digital Multi-Track recorder. Its unbeatable sound quality takes recording performance that much closer to reality.

With punch-in/punch-out recording, variable speed and splice edit, the PCM-3324 has all the features you'd expect from the ultimate analogue machine. But with the added benefits of Digital technology.

Because with a dynamic range in excess of 90 dB, undetectable wow and flutter, no print-through, no loss of quality on repeated copying, and variable cross-fade, Sony are revolutionising multi-track mastering.

Giving the recording engineer unprecedented control over the creative recording process and making better use of valuable studio time.

And because the PCM-3324 is fully tape compatible with comparable 24-track systems from other leading makers such as Studer and MCI, Sony have ensured you won't be tying up capital in a format which doesn't even cross the boundaries of your own studio.

If you'd like to know more, contact Keith Smith or Mike Jopp at Sony (UK) Ltd, Pyrene House, Sunbury Cross, Sunbury-on-Thames, Middlesex, or telephone Sunbury (09327) 81211.

And find out how Sony are making even the newest analogue recorder out of date.

COMMUNICATION SYSTEMS DIVISION

SONY

...obsolete.

product guide — VCAs, faders & panpots

channels, balanced fader audio taper 1 or 2 channels. Also special laws for VCAs.
Resistances: linear 1 kΩ or 10 kΩ, log 600 Ω, 5 kΩ or 10 kΩ.
Maximum attenuation: varies with models, 105 dB generally, 90 dB for some 2 and 4 channel models.
Effective travel: 104 mm.
Switching: three choices of switches, aux equipment (operates within 4 mm of infinity marking), infinity cut-off (short the wiper to common within 4 mm of infinity marking), overpress (when knob is pushed back against an internal spring beyond the infinity marking).
Knobs: available standard as white or black, options for red, blue, green and yellow, 16 mm or 11 mm wide. Also ganging bar to link two adjoining faders.
Dimensions: fascias/top plates 174 × 45, 178 × 38, 190 × 40, 174 × 19 and 148 × 19 mm (back mounting), 46 mm high.
1100 Series: conductive plastic linear motion faders, available in mono or stereo. Similar to the *1500 Series* but 'cost effective' with a more limited accuracy specification and without some of the features of the *1500 Series*. Normally backmounted, but optional fascias, dimensions being identical to the *1500 Series*. Only available in linear unbal and unbal log fader audio taper, infinity cut-off switch, flying leads (no connector), 85 dB attenuation.
Digital fader: conductive plastic linear motion fader providing digital output in 8-bit non-ambiguous Gray code, decimal 0 to decimal 255, 0.4 mm resolution providing an electrical resolution of 1/6 dB per step at 0 dB when the fader output is used to generate the digital equivalent of an analogue audio taper law. Physically interchangeable with the *1100* and *1500 Series* faders and accessories.
1900 Series: conductive plastic linear motion fader, available in mono, stereo or four gangs, basically similar to *1500 Series*, but a longer 128 mm travel. Available in linear or unbal log fader audio taper. Fascia calibrated 0 to 85 dB and infinity.
Dimensions: fascia top plate 190 × 40 mm, backmounted 173 × 46 mm, 46 mm high.
2100 Series: conductive plastic linear motion fader, similar to *1900 Series*, but includes two separately operated faders in one housing with an overall width of 15.4 mm, designed for backmounting. Both faders are stereo tracks, infinity and aux switches.
900 Slimline Series: slimline conductive plastic linear motion faders, similar to *1100 Series*, but shorter 65 mm travel and 12.7 mm width. Available as mono or stereo, linear or audio.
1000 Series: low cost conductive plastic fader, with compact design, similar tracks and feelers to *1100 Series*, but simplified mechanical construction. 90 mm linear stroke, available in single or two channel types, a switch contact may be fitted in single output units, 10 kΩ tracks, linear or audio taper log types, VCA laws to order, designed for backmounting, seven colour knobs.
Joystick quad pan pot: quadraphonic pan potentiometer using joystick to divide a single audio input to four outputs. Uses conductive plastic tracks with a special law to provide a balanced sound effect with sensitive adjustment in the central position. Cut-off in corners 40 dB, six tracks internally, 600 Ω input, 3 kΩ output.

PREH (West Germany)

Jacob Preh Nachf. GmbH & Co, Po Box 1540, D-8740 Bad Neustadt. Phone: 09771 921. Telex: 672503.
UK: Eardley Electronics Ltd, Eardley House, 182/4 Campden Hill Road, London W8 7AS. Phone: 01-221 0606. Telex: 299574.

Film slider resistor 58: carbon film linear motion fader, mono or stereo, available for PC mounting or with wire terminals, options screened, ½-taps, detents.
Resistances: linear 47 Ω to 16 MΩ, log 220 Ω to 10 MΩ.
Effective travel: 58 mm.
Knobs: plastic.
Dimensions: 86 × 10 × 16 mm high.
Film Slider resistor 70: carbon film linear motion fader mono only. Similar to *58*, but 70 mm effective travel, red knob.
Film Slider resistor 100: carbon film linear motion fader, mono or stereo. Metal housing.
Resistances: as for *58*.
Effective travel: 100 mm.
Knobs: red.
Dimensions: 145 × 30 × 12 mm high.

SPHERE (USA)

Sphere Electronics Inc, 20201 A Prairie Avenue, Chatsworth, California 91311. Phone: (213) 349-4747.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.

Travis Fader: digital encoding device with no moving mechanical parts. Level changes are made by finger tip being placed in the fader trough which is an infra-red field. A vertical row of LEDs indicates level and is analogous to fader knob position. Level changes can be made by finger movement anywhere in the field. Slow movement can give resolution as fine as 0.375 dB. Each fader has two preset level memories and a solo/mute switch.
Digital Attenuator: resistive ladder CMOS switching device. The audio remains analogue and the attenuation is essentially passive. Input is 8-bit word from the fader or automation system. Attenuation in 224 discrete steps.
Frequency response: – 0.5 dB at 50 kHz, – 0.2 dB at 20 kHz, – 0.2 dB at 30 kHz.
Distortion: 0.001% at 1 kHz THD, 0.002% IM.
Attenuation; 68.2 dB max and off below 100 dB.

VALLEY PEOPLE (USA).

Valley People Inc, PO Box 40306, 2821 Erica Place, Nashville, Tennessee 37204. Phone: (615) 383-4737.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

EGC-101: electronic gain control cell for OEM applications. Requires a minimum of external circuitry for precision VCA applications.
Gain control range: – 100 dB to + 50 dB.
Noise: (with *5534* op-amp) 0 dB gain – 88 dBV, + 20 dB gain – 80 dBV, – 20 dB gain – 99 dBV.
Distortion: any control voltage within normal operating conditions, SMPTE, IMD or 1 kHz THD, 0 dBV output level 0.0025%, – 20 dBV output 0.0025%, + 20 dBV 0.009%.
Package: 8-pin miniature which plugs into standard 18-pin DIP socket.
EGC-202: voltage controlled amplifier using the *EGC-101* electronic gain control element designed to replace the dbx *202* VCA, physically identical, minor electrical change required to interface circuitry.
Gain control range: – 100 dB to + 50 dB.
Noise: W/R in = R out = 12.5 K —87 dBV, 20 Hz - 20 kHz.
Distortion: THD 0.005%, IMD 0.015%.
Modulation noise: increases 3 dB with application of 1 mA signal.
EGC-205M: voltage controlled amplifier using *EGC-101* electronic gain control element, direct physical and electrical replacement for the Allison *VCA-5M*.
Gain control range: + 50 dB to – 125 dB with 13 kΩ input resistor.
Noise: output noise at max attenuation – 110 dBV, unity gain – 87 dBV, + 20 dB gain – 78 dBV, 20 Hz – 20 kHz.
Modulation noise: – 100 dB.
Distortion: 1 kHz THD 0.008%, 10 kHz 0.015%, SMPTE IMD 0.015%.
EGC-2500: voltage controlled amplifier which is a direct physical and electrical replacement for the factory VCA module installed in MCI *500 Series* consoles.
Noise: output noise – 102 dBV normal fader setting, – 110 dBV full attenuation, 20 Hz – 20 kHz.
Distortion: 1 kHz THD 0.005% SMPTE IMD 0.015%.
FADEX: programmable fader system designed to be fitted in new or existing consoles, which forms a direct interface to the Allison *65K* automation programmer. Available in mechanical form to replace conductive plastic faders in USA and European formats. In addition to programmable level, *FADEX* offers up to VCA subgroups, channel mutes and solos, group mutes and solos, VCA grand master, precision nulling indicators, mute sensing LED and relay drive, 145 dB gain control range. Employs Penny & Giles linear conductive plastic faders and Allison *EGC-205M* VCA. The *FADEX* system includes a modules for each channel, one VCA grand master and stereo master, one master logic card, power supply, console interconnection package, and one *65K* programmer.

WATERS (USA)

Waters Manufacturing Inc, Longfellow Center, Wayland, Massachusetts 01778. Phone: (617) 358-2777.
UK: Variohm Components, The Barn, Wood Burcote, Towcester, Northants NN12 7JR. Phone: 0327 51004. Telex: 311754.

LM8 series: conductive plastic linear motion faders, available in mono, stereo or quad, metal construction, edge connector.
Resistances: *600* Ω, linear, modified audio or log tapers.
Maximum attenuation: 90 dB.
Effective travel: about 125 mm.
Switching: cue switch with detent.
Knobs: round.
Dimensions: 179 × 32 × 59 mm.
LM4/6: conductive plastic linear motion faders, available in mono, stereo or quad, metal construction, terminal connectors.
Resistances: as for *LM8*.
Effective travel: *LM4* 70 mm, *LM6* 104 mm.
Knobs: round.
Switching: cue switch and detent.
Dimensions: *LM4* 108 mm long, *LM6* 151 mm long, 29 mm wide, 13 mm high.
MN4/6: Similar to *LM416* but reduced width, 19 mm, mono only, no switches. ■

product guide — Autopanners

This short guide includes units that can be used to perform the functions of units in the faders, VCAs and panpots product guide while not belonging in that category. Their programmable and pan/fade effects capability also makes them quite capable of being looked at as effects units in their own right ans so they are listed under their own section for reference puposes.

AUDIO & DESIGN (UK)

Audio & Design (Recording) Ltd, North Street, Reading, Barkshire RG1 4DA. Phone: 0734 53411. Telex: 848722.
USA: Audio & Design Recording Inc, PO Box 786, Bremerton, Washington 98310. Phone: (206) 275 5009. Telex: 152426.

Panscan: automated pan effects unit with stereo capability. A mono signal will be moved in a stereo field and a stereo signal will cross-pan and reverse the image. An internal clock can control the rate of pan-action from one to ten beats to the bar (or pan). This rate can also be externally controlled or used to feed external equipment. Additional controls include adjustment of the pan width with the image position being shown on an LED bargraph. It is possible to pan the image round any selected 'centre' in the stereo field not just the mid-way point. The rate of pan can be controlled with adjustment variable up to a stereo vibrato effect. The image may also be frozen at any point in a pan action by pressing the freeze button.
Frequency response: + 0 dB, – 1 dB 20 Hz to 25 kHz referenced 1 kHz.
Noise: – 80 dBm 20 Hz to 25 kHz.
Distortion: 0.1% at + 15 dBm.
Panning depth: adjustable from 0 dB to – 35 dB with respect to the other channel.
Dimensions: 1¾ ins × 19 ins × 7½ ins, 5½ lbs.

REBIS (UK)

Rebis Audio Ltd, Kinver Street, Stourbridge, West Midlands, DY8 5AB. Phone: 0384 71865. Telex: 335494.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, NY 11735. Phone: (516) 249-3660.

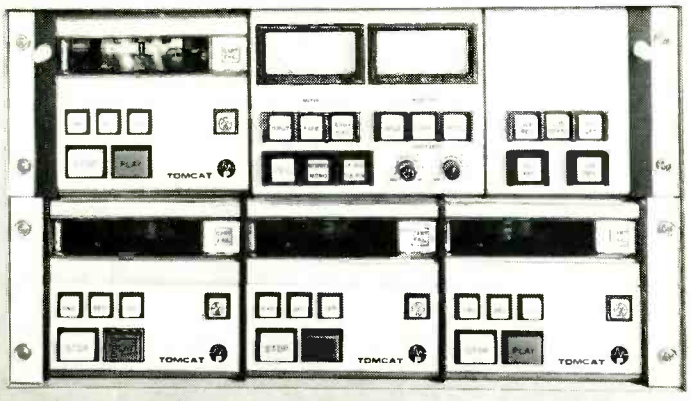
RA215 Programmable Fade/Pan Controller: modulator to suit *RA216*, programmable for fade out, fade in, level change, and wide range of autopan effects. Trigger buttons for one-shot, free run, random hold or programmed hold modes, pan range and flyback switch. Hold selectors programme image hold at any combination of ¼, ½, ¾ and full cycle positions to define one-shot pattern or final status after free run. Full LED indication of signal status. All functions can be externally triggered.
Output: 0 – 15 V dc.
Rate: 30 secs per sweep to 10 cycles per sec.
RA216 Dual VCA: two VCA's in *RA200* Series 1 inch module format. Front panel controls for range (0 – 100% attenuation), and slope (audio taper continuously variable from lin to log). Pan/fade switch inverts voltage to second VCA.
Distortion: less than 0.02% THD at 1 kHz + 10 dBm.
Noise: less than – 85 dBm.
Max. output: + 20 dBm into 600 ohms.
Attenuation: up to 90 dB.
Control voltage: 0 to 15 V dc.
Stereo tracking: ± 1 dB.

SURVIVAL PROJECTS (UK)

UK: Scenic Sounds Equipment Ltd, 97-99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Autopanner: quad/stereo automatic panning unit which will slave or be slaved by other effects units to produce simultaneous panning and flanging, phasing or pitch shifting. Will also duck and vibrato. Standard rack mounting unit. ■

TOMCATS PUT THE ART IN CART



A cartridge system which matches and often exceeds the performance of reel to reel recorders, the TOMCAT has been developed using computer-assisted design to the very highest standards.

MAXTRAX™ wide-track tape heads and precision stainless steel tape guides are fixed to give the lowest possible phase error. A fast-start DC servo-capstan motor, microprocessor control system and true mono/stereo compatibility make the compact TOMCAT system a leader among cartridge machines.

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Leevers-Rich Equipment Limited
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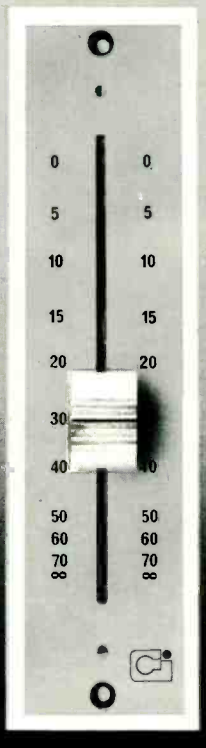

The Citec 100SF Professional Studio Fader

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The Professionals require . . .

Uniform smooth movement
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Essentially infinite resolution
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- Comprehensive servicing and support. Fully modular power supply, audio and transport electronics card frame. A highly dedicated distributor service network with factory trained and supported technicians.
- Available in full-track 1/4", 2-track 1/4" and 4-track 1/2". Comes with optional DIN-head and overbridge versions.

OTARI

Otari Electric Co., Ltd.
4-29-18 Minami-Ogikubo, Suginami-ku Tokyo 167
Phone: (03) 333-9631, Telex: J26604

business

BARRY FOX

Is 'Fishwick' part of red herring plot?

Finally, just a couple of days before the closing date for submissions on the Government Green Paper on copyright law reform, the Tape Manufacturers' Group crawled out from under a stone and talked to a few journalists about its own, singularly unimpressive, submission. This followed some talk by the BPI about its submission, which claimed the curiously exact figure of £304.9 million for record sales lost to home taping last year.

The Tape Manufacturers' Group is supported by BASF, 3M, Maxell, Memorex, Sony and TDK, who all pay £10,000 each, each year, to fight the record industry's claim for a tax or levy on blank tape. So what has the tape industry got for its £120,000? Well, frankly, precious little. The TMG submission simply says it would be superfluous to argue against the record industry's figures. The TMG submission doesn't even query what kind of market research the record industry used to arrive at a loss of exactly £304.9 million. The TMG submission doesn't even take up the interesting point made by a TMG spokesman, in discussion, namely that tape sales don't match the alleged increase in home taping. This may be true, but why not submit a graph or table of tape sales against alleged home taping losses?

The TMG admits that it may have become complacent. Too right! Over a year has passed without any press release from the TMG and all they've done during that time is chat to a few MPs and 'policy makers'. The House of Lords debate, which gave guarded approval to the idea of a levy, produced no TMG response. Naïve articles in the popular press, which rolled home taping and commercial piracy, both audio and video, all into one ragbag of criminality, also produced no response from the TMG. They haven't talked to any journalists to give them the other side of the picture. And they haven't written any letters for publication. Even a curious letter that appeared in the *Daily Mail* from a 'Mr Eric Fishwick of Wigan' produced no TMG response. 'Mr Fishwick's' letter was riddled with factual errors, like reference to the amateur recording licence as being ". . . withdrawn due to lack of public support". It wasn't. It was withdrawn because the BPI withdrew its support for the licence in 1980 when the BPI embarked on its expensive blitz campaign for a tax on blank tape. 'Eric Fishwick', who didn't declare any connection with the record industry, ended his letter with a gratuitous suggestion: "A levy could be placed on the sale of blank cassettes and recording equipment and distributed to record companies and song writers . . .". If I'd been the TMG I'd have spent some of that £120,000 finding out exactly why 'Mr Fishwick' wrote his letter and where he got his facts from. I'd also have spent a little more of that £120,000 on writing a follow-up letter to the *Daily Mail* for publication. *[Not to mention finding out if 'Fishwick' really exists. He is not in the 'phone book—Ed.]*

. . . the Swedes put it into practice

Hard news from Sweden. From September 1, blank tape has been taxed. It is easy to see why the Swedish Government went along with the industry's lobby for a levy. Two-thirds of the tax collected on tape is going straight to the Swedish Government treasury, for social security, defence, roads and whatever else they want to spend it on. The remaining third is going to be split between composers, performers and cultural bodies like the Swedish film industry. The tax on audio tape is one fifth of a British penny per minute. This is on the factory or import landed price of the tape, so by the time it's got to the shops, and had VAT added, the price is much higher. But it's still relatively low. The Swedes went for a low audio tax because they knew the manufacturers could and would avoid a high tax by recording non-copyright music at the beginning of every blank tape, thereby making it no longer blank.

The video tax is much higher, 2½ British pennies per minute on the factory or landed price. This means that the retail cost of video tape in the shops, including VAT, is now almost double the true cost. At current tape sales in Sweden the new tax will produce around £12 or £13 million for the Government, and £4 million for the people who lobbied for it. They've even levied the tax on blank tape sold for duplicating! In Britain there are now around 55 million blank audio cassettes sold every year and around 15 or 16 million blank video cassettes. If the British Government taxes that little lot, and picks up the Swedish idea of collaring two thirds for the treasury, just think of all the extra money it will have to spend on Trident.

Last year, when the record industry held its conference to plug the idea of a levy on blank tape, one of the speakers was John Butcher, MP, a self-confessed 'rapidly ageing ex-rock and roller'. Needless to say he spoke in favour of the levy. Recently the ambitious Mr Butcher got himself a junior ministerial job in the Government's Department of Industry and the record industry was highly pleased, seeing him as a useful ally in a high place.

Maybe, maybe not. Recently I met Mr Butcher in a lift. "Interesting about the Swedish tape levy," I said by way of chat. Butcher obviously hadn't heard about the Swedish scheme for a small levy on audio tape and a larger levy on video tape. "They haven't gone and done it, have they?" he asked. So I started to tell him how the Swedes had tackled the problem he'd been talking about at the industry conference. But I didn't get very far. The lift reached Butcher's floor and off he went, apparently without a spark of interest kindled by the news that Sweden had decided to practice what he had recently been preaching for the record industry.

Compact Disc delay

In mid-August Polygram opened its *Compact Disc* digital audio record pressing plant in Hanover. It was a proud moment. The local mayor was there and so was a not-very-representative cross section of German and Dutch press. As one German hi-fi journalist put it, it was as if the invitations had been sent out by someone who doesn't read the hi-fi press, and was using an address list that was years out of date. Some Polygram people wanted to invite journalists from France and Britain, and more from Holland. So did Philips because when *Compact Disc* is launched it will be first sold in Holland. France and Britain as well as Germany. But here beginneth the intrigue.

The ceremony to open the Hanover plant was also taken as the opportunity to announce a delay of the *Compact Disc* launch. Instead of going on sale in Europe before the end of 1982, it is now due to be launched in March 1983. There are all manner of explanations for this delay. Take your pick from the following. The Japanese launch, in October 1982, relies on a supply of discs from Polygram in Hanover as well as from CBS Sony in Japan. So there wouldn't be enough discs left over for a European launch soon after. But *Compact Disc* players being shown in August at hi-fi exhibitions, and submitted for review, were nothing more than pre-production prototypes. But the Japanese manufacturers said that they certainly would be ready to produce *Compact Disc* players in bulk, well before Christmas. Hitachi even took advertisements in the European press puffing the world's first commercially available *Compact Disc* player. "We are only waiting now for the software from Polygram," a Hitachi spokesman told me at the Düsseldorf hi-fi show.

"It's even more complicated than that," said the others. It seemed Philips weren't ready to produce players so they told Polygram (which is half owned by Philips and half owned by Siemens) to create some artificial delays. And so it went on, with every new story undermining confidence in Polygram's Hanover plant.

Unfortunately Polygram then made a disastrous mistake. First a decision was taken not to invite the French press. Then Polygram UK vetoed any invitations to the British press. This was despite exhortations from Polygram's international headquarters, and from Philips in Britain, that a contingent of British press should be invited to the Hanover opening. That way they could see for themselves that the plant really did exist and really was turning out *Compact Discs*. But 'wiser' factions inside Polygram UK prevailed. They were worried that puffing *Compact Disc* would ruin Christmas sales of ordinary records.

To confuse the issue still further, Polygram in Hanover then over-rode Polygram UK and insisted that *The Gramophone*, Britain's oldest established monthly audio interest magazine, be invited. And one other British journalist, who was on the Continent at the right place at the right time, also rowed himself in. But still Polygram UK vetoed the idea of even telling the rest of the British press about the Hanover conference in advance. Although English language data sheets were given out at Hanover, it wasn't until a week later that Polygram UK reluctantly agreed to release one to me.

I'll bet they now wish they hadn't. It contained a fascinating clanger. Amongst the *Compact Disc* licensees Thorn was listed bold as brass. Thorn-EMI, of course, is the one company which has so far been holding out against *Compact Disc*, because of the 3 cents royalty payable on every record. Sorry, it's a mistake to list Thorn, said Polygram. If you can make a mistake on something as basic as that, says I, what price everything else in the press release?

Frankly the whole business smells fishy. I personally don't believe that the Hanover plant is anywhere near ready to produce *Compact Discs* in bulk. Why am I so sure? Well the journalists who saw the Hanover presses rolling were duly impressed by the large number of discs being produced. But if they'd actually had an opportunity to look at those discs closely they would have seen that they carried the small print disclaimer: "This is an exhibition sample which cannot be played". The only discs coming out of Hanover that *can* be played are individually checked discs. And these cost $100 each. This is the price that the hardware companies are having to pay Polygram in Hanover for playable demo discs! ■

Soundcraft automation

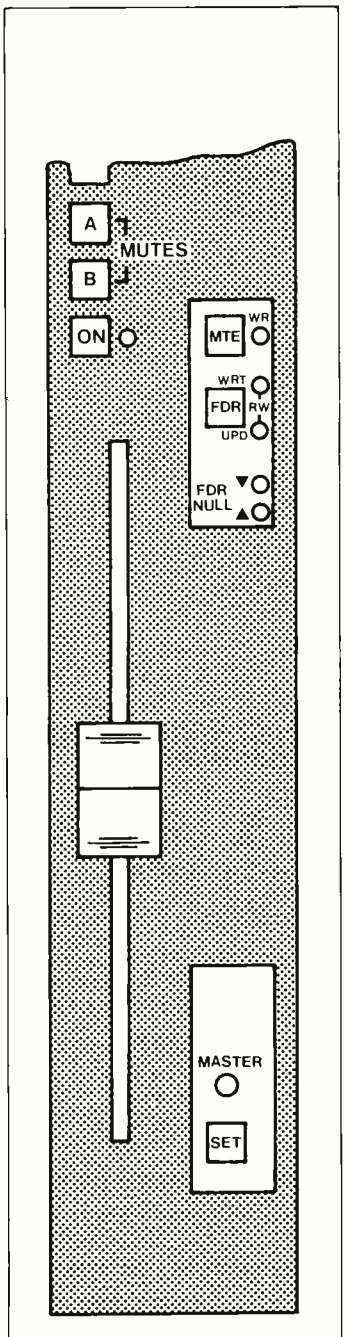

FIG 1.
LOCAL CONTROLS

TOWARDS the end of 1979 Soundcraft started to design an automation system for their *3B* mixing console. In order to compress the time scale of the project and at the same time use a standard format on tape, they limited the design objective to that of making the Allison *65K* system more intelligent. This additional intelligence was obtained by using a very simple Motorola *6802*-based microcomputer of their own design to serve each fader and mute module, each computer talking to the Allison system along the *65K* digital I/O buss. The resultant benefits over the standard *65K* system were:

- independent fader and mute information;
- ability to change automation mode from READ to UPDATE, UPDATE to REWRITE, READ to REWRITE without having to care about the fader position. This is called 'auto-nulling' and, with the exception of REWRITE which will be described later, is a feature of the MCI/Soundworkshop system.

This system was exhibited at the 1980 AES Convention in Hamburg but it was never put into production for two reasons:

- cost – major factors were the price of the Allison parts, the multiple microcomputers and the cabling (each module connecting to its own microcomputer via ribbon cable);
- technology – having wetted their feet in the microprocessor puddle Soundcraft felt that one processor per module was gross overkill and they could easily duplicate, in a few lines of software, the Allison priority system hardware. A 32-channel system would have retailed at about £12,000 which is not so bad if your console costs £50,000, but looks a bit crazy if it is less than £20,000 which is the price range of our *3B* and *2400* consoles.

Alternative systems

Before launching into the 'guess what happened next' section, a brief explanation of the principal system approaches to console automation would be helpful.

Tape based – Here the data resulting from console activity is recorded on to tracks of the multitrack synchronous with the music. Normally two tracks are used, one being what happened last time through the song, another being last time modified by new activity to improve the last attempt. There are two approaches to compiling the data to be recorded – sequential scanning and priority encoding.

Sequential scanning – The best known here is the MCI/Soundworkshop system in which every fader and mute is scanned in turn and the data recorded on to tape whether any movement has occurred since the last scan or not. Depending on the size of console, each scan will take up to 100 ms, therefore average initial response time to record an action will be 50 ms. For each successive pass through the system (each time you have another go at the mix) an additional 1.2 ms of bounce delay will be added to the initial response time. The problem here is that if a drop-out occurs during replay of a scan, the rest of the scan has to be aborted since the computer momentarily loses track and waits to find the sync word signifying the start of a new scan. Even this is not a disaster as far as fader movements go, but a mute which was already a bit late in response is now unacceptably late for no apparent reason.

Priority encoding – The best known here is the Allison *65K* system in which every fader (the Allison *Fadex* module sees mute as fader down) is looked at very quickly and, if it has moved, is recorded on to tape immediately as address plus data. In order to be able to recover the full console status rapidly if the tape is stopped and run back or forward to a point not right at the beginning, the system forces a routine sequential scan to be recorded in the ratio of three priorities to one sequential. When no priorities are found, the system executes a sequential scan, recording all fader positions. Initial response time is on average 6 ms and bounce delay about 8 ms. The Allison system is a very fine piece of hardware design and if used in its digital I/O form, capable of handling a large console and many switches fairly quickly. The problems are twofold. Firstly an 8 ms bounce delay accumulates fairly quickly to a point where a mute, precisely placed, is noticeably not where it was. Secondly, using the *Fadex* module system where mutes are written as a fader at maximum attentuation, muting any more than one channel at a time creates horrendous delays between the mute of the lowest number channel to the mute of the highest number. Drop-out recovery is rapid in that you only lose data for slightly longer than the length of drop-out.

Disk based – All disk based systems are essentially priority encoding. Usually SMPTE timecode is recorded onto one track of the multitrack recorder, the disk operating system reading this code and, if any priorities occured during a certain frame or half-frame of code (40 or 20 ms), recording on disk the time and the priority data that occurred at that time. These events are locked together for evermore, the resolution of the system being determined by the decision to use half or full frame coding, and by any interpolation algorithms used to smooth out ragged fader steps at 20 or 40 ms intervals. There is no need to record any routine sequential data since the operating system always knows where the tape is positioned and will therefore scan the disk from start point to present tape position to restore the correct console status. Fundamentally, all disk systems are the same. They differ greatly however, in the way in which communication is established between engineer and system. Good examples of such systems with widely differing user interfaces are SSL, *NECAM* and Melkuist, the latter being of particular interest since it can be fitted into many console designs. Another common feature of disk based systems is that none of them are cheap. They range in price, from an average studio owner's viewpoint, from barely affordable to pure fantasy. It is not unreasonable that this should be the case since software design tends to be expressed in man years and must be amortised over the expected sales of the product.

It should be stressed that any apparently derogatory comments made on the tape based systems used as examples should not be viewed as competitor bashing. Afficionados of this sport will recall the heated correspondence in *Re/p* in 1978 on the relative merits of sequential scanning versus priority encoding.

a design study—part one

Graham Blyth (Soundcraft)

This correspondence is well worth reading as it goes into far greater detail than there is space for here. These systems were designed some time ago and have given considerable satisfaction to many studios at a reasonable price. Their drawbacks are mentioned here purely to illustrate why Soundcraft felt that a more modern system was required and the design parameters that it should fulfil.

Design parameters

The broad requirement was for a low cost, high speed, tape based automation system with considerable expansion possibilities for additional functions, the initial functions being fader and mute plus VCA sub-grouping. An additional requirement was that the system could be converted to be disk based with negligible changes to the system hardware. The first console chosen to be automated was the Series *2400*. More detailed parameters were:

Method – which had to be priority encoding because upgrade to a disk based system requires priority data and with careful design the method is considerably faster than sequential scanning.

Speed – since it was necessary to be appreciably faster than previous designs, particularly as far as the mute function was concerned. The *2400* is a very mute-orientated console having effectively three group muting busses. It is therefore very easy to mute many channels simultaneously, thus creating instantaneously a large number of priorities.

Expandability – Soundcraft were most anxious to avoid re-designing the system every time a more advanced console was designed. This system had to be capable of supporting a totally automated console, most of the re-configuring to be done with software. A microprocessor based design was obvious. They opted to use the Motorola *68B09*, a very fast 8-bit processor with 16-bit internal architecture.

Cost – It was important that the price of the system should have a sensible relationship to the price of the *2400* console, currently just under £18,000 in maximum configuration. The target was between £6,000 and £8,000 for a 28-channel system and the approach was to give most attention to the cost of the individual channel electronics. This problem was greatly aided by the availability of the dbx *2151* VCA chip, requiring negligible external circuitry, occupying little PCB space and giving excellent performance.

Retrofit – When the *2400* was designed two years ago, future automation was borne in mind and the necessary panel space allocated. Since over 90 of these consoles will have been installed by the time of publication, ease and economy of retrofit was a prime consideration. One of the classic problems is the need to substitute VCA taper faders for conventional log taper units, causing aggravation between client and supplier as to how much is paid to replace the old faders. Soundcraft decided to use the same log taper fader, do the necessary translation in software and at the same time offer the ability to bypass the computer system either locally or totally by putting the audio through the fader in the usual way. In fact the system will translate any audio taper fader into the VCA dB/volt domain.

VCA Sub-grouping – The *2400* has no space anywhere for separate VCA sub-master and grand master faders. It was therefore necessary to let any channel fader become a master of itself and any number of others. This assignment had to be very simple and clearly visible and 'seen' by the computer. This system will be described in detail later but briefly any number of VCA sub-groups may be formed and any one un-grouped fader may become the grand master, though this need not be assigned.

Ergonomics – Any modern sophisticated multitrack recording console is covered in knobs and buttons; the Soundcraft *2400* is no exception. It was important to reduce to an absolute minimum any additional controls for the automation system.

Controls

Local (see **Fig 1**) – It was decided to keep the mute control independent from the fader so one momentary switch toggles between MUTE READ and MUTE WRITE, WRITE mode being indicated by a red LED. The fader however, has four modes – READ, WRITE, UPDATE and REWRITE. By logically rationalising the process that an engineer will go through in performing an automated mixdown, Soundcraft were able to implement and display all these modes by a single momentary switch and a red and green LED. Without further explanation at this stage, this seems like *reductio ad absurdam* but in practice the process is very clear and simple. Two additional LEDs indicate whether the fader is above, below, or at the same position as the data being received from the tape. Due to the almost total transparency of the system these LEDs are largely irrelevant but they do have one useful application which will be explained later. The VCA sub-group system is handled by one momentary switch and a large red LED.

Master (see **Fig 2**) – The five master mode switches force all channels to jump into the mode selected, MUTE READ or WRITE, FADER READ, WRITE or UPDATE. SOL SFE (solo safe), action indicated by an adjacent red LED, prevents the computer from responding to mutes caused by operating channel solo switches when the solo system is in 'Solo In Place' mode ie, all channels except those soloed are muted (unless locally switched to Safe). This allows 'wet' solo of any channel(s) during the execution of an automated mix. If however, the Solo In Place system is used as a third mute group, SOL SFE is not pushed. SET/CLR is used with the VCA sub-grouping system. TRK switches the data from and to the console between two tracks (say 1 and 24). A red LED indicates the track on which data is being recorded, data is therefore being received from the other track. The engineer still has to put the correct track into record, a certain amount of concentration being required at this point. This problem is removed with a disk based system, in which the computer is able to store data attached to time code without the need to press any buttons.

The second part of this article will explain how the system operates and how all the modes function. ■

To be continued

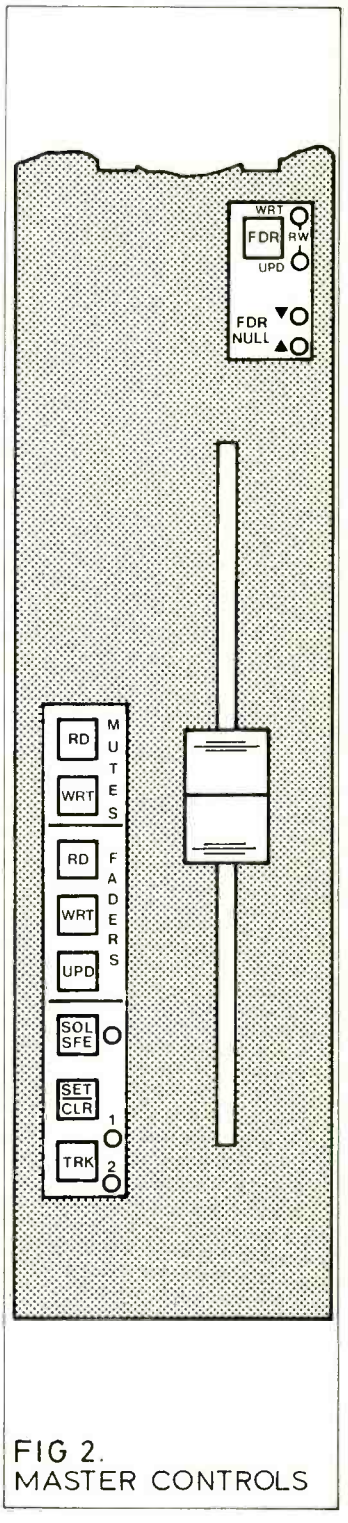

FIG 2.
MASTER CONTROLS

Sony PCM-F1 System

MANUFACTURER'S SPECIFICATIONS

Signal system: Conforms to CCIR television standard, PAL/SECAM colour; NTSC version also available.
Code format: Conforms to the technical specifications of the EIAJ (standard format using 14-bit quantisation), or 16-bit quantisation format
Number of audio channels: 2 channels
Sampling frequency: 44.1 kHz
Quantisation: 14-bit linear quantising, or 16-bit linear quantising
Frequency response: 10 – 20,000 Hz ±0.5 dB
Harmonic distortion: Less than 0.007% (14-bit format). Less than 0.005% (16-bit format)
Dynamic range: More than 86 dB (14-bit format). More than 90 dB (16-bit format)
Channel separation: More than 80 dB
Wow and flutter: Below measurable limit
Error correction: Error correction and concealment using CRCC and parity
Emphasis: Pre-emphasis (in recording): fixed On. De-emphasis (in playback): automatically switched to On or Off (by detecting pre-emphasis identification code). Time-constant 50 μs, 15 μs

GENERAL
Power requirements: Operating voltage: 12 V DC.
Usable power sources: 12 V DC with the Sony *NP-1* rechargeable battery pack (optional) 220 V AC (or 240 V AC adjustable), 50 Hz (type 1) or 240 V AC (or 220 V AC adjustable, 50 Hz (type 2) with the supplied *AC-700* AC power adaptor. 12 V car battery with the Sony *DCC-2500* car battery cord (optional).
Power consumption: 17 W DC
Dimensions: PCM-F1: Approx. 215 x 80 x 305 mm (w/h/d) (8½ x 3¼ x 12⅛ inches). AC-700: Approx. 107 x 80 x 305 mm (w/h/d) (4¼ x 3¼ x 12⅛ inches) not including projecting parts and controls
Weight: PCM-F1: Approx 4 kg (8 lbs 13 oz) net AC-700: Approx. 3.2 kg (7 lbs 1 oz) net
Total weight: Approx: 8.1 kg (17 lbs 14 oz) in shipping carton, including PCM-F1 and AC-700
Accessories supplied: AC power adaptor AC-700 (1). Video connecting cord VMC-110C (1). Video connecting cords with a BNC-type plug and a phono plug (2). Audio connecting cords RK-112 (2). Shoulder strap (1)
Optional accessories: Rechargeable battery pack NP-1. Car battery cord DCC-2500. Carrying case LC-170. Carrying handle AH-330.

Inputs

	Type	Reference input level	Impedance	Minimum input level
Mic	¼ in jacks	–	Accepts low impedance microphones 40 kΩ 75 Ω	0.435 mV (– 65 dB) 95 mV (– 18 dB)
Line in	Phono	– 10 dB*		
Video in	Phono	1 Vp-p		

Outputs

	Type	Reference output level	Load impedance
Line out	Phono	– 10 dB**	More than 10 kΩ 75 Ω
Video out	Phono	1 Vp-p	75 Ω
Copy out	Phono	1 Vp-p	75 Ω
Headphones	Stereo ¼ in jack	– 24 to – 48 dB Attenuation: 5 steps (24, 18 12, 6 and 0 dB)	Accepts low impedance headphones.

* Input level when the peak programme meters deflect to —15 dB.
** Output level when the playback level is – 15 dB as shown by the peak programme meters.

THE Sony PCM system supplied for review consisted of four units, namely the PCM adaptor proper (*PCM-F1*), a matching 'Betamax' style video recorder (*SL-F1UB*) and two power units. Both the recorder and the PCM adaptor may be powered by replaceable NiCad battery packs. The *PCM-F1* is able to convert the sound into one of two formats – either the standard EIAJ 14-bit format or a compatible 16-bit format which allows increased resolution and fidelity at the expense of some of the system's ability to cope with drop-out errors.

When recording audio, the analogue signal is first 'quantised'. This involves sampling the signal at frequent intervals and measuring the voltage of the sample. The numerical value of this is then recorded, so that the samples may be re-constituted during playback. Low-pass filtering of the stepped waveform thus produced will reconstruct the original signal. The degree of fidelity obtainable by such a process will depend on the number of 'significant figures' to which the sample can be measured, and recorded, while the highest frequency that can be treated will depend on the rate at which samples can be taken. In practice these two functions are, of course, related. The almost notorious theorem by Nyquist, Shannon and no doubt many others, states that a bandwidth-limited signal can be completely characterised by a minimum of two samples, so the sampling frequency must be at least twice that of the highest frequency component in the original signal. The advantage of using such a system is that numbers are all that is recorded, and this may be done by binary methods so that the signal on the tape at any point must correspond to either a 0 or a 1. Thus a degraded signal due to imperfect recording can easily be corrected so that the original can be restored – further standard error correction

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techniques and multiple recording strategies can be used to correct or re-create data when the degradation is such as to reproduce totally erroneous numbers. The disadvantage of the system is that the data stream thus created contains frequencies into the MHz region, so that, video recording methods become obvious candidates.

While several methods can be used to enhance the resolution of a system, the *PCM-F1* conforms to EIAJ standards by using a 'linear' encoding format although high frequency pre-emphasis is used to take full advantage of the system. Sampling is performed at 44.1 kHz and recording can be to either 14 or 16-bit resolution. As the *PCM-F1* is designed to be used with standard video recorders, the data stream is converted into a 'TV picture' for recording. It will be seen from the diagrams that the 16-bit mode is achieved within the 14-bit standard by using one of the error correction words to hold the last two bits of data from each sample on that line. The diagrams also show how this is achieved, including the interleaving that is done to disperse drop-out errors. In the 14-bit mode, 'burst' errors of up to 32 TV lines can be corrected; the 16-bit mode reduces this to 16. When an error is detected, the circuitry attempts to reconstruct the damaged data word from redundancy within the data structure. Should this capability be exceeded, an average, generated from the preceding and succeeding samples, is substituted. The interleaving of data mentioned above helps to assure that these last will be error free.

Setting up

Interconnection of the units is simplicity itself. The two power units can be placed side by side forming a stand upon which the *PCM-F1* can be placed, while the *SL-F1UB* can be placed on top of the *PCM-F1*. The top of the cabinets have locating areas for the feet of the unit above to aid physical stability of the stack so formed. The manufacturers do not recommend this form of stacking as the ventilation of the power supplies is naturally restricted. The power units have captive twin-core mains leads supplied, and captive multiway leads with DIN connectors carry the DC to the *PCM-F1* and the *SL-F1UB*. An accessory lead fits into the multiway socket on the rear of the *SL-F1UB* and leads to two phono sockets labelled 'video in' and 'video out'. These must be connected to the sockets respectively marked 'video out' and 'video in' on the *PCM-F1* (surely an additional colour coding would be less confusing) which completes the interconnection of the machines, although an extra phono socket exists on the *PCM-F1* to facilitate copying. A switch to select 14 or 16-bit resolution is also placed (hidden?) on the rear panel – this is only operative in the record mode, correct playback being automatically selected through information carried in the signal. Other rear panel connections to the *PCM-F1* consist of the phono sockets for line level audio inputs and outputs; these operate at −10 dB nominal for interface to domestic audio equipment. With these connected to your audio system, recording may commence. Two mono ¼ in jack sockets are provided on the front panel for low impedance microphones, plus a ¼ in stereo headphone socket. When operating in the portable mode with the NiCad batteries, these are slotted into recesses in each machine; the accessory lead links the two units which may then be carried on each shoulder.

When recording, a 'straight through' signal is provided from the output of the *PCM-F1*, but it is interesting to note that this signal has in fact been through the complete process – turned into digital, from digital to TV format, been through the video recorder circuitry before being returned to be decoded. Thus this signal provides a good check that all is properly connected up and that the vast majority of the circuitry is functioning correctly. This carry-over of a fairly normal video recorder practice seems an excellent idea in view of the complexity of the internal circuitry of the system. Comparison of this signal with the input signal allows one to appreciate, with great accuracy, any differences that may exist.

Controls

Front panels controls on the *PCM-F1* consist of independent left and right channel level controls, a headphone attenuator switch (five 6 dB steps) and several toggle switches. These provide line/microphone input selection, a mute switch which mutes the output unless the data received is valid, and a switch labelled 'copy' which provides a second video output for another recorder. This switch apparently degrades the audio performance during the copy process but allows the user to make copies with 'absolutely no deterioration in quality'. Use of the copy switch with the resolution switch also allows the user to change between the two standards. LED meters with a 50 dB scale are provided; at the top end of these a red panel saying OVER illuminates if the signal level is too high. These meters have twin moving bars, one of which acts as peak level indicator, the other follows the signal excursions. Two push buttons allow the user to select between the peak meter holding its level (unless exceeded) for 1.7 s or holding the peaks until reset. Two further push buttons allow the user to check the battery level and check that the video recorder is set for optimum tracking. Finally, there is an on-off switch and a record mute switch, each with their own indicators. Tell tale windows under the meters indicate whether the mute is functioning, the fact that pre-emphasis is being used, and whether the tape being played is 'copy-prohibited'.

Recorder

The *SL-F1UB* is a relatively standard portable Betamax video recorder, capable of running for about an hour from the internally contained NiCad battery pack which is identical to that

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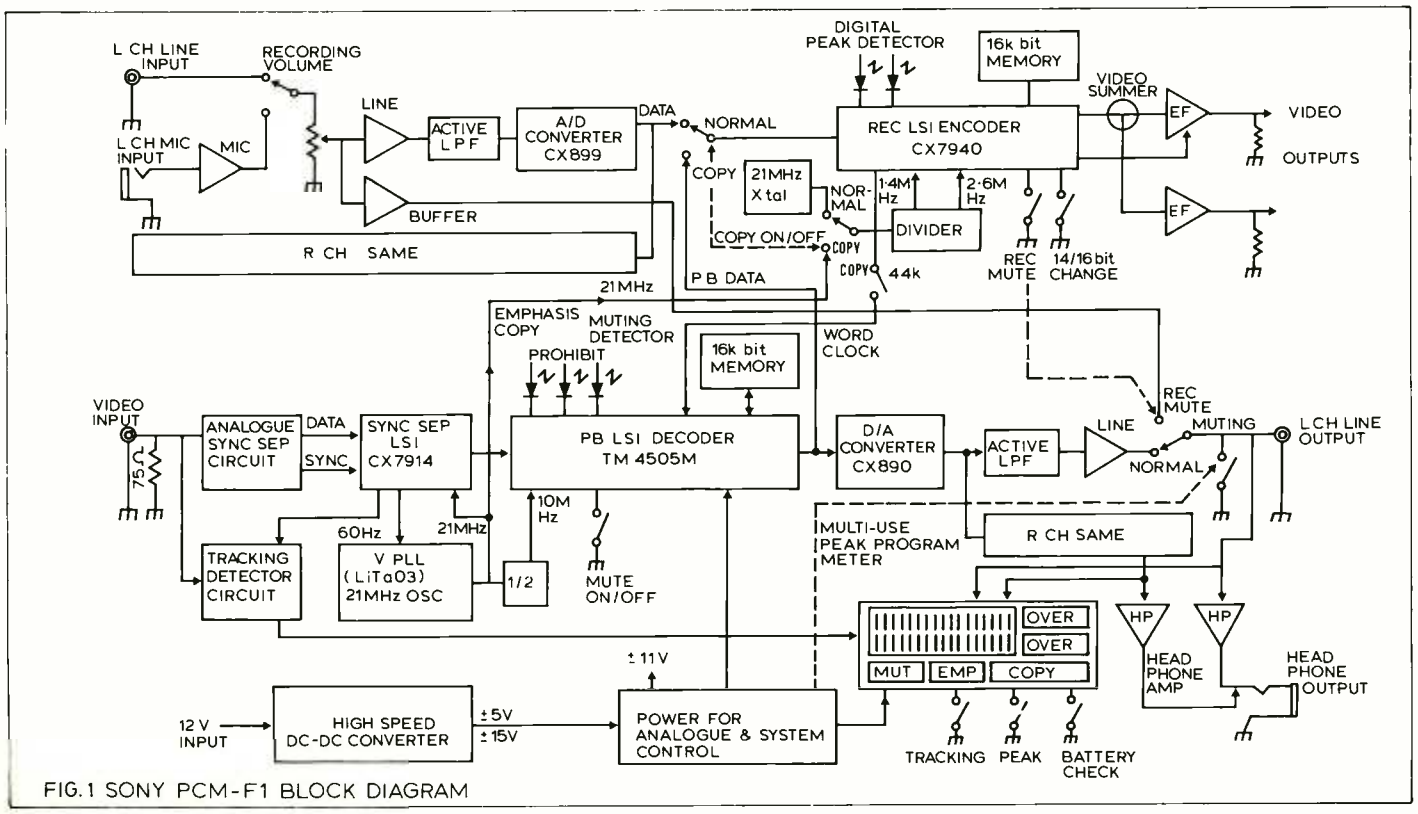

FIG.1 SONY PCM-F1 BLOCK DIAGRAM

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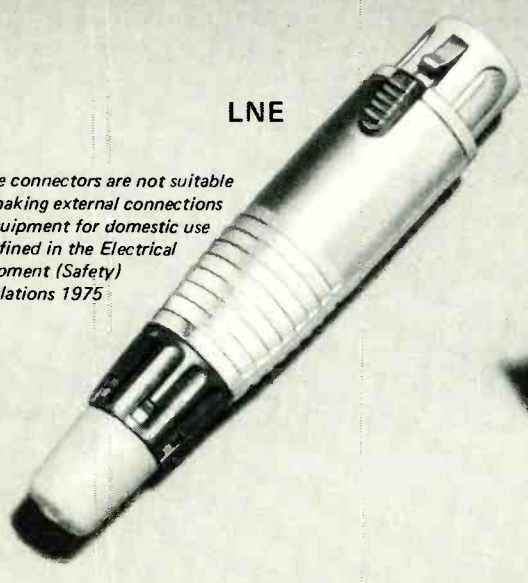

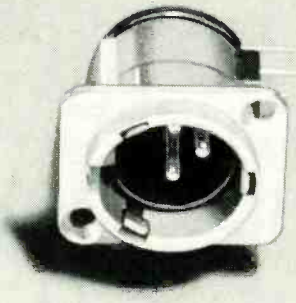

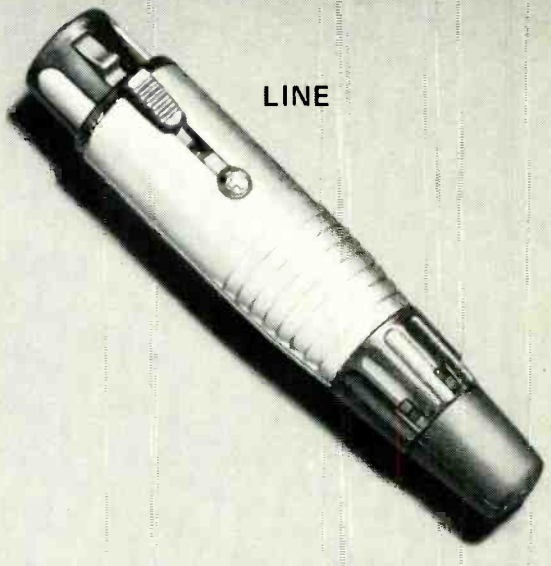

These connectors are not suitable for making external connections to equipment for domestic use as defined in the Electrical Equipment (Safety) Regulations 1975

Part of the
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range

used by the *PCM-F1*, so that these may be substituted. A camera and mike connector exist on the side panel of this machine for video recording, which does not concern us here. The accessory, power and UHF signal output sockets on the recorder are behind a plastic door covering a slightly recessed panel on the back of the machine. Although this looks quite neat, it has drawbacks as will be seen later. All other controls are on the front panel, touch buttons being used for all the tape transport functions, power on/off and eject (which is electronic – a good idea, it means that the tape won't suddenly be ejected into the middle of the road because your coat caught the button, if you have the power off!). An LCD tape counter/timer is provided – this shows through a window in a plastic door that hides a switch to provide a test signal and the tracking control. Small press buttons are provided to allow one to reset the counter, illuminate it and persuade the transport to return to zero or thereabouts. Function controls on the recorder include fast foward and reverse, stop, play, record and audio dub. Extended features include a pause control, a double speed play, and 'swing search' buttons. These allow you, once in the pause mode, to go in and out of normal speed forwards and reverse (to find your place) and two others allow you to creep in either direction. When released the recorder returns to the pause mode. The swing search is very effective in the audio modes, allowing one to find a note in the music quite easily, although one has to get used to the peculiar sound of the same frame being repeated. As with most modern video recorders, one can drop into record from pause; if this is done, the unit will actually go into record at the next frame line when the pause mode is released. This allows the PCM user a limited editing facility, probably sufficient to assemble a series of tracks, but regrettably not precise enough for 'real' editing.

A 'battery flat' warning appears in the bottom left corner of the LCD display about 30 seconds before the death of the battery. It cannot be said that this draws the eye at all, especially when used 'in the field'. The battery indicator is more conventional on the *PCM-F1*, consisting of an alternative meter scale. This is also fairly useless, although to be fair to the manufacturers this is due to the characteristics of NiCad batteries, which give no convenient indication of their status until it is too late. Another indicator on

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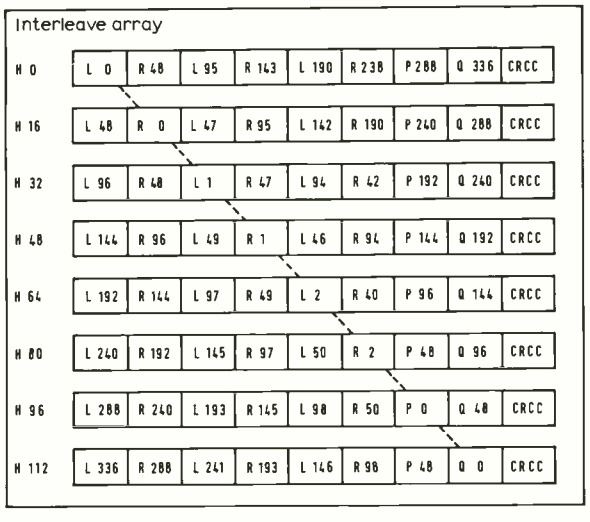
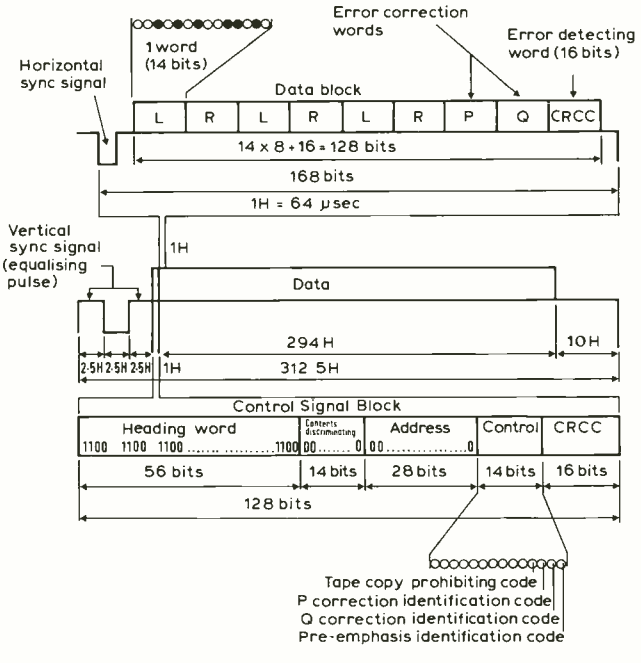
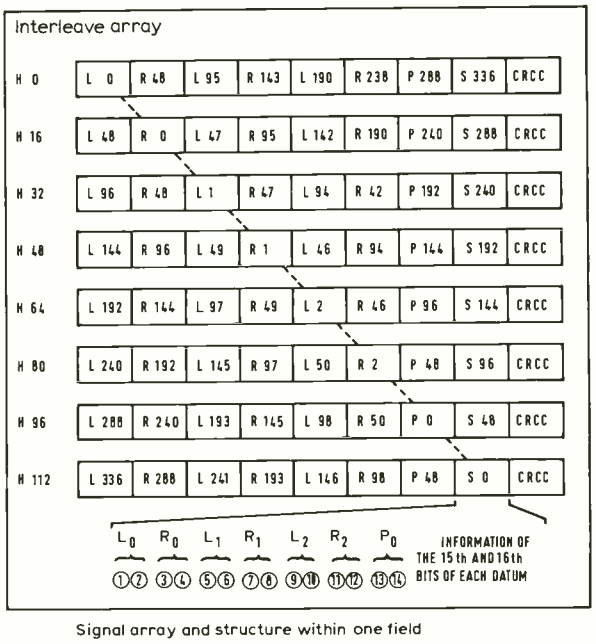
14-bit format of the EIAJ



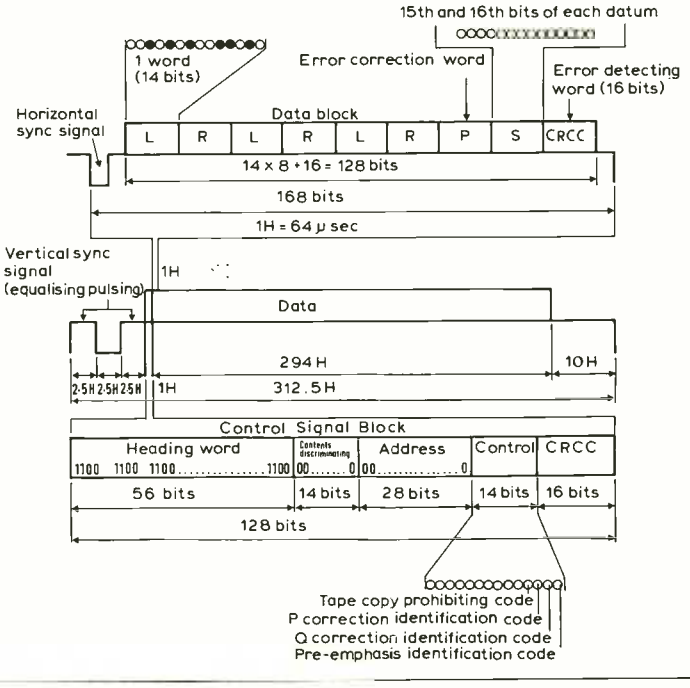
Signal array and structure within one field



16-bit format of the PCM-F1



Signal array and structure within one field



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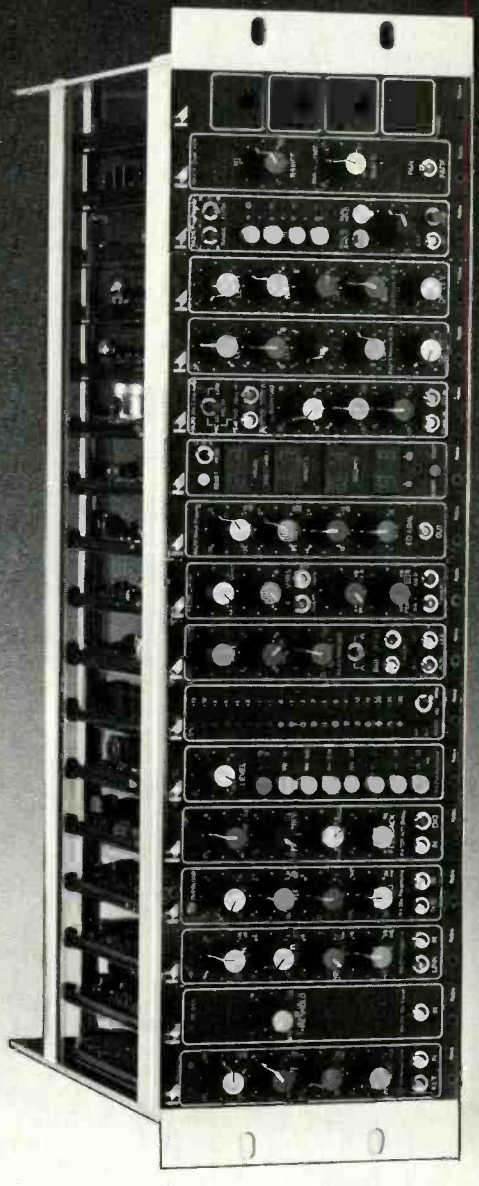

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the LCD display of the *SL-F1* shows if the internal humidity is too high – say through condensation – to allow the machine to operate properly without the tape sticking to the head drum. If this is the case, all the function keys are locked out until it is safe to operate the machine.

Operation

The units under review were subjected to various test recording situations, unfortunately limited in scope by the time and equipment available. The latter was a major factor, as the system cheerfully outperforms almost any other piece of audio equipment. Thus, it was really quite difficult to find a signal source which was not going to be far more suspect than the machine. In view of this, a certain amount of time was spent trying to discover the weak points in the system by intentionally misusing it, so as to have an indication of what should be listened for. Some outside recordings were done, using the machine in a portable mode, for both sound effects and music. In addition, a direct feed from a studio console was used during an orchestral session as a high quality source of music which had not been previously recorded on analogue tape.

It seemed that there were two major areas to investigate vis-a-vis the performance of the system. One was the performance of the actual recording and reproduction of the digitised signal, while the other involved the performance of the conversion and re-conversion of the analogue signal.

To take the recording process first. One trick that one has to get used to is the fact that, no matter what level is being recorded, this bears no relationship to what is going on on tape. Thus, whatever the level used, the tape should behave well. While this is obvious, it is surprising how much one has a gut-reaction that a low level signal must be in some way more prone to drop-out, or in some way more finnicky, than a normal level one. Obviously this cannot be true, and luckily experience confirmed this! The manufacturers recommend that a high quality, high density tape no longer than an L-500 (130 minutes) is used. For the majority of the tests such a tape was used, but the system was also tried out using a previously-used normal grade cassette. No audible difference was noticed between the two. Recordings were also made on a VHS recorder – also using 'old' tape – and no problems were found here either. In fact, the performance of the recording and error correction systems can be summarised very briefly: no audible difference between the 'line in' sound and that played back was discerned at any time during the tests, using either the *SL-F1UB* recorder provided or the home VHS machine. It is obvious that, if any drop-out or similar problems actually occurred, the various error correction systems were quite capable of handling them. It is difficult to know which part of the system is to be complemented for this, but it is most remarkable that whether the signal heard was time shifted (recorded) or not was totally irrelevant to the results – surely an extraordinary comment on a tape recorder!

When investigating the functions of the PCM conversion, the effect of over-recording was tried – this simply proved that the limiter circuits provided did nothing to improve the sound, but were sufficiently good to rescue the situation. What was not quite so clever under these circumstances was that even when the meters are set to 'peak hold' mode, the indication that the system has been overloaded is not retained. Thus, if you were to come back and examine the readings after a time you might think that the level was perfect, whereas in fact quite a lot of overloading could have taken place. Digital recording is the most incompressible of media: no overload is possible, which is why the limiter is so essential. If you do 'overload', the meters cannot register this on playback (it hasn't been recorded any louder!), so no-one will ever know! Obviously in view of the other characteristics of the system there is absolutely no point in 'pushing' the tape, and recording levels can be set 15 or so dB down to allow room for unexpected transients. Nonetheless, it was felt that the peak hold circuits should, to be really useful, tell you that an overload had actually occurred, and by how much. It is interesting to note that the overload indicators are totally tied into the signal input circuitry as, if a playback is made while an incoming signal is present, the meters follow the playback signal, but the overload lights follow the input signal.

Listening to commercially available digitally mastered recordings had revealed a certain 'glassy harshness' in the high frequency range while the treatment of reverberation was most suspicious, many recordings seeming unnaturally dry. It seemed possible that these effects could be partially due to quantisation of the signal, and to the nature of the anti-aliasing filter – which demands an extremely sharp cut off at about 20 kHz – plus, it has also been suggested, to HF cutter-head resonances. Thus investigations were carried out by comparing the sound through the system with the source when the signal was being passed through the PCM adaptor at various (low) levels. Arrangements were made so as to maintain an exact equivalence of listening level between the two signals. As the level through the *PCM-F1* was reduced, noise – up to now uncannily absent – began to make its appearance when the peak level was less than about −40 dB. Obviously, some of this noise can be attributed to the noise of the amplifiers used to make up the gain on playback. I found that, compared with, say, a cassette recorder using metal tape, adequate recordings could be made at this sort of level without any particularly disturbing effects. Downwards, ever downwards! Quantising noise became evident at much lower levels – you can begin hear what it is doing when the input level no longer registers on the PCM's meters if you are in the 14-bit mode; while if you are in 16-bit you have to turn it down even more! In the interregnum below −40 dB the noise – partly quantisation noise and partly that due to the make-up gain – becomes increasingly apparent, being substantially rougher in the 14-bit mode.

What is clear – and to me, at least, unexpected and good – is that the music is not destroyed by the noise at low level as it would be on an analogue recorder. It is still quite easy to listen to the signal which appears quite normal except that it happens to have the Niagara Falls overdubbed on it. Curiously, what reverb one can discern under these conditions still does not sound abrupt or foreshortened as I might have expected. 14-bit is obviously worse than 16-bit although this might be because I was unable to achieve the same relative levels with 16-bit (I had run out of make-up gain by then). Of course this is all a little silly – after all, you try recording anything at −45 dB on tape, just try! On this system you can still listen with pleasure, getting involved with the music, whereas on many other systems you might be lucky to decipher the fact that music was present at all. During this test, frequent use of the recorder was made to prove that the apparent 'smoothness' of the signal was not in fact due to breakthough in 'line in' mode.

I found the above test quite fascinating, especially as the 'sound' of quantising noise was quite unlike what I had previously imagined. It sounds, very boringly, just like perfectly ordinary noise and the experience convinces me that this is not the cause of 'glassiness' or 'harshness' in the top end of digitally-mastered conventional discs. This is all the more significant when one realises the full implications of the signal levels involved: any quantising noise effects that might occur would be well within other dubious areas of the playback chain, cross-over distortion for example.

Regrettably, all the outdoor recordings made were noisy due to the non-availability of quiet enough microphones. In addition, the fact that phantom power and balanced microphone inputs are not supplied on the *PCM-F1* ruled out a number of better microphones that *were* available. These ommissions are presumably due to the fact that the unit is classed as a 'high-end' consumer product. However, the major users of the *PCM-F1* will almost certainly be either professional recordists or financially well-endowed audiophiles, who will already have available – and will most certainly need to use – studio quality microphones. Even modern self-power electret types may not be sufficient. Despite these shortcomings, the results were exceptional. At no time did the fact that the recorder was being moved appear to affect its performance. Physically, the package is quite heavy to carry, and a backpack would be a wise investment for this sort of use.

The potential advantage of a long recording spell on one cassette is offset rather brutally by the fairly short battery life, although the car battery adaptor lead could help a lot (this item was not supplied for review). When used portably, straps are attached to the side of the machines so that they hang on either side of your body, front panels upwards. This means that all the connections face downwards. In addition, the plastic door covering the sockets on the rear of the *SL-F1* must remain open when leads are connected. As a result, the user cannot lift the machines off his (aching) shoulders to put them down for fear of damaging them. I suspect that the plastic door will have a severely limited lifespan too. Nor is it that easy to put one machine down, due to the relatively short interconnecting lead! I also found that if I had the *SL-F1* on my left and the PCM on my right it was extremely difficult to plug the microphones in and adjust the level; if I had them around the other way it was difficult to see the level (this doesn't actually matter much in view of the 'spare' range) or operate the transport controls. I also found it hard to achieve a balance where I did not feel that one or other of the units was going to slip off my shoulder! Still, given the slightest opportunity, I would suffer all the above, though possibly not in silence.

Some recording was also done in the professional environment of a major recording studio, during a small orchestral session. Interfacing the equipment with the mixer was quickly achieved, and no problems at all were encountered – there was not the slightest trace of RF interference in this most sensitive situation where no special precautions had been taken to avoid it, a tribute to the excellent screening incorporated in the system. The return-to-zero feature was found to be most useful in these circumstances, but I found that the relatively slow startup time of the

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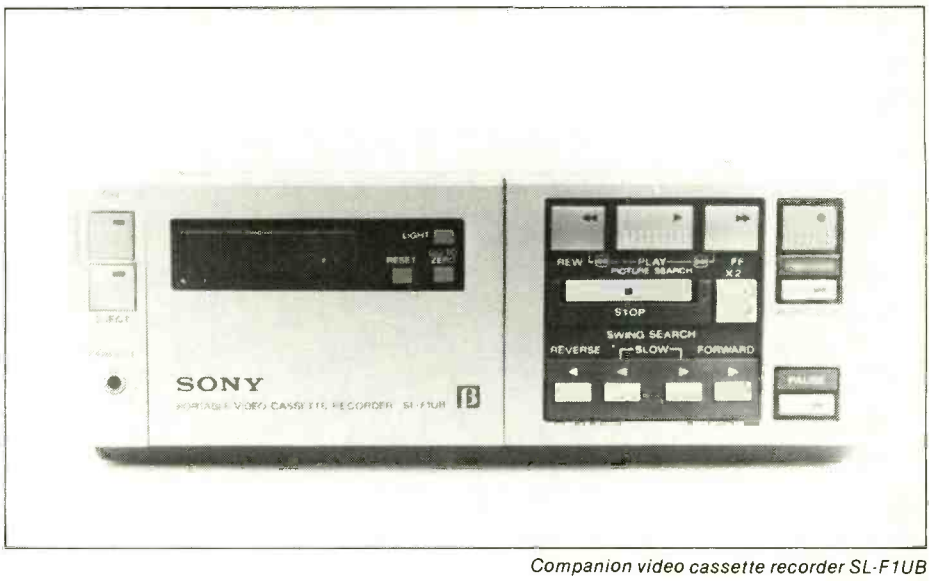
machine made it tempting to leave it in the pause mode, which cannot be good for it.

The music was modern in nature, destined to be used for a film sound track and featuring harpsichord, piano, harp, woodwind and percussion instruments. In addition, all sorts of musical effects had been written into the score, making it ideal test material. Inevitably the constraint of time made it difficult to do critical or analytical tests *in situ*, but it was clearly apparent that the *PCM-F1* was up to its usual tricks – outperforming everything in sight. A brief playback over the monitor speakers was more than sufficient to convince all present, and produce the comment, "There's nothing wrong with that! Where do I get one?" A more detailed and leisured appraisal after the session demonstrated the convincing solidity and transparency of the system, revealing detail and a presence in the music that is almost embarrassing in its intimacy. Once again it became evident that the limiting factors now lie in the source of the sound, specifically in the microphones and in their positioning, and it is felt that many lessons will have to be learned, and equipment evolved, before the system is fully exploited. It is also clear that the person who invents a new form of music paper (that doesn't rustle) and reflective plush carpet (for live rooms) will make a comfortable living.

This extraordinary quality of the PCM system, (in one test it was found that the feedthrough signal was closer to the original than the line-in signal on a certain well-respected tape recorder!) makes it extremely difficult to find anything to say about the positive aspects of the quality of the machine. A very brief listen convinces everyone who has heard it that it 'sounds as if it isn't there', which, of course is as it should be. Extended listening only confirms this impression, and should make the dedicated listener grateful that such a large area of compromised performance can be now be forgotten about – or would you really want to be listening to a mechanical gramophone, however refined? In essence one immediately looks on the *PCM-F1* as if it were some in-line bit of processing circuitry, like a line or mix amp, and judges it accordingly, and even by these criteria it is winning all the way! It is most interesting to note how such items as the sharp cut-off filters required – regarded by many as a severe limitation – in fact turn out to be quite acceptable despite their very obvious visible (on a scope) effect on square wave inputs. Actually, all listeners have commented on the openness and 'ease' of the sound, very far from being 'contained' by the filter bandwidth. It seems evident that the new LSI A/D and D/A converters whose creation have made this machine (and the new Sony *PCM 3324*) possible are an eminently successful design, and that the 'dither' circuitry employed has most successfully dealt with problems that some previous systems had with reverberant fields. I also feel that the use of pre-emphasis and the consequent de-emphasis do quite a lot to 'tame' the abruptness of the anti-aliasing filters.

Conclusions

Although the dynamic range/lack of noise will undoubtedly feature prominently in the advertising for this machine, I feel that the less obvious features are all just as important. The lack of any form of modulation noise to be 'masked' by the signal and the lack of any measurable wow and flutter, all contribute to the clarity and stability of the image produced. And last, but not least, is the flatness of the frequency and phase response which make the listener immediately recognise that the sound is as it should be. Stereo images are rock solid, inspiring immense confidence – I think that it is these properties far more than the immense (and largely unnecessary) dynamic range that will quietly make this system a winner.



Companion video cassette recorder SL-F1UB

Unfortunately, nothing in the world is perfect, and there remain some little niggles about operator convenience that have been mentioned in the body of the text. I am convinced that a machine in this price bracket should have balanced microphone inputs to a professional standard – after all these are also designed to reduce hum and noise, especially as the system is quite good at recording the (acoustic) whirring of the video recorder. In addition, it would be most convenient in view of the portability of the package if 48 V phantom powering could have been provided, allowing the user conveniently to use the very best microphones that he could afford without having to carry extra battery packs and the like. This is all the more necessary when you consider that the amount of gain available with the mike amp provided is not really sufficient when used with a good quality dynamic or ribbon microphone. I also feel that a tell-tale should have been provided to inform the user which mode he was operating in – the difference between 14 and 16-bit modes is not immediately obvious, especially in the field – and as the switch is out of sight it would be quite easy to inadvertently use the wrong one.

Apart from these points, Sony have undoubtedly produced a most useful machine for the audiophile – for whom the manufacturers admit they have aimed the system – enabling him to record and archive material with unsurpassed quality and very low tape costs. It is also possible to perform tape-to-tape 'overdubbing' with only one *PCM-F1* and two video recorders. The neatness and simplicity of the machine is excellent – perhaps it should be emphasised that on playback, no setting up other than the volume control needs to be considered as the correct decoding, pre-emphasis etc. are all done automatically. Cassette handling is pretty foolproof too, so that when one wishes to enjoy music it is far less fuss using this system than putting on a disc, let alone using some of the more exotic audio cassette machines. It is therefore a job that can safely be entrusted to those less technically minded. On top of this, I am sure that the portable aspects will prove most useful to the more energetic enthusiast, bringing him a quality that up to now would not have been available at any price.

The *PCM-F1* could also be considered as a cost-effective way of providing stereo digital audio capability in the professional field, but here a number of factors should be taken into account. First, the system is orientated around the 'domestic' (– 10 dB) line level, and although this presented no problems during the review, difficulties could possibly arise here. Second, and more importantly, the lack of true editing facilities would seem to seriously restrict its usefullness in stereo mastering applications. A digital audio editing system could be used, but if this is contemplated, it should be borne in mind that only tapes made on the NTSC version of the *PCM-F1* will interface successfully. Analogue-to-analogue (eg from *PCM-F1* to *1610*) copying would be possible although some degradation in quality may occur. It is interesting to speculate on whether the fact that this would enable you to use the *PCM-F1* in 16-bit mode rather than the 14-bit editor-compatible mode would leave you better or worse off. Additionally, in the studio it is likely that a U-Matic machine (possibly already present) would be a more appropriate recorder to use, especially as a 14-bit master tape produced in this way would be directly useable in an editing suite, and for disc mastering.

Finally, the system has obvious applications as a location sound recorder. The lack of professional mike inputs may be a disadvantage, but the fact that the VTR's audio track could well be used for timecode – or any other synchronisation method – means that it is ideal for use in the film and video fields. Indeed, I find it really hard to imagine that currently available high quality portable analogue recorders will continue to be marketable! Here again, if digital audio editing or disc mastering is envisaged, the NTSC versions of the *PCM-F1* and video recorder may well prove the best choice. And, after all, if you do this, given a suitable monitor you could also watch American (or Japanese) movies!

George Chkiantz

We recommend intending users of the PCM-F1 system to consult their dealers to establish which of the various configurations is most appropriate for their application, in view of the numerous possibilities available – Ed.

The F1 System by Sony

The PCM-F1 digital audio processor:

Quantization 16 bit linear or
14 bit linear (switchable)
Frequency response: 10-20,000 Hz ± 0.5 dB
Dynamic range more than 90 dB 16 bit
more than 86 dB 14 bit
Harmonic distortion less than 0.005% 16-bit
less than 0.007% 14-bit
Wow & Flutter beneath measurable limits

Available in two versions, PAL or NTSC the PCM F1 can be used in conjunction with any V.C.R. recorder.

Coupled with the SL.F1 portable Betamax, the PCM F1 provides a portable digital recording system with phenominal performance.

The Servo control system of the SL.F1 Betamax will allow absolute synchronization of the recorded material at all times and enables subsequent transfers to be made directly to 35 or 16 mm film or video.

For Studio use, interface the PCM F1 with any of the new Type 5 SONY U/Matic recorders.

Editing can be performed using a standard U/Matic video editing facility to an accuracy of ± two frames i.e. 80 ms-P.A.L.

For serious audio editing and subsequent transfer to disc, the recorded signal from the PCM F1 can be transferred to the Sony PCM 100 or PCM 1610 digital audio systems and edited using the DAE 1100 digital audio editor.

Any number of subsequent copies can be made with no loss of signal quality.

The PCM F1 features a digital to digital dubbing facility.

Your discerning customer can make V.C.R. copies of studio masters and play them back on their own PCM F1 system and experience exactly the same quality of sound as the original master.

For further information or demonstration – please contact Roger Cameron or John Didlock.

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Sony PCM 3324 digital recorder

The review sample of the PCM-3324 was a production prototype. A number of changes have already been made in the production version, and probably more will follow, before the machine is available for sale later in 1982.

Resulting from the activities in the Audio Engineering Society and other areas the sampling frequency for professional digital audio is now hopefully standardised at 48 kHz with the consumer digital audio disc and video cassette-based systems using a sampling frequency of 44.1 kHz. The Sony *PCM-3324* can operate at either of these frequencies.

MANUFACTURER'S SPECIFICATION
Tracks: 28 comprising 24 PCM, CTL (control), EXT data, and two for analogue audio signals.
Tape speed: 665 mm/s (at 44.1 kHz sampling rate) 723.8 mm/s (at 48 kHz sampling rate).
Tape: digital audio master tape – D-1/2-2920 (68 min at 44.1 kHz) D-1/2-1460 (34 min at 44.1 kHz).
Sampling rate: 44.1 kHz and 48 kHz (switchable).
Quantisation: 16 bit linear/channel.
Dynamic range: more than 90 dB.
Frequency response: 20 Hz to 20 kHz, +0.5 dB, −1.0 dB.
Total harmonic distortion: less than 0.05%.
Emphasis: on/off switchable.
Wow and flutter: undetectable.
Rec and play time: 68 min with 14 in reels at 44.1 kHz; 60 min with 14 in at 48 kHz.
Rewind and fast forward time: 5 min (with 14 in reels) 2½ min (with 10½ in reels).
Channel coding: HDM-1 (High Density Modulation).
Error correction: CRCC and Cross Interleave Code.
Editing: Sync, recording, punch in/out and splice free (cross fading at every editing point).
Dimensions: 830 × 990 × 740 mm (whd).
Weight: approx 220 kg.
Power supply: AC100/120/220/240 V – 50/60 Hz, approx 3 kVA.
Manufacturer: Sony Company Limited, 7 – 4 Konan, 1-Chome, Minatu-Ku, Tokyo 108, Japan.
UK: Sony (UK) Limited, Communication Systems Division, Pyrene House, Sunbury-on-Thames, Middlesex.
USA: Sony Corporation of America, 9W 57th Street, New York, NY 10019.

THE *PCM-3324* is an extremely complex machine with numerous special facilities and features. As a result, in the space available for this review it is not possible to fully describe all the features of the machine.

The tape transport, working at over 40,000 bits/in with 28 tracks on thin ½ in tape, requires very precise control of the tape not only in the record/replay modes but also in the fast modes to avoid tape damage or stretching, especially at cut-and-splice edits. The result of this is a tape transport of very unusual design. Two separate castings are used to form the basis of the transport: one with finned sections supports the two servo-controlled DC reel motors with their solenoid-operated band brakes and a second substantial casting carries the rest of the tape transport. Heavy square section bars join the two castings and support them at the front and rear of the machine with twin tangential cooling fans at the rear drawing air over the finned sections at the reel motors.

The spool motors drop into the top of the rear casting making replacement very simple, the spool capacity being a maximum of 14 in. Tape and the empty spools supplied with the machine are of a design similar to the precision spool used for instrumentation work (precision NAB type) but with a reduced clearance between the flanges of 13.306 mm against the standard of 13.462 (+0, −0.051) mm. These are a nice fit on the precision hold-downs which fit directly on to the reel motor shafts with a central screw adjusting height. At a tape speed of 723.8 mm/s using the 48 kHz sampling rate this gives a recording time of 1 hr 7 min using 28 μm thick tape. At the 44.1 kHz sampling rate the tape speed reduces to 665.0 mm/s with a resulting available time of 1 hr 13 min maximum. Sony conservatively specify the times at 60 and 68 min for the two sampling frequencies.

From either reel the tape passes to a precision guide roller which guides the tape with fixed metal edge guides. The roller assembly is on a small sub-casting which is shimmed on to cast pillars on the main casting. From here at either side there is a rather complicated tension sensing arm system. Each arm mounts onto a spindle in a small casting which bolts on to the main casting, and the spindle has a potentiometer on its lower end. At the other end of the arms there is a precision roller.

The system uses three arm positions. In the ready-to-load condition the arms are retracted by a

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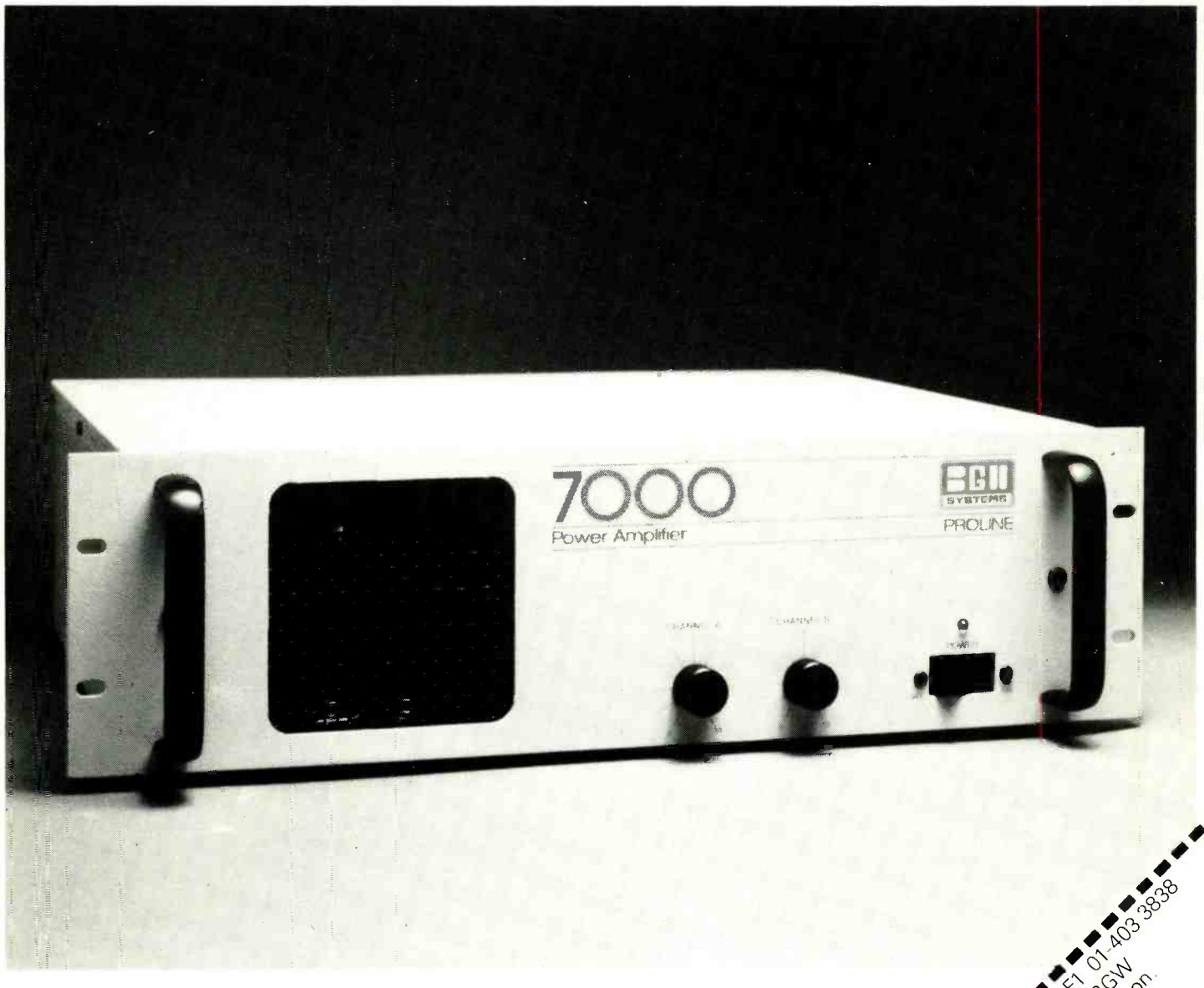
BGW SYSTEMS

The Model 7000 is a 2 channel amplifier, rated at 200 watts average continuous power per channel into 8 ohms, 20Hz-20kHz, at no more than 0.1% Total Harmonic Distortion (325 watts/channel into 4 ohms).



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motor driven linkage and locked in their outer positions. After lacing the tape, pressing the head cover forwards results in the two arms motoring to their other extreme position thus taking up any slack in the tape. Once the inner position is arrived at, a solenoid releases the motor drive and the lightly spring loaded arms become tape tension sensors.

On the pay-off side the tape then passes to a large diameter roller guide which has a tachometer that drives the tape timer and feeds the control processor. Following this there is a lightweight 'shift roller' which has three positions. In ready-to-load state the shift roller is fully retracted to give a straightforward tape path. Once the tension rollers are released to their operating position the shift roller is solenoid driven to its ready position where it remains in all but the play and record modes. When entering these modes the shift roller is again solenoid driven, this time to its inner position where it applies the tape to the heads, at the same time putting the head shields in position and removing lifter pins from the tape path. the complete operation is damped by an air dashpot.

The rather complex plug-in headblock is shimmed on to three pillars which are secured by screws on to a subsidiary casting which itself is bolted on the main casting.

Upon entering the headblock the tape passes over ceramic edge guides to the digital advance record head, the digital playback and the analogue erase heads before passing a second pair of ceramic edge guides. It then continues past the analogue record/replay head, to an optical tape presence detector, the digital record (sync record) head and a third pair of ceramic edge guides. All these components are suspended from a 3 mm thick plate with spring loaded head position adjustments.

Upon leaving the headblock the tape passes over the large diameter pinchroller-less capstan directly driven by a DC servo motor. A large-diameter knurled knob on the capstan allows the tape to be manually shuttled at sensible speeds for locating edit points. From here the tape passes to the take-up tension arm via the guide roller to the take-up reel.

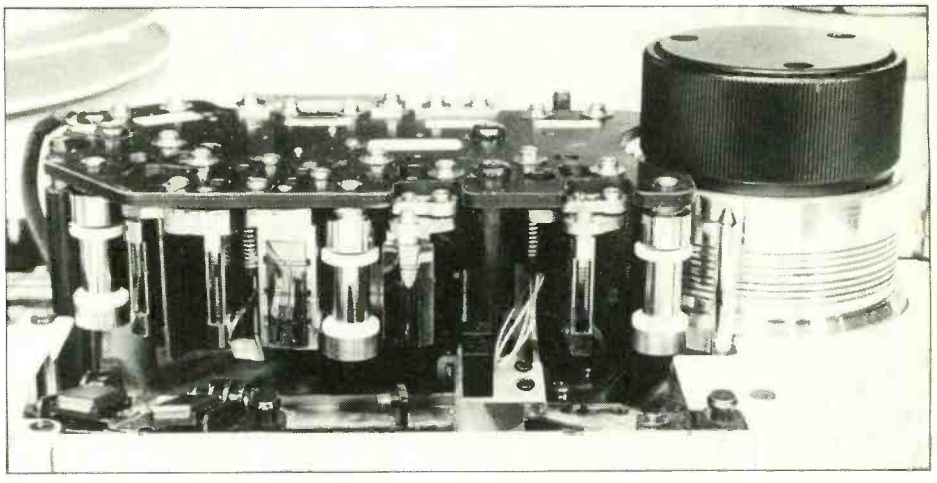
Access to the analogue replay heads for editing is gained by sliding the head cover to the right when it can be lifted off, the production version having a more substantial head cover and, hopefully, a more substantial mechanism. Once removed, access to the tape is excellent.

Tape tension remained remarkably constant at 200 g throughout the tape path in all modes, the drive functions being controlled by a *Z80* microprocessor.

Whilst a number of higher power printed circuit boards are on the transport surfaces the main electronics mount into two racks beneath the tape transport. To the right of the bottom rack are the power supplies with six each A/D and D/A converter boards occupying the left hand side, each board handling four channels. The A/D boards have front panel switches putting pre-emphasis on or off for each channel – more about this later. Switching of de-emphasis is automatic with the D/A boards having front panel indicators for de-emphasis on/off.

Within the top rack are 34 printed circuit boards fitting into sockets which are wire-wrapped on the back plane. About half of them are concerned with normal signal processing and the remainder with system control, editing, interfacing, tape transport control, etc.

Numerous LED indicators are fitted to the front panels which will be secured to the frame by screws in the production units. There are front panel



toggle switches which allow 'dither' to be switched on/off and will switch each channel between analogue and digital input with potentiometers to allow adjustment of the analogue output level from the two analogue tracks. A 16-position switch adjusts the crossfade time during electronic editing, from 1.33 ms to 341 ms at the 48 kHz sampling rate or 1.45 ms to 375 ms at 44.1 kHz sampling rate, the times when splice editing being 5.20 and 5.66 ms respectively.

A further feature of particular interest is an array of seven LED indicators giving error messages. These deal with such conditions as power supplies out of tolerance, excessive motor temperatures, servo problems, etc.

The final user feature in the card frame is a series of DIL switches which select the brightness and flashing rate of the keyboard indicators.

Despite the fact that the review unit was a production prototype all boards were very well made with socketed integrated circuits and full component identifications.

At the rear the analogue inputs and outputs take the form of *XLR* connectors in the lower section of the rack on a hinged panel with the remaining connections being on a hinged panel in the upper section. Here *XLR* connectors give access to the two analogue tracks and the timecode track.

Two multi-way 'D' connectors provide the digital signal inputs and outputs with a further 'D' connector being provided for remote control. Eleven BNC sockets, some with switched 75 Ω terminations, give various synchronisation inputs and outputs with large terminals being provided for chassis and signal grounds.

Within the machine are numerous cooling fans which in the review sample were very noisy (54 dBA at 1 m in the lab), however a 10 dB reduction in noise is anticipated in the production version.

Control of the machine is divided into several sections in two areas, one a sloping panel at the front of the tape transport and the second a horizontal sill at the front of the machine. The former in the pre-production machine has the power on/off switch to the left followed by 24, 16-segment level meters indicating overload (red) to −34 dB (green) below overload. In the production version the power switch will be relocated and the analogue track level indicators will be 8-segment displays.

There follows an 8-segment timecode level indicator with red segments at the extremes, two 16-segment analogue track audio level indicators, and the 8-segment control track level indicator with red at the extremes with three LEDs below the indicator. The latter red, green and yellow LEDs show control track record, replay or a jump in control track data. Further to the right, two horizontal rows of five LEDs provide warnings – the upper row about formats and the lower about alarm conditions. The format warnings are illuminated for an unacceptable tape format, incorrect sampling frequency, incorrect de-emphasis, master safe selected and remote control selected.

Alarms are illuminated for abnormal power supply conditions, abnormal servo conditions, irregularities in the digital control system, excessive dropout and failure of the capstan servo system.

Finally in this section, under a hinged transparent cover are eight miniature toggle switches with associated LED indicators. The first two provide a master safe function and select the sampling rate from 48 kHz or 44.1 kHz. The next has three positions which switch emphasis master on/off or to the individual conditions selected on the printed circuit card front panels. Likewise the analogue or digital inputs can be master selected or individually selected. Monitoring can be automatically switched to the input in the fast and stop modes, or manually selected for individual tracks, and the time can be absolute from the tachometer or read control track time. The synchronisation clock may be internal only or automatically switched to external when an input is present and finally the overall control can be switched local or remote when the local controls are inhibited.

Movement control of the tape is on the horizontal section which includes the usual fast, replay, record and stop buttons which become illuminated to show which action is current. Other functions are provided including a finger operated shuttle roller which is spring loaded to the stop position.

Movement of the shuttle roller in either direction initiates variable speed tape motion with speed depending upon the roller position. As with the fast modes the tape motion is rather sluggish, but this is a requirement of the gentle tape handling.

Pressing the record and play buttons enters the record mode, however there are other options. Pressing the 'record rehearse' and the play buttons, memorises the positions of punching in and out without actually recording. The punch in/out can then be auditioned by pressing the record rehearse and 'auto punch' buttons when the machine automatically switches between the reproduce signal and the input signal at the selected points. Once satisfied that the edit is correct, pressing the auto punch and record buttons executes the edit – a very nice system. Two further pushbuttons are located in the section, 'locate' which works with the tape timer and 'edit stop' which if pressed in the replay mode rolls on for 5 s

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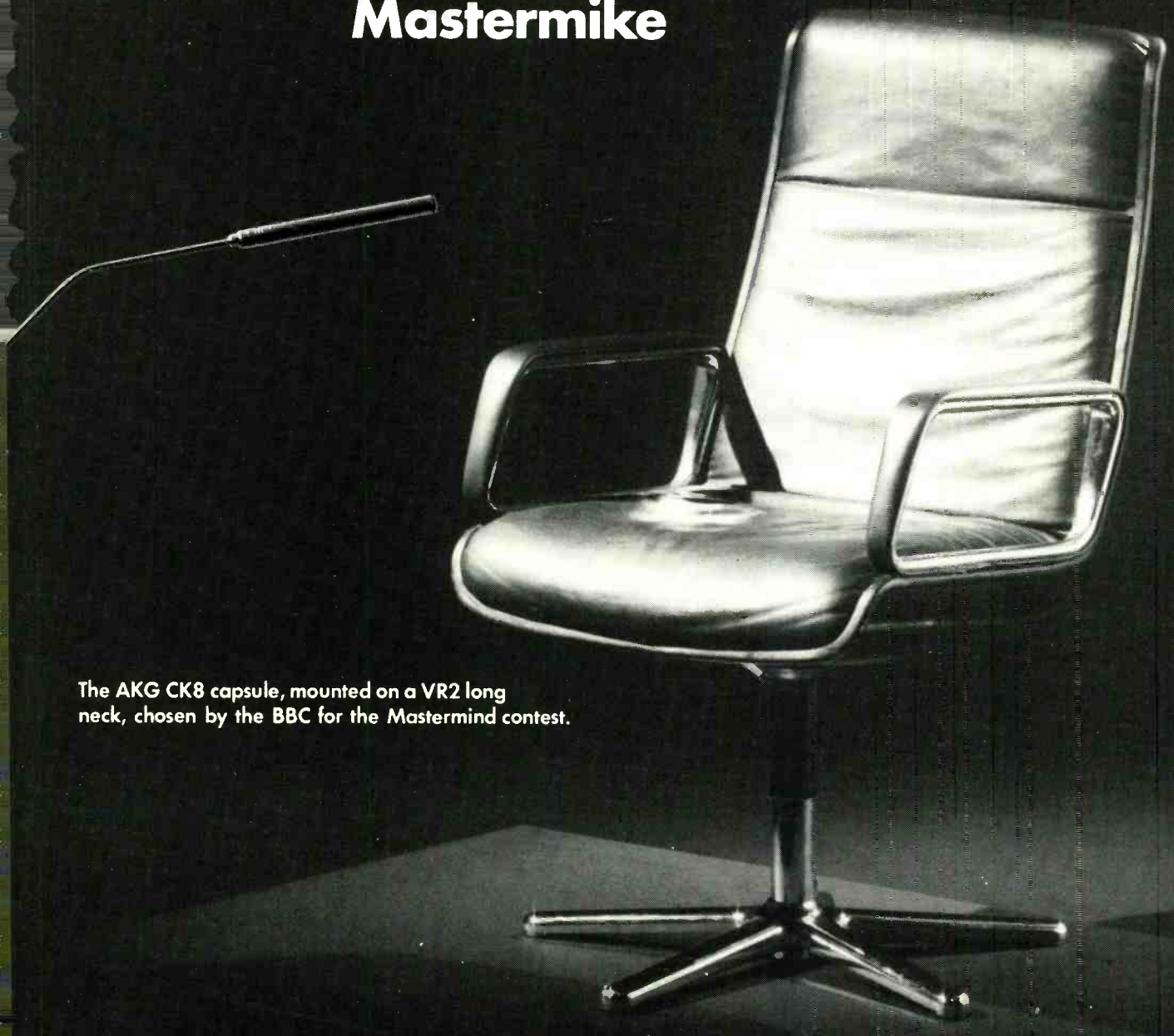
Mastermike



The AKG CK8 capsule, mounted on a VR2 long neck, chosen by the BBC for the Mastermind contest.

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and then rewinds to the tape position where it was pressed.

The tape timer, which counts in minutes and seconds, can be set in a number of ways. Three membrane switches with LEDs perform 'set timer', 'set cue point' and 'set speed'. These work in conjunction with 'up' and 'down' switches which when pressed count the appropriate display up or down starting at a low rate and getting faster the longer the button is pressed. Pressing the 'set' button a second time remembers the setting and extinguishes the LED next to the button. A further way of setting the time is from five cue stores. Pressing the cue store button remembers the tape location and automatically steps to the next store indicated by one of five LEDs. Subsequent pressing of the recall button steps through the stores indicating their contents whilst the button is held. These cue points may be entered as locate points in the timer by pressing 'set cue point'.

Varispeed can be accomplished in record or replay by pressing the varispeed membrane switch which illuminates a flashing LED. The speed may then be set as a percentage or in semitones with the previously mentioned 'up' and 'down' membrane switches.

All signal routing controls are membrane switches with 27 vertical arrays of LED indicators for the 24 digital audio tracks, the two analogue audio tracks and the timecode track. To the left of the displays are record ready, safe, reproduce, input, mute on and mute off buttons, the state of each track being shown by LEDs working in conjunction with a track selector button at the bottom of the arrays. Green LEDs at the top of the arrays indicate the tracks selected, followed by three LEDs in a square opposite the record ready button, green indicating safe, orange record ready and red record.

Below these, further LEDs indicate input or reproduce, thus individual tracks may be set as desired with a clear indication of the current set-up. Alternatively there is an 'all' button which allows all digital tracks to be set the same.

Four grouping pushbuttons allow versatile grouping of any combination of the digital tracks with a memory backup retaining the grouping during power off. Similarly there are four set-up memories which can be loaded with the current status of the digital tracks and recalled as desired.

The final controls are for the selection of the record mode – this requires the simultaneous pressing of the 'record mode set' button and one of four other buttons. The normal mode is the advance record mode where digital recording takes place at the advance record head with digital monitoring at the digital replay head. A record mute setting effectively erases the tape with the other two settings being sync modes using the digital replay head and the second digital record head located after the replay head. There are two of these conditions, 'insert' where all tracks other than the control track are recordable and 'assemble' where the control track is also recorded.

Whilst the audio signal routing is easy to master as is the grouping and the set-up store, the cue system needs good driving lessons and I think it could be simplified.

Tape loading on the review sample was straightforward with one proviso – if the tape was not properly located in the optical sensor (which could happen accidently) the machine refused to load without any indication of the error.

Inputs and outputs

Investigation of the analogue inputs to the digital audio system showed the inputs to be balanced with an impedance of 10.3 kΩ. Meter zero corresponded to an input level of +4 dBm with the input at clipping being +24 dBm. Common mode rejection varied widely between channels with a worst case of 60 dB from 20 Hz to 20 kHz. The inputs to the analogue tracks also had a sensitivity of +4 dBm for meter zero indication with the maximum input capability of +24 dBm into 9.8 kΩ. Common mode rejection at the analogue inputs was not good as shown in **Fig 1**.

All the audio outputs, with the exception of the

FIG.1
SONY 3324
ANALOGUE
INPUT CMRR

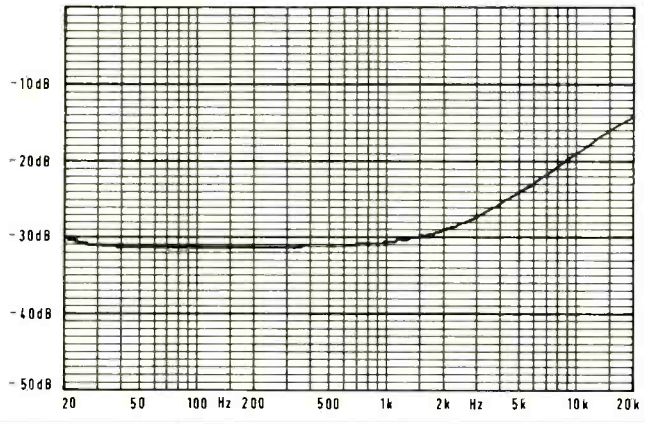


FIG.2
SONY 3324
ANALOGUE TRACK
FREQUENCY RESPONSE

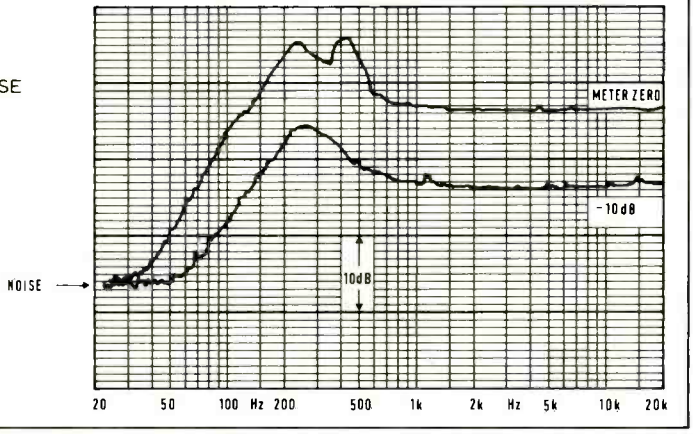


FIG.3 SONY 3324 DIGITAL FREQUENCY RESPONSE 10dB BELOW PEAK

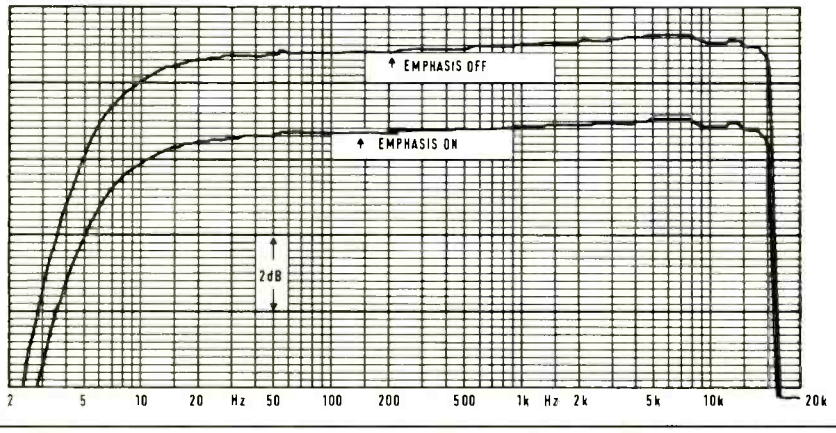


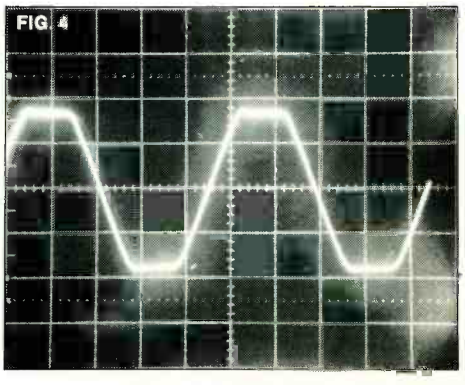
TABLE 1

	With tape	Machine
A-weighted RMS	– 39 dBA	– 49.5 dBA
CCIR-weighted peak	– 28 dB	– 45.5 dB
CCIR ARM	– 42 dB	– 56.0 dB

TABLE 2

Measurement method	Emphasis on	Emphasis off
22 Hz to 22 kHz RMS	– 92 dB	– 87 dB
A-weighted RMS	– 92 dB	– 88 dB
CCIR-weighted RMS ref 1 kHz	– 85 dB	– 80 dB
CCIR-weighted quasi-peak ref 1 kHz	– 81 dB	– 77 dB
CCIR-weighted ARM ref 1 kHz	– 92 dB	– 87 dB

FIG. 4



headphone output, had a drive capability of +24 dBm from a source impedance of approximately 47 Ω. The maximum output available to the headphones was too low at 0.3 V from a satisfactory impedance of 10 Ω which remained constant with the output gain setting.

Various synch and other digital outputs were either not connected or thought not to correspond to the production machine, but many are destined to be TTL compatible.

Analogue tracks

Whilst I was informed that the analogue tracks were out of specification on signal to noise, this appeared to be an understatement. Of course the analogue tracks are only used for editing and thus at first sight do not need good quality. However, experience shows that editors do require a reasonable quality which is certainly lacking in the prototype machine.

Fig 2 shows the frequency response of the analogue tracks at meter zero and −10 dB, both being flat at high frequencies and having an unhealthy boost around 250 Hz.

Harmonic distortion was poor with the following at meter zero: second harmonic −15 dB, third −30 dB, fourth −30 dB.

Noise related to meter zero was also poor as shown in **Table 1**.

Erasure was satisfactory at > 70 dB for a 1 kHz tone, but erased tape had a 'snap, crackle and pop' noise problem. Finally the IEC weighted peak wow and flutter was satisfactory at 0.14%.

Analogue/digital frequency response and noise

Without emphasis switched in the frequency response was independent of recording level. At lower levels the frequency response was as shown in **Fig 3** with the anti-aliasing filter being the same for both sampling rates. The extremely steep filter had a −3 dB point at 20.527 kHz falling to −50 dB at 21.758 kHz.

Fig 3 demonstrates very good matching of the pre-emphasis and de-emphasis components, the time constant of the emphasis being in the order of 20 μs.

Noise relative to the maximum output was measured with and without emphasis and the switchable dither. The latter degraded the noise by only about 0.5 dB with **Table 2** showing typical results at the 48 kHz sampling rate.

Noise when using the 44.1 kHz sampling rated was 1 to 2 dB worse, however there was no problem with mains hum or its harmonics and sampling rate artifacts were at a very low level.

Distortion

Overloading the system resulted in symmetrical clipping as shown in **Fig 4**; the metering, being after pre-emphasis, gave 'safe' results when pre-emphasis was used.

Total harmonic distortion measured at 1 kHz and at 10 kHz was less than 0.01% at meter zero level and above until clipping was reached. At lower levels the measurement of the second and third harmonic distortion produced **Fig 5** which shows that distortion is extremely low, down to −50 dBm output, corresponding to 74 dB below maximum output.

As shown in **Fig 6**, for an output level of −50 dBm the harmonic distortion was not particularly sensitive to frequency.

Measurement of intermodulation distortion to the CCIF twin tone method showed a worst measurable case at −30 dBm output, this being shown in **Fig 7** for the first and second order difference frequency components.

70 ▶

As with other digital machines the ringing on squarewaves was rather poor as shown in **Fig 8** for a 1 kHz waveform, the ringing being irrespective of level.

Metering

The meters in the digital channels were very fast reading peak meters which responded in less than 10 μs and had a sensible fall time of 700 ms giving good readability. The ability to read in 4 dB steps down to −12 dB and then −22 dB and −34 dB makes good sense for digital recording.

Metering of the two analogue audio channels was with a slightly more restricted range, the meters appearing to simulate a VU meter with a 300 ms rise time and 1.4 s fall time. As has been mentioned the meter zero corresponds to far too high a distortion level.

FIG.5

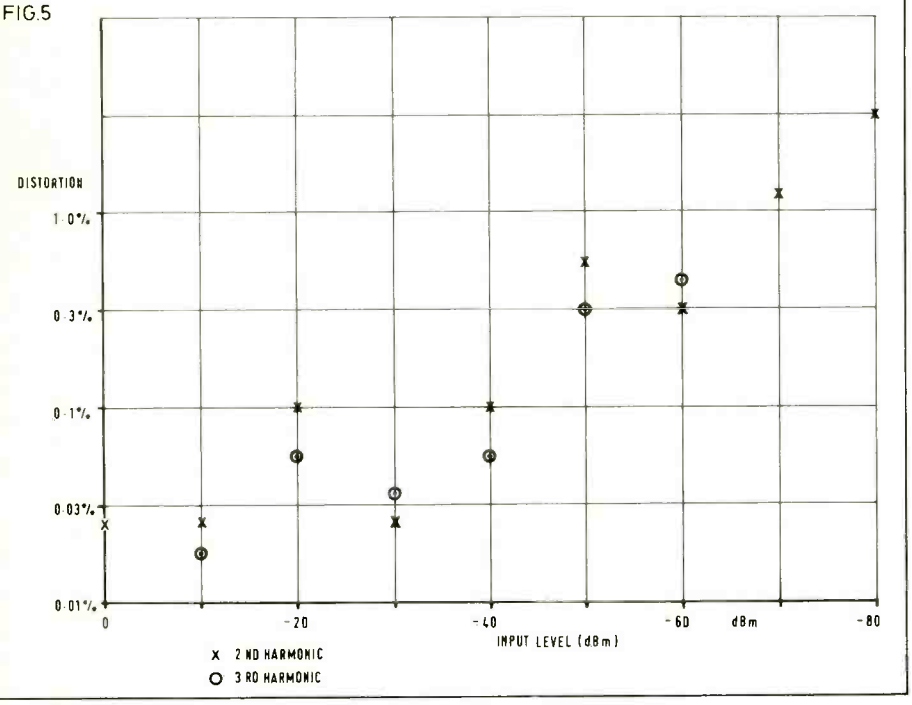


FIG.6
SONY 3324
HARMONIC DISTORTION
AT -50 dB

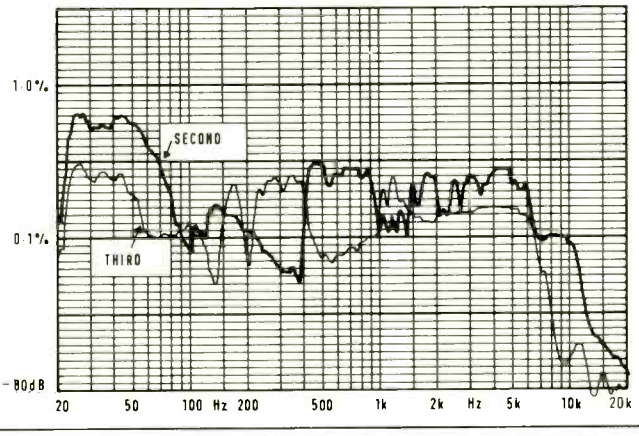
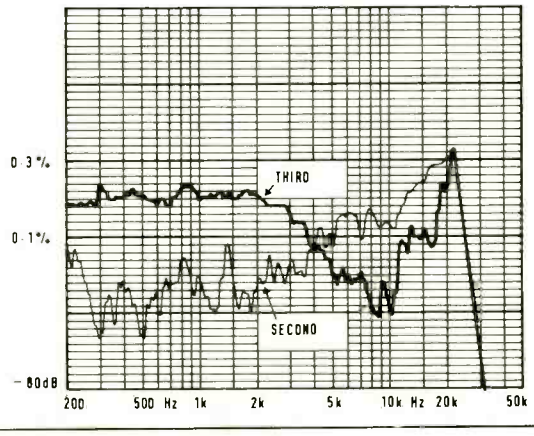


FIG.7
SONY 3324
WORST CASE
INTERMODULATION DISTORTION



Editing

Normal dropping in and out of record produced clicks but this is not the intended method of editing. Using the 'auto punch' button in conjunction with the assemble and the insert record modes gave excellent results.

The nature of the edit achieved depends of course upon the setting of the crossfade time, this feature operating at both drop-in and drop out. **Fig 9** shows a particularly nasty situation for the machine – a 100 Hz tone has been recorded over with another 100 Hz tone and dropped out where the waveforms are 180° out of phase. In spite of the minimum crossfade time setting a smooth result was obtained. The waveform in **Fig 10** shows the results of dropping out a 200 Hz tone recorded over an existing 100 Hz recording at the maximum crossfade setting, it being seen that there was a smooth transition lasting over 100 ms.

Crossfade operated in a satisfactory manner in the preview and the execute editing modes, but splice editing was another story.

Quite rightly Sony recommend great consideration for cleanliness and the wearing of gloves while handling the tape, but there was not found to be a severe problem and only bad contamination led to a dropout. The machine then muted without clicks as shown in **Fig 11**, the length of the mute depending upon the length of the contamination.

Bearing in mind that a gap at a splice of less than 100 μm is recommended, this is a hint that a satisfactory edit is not easy. In practice, with the thin 26 μm recording tape, it is extremely difficult not to damage the tape near the edit point. In addition, 16 μm splicing tape is not easy to handle, particularly when wearing cotton gloves.

Having managed the joining of the tape the next problem is trimming the splicing tape on the recording tape – again not an easy matter requiring a very sharp razor blade if the splice is not to be damaged.

The final problem is removing the splice from the splicing block. If an *EDITall* block is used the tape is very easily damaged on its edge and I found my *BIB* block far better in this repect.

Having overcome these hurdles it is then very easy to damage the tape on the optical tape presence sensor when re-loading the machine, and if you forget to carefully place the tape in the slot in the sensor the machine attacks the tape when it tries to load.

Overall I feel that splice editing is far too critical to be of any practical use and I certainly wouldn't dare to attempt a splice edit without back up copies.

Other matters

It is generally accepted that digital recordings have the advantage of a good dynamic range without degradation in copying, no modulation noise and no wow and flutter – all OK in this department. However, in a multitrack machine, there are other

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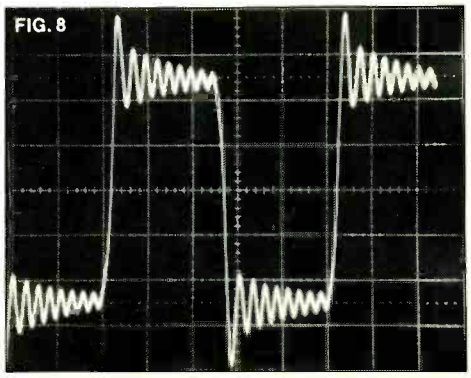
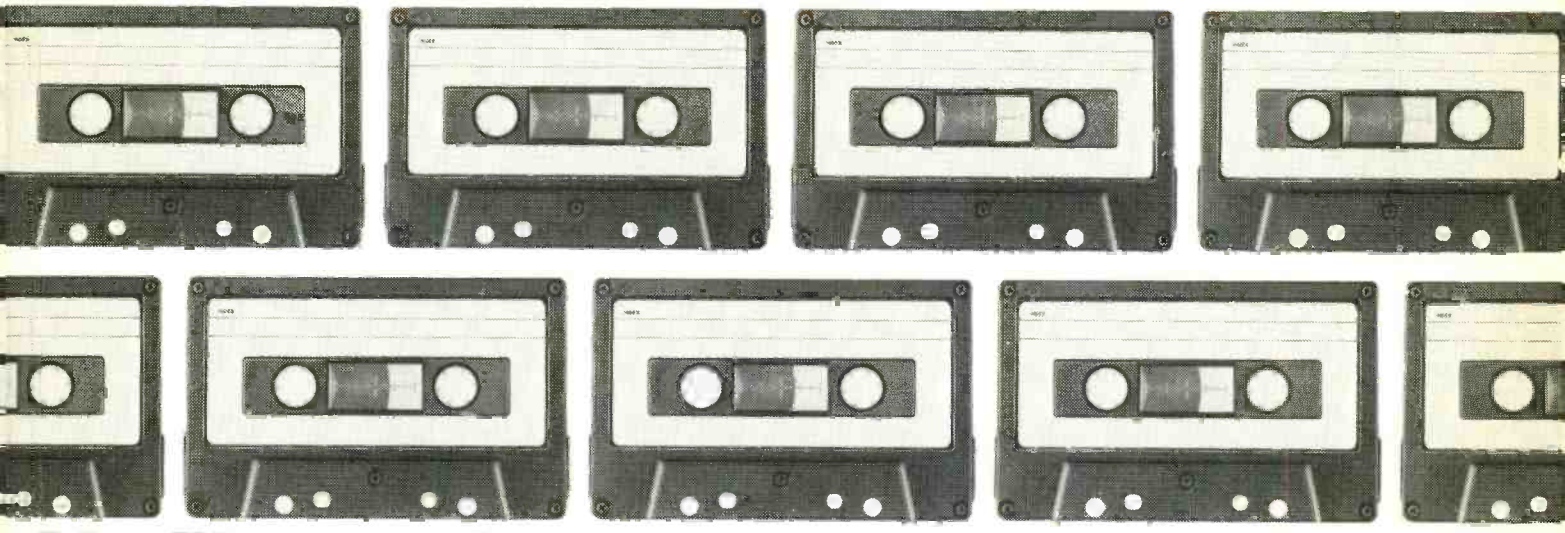
FIG. 8





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very important advantages.

Firstly, the crosstalk between tracks is virtually non-existent in the record/replay mode as shown in **Fig 12** for adjacent tracks. Secondly, there are two great advantages in the sync mode. When recording on one track and replaying in the sync mode the crosstalk between adjacent tracks is also shown in **Fig 12**. Of course print through doesn't exist so far as the audio is concerned and neither does phase jitter between tracks.

Whilst the pitch readout in the varispeed mode was accurate to within its readability the setting of both the pitch and the tape time was far too long-winded.

FIG.9
SONY 3324
DROP OUT AT 100Hz, CROSSFADE MINIMUM SETTING

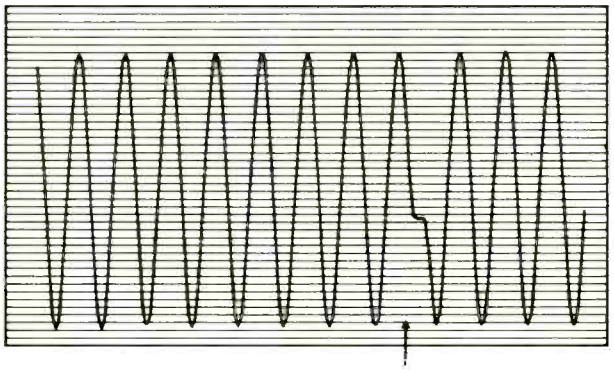


FIG.10 SONY 3324. 200Hz TO 100Hz DROP OUT, CROSSFADE MAXIMUM SETTING

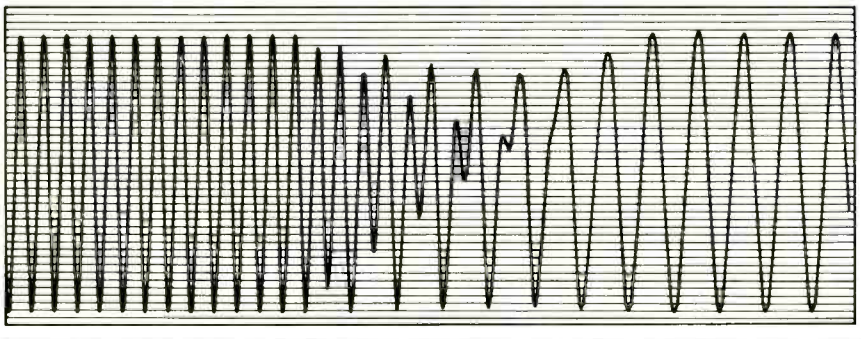


FIG.11 SONY 3324. DROP OUT MUTE - 100Hz TONE

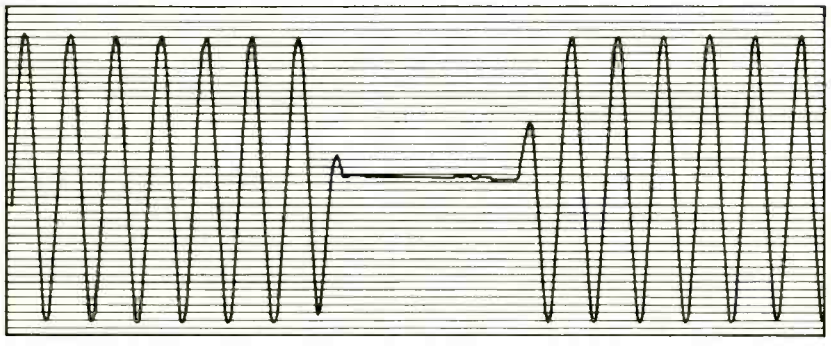
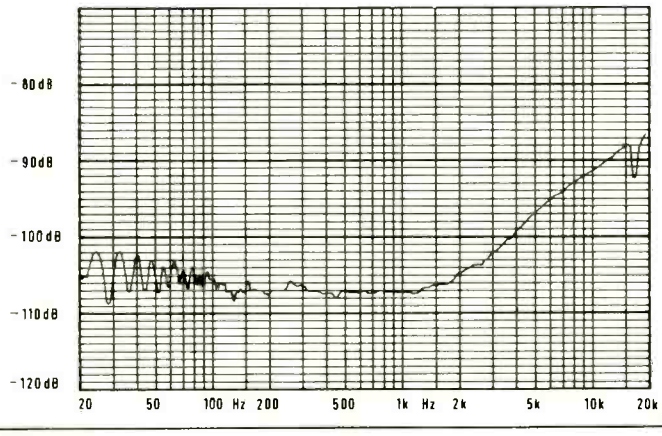


FIG.12
SONY 3324
CROSSTALK, TRACK 18 RECORD, TRACK 19 PLAY



The quality of the tape winding in the fast modes was not good as the tape blocked and leafed – this is a common complaint with Japanese recording tapes and maybe the machine would behave better with American tape. The production machine will wind slower so this may help.

The locate function was found to be very accurate and smooth in operation but the tape timer arrangement was rather tricky to master and could produce very unexpected results if operated incorrectly.

Whilst it is known that the production head-block will be modified I hope that the rather flimsy head shield arrangement will also be modified or removed – I don't see its purpose. Similarly, the mechanics of the moving optical tape sensor are not very robust and serve little purpose.

Summary

This prototype 24-track machine has very many novel and innovative features. The tape transport handled tape extremely gently with excellent control of tension, although the fast winding of the tape left something to be desired.

In the audio signal department the machine behaved very well with an exceptionally good distortion performance at low signal levels. Whilst noise is very good, possibly an extra dB or so could be achieved with the 16 bits.

Operationally the signal routing was a delight, as was the electronic editing, but there is room for improvement in the tape timer system.

As things stand, splice editing is not to be recommended but is it really necessary anyhow?

There is no doubt that this digital machine offers very many significant advantages over any analogue machine. Of course there is a price to be paid, but this also applies to a premium 24-track analogue machine with 24 tracks of noise reduction which is still no match for digital technology.

For those interested in the detailed design of the Sony *PCM-3324* I recommend the following written by Dr Toshi Doi and others of Sony:

1) *A Format of Stationary Head Digital Audio Recorder Covering Wide Range of Applications* (AES pre-print 1677);
2) *Channel Codings for Digital audio Recordings* (AES pre-print 1856);
3) *Cross Interleave Code for Error Correction of Digital Audio Systems* (AES pre-print 1559);
4) *A Design of Professional Digital Audio Recorder* (AES pre-print 1885).

Hugh Ford

MANUFACTURER'S COMMENTS

Sony has made the following comments on Hugh Ford's review of the *PCM 3324*:

- *The head-cover mechanism and opto-sensor have been completely redesigned on the production models, and the head-phone level, which is adjustable, has been increased by 10 dB.*
- *Common mode rejection is adjustable and meets spec on all production models.*
- *The analogue track performance on the review model was out of spec as stated; however, production models meet published data.*
- *Splice-editing does require more care than with analogue, and because the tape base is thinner than with standard-play analogue tape, the edges are more susceptible to damage. Sony will be able to supply a specially-designed, precision block as an accessory to improve handling. In independent trials under practical editing conditions at CBS/Sony studios, 100% success rate has been achieved. Also, because the machine deals with a splice by automatically correcting for it as an error, the resulting transition is excellent.*

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The SM81 has been quite a shock to me, not only from when I first tried it out, liked it, and decided to buy a pair, but also a year later when I discovered from the brochure that the mic. was an electret.

Shure Brothers have always had a good name for robustness and reliability, and electrets are usually thought of as a low cost alternative to regular capacitor mics. with some sacrifice in sound quality.

With the SM81 Shure have produced an unique combination - together with a transparency of sound and freedom from coloration, distortion and noise comparable with other manufacturers' traditional condenser models costing a lot more. The switchable bass roll-offs and attenuator are helpful extras as well, and missing from my other favourite choice of cardioid costing around double the price.

Recording classical music is a tough test for microphones and my SM81s earn their keep successfully as very useful additions to my kit of mics., both for distant and close pickup if required.

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dbx 2150 series VCA

ABBREVIATED SPECIFICATION
Electrical characteristics at 25°C ambient

		Min	Typ	Max	Unit
Equivalent input bias	No signal	–	5	8	nA
Gain linearity	– 60 dB to + 40 dB	–	±1	±2	dB
Output noise	0 dB gain				
	Rout = 20 kΩ	–	– 95	– 90	dBV
Gain control constant	– 60 dB to + 40 dB	5.8	5.9	6.0	mV/dB
Output offset voltage (Rout = 20 kΩ)	0 dB gain	–	±1	±3	mV
	15 dB gain	–	±2	±3	mV
	40 dB gain	–	±10	±15	mV
Intermodulation distortion	15 dB gain	–	.01	.02	%
Total harmonic distortion	0 dB gain	–	.01	.02	%
(I_{tot} = 175 μA 1 kHz)	±15 dB gain	–	.05	.07	%

Manufacturer: dbx Inc, 71 Chapel Street, Box 100C, Newton, Massachusetts 02195, USA.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA.

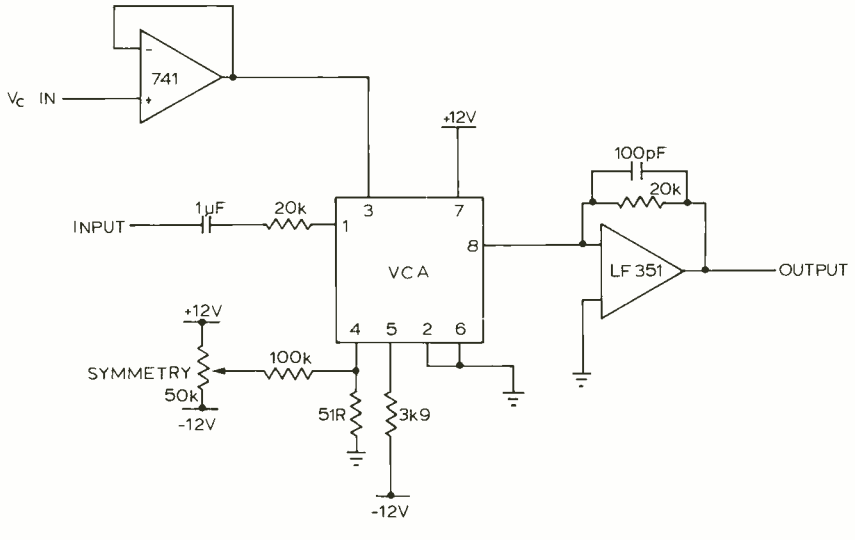
TABLE 1

Control voltage	Gain Sample 1	Gain Sample 2
– 300 mV	+ 49.84 dB	+ 49.81 dB
– 240 m	+ 39.92 dB	+ 39.91 dB
– 180 m	+ 29.93 dB	+ 29.94 dB
– 120 mV	+ 19.92 dB	+ 19.94 dB
– 60 mV	+ 9.95 dB	+ 9.87 dB
0 V	– 0.01 dB	+ 0.01 dB
+ 60 mV	– 10.02 dB	– 9.99 dB
+ 120 mV	– 20.04 dB	– 20.01 dB
+ 180 mV	– 30.02 dB	– 29.99 dB
+ 240 mV	– 40.03 dB	– 39.99 dB
+ 300 mV	– 50.04 dB	– 49.99 dB
+ 360 mV	– 60.08 dB	– 60.04 dB

TABLE 2

Measurement Method	Noise at a gain of (dB)				
	40 dB	20 dB	10 dB	0 dB	– 20 dB
22 Hz to 22 kHz RMS	– 62.5	– 82.0	– 92.0	– 105.0	– 100.0 dBV
A-weighted RMS	– 68.0	– 86.5	– 94.5	– 100.5	– 106.0 dBV
CCIR-weighted RMS	– 60.0	– 78.0	– 85.0	– 91.5	– 97.0 dBV
CCIR-weighted quasi-peak	– 56.0	– 73.5	– 78.5	– 87.0	– 93.0 dBV

FIG.1
dbx VCA
BASIC CIRCUIT USED FOR TESTING



THE dbx *series 2150* voltage controlled amplifiers are of remarkably small dimensions, being housed in an 8-pin plastic single-in-line package. Positive and negative supply rails are required with the manufacturer recommending ±12 V rails with operation being possible between ±4 V and ±15 V maximum.

The single input is a current port which must be isolated from DC – the input therefore requires a series capacitor and a current ranging resistor. If the source impedance feeding the input exceeds 20 kΩ at 100 kHz, instability may occur but this problem can be rectified by applying a further series RC from the input to ground, 100 pF in series with 20 kΩ being recommended by the manufacturer.

At the recommended bias current of 3 mA (which is controlled by an external resistor) the maximum input current is limited to 1.5 mA instantaneous.

Like the input, the output is also a current termination intended to feed a virtual earth. A current-to-voltage converter can be simply accomplished with any reasonable operational amplifier such as the *LF351* with a single RC in the feedback controlling the current-to-voltage ratio and stability.

The basic circuit used for testing the samples is shown in **Fig 1** where the audio input is applied to pin 1 via a 1 μF DC isolating capacitor and a 20 kΩ scaling resistor giving a – 3 dB point at 8 Hz. The audio output at pin 8 is fed to a virtual earth amplifier in the form of an *LF351* with 20 kΩ in the feedback giving a current scaling factor of 50 μA/V, thus giving unity overall gain when the VCA gain is unity.

The control voltage Vc is fed via a buffer *741* op amp to pin 3, which is a voltage control input with a sensitivity of 6 mV/dB, the buffer being necessary to provide a very low impedance (recommended to be < 1 Ω) to stop audio crosstalk entering the voltage control input. This arrangement gives negative gain with positive control voltage, but grounding pin 3 and applying the control voltage to pin 2 (with the bottom of the 51 Ω resistor also connected to pin 2) gives positive gain with positive control voltage.

In the circuit shown, the voltage applied to pin 4 adjusts symmetry and is set up for minimum second harmonic distortion at 1 kHz and unity gain. This was found to be a critical adjustment and I strongly recommend a multiturn potentiometer for the symmetry control.

Finally there is the input to pin 5 which controls the overall bias current, pin 5 being four diode-drops below ground (measured at 2.65 V).

As has been seen this VCA requires the minimum of external components for its basic

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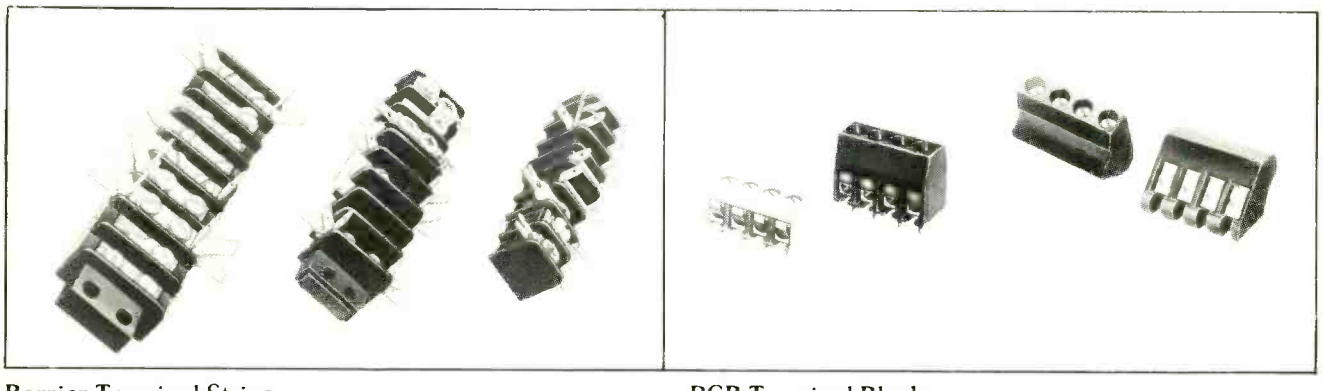

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applications, one more resistor being recommended when rapid jumps in control voltage occur, as it is stated that they may produce transient DC offset at the ouput (not that I found this complaint).

Frequency response

When working at unity gain, the relation between frequency response and gain is shown in **Fig 2**, from which it is seen that up to 20 kHz there is no difference in frequency response up to levels of +20 dBm with the feedback in the output stage giving a −3 dB point at 80 kHz at levels of +10 dBm or less.

The relation between gain and bandwidth is shown in **Fig 3** for gains of 0 dB, 20 dB, 40 dB and 60 dB with the frequency response up to 20 kHz being completely flat up to gains of 40 dB with a gain bandwidth product of about 8×10^6. This means that 50 dB gain can reasonably be used in audio applications.

Gain control

Two samples of the VCA were evaluated, both giving an output clipping level of +20 dB.7 V with ±12 V rails with the output offset voltages being −1 mV and +200 μV at unity gain and shifting to 5 mV at 50 dB gain.

As is to be seen from **Table 1** the linearity of the gain control law was amazingly accurate when using the pin 3 input.

These figures are very close to the nominal control law of 6 mV/dB and show that excellent stereo matching can be obtained if this performance is typical.

Using the inverse control law at pin 2 gain control input the law was also excellent, but slightly different, at 6.04 mV/dB.

Noise

Noise was measured by various methods in the outputs for various gains and found to perform as shown in **Table 2** with no significant signal related noise breathing effects or other objectionable subjective qualities. The table shows that an excellent dynamic range is available bearing in mind the output capabilities.

Distortion

Second harmonic distortion at 1 kHz could be nulled to be 0.05% at lower levels by careful adjustment of the symmetry control with both the second and third harmonics up to 20 kHz input frequency (higher harmonics being effectively absent) remaining at less than 0.01% at 0 dBm input and any gain setting.

At +20 dBm input and unity gain the performance is shown in **Fig 4**, where harmonic products remain below 0.03% up to 20 kHz, rising sharply above 50 kHz.

Similarly, the intermodulation distortion performance to the CCIF twin-tone method was good, the typical performance for the second and third order products being shown in **Fig 5**, for 0 dBm output at either unity gain or −20 dB gain.

Summary

This is a very attractive VCA offering attenuation or gain over a wide range with an incredibly accurate control law.

Overall the performance as measured on the two samples was very good and a particular attraction of this VCA is its small size and the requirement for very few external components.

The final incentive is a reasonable price and a single control for quick alignment.

Hugh Ford

FIG.2
dbx VCA
UNITY GAIN-RELATIONSHIP BETWEEN FREQUENCY RESPONSE AND GAIN

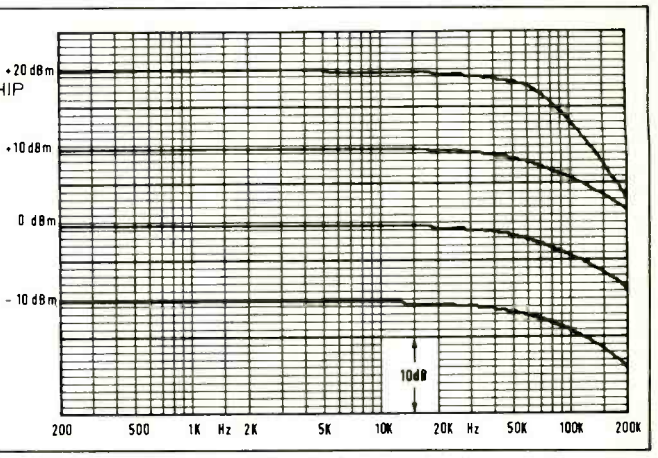


FIG.3
dbx VCA
RELATIONSHIP BETWEEN GAIN AND BANDWIDTH

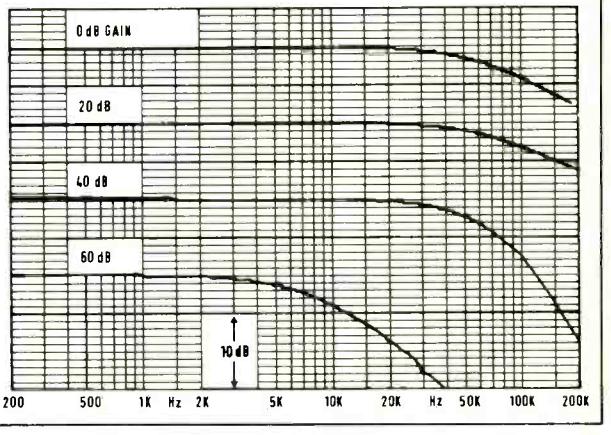


FIG.4
dbx VCA
HARMONIC DISTORTION WITH +20dBm INPUT AND UNITY GAIN.

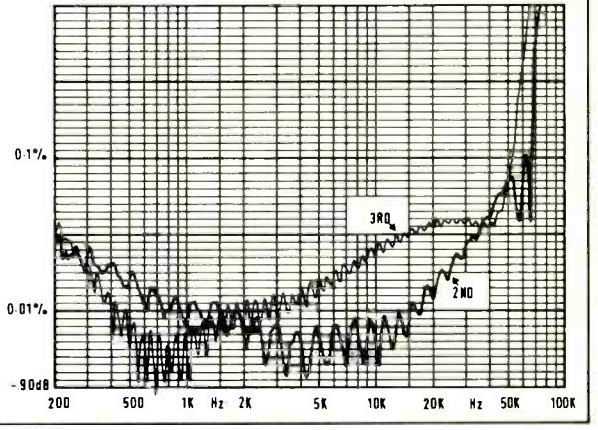
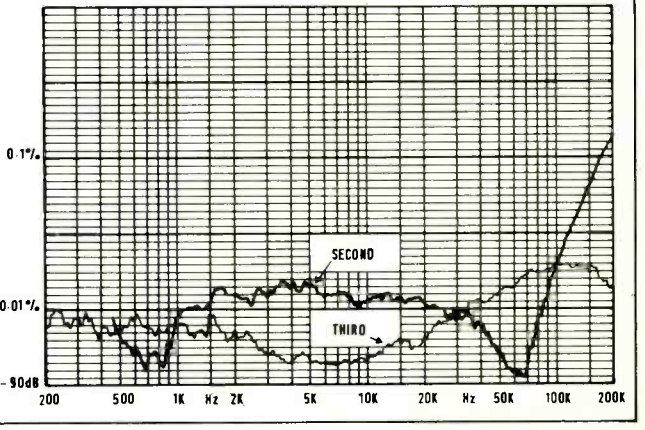
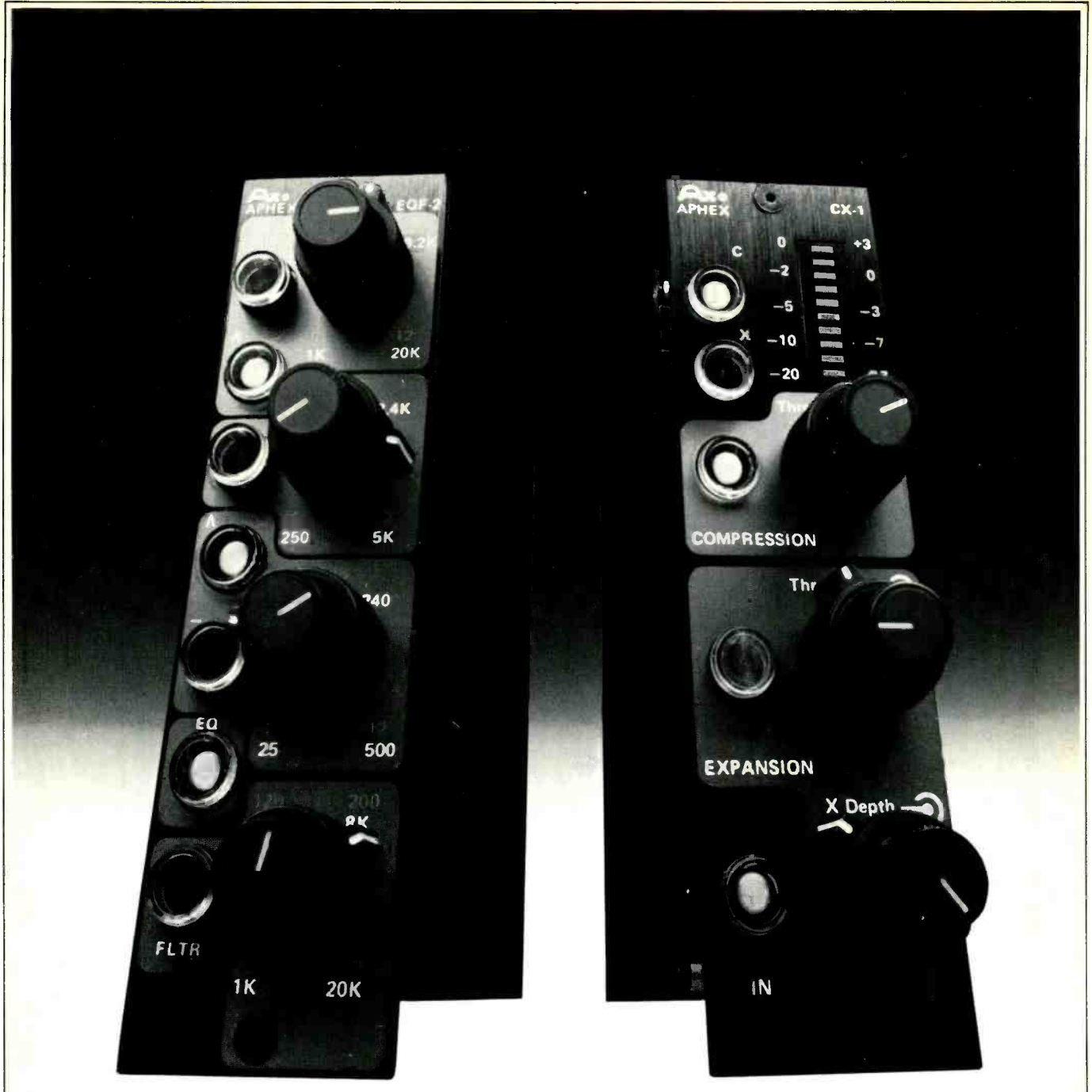


FIG.5
dbx VCA
INTERMODULATION DISTORTION FOR 0dBm OUTPUT AT UNITY OR -20dB GAIN.





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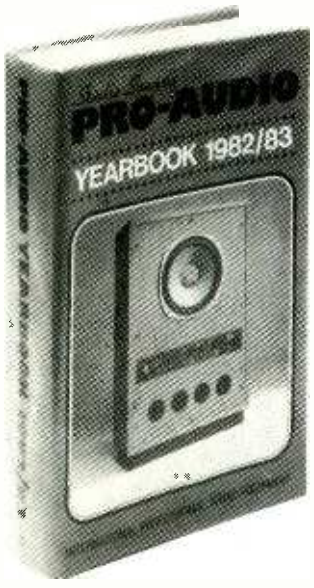

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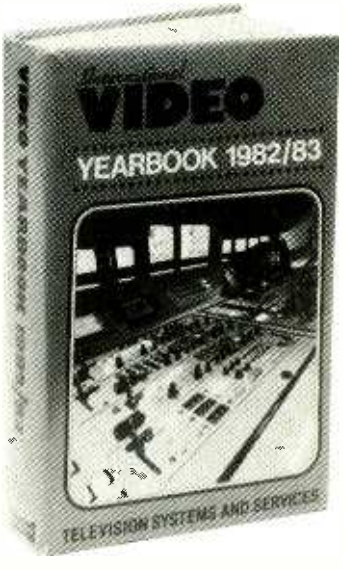

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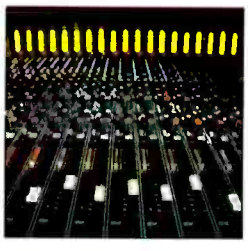
The MR-3 is supplied complete with integral patchbay and varying frame sizes to accommodate up to 56 input channels.

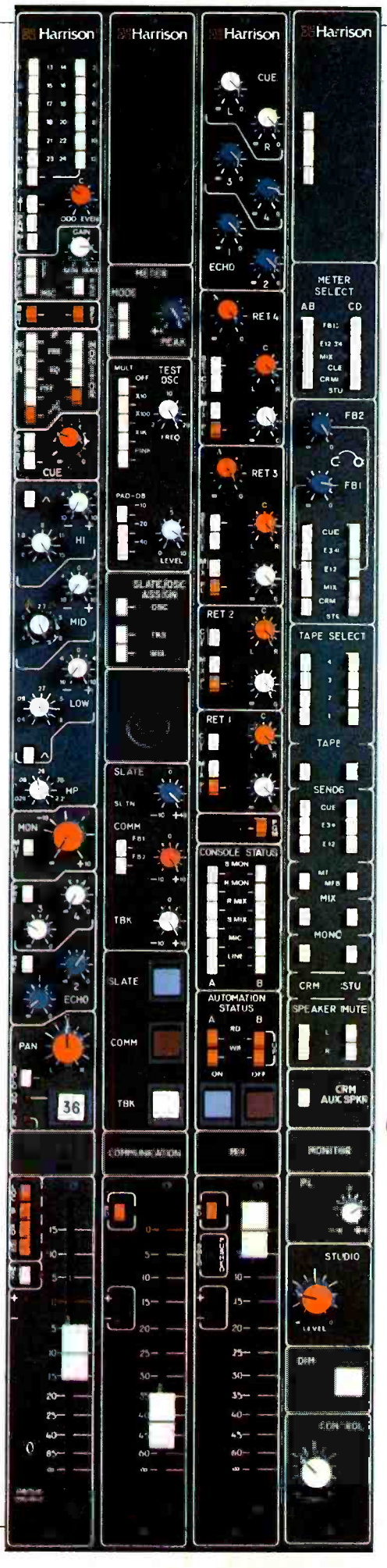
Each input module has a full 24-track output-assign matrix and three bands of parametric E/q, with a high pass filter and optional variable 'Q' on each band.



In addition, each module offers six auxiliary sends and a direct assign button for multitrack recording.

Major console status changes are effected with one-button ease. Six modes of operation are available including two new statuses for broadcast and video post production.

The standard VCA faders enable the operator to establish VCA groups when recording and mixing. The console is prepared for rapid installation of three proven automation systems: Melkuist, Allison and Harrison's own Auto-set.



The Harrison MR-3. High on Features, Low on Price.

You can never afford to buy cheaply. What you must do these days is buy economically.

Recognising this, Harrison has developed the MR-3 music desk. High on impressive features, but not on cost, the MR-3 represents excellent value and a unique opportunity to invest in Harrison performance.

In other words the MR-3 guarantees maximum efficiency in 24-track recording.

The Harrison MR-3. Underpriced it may be, undermade it's not.

To find out more about the Harrison MR-3 contact F.W.O. BAUCH at the address below.

Harrison

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