studio sound

AND BROADCAST ENGINEERING

January 1984 £1.20

A LINK HOUSE PUBLICATION

Effects
Equalisers
Reverberation

Pretty Impressive

Over the past couple of years our Series 2400 studio mixing consoles have been impressing more and more people. So many, in fact, that over 200 consoles have been installed already in studios throughout the world.

Studio owners have found that the 28/24 version of the Series 2400 is the most cost-effective way to get 52 equalised inputs during mix down.

While their clients love the flexibility of the split console design, with its logical format and easy to follow signal paths.

All sound professionals are impressed by our attention to functional detail. Like the three solo modes, two programmable mute buses, six cue sends and of course semi-parametric equalisation on input and monitor channels. And the new optional Soundcraft Automation system makes Series 2400 even more versatile.

Richard Elen of Studio Sound recently described the Series 2400 as "a very impressive console that is a pleasure to use."* Which is praise indeed.

So it's no surprise that the Series 2400 is the most popular 24 track studio console in Europe. And that's a pretty impressive achievement—even for a Soundcraft.

*Studio Sound – September 1983. Reprints available.

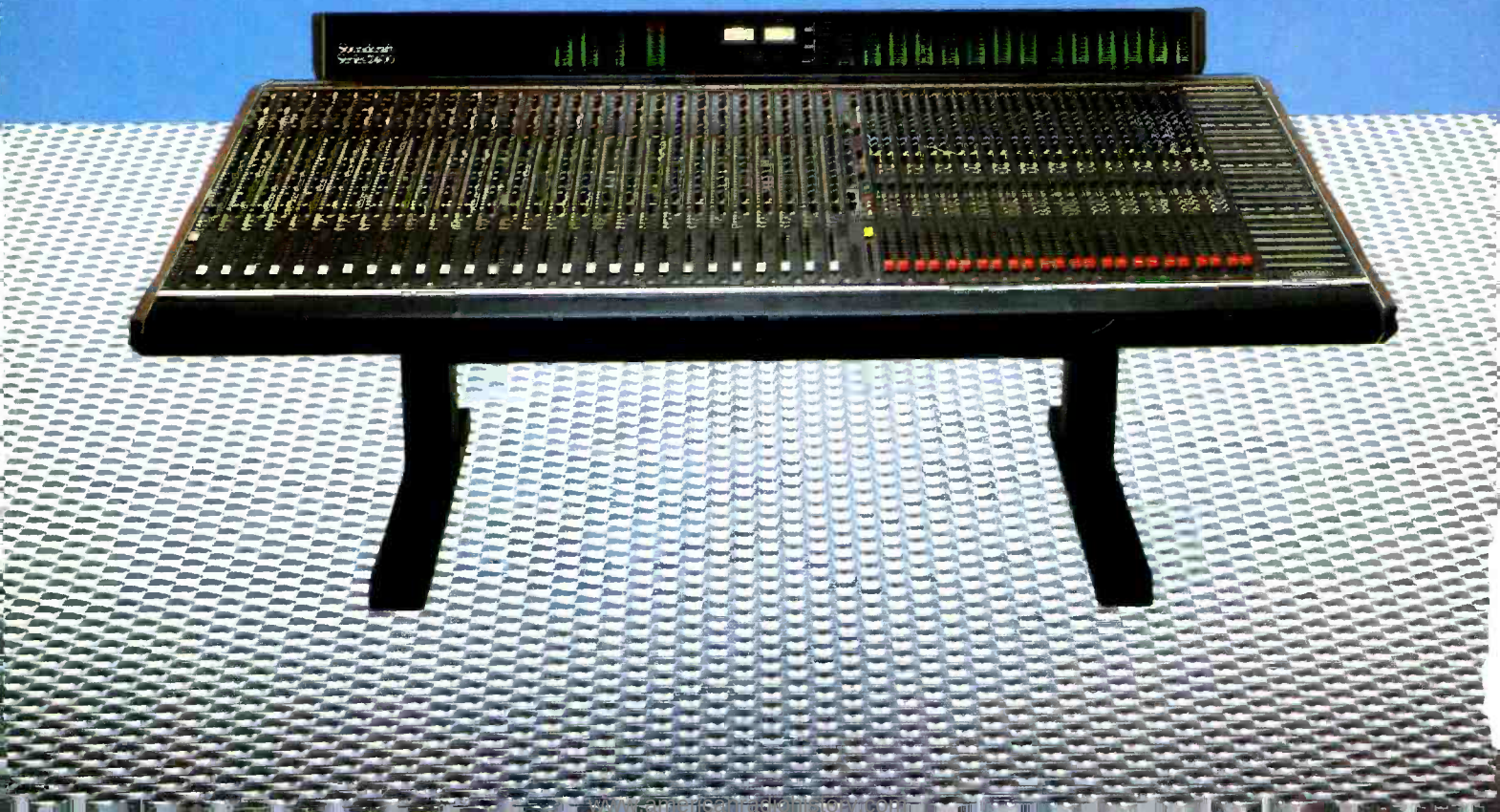
Soundcraft

The Professional's choice

Soundcraft Electronics Limited, 5-8 Great Sutton Street, London, EC1V 0BX, England. Tel: 01-253 6988 Telex: 21198.

Soundcraft Electronics USA, 1517 20th. Street, Santa Monica, California 90404. Tel: (213) 453 4591. Telex: 664923.

Soundcraft Canada Inc. 1444 Hymus Blvd. Dorval, Quebec Canada H9P 1J6. Tel: (514) 685 1610 Telex: 05 822582.



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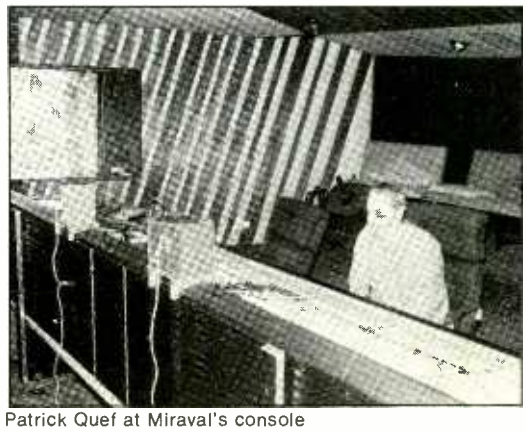
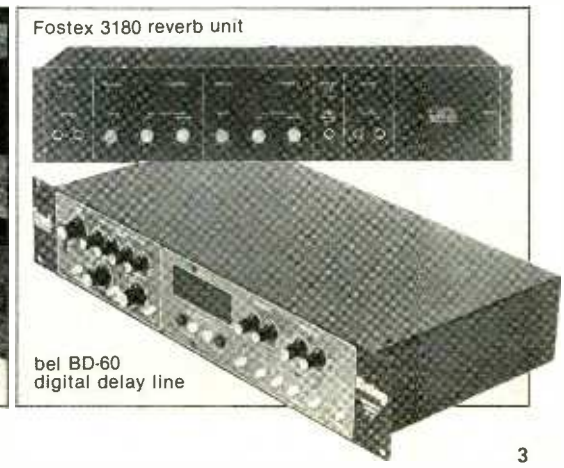


Patrick Quef at Miraval's console



Fostex 3180 reverb unit

bel BD-60 digital delay line

Advice, instruction, back-up support. These are the key words in HHB's digital service philosophy.

We feel that with technology this advanced and innovative, these factors are more important than ever before.

We keep a full range of Sony digital audio products, from the low-cost PCMF1 processor to the DAE1100 editor and PCM1610 processor needed to make your CD compatible masters, and of course the CDP101 Compact Disc player.

Additionally, we can supply the video recorders you need, in both U-matic and Betamax formats, and advise you on which machines suit your needs best.

Our comprehensive rental service now includes all these products, delivered and installed, and with full instruction on their operation from trained, experienced personnel.

We can also offer a fully-equipped in-house editing and transfer facility.

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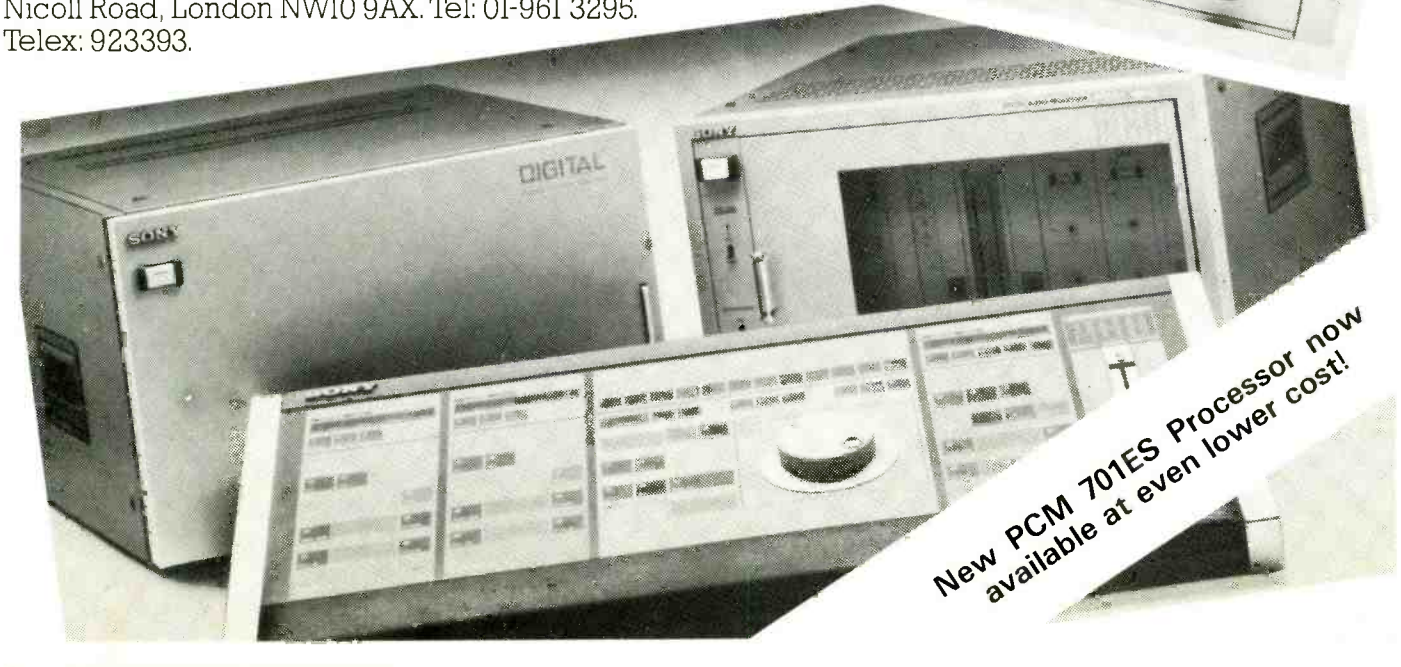

EDITORIAL

studio sound

AND BROADCAST ENGINEERING

STUDIO SOUND is published on the second Friday of the preceding month. The magazine is available on a rigidly controlled requested basis only to qualified personnel (see back page for terms) or for an annual cost of **£14.00** UK, **$40 US** surface mail, **$75** US airmail **£20.25** overseas surface mail or **£32.50** overseas airmail to non-qualifying readers or where more than two copies are required in a studio or small organisation. *All* subscription enquiries, including changes of address (which should be *in writing* and preferably including an old address label or at least the 7-digit label code) should be made to the **Subscription Department, Link House, Dingwall Avenue, Croydon CR9 2TA, Great Britain.**
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News on digital clarification

Since I wrote last month's editorial, there have been some interesting movements in the direction of an accepted standard for the labelling of records which include digital audio techniques in their production. It has been suggested that RCA will state on appropriate CDs 'digitally mastered from an analogue source', and that PolyGram are reconsidering their labelling policy.

At the recent digital audio seminar held in London jointly by the APRS and the music industry trade weekly *Music Week,* Chris Stone, chairman of SPARS (the US equivalent of APRS) put forward their carefully-considered view which they have been proposing to the US record industry for several months. (I discussed these proposals last month: the recording process is split into three parts, recording, mixing and cutting, and the three stages are labelled A or D as appropriate. So ADD would be an analogue multitrack recording, digitally mixed and released on CD, for example. In discussion with Chris Stone it quickly became obvious that my own suggestion of a 4-character label was too complex for the consumer at this time, although thought should be given to allowing the classification system to be expanded in the future to encompass new developments.)

Chris Stone generated a good deal of comment and results include the facts that both APRS and the BPI will be considering the proposals for a 'Digital Audio Recording Code' as one might call it. The APRS is considering the SPARS DARC proposals with a view to endorsing them; the BPI, as the representative of the majority of UK record companies, appears interested in the idea. PolyGram intimated that if SPARS and APRS agreed a unified code, they too would be prepared to endorse it, presumably both at the CD manufacturing and the record company level, although they would probably require some subtle changes to the exact definitions of some parts of the recording process. It would appear that UK and European record companies are more receptive to the arguments for a DARC than their North American counterparts (in the form of the RIAA), although it is notable that the US 'Compact Disc Group' (the record companies producing CDs in the States) have added a couple of paragraphs in their latest CD catalogue contrasting the 'very good' quality which analogue-derived CDs offer with the 'very best' that all-digital CDs can achieve.

It may be that British companies are worried about the possible invocation of the Trades Description Act and similar legislation in other European markets, particularly when it comes to 'digital re-mastering' (the practice of copying analogue masters to digital and cutting conventional lacquers from the digital copy so as to cash in on the enthusiasm of 'digital fetishists' in the record-buying audience). However, it is possible that US advertising legislation may be interpreted as covering the label on a product as well as advertising for that product, which would put American record companies in the same boat as their European counterparts.

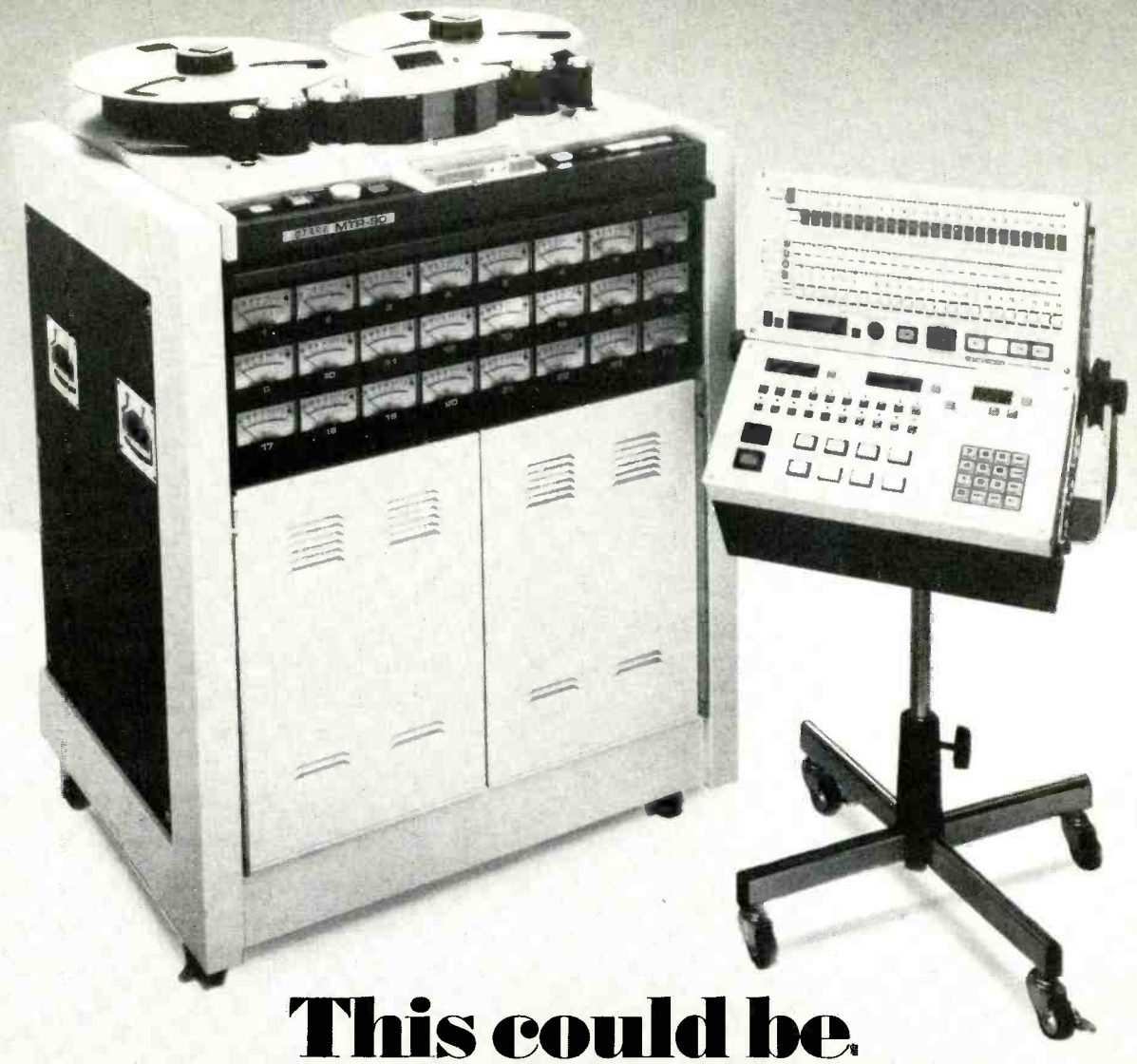
When it comes to 'digital remastering', there are some interesting developments going on. Roger Lagadec of Studer demonstrated a digital noise reduction system at the recent AES Convention in New York: you put your noisy old analogue master in one end and out it comes, pristine and neatly digitised into 16-bit PCM. . . with 30 dB or so of noise removed! The system divides the audio into 512 bands and analyses them, subtracting the noise components, rather like an audio equivalent of the systems NASA use to enhance video images from space probes. So far it is a mainframe simulation and it isn't perfect but if Dr Studer can be persuaded to manufacture it, it could revolutionise audio archiving and do marvellous things to more recent recordings.

Then there is the field of digital signal processing. With a digital console, it is possible to process an already-digital signal (say a digital master or a master which has been stuffed through Roger's box) without leaving the digital environment—with no further degradation of the signal. EQ, limiting and other twiddles can be done to the tape to be released on CD, in much the same way as it is today possible to EQ, compress and otherwise treat analogue masters in the cutting room. I can well see cutting rooms getting into 'Digital Audio Post-Production', as they are going to be the obvious DAPP centres for CD production, CD manufacture being basically an exact copy, soundwise, of the original master supplied to the factory.

With digital signal processing in the cutting room, too, analogue cuts can be 'sweetened' without losing the quality edge of a digital master. Obviously this is why central London cutting room, Tape One, is purchasing a Neve *DSP.* And the *DSP* plus Lagadec's box could mean cleaner 'old masters'—and new ones too. When that, happens, 'digitally remastered' on a record will mean something—but not before then.

Richard Elen

The OTARI MTR90-II from ITA



This could be in your studio tomorrow!

At ITA we know Otari better than anyone, and we know that the best way to sell these superb machines is to let you, as a professional user, try one yourself in your own working environment.

We'll deliver an MTR90-II to you tomorrow and install it in your studio. Here's what we'll want you to do:

EXAMINE IT: Apart from the obvious very high standard of construction, you'll be most impressed with the thought that has gone into the totally modular design. You'll quickly realise just how simple servicing and alignment would be.

RUN SOME TAPE ON IT: You'll find there are no clonks or bangs with the MTR90. No Pinch rollers. No wild accelerations. Just the smoothest possible tape motion, with constant tension in all modes, made possible by the MTR90's advanced computer-based control system.

MEASURE IT: You'll be impressed with how flat the response is. Try the sync response. You'll barely be able to see any difference. Put a squarewave in and see that what comes out still looks like a squarewave. And you'll find that the noise performance of the MTR90 can only be bettered by digital recorders costing many times the price.

FINALLY, USE IT: Work it hard. Use it in the most critical situations you can find. It won't disappoint you. The MTR90's superior transient response will brilliantly bring out those difficult electronic instruments that always sounded muddy. Perform drop-ins, drop-outs and spot erasures that you never dared attempt before. Need to sync to video? the MTR90's extremely responsive servo control makes it the perfect choice for this application. Lastly — you'll find out just how easy and logical the actual operating is, with the MTR90's remote and autolocator.*

ITA have been closely involved with the MTR90 since its introduction in 1979. We are the logical OTARI source for the true professional who expects the best sales and service backup to go hand-in-hand with the best multitrack on the market.

Call Mick Boggis on 01-748 9009 for more information on the MTR90 or any of the other excellent products in the OTARI range.

*Autolocator and stand optional accessories.

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PROFESSIONAL PRODUCTS PAGE

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We are the sole agents for the N.S.F. echo plate. This new British-made plate offers real reverberation at a fraction of the cost of other plates. Its compact size and solid casing mean it can be installed in situations that would normally preclude a plate. Remote control of decay time supplied as standard. If you're fed up with the boinging and twanging that your spring makes, check this one out!

KORG SED3000 PROGRAMMABLE DELAY

This device has created a bit of a stir. It's a digital delay line offering over a second of high quality delay with no bandwidth restrictions and has several useful and unusual features. Selectable input and output levels, high and low pass filters, choice of modulation waveforms including RANDOM, plus an envelope follower coupled to the modulator that allows the modulation to be triggered by the envelope of the input signal. There's also infinite hold, which can be synced to an external source, and on top of all this, the whole device is fully programmable! This means that up to 9 sets of front panel control settings can be stored in internal non- volatile memories. Definitely one of the most impressive devices we've encountered for a long time.

OTARI MX5050B2-II

The alternative to Revox. The 5050B was the best-selling half-track in the USA, and now its successor, the B2-II, has taken over that position. Read the "hands-on" review in September '83 Studio Sound to find out why.
A very robust and hard-working machine, packed with sensible features, and giving excellent results.

T.C. ELECTRONICS PARAMETRICS

Parametrics can reach those parts of programme material that mere graphics can't!
We now stock this range of excellent parametric equalisers from Denmark. Three models are available: the TC1220 is a 2-band stereo unit, the TC1140 and TC2240 are mono and stereo 4-band units respectively.
Most competitively priced, all models offer a range of tuneable overlapping centre frequencies and continuously variable bandwidth. 0dBm operation with XLR connectors.

TASCAM 234

A new four-track cassette recorder from Tascam, giving a different angle on the Portastudio format. Mic and line inputs on each channel, full sync facilities, dbx, stereo monitoring, shuttle operation, varispeed . . . a nicely presented unit packed with features. Use it either stand-alone or with a mixer to make a more flexible system.

DBX NOISE REDUCTION

You don't have to spend several thousand pounds to have noise reduction on your multitrack. Professional Type 1 DBX works out at £110 per track, it's simultaneous, and it REALLY WORKS! 30dB of S/N improvement can be obtained and you don't have to line it up every session to critical levels. And if you're buying 16 or 24 tracks of it, we'll give an extra discount.

The products shown above are mainly recent additions to our range and represent only a fraction of the wide variety of Pro-audio products we supply.

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A mic for all reasons

The Sennheiser modular condenser microphone system provides an answer to every requirement of the modern recording engineer.

The basic element is the K 3U powering module. This can be powered by a 9-52 volt phantom supply or an internal battery and has integral variable bass attenuation. The alternative K 30A V is available for use with video cameras.

The ME 40 cardioid capsule has a smooth extended frequency response and near ideal polar pattern making it suitable for any application.

The ME 80 is a super-cardioid capsule. Its tight pattern and high output make it the right choice for "live" situations, interview work and p.a. systems.

The system has recently been extended by the addition of two new products. The MKE 212 acoustic boundary microphone gives a clean, balanced, natural sound that adds a new dimension to sound recording.

The MKE 2 is a subminiature tie-clip microphone. Using the same capsule as the MKE 212 it provides a clear, crisp sound from a package only 5mm. (dia.) x 10mm.

For full details of Sennheiser products and our guide "How to choose, use and abuse microphones" please contact us on 0753 888447

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ITA THE ONE SOURCE for 8-track.

At ITA we can always offer the most attractive deals to purchasers of complete systems. At the moment we are offering a choice of 3 systems to those ready to go 8-track, all at special discount prices.

OTARI Mk III-8 and ITAM 12.4.2 mixer

The Itam 12.4.2 was designed specifically to accompany the Otari, and this combination of equipment would form the basis of a really top-quality facility for demo, A-V, or video post-production work.

The Mk III-8 is still the best ½" 8-track around, and offers all the features and performance you expect from a studio multitrack. Full syncroniser interface as standard, all functions (including audio) remotable, and an autolocator is now available.

The Itam 12.4.2 has a level of performance and a standard of construction that does the Otari justice. 3 aux. sends, 3-band eq with sweepable mid, full 8-track monitoring, much more. A very popular system.

SPECIAL PRICE—£3,900

TASCAM 38 with ITAM noise reduction and P.E.P. 1200 mixer

A nice mid-range system for the serious songwriter and smaller demo studios.

When the Tascam 38 was introduced last year, it set new standards in cost-effectiveness. Combined with the Itam DB30/8 simultaneous noise reduction, excellent results can be obtained.

The PEP 1200 is a budget 10/4 mixer with a tremendous range of facilities for the price. Full 8-track monitoring and metering are fitted, there are 2 aux. sends and 4-band eq on each channel.

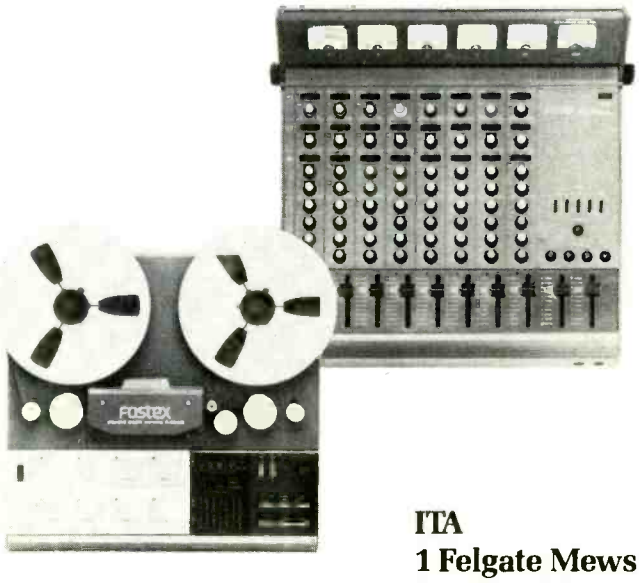
SPECIAL PRICE—£2,250

FOSTEX A-8 with 350/3060 mixer

Many home recordists prefer to stick to equipment from one manufacturer. Fostex cater for this by having a range that covers a lot more than just recorder and mixer; there are all kinds of outboard effects too. We're offering the most popular items in the Fostex range as a special offer.

The Fostex A-8 is still the world's smallest 8-track, but many users consider its performance at least the equal of some ½" systems. Combined with the 350 mixer/3060 meter bridge combination, as Fostex intended, a very compact, neat, yet powerful system is the result.

SPECIAL PRICE—£1,350



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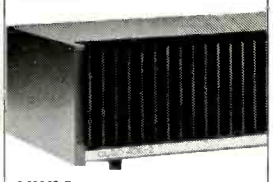
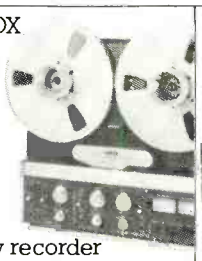
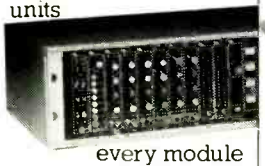
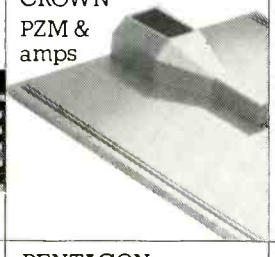
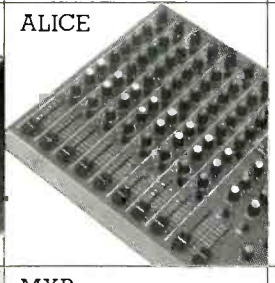
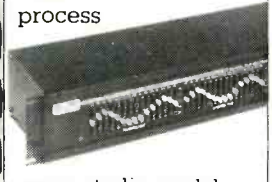
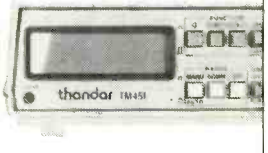
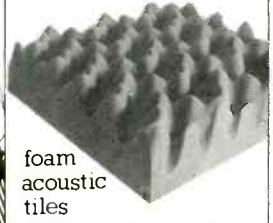
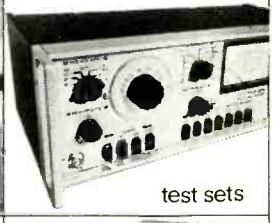
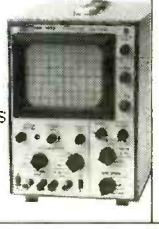
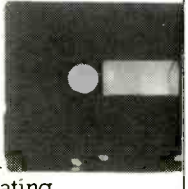
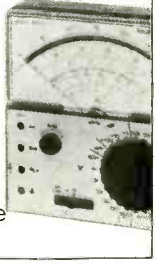
AKG SENNHEISER dynamic condenser lavaliere 	QUAD  amps speakers every model	REVOX  B77 PR99 B710 every recorder	DBX noise reduction compressors  every model	TANNOY  monitor speakers 7 models
REBIS rack effects units  every module	CROWN PZM & amps 	AURATONE monitors  3 models	ALICE 	SPENDOR  BBC broadcast monitors
D&R  signal processors reverbs every model	PENTAGON  digital & audio cassette duplicators	AMPEX AGFA WEBBER  audio & video test tapes & cassettes	MXR signal process  every studio model	THANDAR  7 models
CABLES  Connectronics multiway mic speaker power	ACOUSTILES  foam acoustic tiles	LEADER  oscillators chart rec scopes millivolt DVM etc 16 models	BULLDOG  leads 70 versions	FERROGRAPH  test sets
G.W. INSTRUMENTS  scopes oscillators DVM's UVM's etc	PLUGS  adaptors sockets audio video	DE MAGNETISERS  Ferrograph cassette Teac	I.T.D  audio digital duplicating 20,000 in stock	PANTEC  digital & analogue meters

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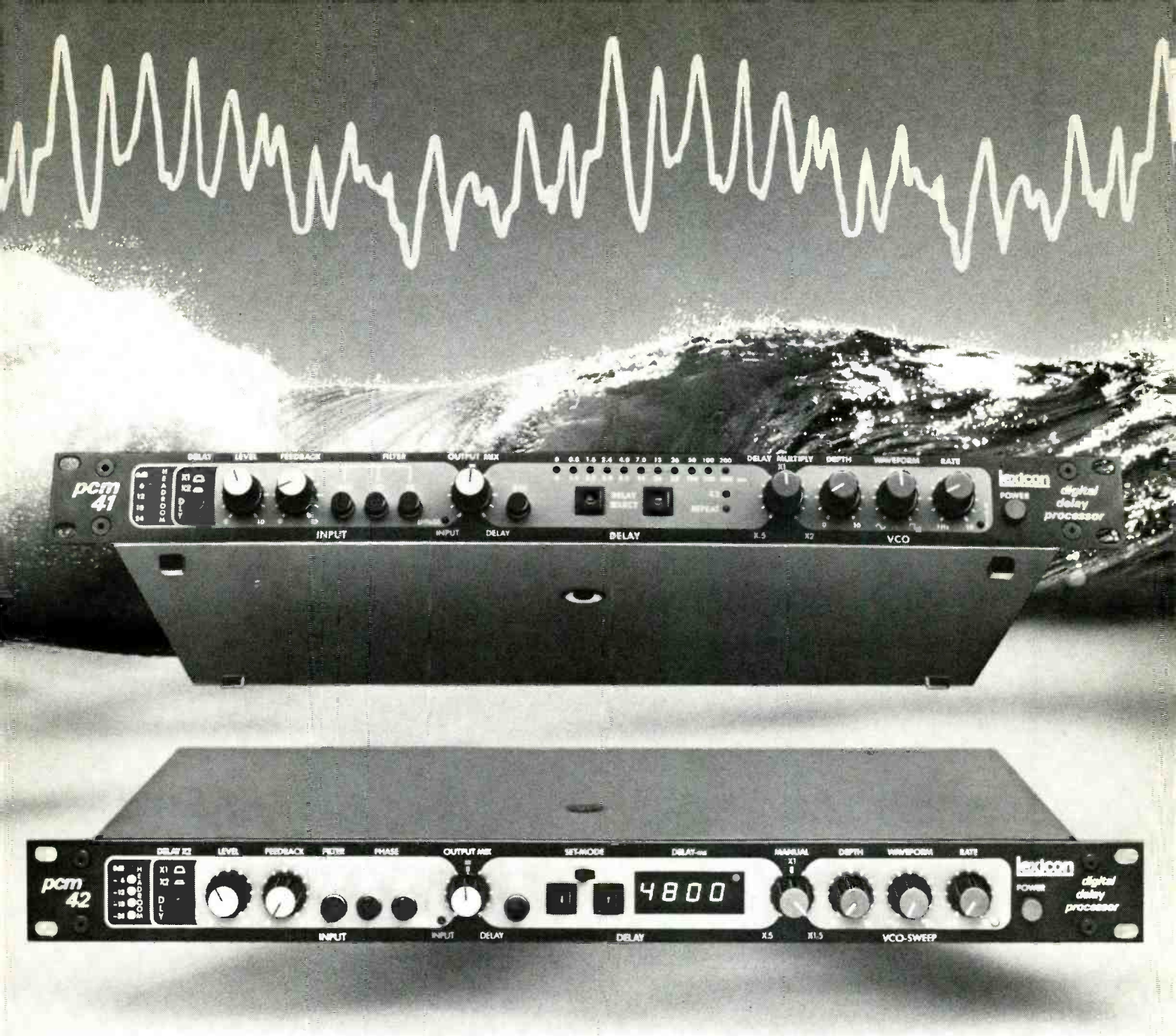

The Image Builders.

The PCM series of Lexicon digital delay systems has revolutionised the creation of dramatic sounds on stage with studio precision, enabling live performers and recording artists to explore new dimensions of musical expression.

Both systems expand the horizons for digital audio processing, drawing on the Lexicon technology which is the first choice of world-class recording studios and entertainers.

The compact, cost-effective PCM41 puts full professional quality within the reach of almost all entertainers and small studios. Effects include automatic double tracking, flanging, slap echo, infinite repeat, doppler pitch shift and many others. Virtually distortion free over the entire power bandwidth, the PCM41 is easy to understand and simple to operate. The only modest feature is its price!

The stunning repertoire of the PCM42 has all the functions you need from a high quality delay line, enhanced by Real Time over-dubbing of musical, vocal or instrumental phrases. With a device that lets you lock delay times precisely top musical tempos, comprehensive foot pedal control for live performance, and a numeric display that tells you the exact delay time at any instant.

Lexicon PCM Series. The cornerstone for improved performance.

lexicon

International: Gotham Export Corporation
New York, NY 10014 TLX: 129269

F.W.O. Bauch Limited

49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ
Telephone 01-953 0091, Telex 27502

QUESTION:

Which range of studio monitors, apart from the new Tannoy SRM Series with SyncSource,™ provides a single point sound source for all monitoring applications?

ANSWER:

The New SRM Series with SyncSource.™ A range of time compensated single point sound source monitors from:

TANNOY®

Bringing Music Back to Life

Tannoy Ltd., 21 Canterbury Grove, West Norwood, London SE27 0PW
Telephone 01-670 1131 Telex 291065

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HHB London NW10 01-961 3295
ITA London NW1 01-724 2497
Don Larking Luton 0582 27195
Music Labs, London NW1 01-388 5392
Sigma Sound, Nottingham 0602 783306
Michael Stevens & Ptns., Bromley 01-464 4157
Turnkey, London NW9 01-202 4366
and in the **USA & Canada: Tannoy Crown,**
97 Victoria Street N, Kitchener, Ontario, Canada
N2H 5C1. Telephone (519) 745 1158 Telex 069 55328

Individually, they're amazing. Together, they're unbeatable.

Recording studios, radio and television production suites, video and A/V facilities and serious musician/composers have discovered that Valley People equipment improves productivity, offers ease of operation, requires little or no maintenance and provides excellent signal quality. Standing alone they offer unrivalled processing power and versatility: used together by the creative engineer/producer, they increase variety and control dramatically.

QHZ Quad Preamp

Provides all the equipment below with instant access to the full range of high impedance sources, such as electronic musical instruments, microphones and even hi-fi kit. The clarity of audio is unsurpassed.

QLZ Quad Mic Preamp

The low impedance unit for transparent, distortion-free recording as well as on-the-air or live performances. Each of the four inputs is adjustable from 20dB to 60dB of gain.

Maxi-Q Equaliser

Provides maximum flexibility: the seven octave range on each of the three frequency bands gives a six octave overlap. Ideally suited to pre-conditioning of signals when used in-line with the Kepex II and Gain Brain II.

Kepex II Expander/Gate

Spectacular performance in noise gating, reduction and elimination is complemented by its creative value in 'loosening' drum effects, removing excess cymbal ring or performing electronic musical effects.

Gain Brain II

Widely adjustable limiter/compressor and ducker, specially designed for production work. Major advantages include 'natural' quality of sound, flatness of VU and improved low frequency performance.

Five superior products. Each amazing. All unbeatable.

Valley People Inc., Nashville, Tenn 37204
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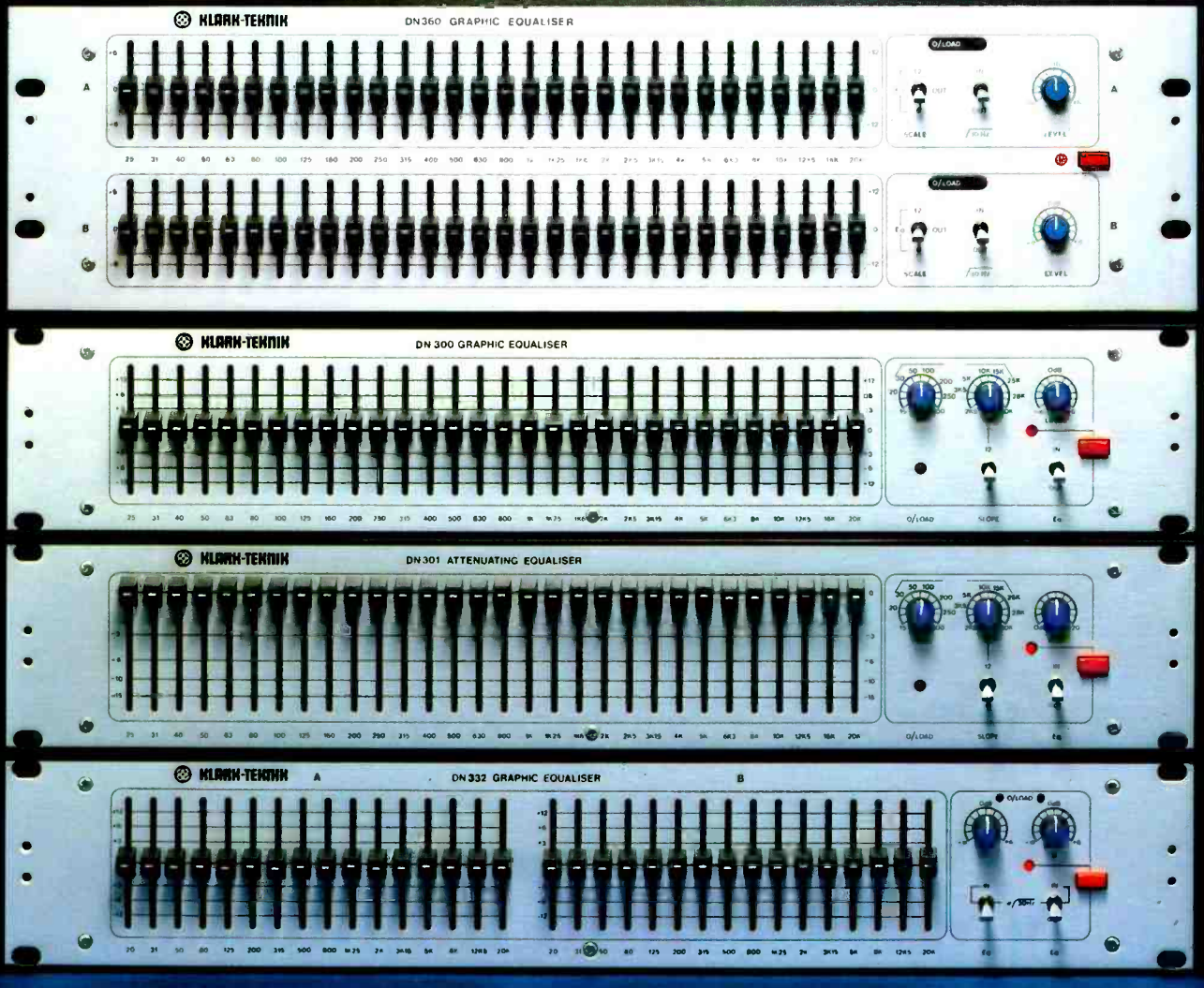
Klark-Teknik Series 300

GRAPHIC EQUALISERS

Reliable performance – at reasonable cost.

Introducing a whole new family of audio equalisers combining active-circuit design with thick-film engineering ... to reduce the component count and set new standards of reliability and performance.

KLARK TEKNIK
sound science



Dedication is the soul of good design. Klark-Teknik are dedicated to making every product a classic. *Terry Clarke*

Cutting the cost of certainty

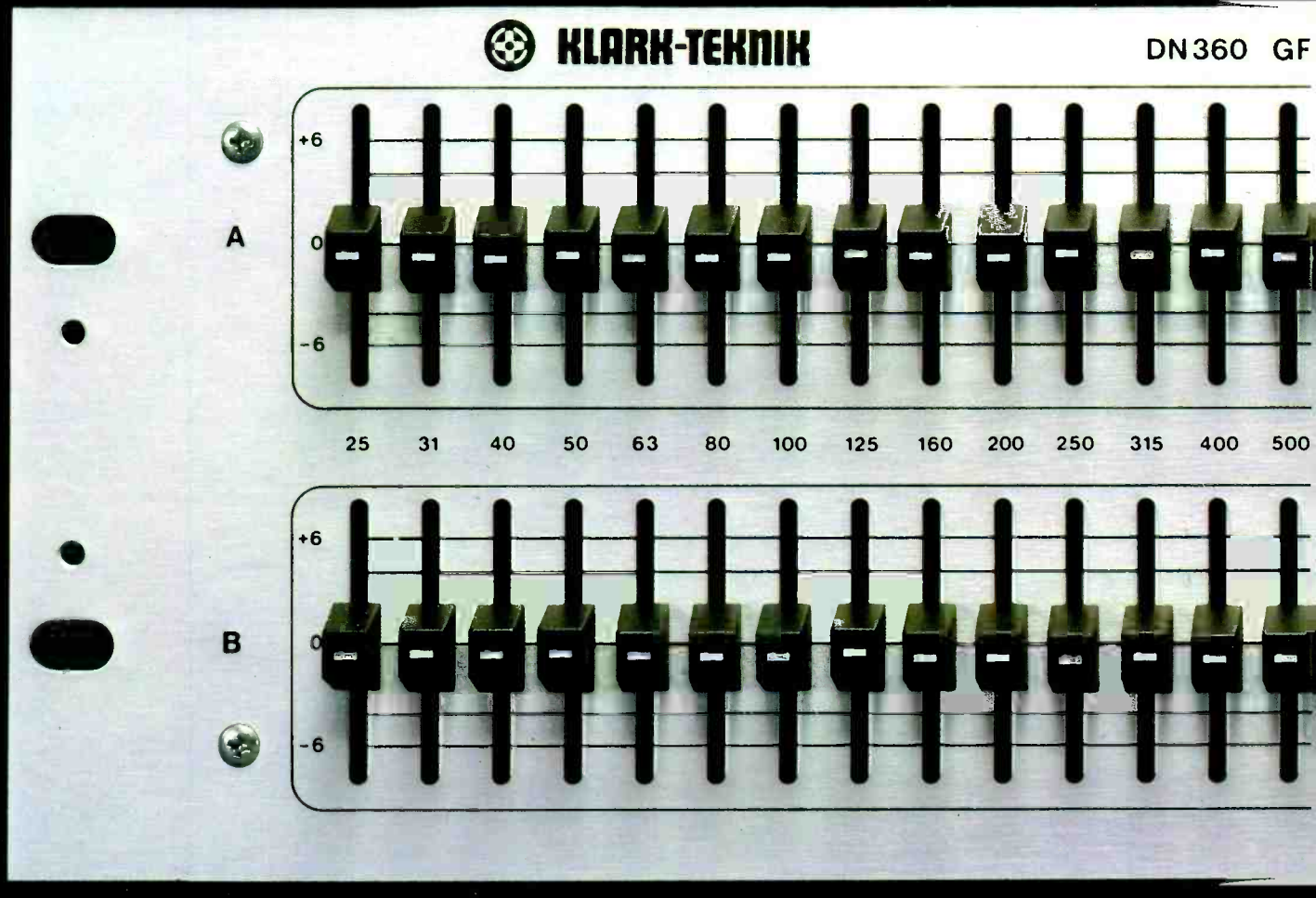
In 1975 Klark-Teknik launched the DN27 – the graphic equaliser that brought the price of high reliability and precision performance down to a new affordable level. Now our policy of constantly watching new advances in technology has led through a series of design breakthroughs to yet another generation of graphic equalisers. Every equaliser in the Series 300 from Klark-Teknik has been given extra heart by innovative use of microelectronic circuitry engineered with thick-film technology.

Result: our Series 300 equalisers don't only equal the performance of the DN27 – they actually improve on both its impressive reliability record and its unrivalled price: performance ratio. The whole family of related instruments show greatly increased capability over a wide variety of applications ranging from monitor and recording equalisation to sound system tuning. But that's not all ... Klark-Teknik have designed the Series 300 around a philosophy of maximum affordability.

FITTING INTO YOUR PLANS

How did we do it? By pursuing a policy of dedicated attention to our design objectives, we were able to fulfil the professional's need for greater control over sound. By giving attention to production engineering detail we have been able to maximise the economic as well as the performance benefits from the new technology, so that this new family of instruments fits in with your needs and your budget better than ever before. As an added benefit, all the instruments in this prolific family now fit into just 2U of rackspace – except the DN360 that fits *two whole channels* of equalisation into 3U of rack.

The photograph of the DN360 below, is ACTUAL SIZE.



RELIABILITY CONTROL!

Even with the advanced technology incorporated, these instruments are given the full backing of Klark-Teknik ***'reliability control'***, which proves each equaliser against a specification consistent with the highest professional standards. Only top quality components are used, and every unit is bench-tested and aligned before a burn-in period and final performance test.

THE INVISIBLE EXTRA

Careful design of microelectronic filter circuits and the application of thick film technology have effectively raised our previously outstanding reliability standards by a measurable amount. This makes it possible for Klark-Teknik to back every equaliser in the series with a unique ***five year warranty.****

*Parts only.

Specification

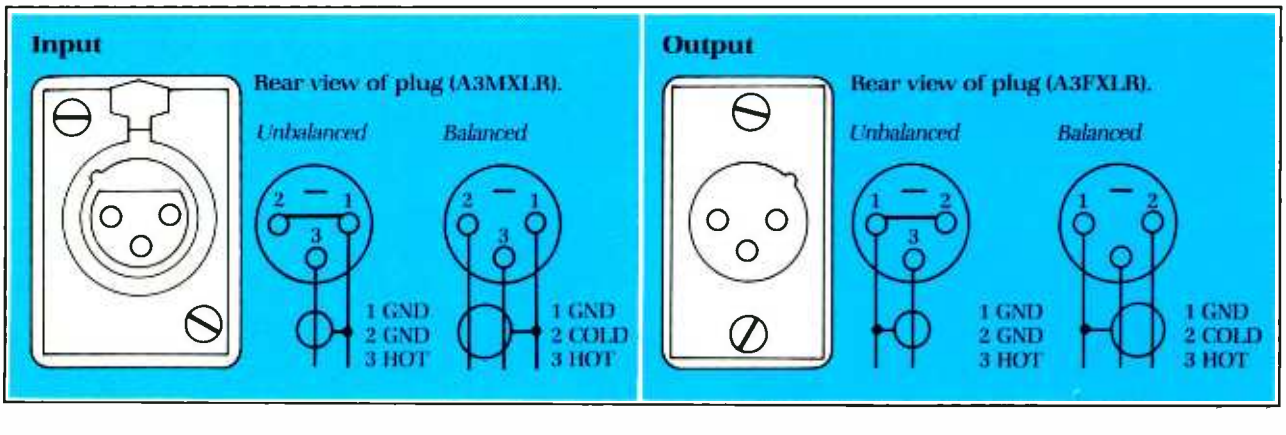
	DN360	DN300	DN301	DN332	DN27A
Input					
Electronic balancing	Balanced	Balanced	Balanced	Balanced	Unbalanced
Impedance (ohm)					
Balanced	20k	20k	20k	20k	10k
Unbalanced	10k	10k	10k	10k	10k
Output					
Type	Unbalanced	Unbalanced	Unbalanced	Unbalanced	Unbalanced
Min. load impedance	600 ohm	600 ohm	600 ohm	600 ohm	600 ohm
Source impedance	<60 ohms	<60 ohms	<60 ohms	<60 ohms	<60 ohms
Max. level	+22dBm	+22dBm	+22dBm	+22dBm	+22dBm
Frequency response (20Hz-20kHz) Eq out	±0.5dB	±0.5dB	±0.5dB	±0.5dB	±0.5dB
Eq in	±0.5dB	User defined	User defined	±0.5dB	±0.5dB
Distortion (@+4dBm)	<0.01%@1kHz	<0.01%@1kHz	<0.01%@1kHz	<0.01%@1kHz	<0.01%@1kHz
Equivalent input noise (20Hz-20kHz unweighted)	<−90dBm	<−90dBm	<−90dBm	<−90dBm	<−90dBm
Channel separation	>75dB@1kHz	N.A.	N.A.	>75dB@1kHz	N.A.
Overload indicator	+19dBu	+19dBu	+19dBu	+19dBu	N.A.
Auto-bypass (failsafe)	YES	YES	YES	NO	YES
Gain	+6dB	+6dB	+20dB	+6dB	+6dB
Filters					
Type	MELT	MELT	MELT	MELT	LCR
Centre frequencies ISO	2×30 25-20kHz ⅓ octave	30 25-20kHz ⅓ octave	30 25-20kHz ⅓ octave	2×16 20-20kHz ⅔ octave	27 40-16kHz ⅓ octave
Tolerance …	±5%	±5%	±5%	±5%	±2%
Maximum boost/cut	±6/12dB	±12dB	−15dB	±12dB	±12dB
Subsonic filter	18dB/octave −3dB @30Hz	N.A.	N.A.	18dB/octave −3dB @30Hz	N.A.
High pass filter slope	N.A.	15Hz-300Hz 12dB/octave	15Hz-300Hz 12dB/octave	N.A.	N.A.
Low pass filter slope	N.A.	2k5Hz-30kHz 6/12dB/octave	2k5Hz-30kHz 6/12dB/octave	N.A.	N.A.
Power requirements					
Voltage	110/120/220/240V 50/60Hz	110/120/220/240V 50/60Hz	110/120/220/240V 50/60Hz	110/120/220/240V 50/60Hz	110/120/220/240V 50/60Hz
Consumption	<15 VA	<15 VA	<15 VA	<15 VA	<15 VA
Weight					
Nett	4.5kg	3.5kg	3.5kg	3.5kg	6.5kg
Shipping	7kg	6kg	6kg	6kg	8kg
Dimensions					
Width	482mm (19 inch)	482mm (19 inch)	482mm (19 inch)	482mm (19 inch)	482mm (19 inch)
Depth	205mm (8 inch)	205mm (8 inch)	205mm (8 inch)	205mm (8 inch)	205mm (8 inch)
Height	133mm (5¼ inch)	89mm (3½ inch)	89mm (3½ inch)	89mm (3½ inch)	133mm (5¼ inch)
Terminations					
Inputs	3 pin XLR	3 pin XLR	3 pin XLR	3 pin XLR	3 pin XLR
Outputs	3 pin XLR	3 pin XLR	3 pin XLR	3 pin XLR	3 pin XLR
Power	3 pin CEE	3 pin CEE	3 pin CEE	3 pin CEE	3 pin CEE

*MELT – Proprietory Microcircuit.

The whole Series 300 family of graphic equalisers comply with standard 19 inch rack mounting requirements.

As part of a policy of continual improvement, Klark-Teknik reserve the right to alter specifications without notice.

PIN CONFIGURATIONS



Options

Security covers

An optional perspex security cover is available to prevent unauthorised interference with calibrated equaliser settings in permanent sound installations.

For DN300	Order number SC30
For DN301	Order number SC30
For DN332	Order number SC30
For DN360	Order number SC36
For DN27A	Order number SC27

Transformer balancing

Retrofittable output balancing transformers:

For all Series 300 models	Order number BU37

Transformer input balancing is available on all Series 300 equalisers but must be specified with initial order.

	Order number BN37
For DN27A (In/Out)	Order number BA27

THE RIGHT PERFORMANCE LEVEL AT THE RIGHT PRICE

Series 300 equalisers are designed with inbuilt capability for a very wide range of applications – including:

In live performance – sound reinforcement, wide-band equalisation and monitor tuning.

In the recording studio – monitoring, system equalisation and 'second thoughts' track clean up.

In the motion picture industry – dialogue sound equalisation ... and B-chain equalisation in the re-recording studio.

In stereo broadcasting – announcer microphone enhancement and stereo channel equalisation to give maximum on-the-air brightness and punch.

In the discotheque to give bass with substance – and aggressive top for increased accentuation.

For contractors and others who need effective equalisation to achieve spot-on public address system intelligibility.

In the theatre for front-of-house system and grouped microphone equalisation.

Ask around, you'll find that Klark-Teknik equipment is known for its ability to take the roughest conditions with the smoothest performance, in the studio or on the road.

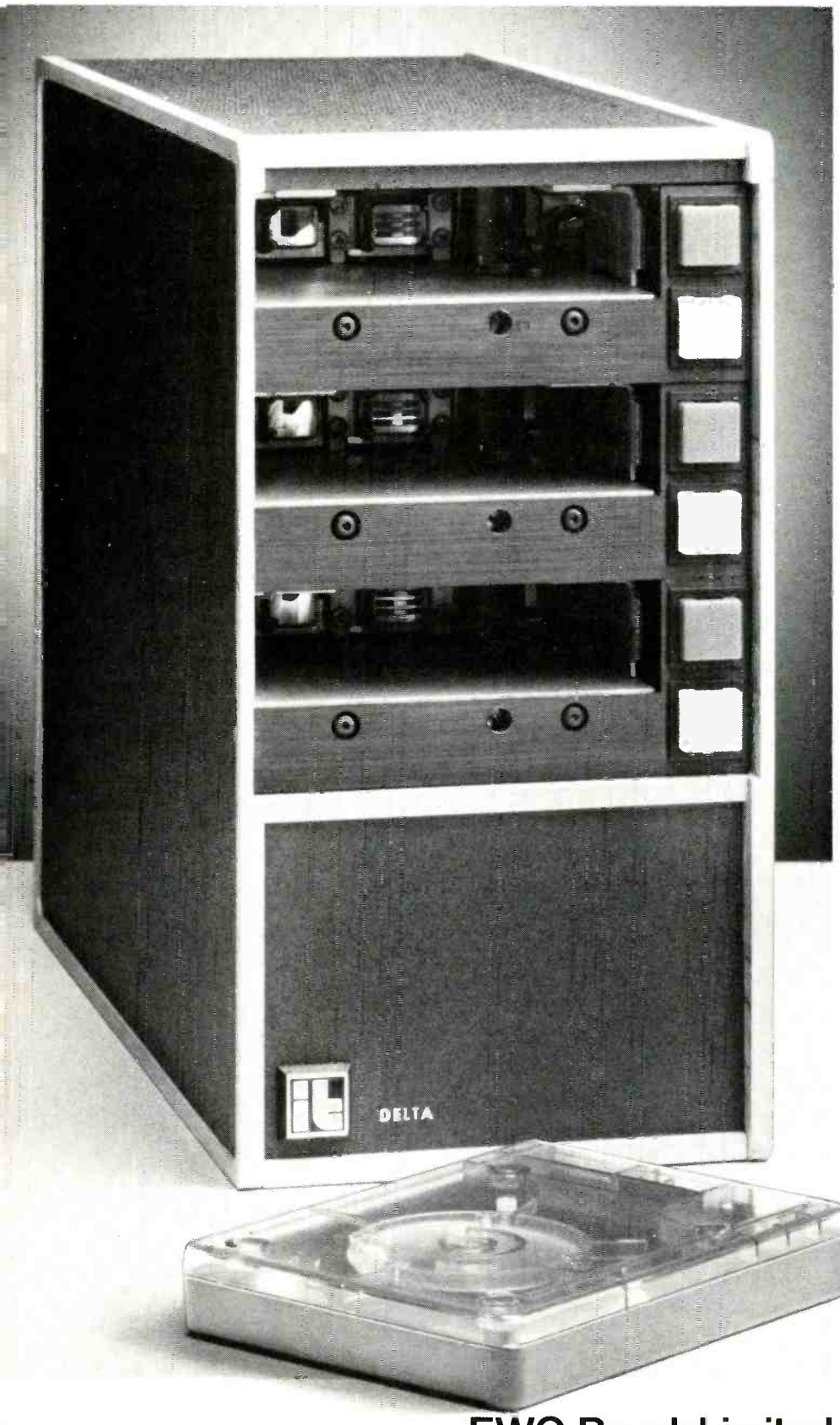
Klark-Teknik Research Limited,
Coppice Trading Estate, Kidderminster,
Worcestershire DY11 7HJ, England.
Telephone: (0562) 741515 Telex: 339821

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

Your supplier is:

Design & Production by Nicholas J. Jones Graphics, Gloucestershire, England. 83743

MAJOR BREAKTHROUGHS DON'T HAPPEN OVERNIGHT.



The new Delta three triple-deck cartridge machine from ITC gives you even more rugged reliability than its well-proven predecessors in the Premium Series, it provides improved performance, is more compact and includes a host of new operating features.

All this has been made possible by major technological advances in the ITC research and development lab. Advances which have created an electronically and mechanically superior unit. A machine that will take all the inevitable day-to-day knocks yet keep on functioning at the high standard you – and your listeners – expect.

The Delta Three has three independently-romovable decks and modular construction makes alignment and servicing simple and convenient. It is clean and smooth in operation, with minimal flutter and optimum frequency response, in mono or stereo. Insertion and removal of cartridges is positive, and there's a microprocessor-controlled digital cue tone detector. A newly designed capstan motor with its own integral "gallows" greatly reduces bearing noise and ensures stability of shaft along its entire length. Record facility on lower deck is made possible by the addition of Delta Four.

Rigorously tried-and-tested in the field, the Delta Three represents a significant breakthrough in practical technology for the studio.

ITC. Sound quality.

International Tapetronics Corp.,
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Illinois 61701, USA

F.W.O. Bauch Limited

49 Theobald Street, Boreham Wood, Hertfordshire WD6 4RZ
Telephone 01-953 0091. Telex 27502

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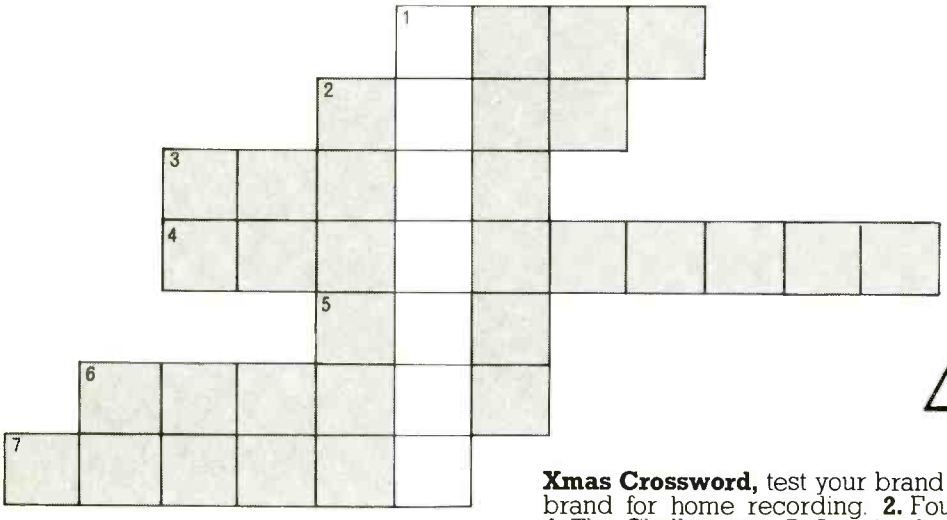
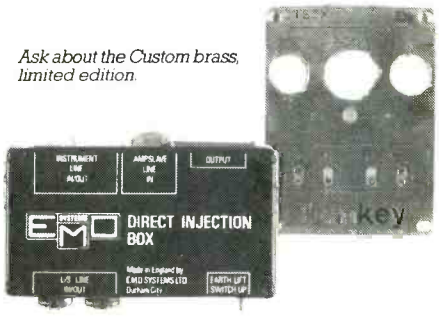
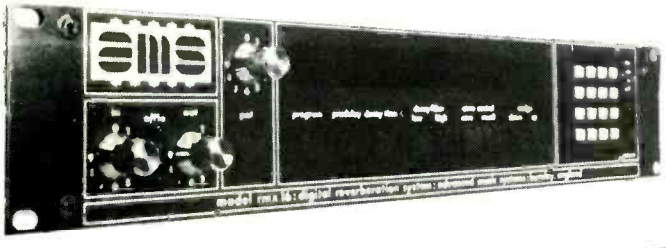


Xmas Crossword, test your brand knowledge. All clues are across. **1.** First brand for home recording. **2.** Four sided amp. **3.** The New Workhorse. **4.** The Challengers. **5.** Austrian best sellers. **6.** Personal multitrack pioneers. **7.** Dual concentrics.

Sony Digital at the price of Analogue
We carry stock of the complete Sony PCM F1 system which includes the following products PCM F1 digital processor and the SL F1 and TT F1 Betamax recorder,timer and remote control.For further details and prices call us on 01- 202 4366

Ask about the Custom brass, limited edition.





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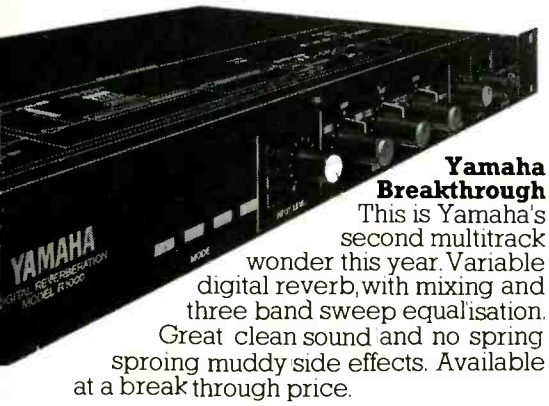
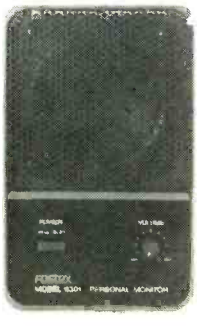


Yamaha Breakthrough
This is Yamaha's second multitrack wonder this year. Variable digital reverb,with mixing and three band sweep equalisation. Great clean sound and no spring sproing muddy side effects. Available at a break through price.

Soundcraft 400B and the Ortari 5050 package.

Microchip Monitor
A power amplifier in this efficient Fostex speaker will produce high yet accurate levels of sound from line level. Just plug in the audio and mains power.For near field monitoring, presentation, or personal studios.Call us to evaluate.



The Outstanding, Fifty Fifty Systems
Price is not the only reason why eight track is such an attractive proposition for production recording. The track count is plenty enough in many cases, easy to work with, in particular as used in self-op installations.

Packaging the Soundcraft Series 400B with the Otari 5050 eight track, shows three distinct advantages. There's the economy of half inch, the bother free reliability of both brands, and the fastest and yet most comprehensive operating features in their class. Find out more about the Fifty Fifty systems from Turnkey, from just £4,999

The Most Challenging Tascams Ever
You may have noticed that they always conquer the market with 'Series'. The recent 30's range includes a half inch eight track to replace and update their classic 80-8 (at lower cost), a two track to challenge the foothold of Revox, and their very first cost effective mixer to match. Call or write and find out more about this latest challenging 30 Series of Tascams at Turnkey.

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Quick Production Audio
The SECK Producer is a stereo, VCA controlled mixer designed for fast audio production. Fully switchable inputs, long throw faders, third output and auto functions. Rack mounting and full studio level compatability. Ask for the 16 page Operators Manual now available.

Ask for a copy of the 16 page operators Manual

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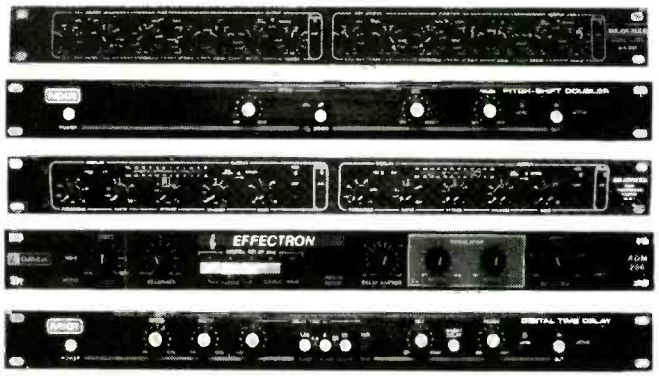
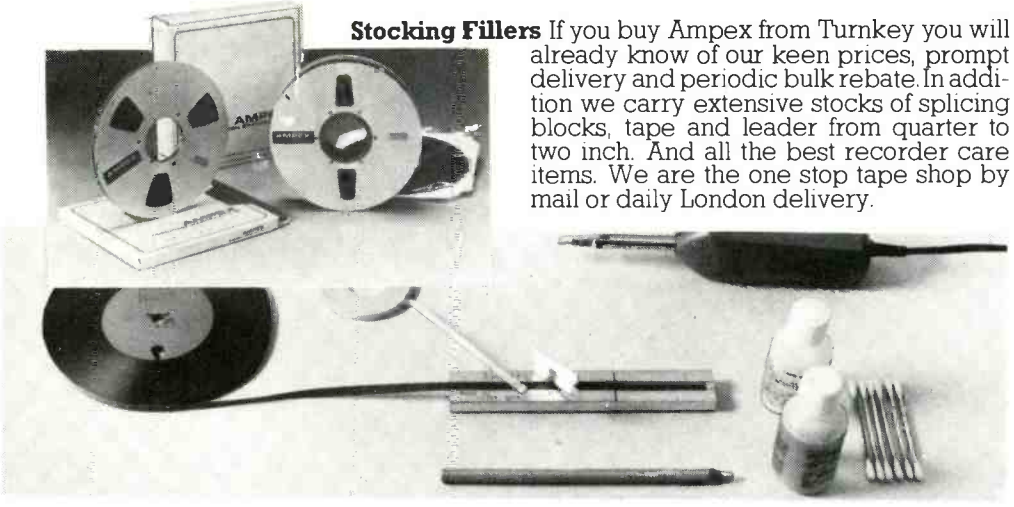
Drawmer Lim/Com every dynamic control facility for tight punchy sound
Effectron One second delay plus all the time wrap effects available from the Deltalab experts
MXR 175 The new long delay from the master of musicians studio effects



Service Calls
The expanded workshop capacity at our new premises means we offer service on a wider range of pro audio products. As our engineers specialise in different products, please name your brand of product to our receptionist when you call.
Duncan Crundwell
Manager Technical Support



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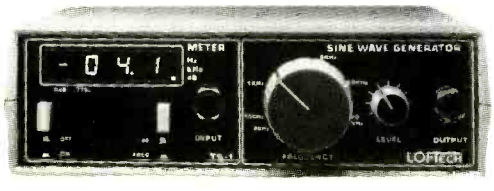

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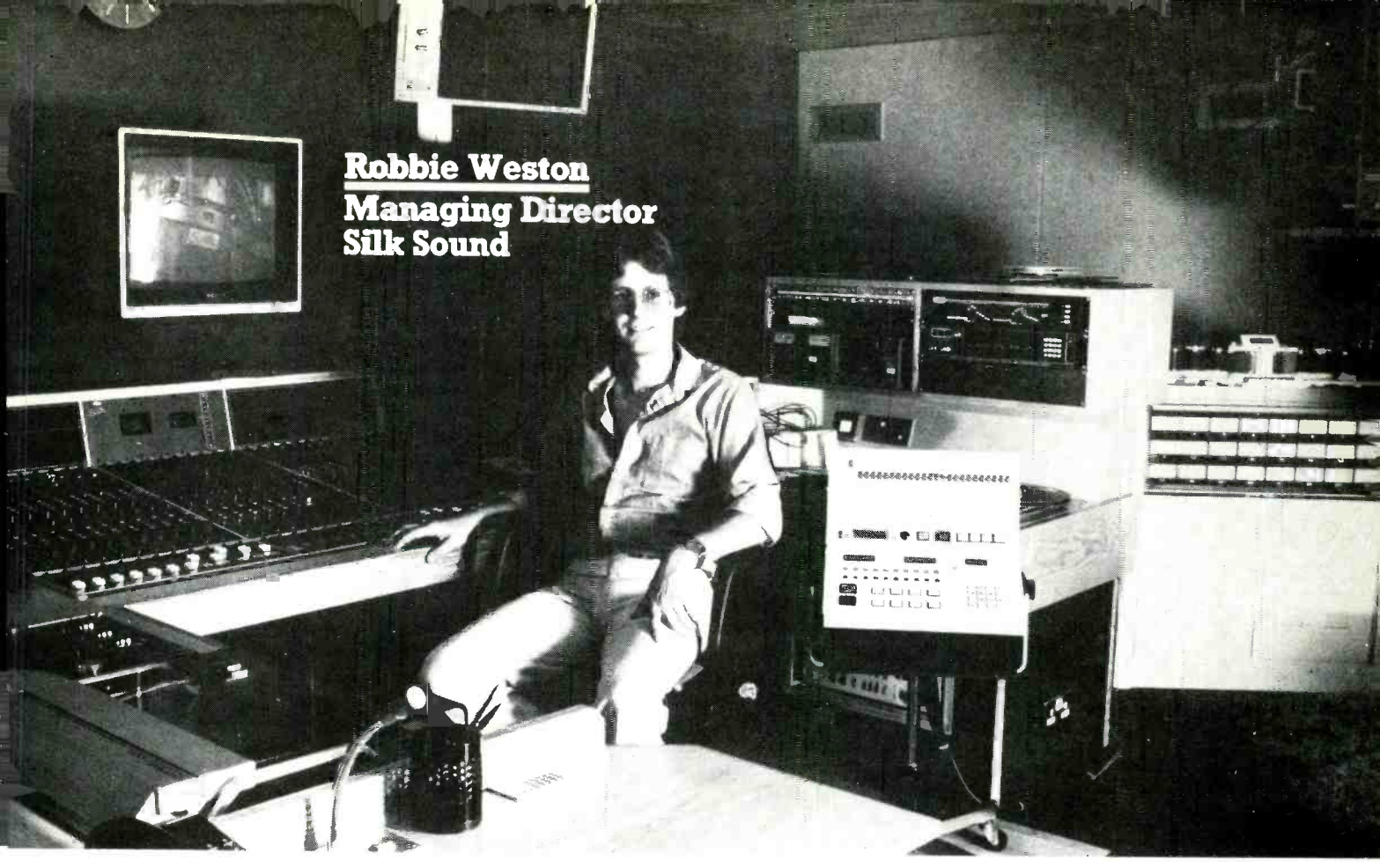

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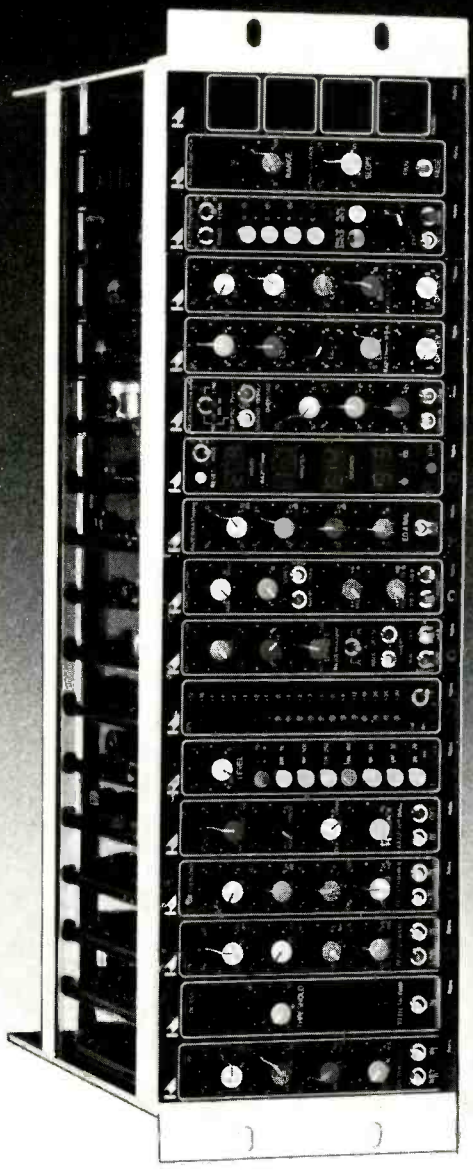
The Nº1 Gate



The Nº1 System

Get the full facts on the most creative modular system in the world.

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The NEW SA 900B and SA 500B:
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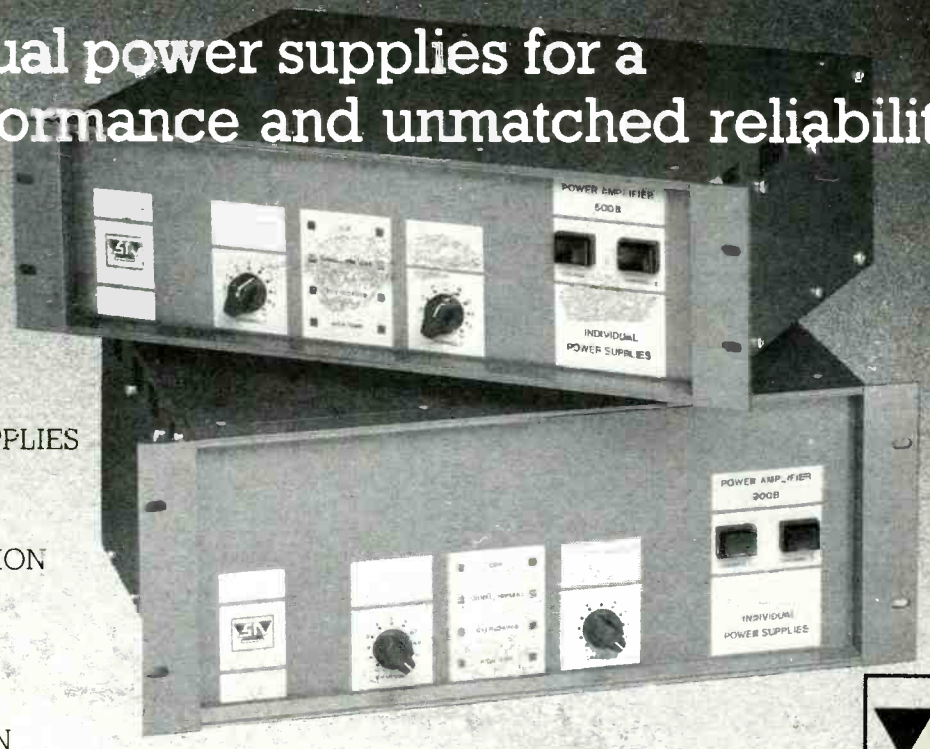

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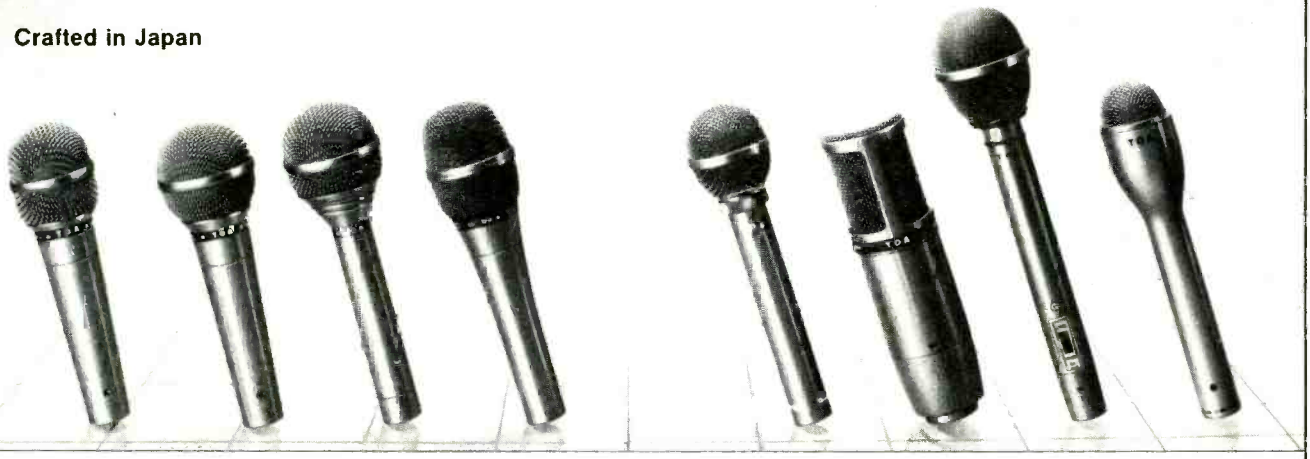
Crafted in Japan



Toa's RD series of professional microphones should prove fine enough to satisfy the most discriminating ear. Strict attention to quality and research into a wide range of varying acoustic needs and environments have allowed us to produce a line of microphones tailored to every kind or circumstance. The series offers a choice between unidirectional and omnidirectional microphones, and for outside use all are equipped with windscreens and offer a high resistance to shock, noise and fumes. To minimize unwanted noise, breath and pop filters are provided on all models, and the mics are all adaptable either for hand-held use or use on a mic stand.
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A precision instrument with **self-sharpening** anti-magnetic steel blades and built-in splicing tab dispenser which eliminates tape wastage, the CAT is chosen by the professionals. Over a hundred major organisations are using the splicer successfully. It is the only system of its kind available anywhere in the world.

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Telephone: 01-749 6911

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Islington was v. nice, but lets face it there wasn't room to swing a cat without stripping the pine off the walls. Parking and loading wasn't much fun either.

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Telephone 01-986 6111

Hardware House

UK distributors for Nady Systems stage radio products.
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NEW From SCAMP®

S30 EXPANDER/GATE

- ★ Log/anti-log release network
- ★ 60dB attenuation range
- ★ Pre-emp side chain
- ★ Keyable
- ★ Variable Expander/Gate slope
- ★ Computer control mute input

The S30 Expander/Gate module sets a new standard in flexibility and quality. This module offers precise SLOPE control from 1:1 through the expander range (1:2 to 1:3) into the GATING range 1:4 (soft) to 1:20 (hard).

The RANGE control, whilst being manually set, will also vary automatically with the slope selected, according to the theoretical maximum for that RATIO. In the GATING modes a maximum of 60dB attenuation will be possible reducing to some 12dB on a slope of 1:2. The THRESHOLD can be varied from +12dBm down to —50dBm. In addition the SIDE-CHAIN can be pre-emphasised by up to 12dB in the HF or LF.

The ATTACK time has three switched positions giving 0·01mS(F), 5mS(M), and 40mS(S). The RELEASE time has a unique anti-log auto mode which speeds up as attenuation increases. The standard RELEASE ranges from 25mS to 4 seconds and incorporates hold facility which can be varied from out to 2 sec.

EXPANDER attenuation is indicated on a 60dB green bargraph with the first 10dB shown in single steps. The second ten segments working in 5dB steps.

S31 COMPRESSOR/LIMITER

- ★ Ratios 1:1–20:1 (continuously variable)
- ★ Separate Limiter Threshold
- ★ Side chain access
- ★ Computer control mute input
- ★ Threshold down to —50dBm
- ★ Log/Lin release selectable

The S31 Compressor/Limiter is a dual function unity gain module which incorporates several new features.

A separate feed-back PEAK LIMITER can be moved by an indexed 20 position pot. over a range of 20dB from 0dBm to +20dBm. The Limiter has an auto attack and release function and operation is indicated by a single red led.

The feed forward compressor has linear RATIOS that can be continusouly varied from 1:1 to 20:1. The THRESHOLD is an indexed 20 position pot and can be operated from +12dBm down to —50dBm (ideal for very soft slopes, 1:2·1 etc). RELEASE time can be switched to a LOG or LIN response. ATTACK time can be varied by a three position switch (·03mS(F), 2·5mS(M), 25mS(S).

MAKE-UP gain is a 20 position indexed pot which will compensate for gain reduction from unity to 30dB of gain. The SYSTEM can be switched IN or OUT as well as the side-chain (SC) being routed to an extrenal equaliser. Gain reduction is indicated by a twenty segment orange led column with a range of indication from 0—60dB. Other features include a MASTER/SLAVE option which is switched on-board but with status indicated by front panel leds. This allows any one unit's MAKE-UP GAIN control to vary several units that are linked to it; thus simplifying stereo or four-channel operation. Control volt linkage for stereo operation is achieved by selecting STEREO link on the board.

These latest modules up date what was the first comprehensive modular signal processing system. SCAMP sets the standard that others emulate.

Other modules in the SCAMP range include: **Gates, Compressors, Equalisers, Dynamic Noise Filters, Pan Effects, Auto Double Track and Time Shape modules, De-Esser, Distribution Amplifiers, Microphone Pre-Amp, Jack Bay, Power Supply and a VCA module.**

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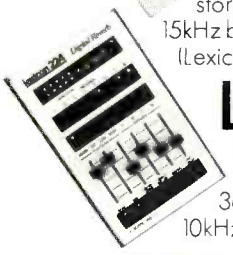
MUSIC LAB HIRE

lexicon 224X

18 program Digital Reverb with 59 variations. Halls, plates and room/chamber algorithms plus chorus, resonant chords, multi-band delays and unique split reverb/effects programs. Cassette data storage, 36 non-volatile memories and full 15kHz bandwidth. Now available with 'LARC' (Lexicon Alphanumeric Remote Console).



lexicon 224

9 program Digital Reverb, Halls, plates and room algorithms plus chorus. 36 non-volatile memories. 10kHz bandwidth.

lexicon 200

3 program Digital Reverb with 30 variations. 10 non-volatile memories. 10kHz bandwidth.

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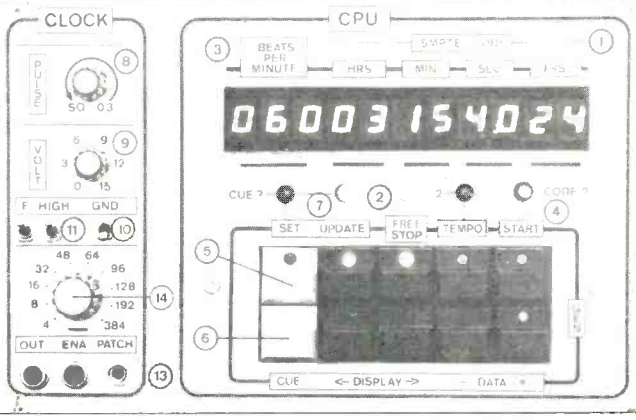
SMPTE READING CLOCK SMPTE READING CLOCK SMPTE

SRC

SYNCHRONISES ANY DRUM COMPUTERS AND SEQUENZERS TO SMPTE CODE

REMEMBERS START AND CUES

SOLVES ALL SYNC PROBLEMS TODAY AND TOMORROW



CPU: 1 SMPTE code writer/reader ±1 octave **2 FREE MODUS**, working on internal clock or clockinputs (see modules) **3 TEMPO** 30–255 BPM, step 1 **4 START TIME** setable in hrs/min/sec/frs **5 CUE POINT** setable in smpte **6 DROP IN CUE-BOTTON** with **METRONOME FLASHER** for online cue setting, press and all outputs will be enabled on the next metronome beat and cue time will be stored! **7 UPDATE** for printing and shifting cue point **MODULES: CLOCK 8 PULSE** 0·3 ms → square **9 VOLTAGE SWING** 0 → ± 15V **10** cuts neg. voltage **11** first edge high or low **12** enable high or low **13** patch for the **SRC** clock delay **14 CLOCK-COUNTER** 4, 6, 8, 12, 16, 24, 32, 48, 64, 96, 128, 192, 256, 384, 512, 768
DELAY clock delay 1 → 64 ms ± any frames alows also to sync ahead
OUT fixed outputs for Linn, Oberheim, Roland, and others
IN reads Linn, Oberheim, Roland, and others

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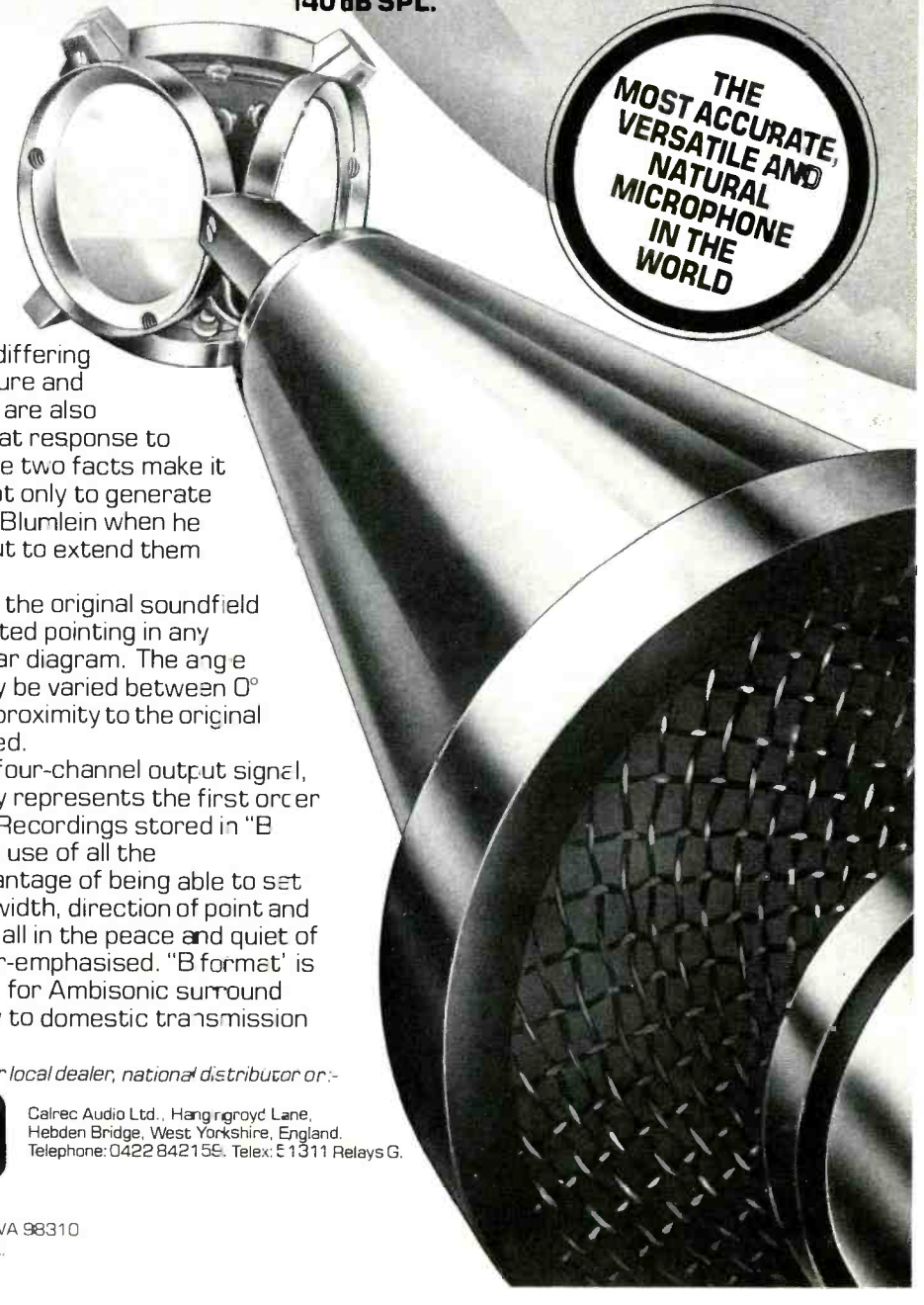
The unique Calrec Soundfield Microphone MK IV

THE FEATURES

- ☐ **Totally steerable both vertically and horizontally in POST PRODUCTION, off tape.**
- ☐ **Variable stereo capsule angle and polar pattern in POST PRODUCTION, off tape.**
- ☐ **Variable zoom, both forwards and backwards, in POST PRODUCTION, off tape.**
- ☐ **The only truly coincident stereo microphone in the world.**
- ☐ **Very low noise performance for the digital era.**
- ☐ **Separate outputs for Ambisonic surround sound.**
- ☐ **Level frequency response both on- and off-axis.**
- ☐ **The most accurate polar patterns in the world.**
- ☐ **Maximum input for less than 0.5% THD 140 dB SPL.**



THE FACTS

The spherical three-dimensional pick-up of the Soundfield Microphone is such that the phase errors introduced by the capsule spacing in normal microphones are effectively eliminated and the resulting stereo output of the control unit has virtually perfect image placement at all frequencies. The differing frequency responses of the pressure and gradient components of the signal are also corrected, thus giving an equally flat response to both on- and off-axis sounds. These two facts make it possible, for the first time ever, not only to generate exactly signals envisaged by A. D. Blumlein when he first proposed the M/S system, but to extend them into three dimensions.

This spherical representation of the original soundfield allows a stereo signal to be extracted pointing in any direction and of any first order polar diagram. The angle between the two microphones may be varied between 0° (mono) and 180° and the apparent proximity to the original sound sources may also be adjusted.

The control unit also provides a four-channel output signal, known as "B format," which exactly represents the first order characteristics of the soundfield. Recordings stored in "B format" allow the POST SESSION use of all the aforementioned controls. The advantage of being able to set such critical parameters as image width, direction of point and tilt, polar patterns and distance – all in the peace and quiet of the dubbing studio – cannot be over-emphasised. "B format' is also the professional signal format for Ambisonic surround sound and may be encoded directly to domestic transmission and consumption formats.

For further information, please contact your local dealer, national distributor or:-

CALREC

Calrec Audio Ltd., Hangingroyd Lane, Hebden Bridge, West Yorkshire, England. Telephone: 0422 842159. Telex: 51311 Relays G.

U.S.A. & CANADA
Audio and Design, Inc., P.O. Box 786, Bremerton, WA 98310
Telephone: 206275 5009 Telex: 152426 AOR USA.

We turn heads..

. . . and the new Series 70 from Trident Audio Developments is certainly stopping everyone in their tracks. It's versatile, it's compact, and the price is very right. Painstakingly and ingeniously designed with the 16/24 studio producer/engineer in mind, this mixing console will be attracting a lot of attention. With it's superb Series 80 styling the Series 70 is configured as a 28-16-24 frame with an additional 4 way echo return. An 8 way monitor module gives you full 24 track monitoring facilities.

The many excellent features include:

- 16 GROUP OUTPUTS AND SEPARATE STEREO MASTER OUTPUTS
- LONG THROW FADERS ON BOTH INPUTS AND OUTPUTS
- MONITORING FACILITIES INCLUDE, MONITOR PAN, MONITOR LEVEL, AUX SENDS AND MUTE FOR EACH GROUP OUTPUT/MACHINE RETURN
- FULL SUB-GROUPING FACILITIES
- V U METERS (LED PPMS AVAILABLE ON REQUEST)
- 4 AUXILIARY SENDS CHANNELS AND MONITORS
- 16 MONITORS ROUTABLE TO REMIX GIVING MAXIMUM 48 LINE INPUTS DURING MIXDOWN

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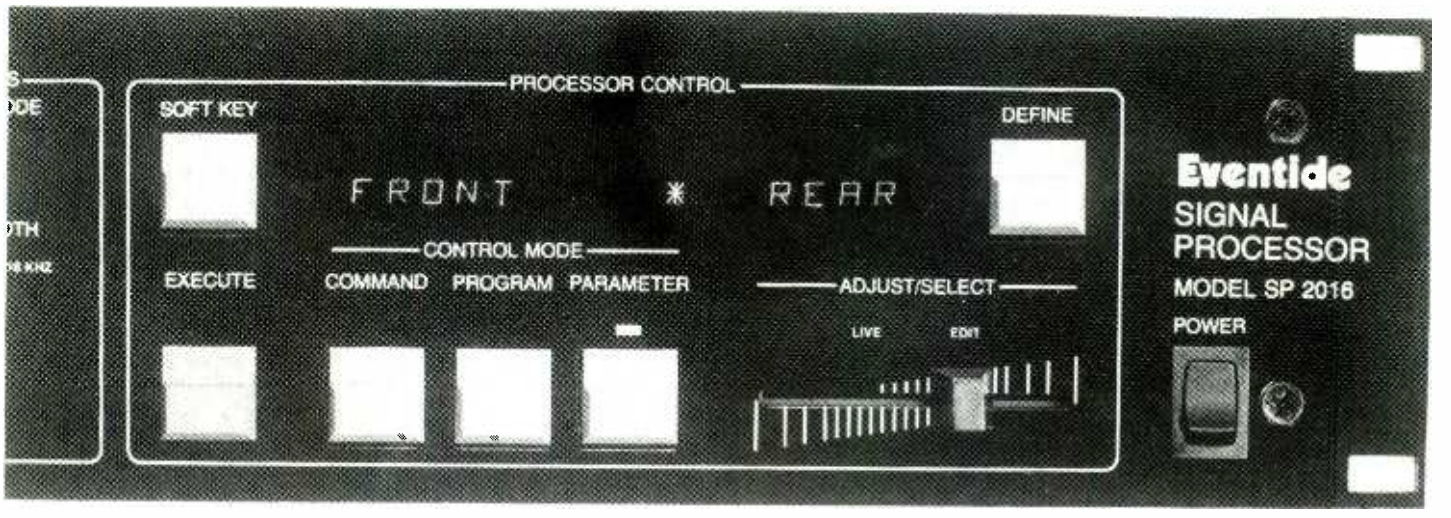
29 Guildford Street, Luton, Beds.
Tel: Luton 450066

Trident (U.S.A.) Inc.
652 Glenbrook Road, Stamford
Connecticut U.S.A.
Tel: 203-357 8337

HOW TO GET FROM THE FRONT OF THE MARQUEE TO THE BACK OF THE ALBERT HALL



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DIGITAL REVERBERATION WITH AN ENTIRELY NATURAL SOUND. A CONCERT HALL THAT ISNT JUST FOUR WALLS. A CLUB ROOM WITH ACOUSTICS. PLATES THAT DONT PING OR WOBBLE. BUT IT DOESN'T STOP THERE.
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Marquee Electronics

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SHOWROOM AT 10 RICHMOND MEWS TELEPHONE (01) 439-8421

diary

AKG Studio Sound Award

Readers are reminded that entries for the first AKG European *Studio Sound* Award are required by March 31, 1984. The categories include professional recording in music, speech, the arts and education, plus broadcast material and developments in live, pre-recorded and production studio applications. First prize is the AKG *Studio Sound* Award Trophy and £1,000. Official entry forms are available from: AKG Acoustics Ltd, 191 The Vale, London W3 7QS, UK.

Quad/Eight purchases Westrex

Litton Industries and Quad/Eight Electronics have announced that Quad/Eight has purchased Litton's Westrex Sound Recording operation in the USA, and all Westrex operations in the UK. The organisation will be known as Quad-Eight/Westrex. Quad/Eight has traditionally sold at least half of its production outside the USA but it will now have a manufacturing base in the UK with the purchase of Westrex's interests here. The company will manufacture consoles and recording equipment in both the UK and USA.

Quad-Eight/Westrex, 11929 Vose Street, North Hollywood, CA 91605, USA. Tel: (213) 764-1516.

Information we could never bring you

There have recently been a number of interesting events that, unfortunately, we were informed about too late for publication before the date had long been and gone. Wherever possible we like to publicise appropriate events and as a leading source of information within the industry, our pages are looked at for just this sort of information. This is a plea to those organising events to send us all the details in sufficient time for the information to be published before the event—we need to know at least two months in advance.

Marquee Electronics

Marquee Electronics is a new pro-audio company formed by Tim Hamill and Graham Middleton (formerly of Feldon Audio), Harold and Barbara Pendleton, Jerry Browse and Simon White. It functions as part of the Marquee group of companies and will be active in areas of sales, servicing, rental and manufacturing. On the sales side they have been appointed a UK distributor for Eventide Clockworks, dealers for JBL, UK distributor for the Ariel Corporation and have many other lines. They are also running a full studio equipment hire service covering a wide range of items including digital mastering systems.

Marquee Electronics, 90 Wardour Street, London W1, UK. Tel: 01-439 84521.

Stolen

REW Video Products Ltd had two items stolen from their Pro-Audio shop on October 1st. These were a Sony *PCM-F1* serial number 650023 and a Sony *SL-F1 UB* serial number 208347. Any information concerning these units should be directed to REW. Tel: 01-836 2372.

Forthcoming events

January 23 to 28
MIDEM, Cannes, France
February 21 to 23
Sound '84, London, UK
March 27 to 30
AES 75th Convention, Paris, France
April 29 to May 2
NAB 62nd Convention, Las Vegas, USA
May 12 to 15
AES 76th Convention, Anaheim, USA
June 13 to 15
APRS Exhibition, London, UK
September 21 to 25
International Broadcasting Convention, Brighton, UK

Studio Sound Pro-Audio Directory

This issue of *Studio Sound* contains some changes that will significantly improve the usefulness of the magazine as a reference source over the next six months. The major change will be the introduction of the *Studio Sound Pro-Audio Directory* the 84/85 edition of which will be published in June 1984 and available by the time of the APRS exhibition. The Directory will be a complete subject by subject, short-form listing of equipment and services for the professional recording industry together with a full index section of the manufacturers and distributors addresses—in short where to get what you want worldwide. This directory will be in a size similar to that of *Studio Sound* and will be distributed free of charge to recording studios who currently receive the magazine. It will also be possible to buy copies at a cover price as yet to be announced. This means that all the information that used to be contained in our product reference will be available in one handy publication—in fact there will be far more information than we have carried before.

As a result of the introduction of the *Pro-Audio Directory*, we will no longer carry the product reference section on topics of the month as this information will be carried in the forthcoming Directory. There will however be an update feature that will review changes in the topic under discussion. This will not be just a listing of new models but will, in a wider context, look at design trends, changes in features, innovations and where possible or appropriate, changes in application and use of equipment. The actual content of the feature will vary depending upon the subject topic but it is hoped that this provides a more generally informative background and complement the Directory. We will also be expanding the New Products section to place more emphasis on new products that are appropriate to the monthly topics and this will then serve as the update to the Directory—providing a more detailed description, a brief specification and usually a picture (when provided by the manufacturer).

We would request that manufacturers and distributors continue to send information relevant to the month's topics as before.

A useful consequence of the removal of the product references is that we will now have room for more features and be able to look at some topics in greater depth and expand our interests into areas that lack of space prevented us doing before. We think that these changes will be beneficial to readers and generally lead to a more informative magazine as we approach our 25th year of publication.

Agencies

● An omission from last month's list of new domestic sales representatives for Ursa Major was Lassers, Sangwin, Lassers of Chicago who will be covering Illinois and Wisconsin.

● Marquee Electronics have been granted manufacturing rights for the Survival Projects stereo panner and the Marquee-Pulse Design TTS 124 digital metronome.

Address changes

● Applied Microsystems Ltd have moved to larger premises and are now located at Town Mill, Bagshot Road, Chobham, Nr. Woking, Surrey GU24 8BZ, Tel: 0905 6267. Telex: 8952022.

Contracts

● Neve have received an order for a 56/48 *8128* console from Los Angeles based Soundcastle recording studios to be installed in a new remix/overdub room plus a custom console for the National Film Board of Canada.

● Recent orders received by Trident Audio Developments include a *Series 80* console for the Pink Floyd's Roger Waters home studio; two *Series 80* consoles for Swedish Radio to add to the 24 they already own; and South African Broadcast have ordered a customised *Series 80* for their Pretoria Music Studio.

People

● Soundcraft Inc. have announced two new appointments at their Santa Monica office. Nick Bogden will assume financial control of the company as accountant/controller. He was previously general manager of Rumbo Recorders. Linda Frank has been appointed as sales administrator. She was formerly of Interlake Audio, Canada.

● Ursa Major have announced the expansion of their engineering staff with the appointment of two new members: Mark Bruckner, previously with Raytheon, becomes a hardware engineer; and Charles Anderson joins the software development side.

● Mick Boggis, formerly engineering manager with Otari UK has been appointed sales manager of ITA.

● Valley Audio have announced the appointment of Emil Handke as general manager. He was previously national sales manager for Sound Workshop Inc.

● Quad/Eight Electronics has announced the appointment of Joe Urbanovitch as chief engineer manufactured systems and will head a team developing new console systems. Previously he was with MCI/Sony.

● Peter Horsman has been appointed director of marketing and sales for Gauss Loudspeakers. Previously he was national sales manager for Phase Linear.

In brief

Syn-Aud-Con have added Neutrik as a sponsor of their educational seminars and workshops... **Audio Engineering Associates** of Pasadena CA, in conjunction with Saz Records, have just completed Pakistan's first digital recording project—a double album by the Sabri Brothers, the foremost Qawwali musicians of Pakistan, recorded on location... **Abbey Road** studios have installed the first Yamaha *PC5002M* 500 W/channel power amplifier in the UK where they apparently drive their B&W monitors 'a treat'... **NASA** are using an MXR *Model 175* digital delay line as part of the system synchronising received audio and video communications from the Space Shuttle during flight time at their Houston control centre... The **Eurythmics** *Sweet Dreams* single and album were mixed on an old Soundcraft *Series 2* 16/8 console... ■

An amazingly versatile, fully programmable Digital Drum Computer...

The MXR Drum Computer. With a full complement of features and functions, it rivals the performance of a high-cost system—at an affordable price. And it's simple to use.

• 12 real drum sounds, digitally recorded in memory: Kick, snare, rim shot, toms 1, 2 & 3, hi-hat open and closed, crash cymbal, claps, block, and bell • Individual level controls and outputs for all voices • Capacity: 100 patterns of up to 99 beats each, 100 songs • 2000 drum beat memory • Seven accuracy levels from 1/8 notes to 1/32 triplets • Infinitely variable time signatures • Tempo adjustable from 40 to 250 beats per minute • Built-in click track (metronome) • Four shift levels for human feel • Pre-panned stereo outputs • External trigger for all voices plus accent • Tape/sync interface • Memory-tape interface • External tempo capability • External voice expansion • Exclusive MXR one-year *full* warranty

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☐ **Please send me information on the MXR Drum Computer.**

SS 1/84

new products

A modified Shure mic is shock mounted inside the drum shell

RMI drum miking system

RMI have recently introduced the *MAY AE* drum miking system. The heart of the system is a modified Shure *SM 57*, shock mounted inside the drum shell. The mic is a low impedance transformerless version of the standard model. This is mounted on a rotation assembly made of high grade steel with a hinged mounting member designed to provide a shock absorbing air membrane around the mic giving a claimed mechanical separation in excess of 20 dB. The assembly can be rotated through 180° by an external shell mounted drum key. The mic output is taken from an *XLR*-type socket mounted on the drum shell.

RMI say that the system is easily installed in any drum, there being another model for bass drums. Further, claimed advantages for the system include reduced set up time (live use), increased sound separation and reduced leakage between drums while retaining a high degree of control of the drum sound through adjustment of the mic angle.

RMI, 8312 Seaport Drive, Hintington Beach, CA 92646, USA. Tel: (714) 536-2505.

A bigger Yamaha

In the September issue new products section, there was a reference to a certain power amplifier being 'probably the biggest production amplifier currently available'. As always if you start making statements like that it is only a matter of time before . . . In this case it is the Yamaha *PC5002M*, a new 2-channel amplifier similar in external appearance to the established *P2000* series.

The *PC5002M* is actually two independent power amplifiers within the same chassis with the separation between the two channels including individual power switches, power supplies and mains cables. Each channel is rated at 500 W into 8 Ω and can be switched to a mono mode with a rating of 1,500 W into 8 Ω. Front panel features include large 'peak'-type meters with a 53 dB range and LED clipping indicators; switched attenuators in 1 dB steps with rubber 'knob-lock' adaptors to protect settings; LED indication of protection and thermal protection systems operation, and mains power switches.

Rear panel features include balanced terminal strip, male and female *XLR* inputs and terminal strip outputs: switches for stereo/mono operation, pin 1 ground lift switch and subsonic input filter. The *PC5002M* has all heatsinks rear mounted with fins running the entire panel, the design requiring no internal fan system.

Specification: power output level 500 W/channel 8 Ω 0.003% THD 20 Hz to 20 kHz, 750 W/channel 4 Ω 0.01% THD, mono operation 1,500 W 8 Ω and 1,000 W 16 Ω 0.01% THD; input sensitivity 1.73 V (+7 dB) at 25 kΩ for 500 W into 8 Ω; power bandwidth 10 Hz to 100 kHz 0.1% THD 8 Ω 250 W; frequency response +0/-3 dB 10 Hz to 100 kHz 8 Ω 1 W; signal to noise 122 dB IHF-A network 8 Ω; THD less than 0.003% 8 Ω 250 W 20 Hz to 20 kHz; IMD less than 0.002% 8 Ω 250 W; damping factor 500, 8 Ω 1 kHz; dimensions 19 × 10⅜ × 17⅞ in; weight 134 lb 8 oz.

Yamaha, Nippon Gakki Co Ltd, Hamamatsu, Japan.

UK: Yamaha Musical Instruments, Mount Avenue, Bletchley, Milton Keynes, Bucks.

USA: Yamaha International Corp, PO Box 6600, Buena Park, Ca 90620.

Orban programmable EQ

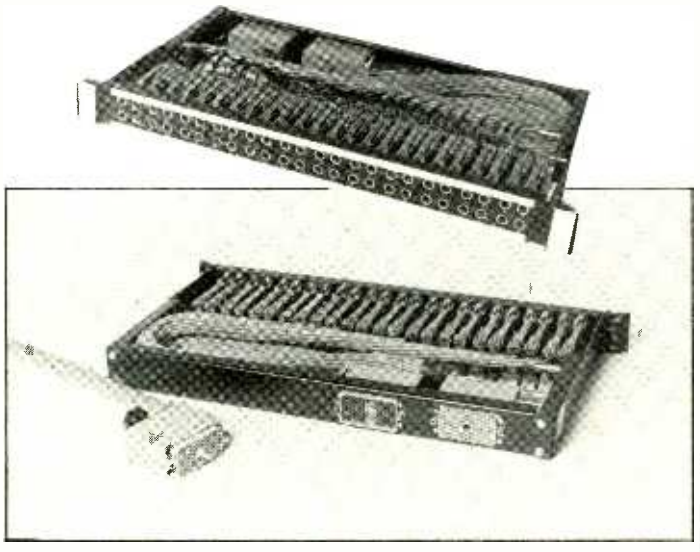
Orban Associates have announced preliminary details of a programmable parametric equaliser. The unit is a standard 19 in rack mount design with two channels and four bands per channel of parametric EQ. The non-volatile memory can store and recall up to 32 different EQ, high and low pass filter, and level settings. Recall of EQ settings is by selection from front panel keys arranged in banks and registers. The front panel provides controls for one band of EQ at a time and the band to be modified must be selected; three LED displays give a readout of frequency, Q and cut/boost. The physical position of the controls corresponds to the recalled value of the EQ setting even if the controls have been moved since storing the setting. The two channels can be made to track accurately in stereo and it is possible to link up another 28 channels and control them simultaneously. The equaliser itself is apparently similar in function to the Orban *622B* parametric and uses no VCA in the audio path. All the EQ parameters are programmable including frequency, Q and cut/boost, high and low pass filter settings and input level.

Provision is being made to allow the system to communicate with a computer via an optional IEEE-488 or high speed serial interface. This will provide an unlimited storage capacity of EQ settings on tape or disk together with remote control for automated mixdown, etc. Intended applications include automated mixdown, disc mastering, video/film/live sound and a host of broadcast applications. Orban say that the unit should be available mid 1984.

Orban Associates Inc, 645 Bryant Street, San Francisco, CA 94107, USA. Tel: (415) 957-1067. Telex: 171480.

UK: Scenic Sounds Equipment, 97-99 Dean Street, London W1. Tel: 01-734 2812. Telex: 27939.

2×24¼ in jackfield prewired to EDAC 516 series connectors



Pre-wired jackfields

Bryant Broadcast & Data Communications have introduced a range of connectorised jackfields. These are available with either bantam or ¼ in long frame, gauge B jack sockets. The units are pre-wired to customer specification and are connectorised with either EDAC/Varelco *516 Series* or *Tuchel* DIN 41622 connectors. Accessories and mating connectors are also available. This series of jackfields offers a wide variety of options and can save installation time. Bryant can also supply a wide variety of jack sockets including PCB mounted types for OEM applications.

Bryant Broadcast & Data Communications, Top Floor, 33 London Road, Bromley, Kent BR1 1JG, UK. Tel: 01-464 4967.

Active M-S matrix decoder

Audio Engineering Associates have developed the *MS-38* Active Matrix Decoder, a transformerless single-control device providing simple decoding of stereo M-S type signals into conventional stereo left and right signals. The transformerless design gives an improved image without shifts in channel balance or level regardless of the stereo width setting, claim AEA.

The *MS-38* is a compact unit with dimensions of 2⅜ × 5¼ × 8½ in. It is designed to interface between the mic preamplifier and the input channel controls. Differential inputs accept balanced line level signals from the M-S mic while the outputs are unbalanced line level. If the M-S signals were recorded discretely, then the unit will accept these as well. Powering is 120/240 V AC with provision for external powering with a bipolar supply between ±12 and ±18 V.

Audio Engineering Associates, 1029 North Allen Avenue, Pasadena, CA 91104, USA. Tel: (213) 798-9127.

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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.

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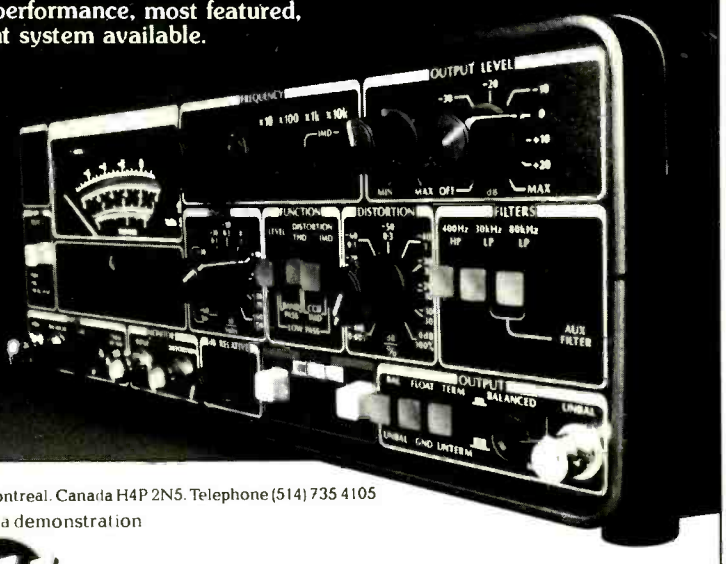

amber Amber Electro Design Inc. 4810 Jean Talon West. Montreal. Canada H4P 2N5. Telephone (514) 735 4105

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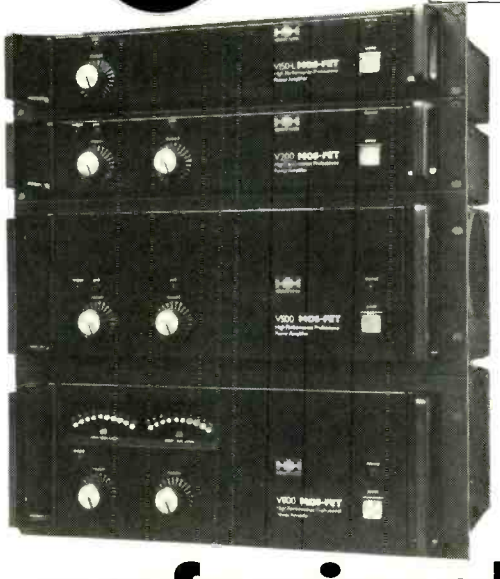

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new products

Beyer 734

Beyer have added a new vocal condenser mic to their range that they describe as being suitable for stage and studio. The *MC 734* has a cardioid characteristic, pop and hiss suppression and is not affected by handling noise. It is fitted with a three-position bass roll-off filter with cut-off frequencies between 80 Hz and 200 Hz. Phantom powering between 24 V and 48 V is acceptable.

Specification: Frequency response: 20 Hz to 18 kHz; open circuit voltage at 1 kHz: 5 mV/Pa; nominal impedance: 150 Ω; rated-load impedance: ⩾1 kHz; max sound pressure level at 1 kHz and distortion factor of K⩽0.5%; 138 dB; signal to noise: 69 dB; 'A' weighted equivalent sound pressure level: approx 18 dB; power supply 24 V to 48 V phantom powering; supply current: approx 0.5 mA; dimensions: length 7½ in, diameters 1 in/1.8 in.

Beyer Elektrotechnische Fabrik GmbH & Co, Theresienstrasse 8, PO Box 1320, D-7100 Heilbronn, West Germany. Tel: 07131 6170.

UK: Beyer Dynamic (GB) Ltd, 1 Clair Road, Haywards Heath, Sussex RH16 3DP. Tel: 0444 451003.

USA: Beyer Dynamic Inc, 5-05 Burns Avenue, Hicksville, NY 11801. Tel: (516) 935-8000.

Beyer MC 734

Rebis equaliser redesign

The Rebis *RA402* dual 4-band parametric equaliser has been redesigned to incorporate a front panel mains power switch and indicator as well as electronically balanced *XLR*-type inputs and outputs. The front panel appearance design has also been updated although the price remains unchanged.

Rebis Audio Ltd, Kinver Street, Stourbridge, West Midlands, DY8 5AB, UK. Tel: 0384 71865. Telex: 335494.

The Infernal Machine

The New York AES was the first showing of a new unit from Publison, the *Infernal Machine 90* stereo audio computer. Published information is at present rather brief but it is claimed that the *IM 90* is designed to handle any digital effect or function.

The front panel has a large alphanumeric display that allows communication between the machine and the user. For each function there is a user guide mode on the machine that explains all the user need know about the selected function. The unit features 18-bit A/D conversion, 32-bit internal signal bus and a 20 kHz frequency response. The principal functions are delay 0.02 ms to 5 min; echo with digital feedback; pitch shifting; automatic arpeggio—a combination of echo and pitch shifting; reverse sound; sound memory depending on the selected options up to 5 min; digital editing capability on internally stored sounds; time compression/expansion; reverberation with frequency dependent selectable reverb times; internal and external automation. The user can apparently order only those functions that he requires according to an options table. Also many other digital functions are under design and updating existing machines will be a simple operation.

Publison Audio Professional, 5-11 rue Crespin du Gast, 75011 Paris, France. Tel: 33.1-357 64 07. Telex: 250303.

UK: Scenic Sounds Equipment, 97-99 Dean Street, London W1 5RA. Tel: 01-734 2812. Telex: 27939.

Whitetower spiders

Whitetower Records have developed an elastic spider mount specifically for the Calrec *Soundfield* microphone. This will allow the mic to be boom mounted with a good degree of isolation from stand and boom borne vibration. The head of the mount is adjustable in angle. The screw fitting for the boom end is a brass insert with ⅜ in × 26 TPI internal thread. The mic multicore cable must be anchored further down the boom with a ½ m cable loop allowed towards the mic. The mic is securely held within the inner-sleeve of the mount and even if inverted, the mic cannot fall out as it is restrained by its multicore cable.

Whitetower Records, 2 Roche Gardens, Bletchley, Milton Keynes MK3 6HR, UK. Tel: 0908 73969. ■

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Everyone knows by now that digital audio doesn't just give you low noise, low distortion, wide dynamics and so on: it also presents its own set of problems, some of which are directly related to the low noise, lack of distortion and so on themselves. A different approach to recording is necessary in which compensations at the recording stage are *not* made for the vagaries of normal tape recorders. It has to sound right before tape, because it will sound virtually identical on playback—as Michael B Tretow discovered recording the recent Abba album, 'The Visitors', with the 3M digital system.

Digital Abba

Michael B Tretow

Michael Tretow (left) with Benny and Bjorn

ALTHOUGH three tracks for Abba's *The Visitors* album had already been mixed on analogue tape, we decided to switch to digital for the remaining songs, when 3M delivered the digital 32-track and the two 2-track machines to Polar Studios in June 1981. We had been very impressed with what we had heard previously on digital and those eight extra tracks certainly helped Bjorn and Benny (producers and songwriters within the group) to make up their minds. Producers just can't resist extra tracks.

However, these 32 tracks turned out to be not more than 30 in practice. Track number 1 serves as a code track for the machine itself, and track 9 is the phase lock track—once recorded, it cannot be erased, since the machine will lose its phase information with a complete breakdown as the very audible result. It took me quite some time to figure out what to record on that track. If I had to do a punch-in on that track, the whole recording would be spoilt! Tell me just *one* instrument that never needs an occasional punch-in! I finally put one of the overhead drum mics on that track.

The punch-in technique differs a whole lot from what we're all used to, since the old recording actually *fades* down, while the new is faded up with the same rate. This makes it possible to perform impossible edits, like in the middle of a steady vocal note, or in a decaying bass note, for instance. You can hit the recording button just anywhere—it's impossible to do a bad edit!

It took me 15 years to perfect my punch-in technique to the point where everybody in the control room says: "Aaah, I never thought that was possible!" All that was ruined in just one week of laying the basic tracks for the album. Everybody learned that this new super machine could do the craziest edits, so naturally everybody played lousy—just to get a chance to be punched in!

By the way, one of the reasons for engineers being so unrhythmic must be the habit of always pressing the record button on the syncopated note before the actual punch-in. Ask any engineer to tap the rhythm on the desk—he's always ahead!

Anyway, back to the 30-tracker. What happens? You guessed it, *everything* is recorded in stereo down to the tambourine. What was said to be a 32-track recorder, is now down to 15. And then there's automation; two tracks for the Allison. That leaves us with 13. That's three tracks less than I had in 1968!

OK, it's not as bad as that, but the extra tracks do not always result in more overdubbing, as you might expect. Instead you allow yourself more freedom to keep everything separated. With Abba there's always a lot of track bouncing since most of the vocals, except for the leads, are double tracked at least once. Since there's no loss in the transferring, we wound up with some extremely good quality vocal tracks. I purposely say 'good quality' because it's a whole different thing, recording digital where the quality is undisputed, as opposed to analogue recording with all its shortcomings. Good quality is not the same as a good sound, right?

For instance, you could not very well go up to Ted Nugent and politely say: "Pardon me, Mr Nugent, but you have exceeded your head room by far. Please sir, try to get a cleaner sound! Forgive me, but your guitar sounds . . . distorted, if you'll pardon the expression."

How does a kick drum really sound? Most of the record buyers have never heard a bass drum without tape saturation! In Germany, 90% of record buyers have never even heard a bass drum without a *Keepex*! Even I don't remember what a bass sounds like without modulation noise!

What I am trying to prove is, that rock music is, and should be, tape saturated.

For the first couple of sessions, I was carried away by the sheer quality of the sound coming out of the speakers, and did not care much for the *sound* of the instruments. Every engineer knows the painful situation when the band comes in to the control room to hear the tape. The drums are always down 6 dB after tape and it's a whole different mix you're hearing. And you always say: "You guys should have heard it before it went on tape. Absolutely the greatest sound I've ever heard."

With digital, however, what goes in, really comes out. No noise, no tape saturation, no nothing. Everybody is impressed, including you. And you start to think that you can

get away with anything. All that hard work on analogue to preserve the sound after tape is no longer needed! You can start to live the life of a producer—you think!

What happens when you get home and listen to the 7½ copy? You get the digital hangover. The 90 dB of dynamic range means that you're actually missing a lot of the music on your 30 dB home system; things that you normally would have compressed or limited to keep over the noise level on analogue tape. When the bass player hits the strings with the knuckle, it becomes *very* loud. The kick drum may vary 10 dB, so you set your level on the loudest and then it's gone on your car stereo.

The lack of noise is actually another problem, for the analogue man, anyway. Every engineer is familiar with the psycho-acoustic mystery that occurs when you're splicing leader tape to a fade out ending of a song. You push the stop button when the fade is over, and you can swear that the music did not stop until then. You're always wrong. The music always stops two turns earlier on the reel. You're actually hearing things that aren't there! The sound lingers in your ears, long after its gone.

This does not come out the same way in digital. When it's quiet, it's off—I mean, *nothing*. Manufacturers of digital reverberation actually substitute the last portion of the decaying echo with white noise, to overcome this effect. The ear does not regard this as white noise, but rather that the continuity is undisturbed as in a natural echo.

On the 32-track master, the noise was in fact lower than the actual 'live' noise. Air conditioning rumble and mic preamp noise that was below the 87 dB 'threshold', simply wasn't recorded. So, when you play the tape back for the musicians, you turn your monitors up a bit as you would normally do when working with analogue. When the tape starts to roll, there's none of that familiar hiss, so you turn the monitors up some more. Then: BBOOM, KAPUTT—go your speakers!

Another thing you have to change, is your miking technique. Most of the time you choose mics for what they sound like on tape, not what they sound like in real time. That peak at 20 Hz you hear when you're setting up the bass drum—you know it won't be there after tape, so why worry about it. In fact you sort that out of your mind, since you're so used to what's going to happen on tape, that you don't even hear it any more!

A few years back, I used to work in a room with a sharp peak at 40 Hz and another at 80. After two years I couldn't hear it any longer and could not understand what people were complaining about. I guess it's the same type of phenomenon as when you live close to the railroad track; after a while you don't hear the trains anymore. The brain prefers not to take any notice, even if the SPL makes your ears wiggle.

Anyway, the 3M picks up everything that's in the mic's response, and nothing is flattened out by tape saturation. Mics you thought you could trust, start to sound different. It's like the switch over from the old tube desks to transistorised consoles. In those days, people believed in mirror impedances, which meant that the mics were loaded with 200 or 600 Ω. When the transistor boards came, they loaded the mics with 1.5 kΩ, so all the dynamic mics started to change character. Even some of the condenser mics sounded different on the transistor preamps. I remember what the old *U47* sounded like on the old tube board. It was nothing like the 'warm' quality they claim today. With some singers it put a shrieky quality on the syllables, that was just impossible to get rid of. Of course, there were no EQ sections on the board in those days. Just one knob, marked 'piercing' and one marked 'dull'. And we only had the Conax de-esser—remember the black boxes that substituted syllables with distortion?

On the Harrison desk at Polar there is a number of transformerless mic preamps. For the stereo piano I run two *U47*s directly from the mic preamp output to the inputs of the 32-track recorder. That's 'warm' quality! Since there's no EQ involved, it becomes a little muddy, so I always place a third mic close to the piano and add the EQ on that, panned to the middle on a separate track.

Polar Studios has an enormous glass-coated room that's just ideal for the Bösendorfer grand piano. Two glass doors seal this room from the rest of the studio, so it is possible to place the mics as far away as you prefer. This room is also great for strings. It can take some 20 musicians, with full separation from the rest of the band.

On *Let the Music Speak* I used another of the Polar rooms. It's a medium-sized all-wood room. I recorded the choir parts with very distant miking, so that the reflections from the walls hit the mic with almost the same level as the direct sound. It gives a special, boxy quality to the background vocals, very similar to a theatre stage recording, a 'Broadway-y' sound, that I like very much.

On the vocals I mostly use the AKG *C34* but due to the confusion mentioned earlier (with all the mics giving a new, unfamiliar sound) I tried a lot of different types, which I deeply regret today, when I listen to the record.

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'The Visitors' album

I try to avoid limiting or compressing on the harmony mics but sometimes you're forced to use it. With lead vocals I have a method worked out with the *C34*. It is a dual capsule stereo mic on which you can turn both capsules in the same direction. If you balance the two capsules on the board to give equal output and then compress just one of them, the un-limited capsule will 'take over' on loud passages, thus minimising the strained character you get with limited voices.

By using different rooms for different overdubs I try to achieve some depth perspective in the recording. I'm not very much for recordings where everything comes from left, right or phantom centre, with equal distance from the mics.

If something is supposed to be in the background, it should be recorded with a background sound to it. Nobody records strings with close up mics, because distant miking has proved to give the richest, most musical sound. With Abba it's very much a question of making room for all the overdubs, soundwise. There can be some four or five different background vocals going on at the same time, three or four pianos here and there, and maybe eight different synthesiser lines, together with the drums, bass and doubled guitars. If I was going to record all that with close mics, it wouldn't work in the final mix. By close miking you get a lot more transients that force you to keep down the levels. Remember, all this is going on at the same time! If the congas are close to the mic, it creates enormous peaks, with lots of energy on the track. Now, if the kick drum and the piano coincide on the same attack as the conga, the transients will be added and make those light meters flash like mad, although the apparent level is not more than if the instruments were played separately. If you back off a little with the conga mics, the congas will still be heard, maybe even better since the room might add a little body to the sound. This also tells the listener that congas is a separate percussion sound and not a sound coming from the regular drum kit.

I always pick up the guitar with one close and one distant mic. Not to get a 'room' sound but to let the guitar amp 'fill' the room before it's picked up by the mic. I believe that the amp has to 'breathe' in order to get a good guitar sound in the control room. Most of the time, I give the room mic a little vibrato from a DDL to make it sing a little.

Benny owns a Yamaha *GX 1*, the *Dream Machine*. It is a 16-voice synthesiser with some extremely rich sounds in it. There are three manuals, bass pedals and a useless drum machine in it. On *The Visitors*, I recorded the *GX 1* in a different way from the previous albums. I did not use the direct outputs to record but routed them through the foldback system, out in the studio speakers. At Polar they have Philips *MF* with built in power amps, all over the studio. These are good quality speakers, with quite some power if you want it, and they cover all the different rooms at Polar. I miked each of the speakers in stereo, some of them very distant and recorded the sum of them on two tracks, without any direct injection. This proved to be very successful—it had none of that 'electrical' sound that's common with synthesised strings. The string and brass sound was extremely natural and even hard to tell from the real thing. On some cuts I used the Publison pitch shifter for octave doubling on the most distant mics. On slow lines it actually works very well and gives an 'airy' sound.

To edit an album on digital tape you have to transfer each mix to a new reel of digital tape. To get rid of the count-in at the beginning of a song, you must record a few turns of silence before the song, since you can't splice leader tape to a digital tape. Then you rewind both machines and electronically edit out the count-in.

Once you've selected the edit point, the machine will perform the edit automatically. It sounds rather complicated compared to splicing a piece of leader between the songs, but it works just fine, once you get used to it. One drawback, however, is that you can't change the sequence of the songs afterwards, without doing the whole album side over again. I first transferred all the mixes to regular 15 in/s tape, and then we fooled around with these copies, until we found the right sequence.

The crossfade editing technique proved to be very useful on 2-track, since there's no definite edit point causing abrupt changes in the stereo image. In fact, it was possible to edit between two takes of different tempos by moving the edit point to a 'wrong' spot and fool the ears. We did a lot of substituting jobs on some of the songs. If the second refrain turned out better in the mix, we substituted the first refrain with the second, and so on. We mostly work with a click track of some kind, like the Linn drum machine but on songs with tempo changes and ritardandos, etc, a click track won't work. But even if the tempos did not match at all, it was always possible to edit between the different parts of the song, by simply moving the edit point.

For the cutting, we had to wheel the machine over to the cutting room, where Peter Strindberg made the cutting, by-passing the regular rack and feeding the cutter head amp directly from the machine's output. 3M manufactures a preview delay that we used in the cutting process. Then the metal father matrices were exported. Thus the actual cutting of the album had to be done some 30 times by poor Mr Strindberg and his crew, since each country was supplied with their own, original cutting—we've had some weird experiences previously, when each country made their own cutting.

I once received a record of foreign origin, where they had compressed the *Voulez-Vous* album a good 20 dB! Strangely enough, this 3 dB dynamic range album sold more copies in that country than the previous albums had. I must have done something wrong, again! ■

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NEWS DEVELOPMENTS INNOVATIONS PRODUCTS TRENDS

EFFECTS

The last 12 months in effects has seen a gradual evolvement, with manufacturers bringing out improved models and updating rather than rapid changes. If there has been a trend, it must be the introduction of low cost effects with performance and facilities that were until recently restricted to considerably more expensive items. Nowhere is this more obvious than with delay lines and the fact that these units can now be made more reasonably priced has brought a number of new manufacturers into the professional field.

A good example of the latter is Ibanez, a Japanese manufacturer of guitars and associated equipment. They have introduced a range of three digital delays—the *DM500* with max delay time of 256 ms, 16 kHz bandwidth and a modulation section; the *DM2000* with the same features and a max delay of 1,023 ms; and the *HD1000* which has a combined pitch shifter/delay line offering a delay time of 504 ms and a pitch shift of ±13 semitones.

Korg are a similar company although their background is mainly synthesisers and they have introduced the *SDD-3000,* a rather more comprehensive unit offering delays up to 1,023 ms with 17 kHz bandwidth. The modulation section has a selection of waveforms and filters also having a programmable capability that will store most functions required.

An example of a complete turn round in product range must be DeltaLab. For all intents and purposes the original *DL Series* has been dropped and the *Effectron* range released. These are digital delay lines with maximum delay time of 64 ms, 256 ms and 1,024 ms being the *ADM-64, ADM-256* and the *ADM-1024* models respectively. All three have modulation sections and the two longer delay models have their delays switchable in preset ranges that allow more immediate settings for echo, doubling and flanging, etc. There have also been some further additions to this range in the form of the *Super Time Line Series* with models *ADM 512* and *ADM 2048* having delay times of 0.5 s and 2 s respectively, each having a programmable memory for four settings. Lastly there is the *Echotron* also known as the *ADM 4096* which has a 4 s maximum delay time, an infinite repeat facility, a hard and soft feedback control as well as a metronome facility that allows synchronisation of a drum machine to the delay setting. All these units are very low cost and in some aspects actually out-perform the original series at something in the region of one sixth of the cost.

Analog Digital Associates have two new digital delay lines available in the form of the *D1280* with seven delay time ranges covering deals of 0.156 ms to 1,280 ms; and the *D640* with delays up to 640 ms with both units having bandwidths of 15 kHz. The *D640* has a modulation section with a capability of a 3½-octave flange sweep at the max depth setting and high cut control on the regeneration path. The *D1280* has all these facilities plus the novel idea of an LED that flashes in relation to the delay time allowing more easy setting of delay times when trying to sync them with a musical beat.

Bel Electronics have developed their first digital delay line and they have benefited from watching everyone else learn the hard way. The *BD60* has a maximum delay time of 2 s with a 16 kHz bandwidth and 4 s with 8 kHz. As well as this main delay there are three sub delays whose levels may be individually set. Regeneration can be added from main or sub delays. As with previous units, delays may be set manually, under internal oscillator control, external oscillator or envelope following of the input signal.

A slightly different approach can be found on the low cost *CD425* from Cutec. This is a 1,024 ms digital delay with all the standard delay line features including a sub delay that is quite independent of the main one in delay time although is recombined for the output.

The Klark Teknik *DN700* is a dedicated delay line providing a maximum delay of 434 ms. It has threee outputs and a non-volatile memory. Applications are obviously many although the manufacturers see live sound as a main use area.

Yamaha chose the New York AES to show the *YDM-2600* digital delay system, a very comprehensive unit with applications from live sound to disc mastering. The unit comprises a rack mounted control unit and a small hand held remote unit. There are effectively three modes of operation—one input and eight discrete outputs with up to 2,660 ms delay on each; two inputs with four discrete outputs up to 1,333 ms each; and four inputs with two discrete outputs each up to 655 ms each. The *YDM-2600* may be used as four discrete units or with different delay times from a common input to create interesting effects.

In the area of updates, AMS have made available a new card for their *DMX 15-80* or *DMX 15-80S* systems that has an improved deglitch capability. It can be fitted in either of these units provided that they are at present equipped with two pitch change cards. This card will also allow a number of new features on the *Loop Editing System* such as 'electronic reel rocking'.

Orban Associates have updated the *245E* stereo synthesiser unit with the replacement being known as the *245F.* Changes include balanced input, output transformer option, RF filtering on the input and AC line filtering. Commendably, the price has remained the same. Studio Technologies Inc have just announced that they will have a stereo simulator available soon with stereo width control and a choice of FM or AM modulation features.

Aphex Systems launched their *Type B* processor which is a simplified version of the *Aural Exciter.* It is intended for a range of applications where the full facilities of the *II-S* are not required. Each channel has drive, tune and mix controls with a tri-colour level LED. The unit is also more compact being only one

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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.

Tony Faulkner
Audio Engineer

VAT No 225514681

Tony Faulkner is a leading freelance independent recording engineer based in London who records around 50 classical music albums each year.

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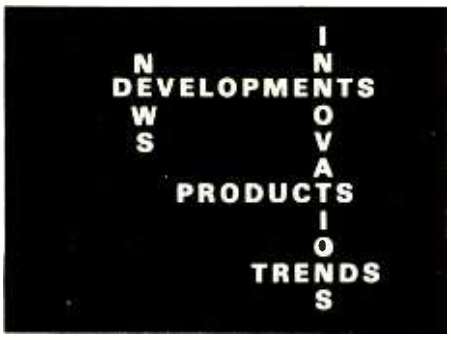

rack unit high. The EXR Corporation announced the *Model EXIV* which contains most of the features of the *EXIII* but with a new enhancement process that affects the bass and lower-mid frequencies when in the 'A' process mode. Other features include a peak level switch which gives a choice of narrow or wide band enhancement.

Lastly, the New York AES saw the first showing of a new unit from Publison, The *Infernal Machine 90*. This is a stereo digital processor that from the preliminary information appears to be able to tackle virtually anything. Claiming a 20 kHz bandwidth, it has a delay capability of 0.02 ms to 5 min and other possibilities including echo, pitch shifting, arpeggio, reverse sound, sound storage up to 5 min, editing of stored memory, time compression/expansion, and fully adjustable reverberation programs. There are modes for user instruction for each of the operating states and a large alphanumeric display for information from the machine. It appears that not all the functions need be purchased—just those needed dependent upon an options table. Publison have further added that a number of other functions are under development and it is intended that these will be retrofitable.

REVERB

In the field of reverberation, most of the major developments at the top end such as the EMT *245*, Lexicon *224X*, Quantec and Eventide *SP2016,* etc, happened last year. This year has largely seen consolidation and updates in the digital reverb area.

The AMS *RMX 16* has seen a number of modifications. A memory extension was made available increasing the maximum delay to 1620 ms from 810 ms, all full bandwidth. There was also a release of software known as *REV.2.1* and this included alterations to the DELAY program resulting in it being renamed ECHO. With this program it is now possible to set different amounts of delay for each output from the unit as well as different amounts of cut and delay to any regenerated signal. In addition there is a CHORUS program and different versions of the NONLIN and REVERSE programs.

Quad/Eight showed a new model of the *System 5* digital reverb known as the *System 5-LC*. The *LC* aspect refers to local control achieved by fitting all the features from the remote control into the front panel of the processor. The advantage of this has been a substantial reduction in price that will outweigh the minor inconvenience caused, with the possibility of adding the remote at a later date.

There has been at least one new release of software for the Lexicon *224X* and the arrival of the *LARC* alphanumeric remote console. This is intended as a unit to replace the standard *224X* controller and was designed to simplify *224X* operation. The alphanumeric displays are equipped with 48 characters that identify programs by title, and sliders are labelled with their function at that time. Users may name banks of stored registers and the capacity for memory is 36 user generated programs although data may be dumped or loaded to/from tape. The *LARC* is also potentially adaptable to other Lexicon computer-driven audio and video equipment.

Also from Lexicon and shown early in 1983, was the *Model 200* digital reverberator. This is a rack mount unit with all controls on the front panel rather than a separate control although such features as reverb time and selection of programs can be operated remotely. It has three main programs—concert hall, plate and chamber with 10 preset variations of each program. Reverb decay is adjustable from 0.6 to 70 s dependent on program, with simulated room size also being variable from 1 to 1,000,000 m^3, again program dependent. There are also a number of additional controls such as pre-echoes, reverb time contour and roll-off, etc, and 10 user program stores. The *Model 200* is intended as a relatively low cost unit with some of the capabilities of the larger units.

Other units to have had a software update include the Sony *DRE-2000* digital reverb and reports from users have described this unit as having 'benefited mightily' from this.

One manufacturer new to reverb systems is Yamaha and they have announced two new digital systems. The first is a low cost unit known as the *R1000* which is a single-channel unit with four reverb modes that differ largely in the length of the reverb time. It also incorporates a 3-band equaliser and comes in a single unit 19 in rack.

The other Yamaha system is rather more complex. The *REV-1* consits of a rack mount processor with a large control unit. The user has control over facilities such as the number of early reflections, the relative timing between reflections, their absolute level and their initial delay. The adjustment of levels of later reflections is possible as is the setting of reverb time in any of four frequency bands. The control unit has an LCD display that will allow a graphic representation of the set reverb pattern. There is memory for 60 user settings and 30 factory presets with provision for editing all programs. The remote control unit has 64 pushbuttons, 10 control knobs, four LED displays, two LED level displays and the already mentioned LCD 'window' and this will give some idea of the control complexity of the unit.

Turning to mechanical reverb systems, there have been at least two new plate reverb systems. NSF is a small British company which has been manufacturing plates for about a year. These are low cost compact type units that could just about be carried by one person (if very strong) with a separate control unit with brightness mode. It is also supplied with a motorised damping system to allow remote adjustment of the reverb time. Another company manufacturing low cost plates is US manufacturer Reasonable Alternatives Inc which has a modular system where you just buy

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the bits that you need to meet your requirements, eg the frame and the plate itself—using your own driver and pickup transducers.

Studio Technologies announced a new model in the form of the *Ecoplate III* which is again a fairly compact system with a pre-tuned plate and a shock mount system that they claim will eliminate tuning problems. The reverb time is variable from ½ s to 5 s and it is equipped with high and low end EQ.

MicMix Audio Products released a device known as the *DC-2* which allows the user to control the decay time of any reverb device. The unit is a simple rack mount device with a single knob for decay time setting. The *DC-2* has the additional advantage of offering a claimed 30 dB of noise reduction on the used reverb device.

Finally, there have been a number of spring line type reverb systems to fill the area below plates and more costly mechanical reverb systems. These include the Fostex *3180* 2-channel unit; the Shiino *Vesta Fire RV-1* and *RV-2* units with limiters, EQ and 'cross' mode on the *RV-2*; and the D & R Electronica Stereo *Model 1* reverb in a single rack unit chassis.

EQUALISERS

In a field of design that has a tendency to be farily static with regard to new developments, the last 12 months has seen a comparative rush of activity. Aside from manufacturers simply updating models from their product range, there has been a relatively large number of new products at the extremes of the price spectrum.

Turning first to graphic equalisers, two of the manufacturers most associated with this type of unit have both recently replaced most of the models in their range. MXR now has three models—the *170* 2-channel 1-octave 10-band; the *171* 2-channel ⅔-octave 15-band and the *172* single-channel ⅓-octave 31-band. All these models have centre detent sliders, a low cut filter and signal present LEDs, being finished in that still unfamiliar MXR black colouring.

The other manufacturer to change their range is Klark-Teknik. They have replaced the complete line of models except for the *DN27A* which will remain available for the time being. The four new models are known as the *Series 300* and include the ⅓-octave single-channel 30-band *DN 300*; the *DN 360*, a 2-channel version of the *DN 300*; the *DN 332* 2-channel ⅔-octave 16-band unit; and the *DN 301*, designed mainly for room tuning with the same features as the *DN 300* but with attenuate control only. All these new units employ a proprietary active filter circuit using thick film technology that they claim will increase reliability. The units also carry facilities that reflect the general direction for graphic equaliser features—additional high and low cut adjustable filters, failsafe bypass in the event of power failure and on some models, switchable slider ranges to match the type of application with either ±6 dB or ±12 dB settings.

Other new graphic equalisers include the Audioarts Engineering *Model 2700* ⅓-octave single-channel 27-band unit with ±12 dB and a *Model 2700C* with 16 dB attenuate only control per band as well as the high pass filters and overload indicator of the *2700*; and the *E27* graphic from Phase Linear. The latter is a 27-band ⅓-octave single-channel unit with boost of +12 dB and cut of −15 dB per band.

Two new units that should be included with the graphics are the Formula Sound *SE1* 2-channel ½-octave 19-band system equaliser contained in a single unit high, standard rack mount case (all controls are screwdriver operated and are normally covered by an anti-tamper panel); and the Japanese Cutec *GS2200*, a 2-channel 10-band graphic that also includes a miniature 10-band spectrum analyser with peak hold facility and a pink noise generator all within a 2 U high chassis. This is a very moderately priced unit and is probably not intended for high precision work but makes a useful package.

NEI have recently introduced the *2711X Digital Equaliser/RTA* unit. The digital aspect of the title refers not to the equalisation process itself but the internal memory of equaliser settings, the computer aided RTA and the auto EQ facility (using a pink noise generator). The EQ section is 27-band ±12 dB with adjustments for each band made by pressing the up or down button for that band. An LED display shows the level set on one of nine LEDs per band. The level adjustments are in 256 logarithmic steps while function switches and a four-figure alpha-numeric display allows control of the unit modes. Full description of this unit is outside of the scope of this update but the *2711X* appears to present some interesting possibilities.

Before leaving graphic equalisers it is interesting to note that no more manufacturers have attempted to harness the awsome power of the parametric graphic equaliser and this may indicate that we have reached the complexity peak for the present market in terms of EQ facilities. Having said that I would like to draw your attention to the Technics *SH-8065* graphic equaliser. This is a hi-fi unit in concept but claims to be the first 33-band graphic with the top band having a centre frequency of 25 kHz. Even more extreme is their *SH-8045* stereo 12-band 1-octave graphic with a top band centre frequency of 32 kHz! Why?

Most of the new parametric equalisers introduced recently have been very moderately priced units. Teac have a new unit as part of the *Tascam Enhancement Series,* in the form of the *PE-40,* a 4-channel equaliser with each channel having four bands plus low and high pass filters. All the bands overlap into the ranges of other bands with the two mid ranges having EQ abilities over four bands.

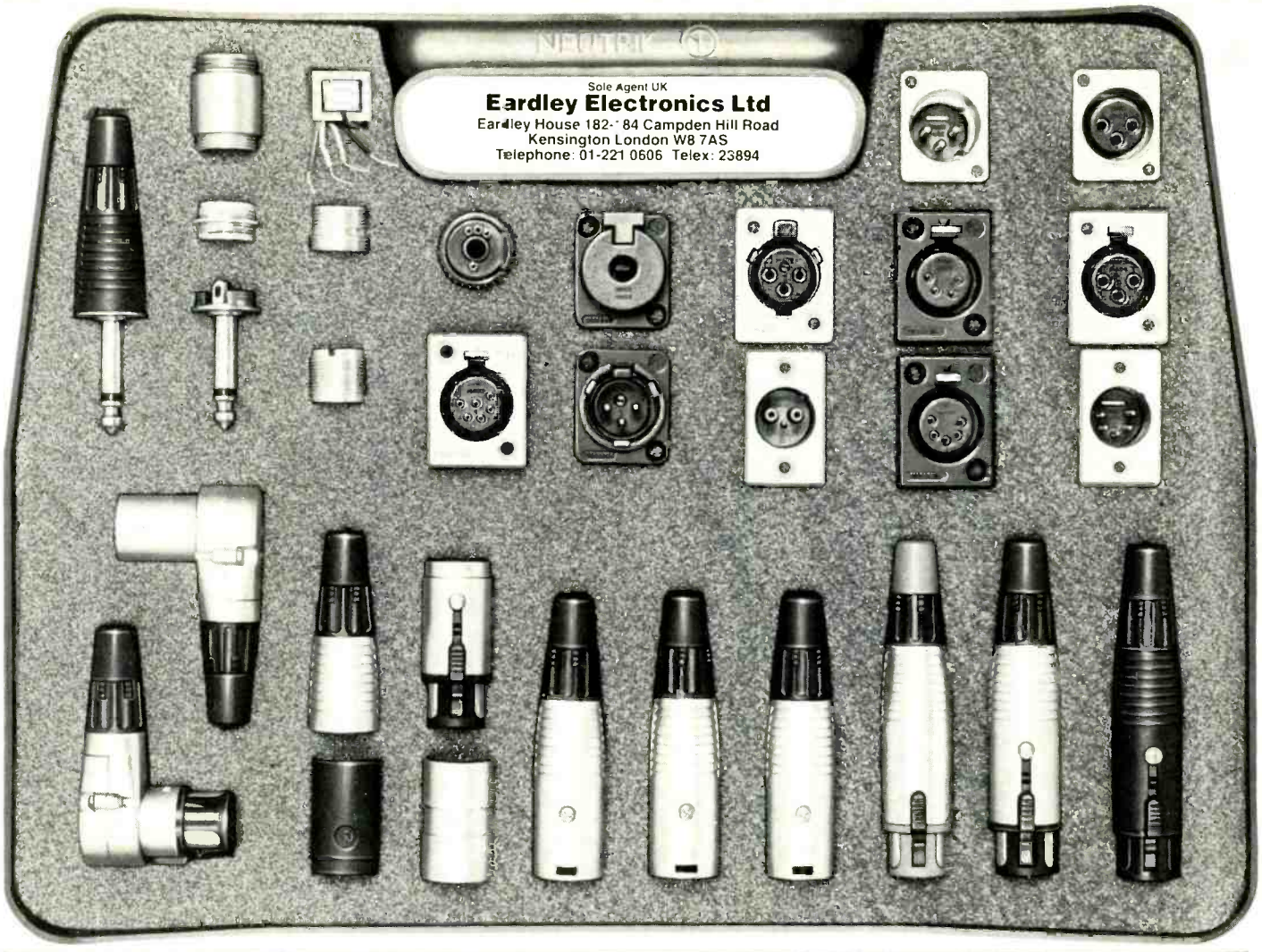
A manufacturer new to professional audio outside of their native Denmark is tc electronic. They have made a name for themselves in Europe and the UK for a sophisticated range of musical instrument signal processors that have a very high standard of performance—perhaps too high for some users! They have recently introduced the *TC 1140* single-channel 4-band parametric, the *TC 2240* parametric with two 4-band channels and the *TC 1220* 2-channel 4-band parametric intended mainly for feedback suppression.

The history of programmable equalisers has not been one of great commercial success so far, unfortunately, for a variety of reasons. The 360 Systems equaliser is no more, the *Param* is very quiet at present and the NTP system disappeared from sight but is now back being marketed as a disc mastering system. There have recently however been three new systems announced.

The Orban *Programmable Parametric EQ* is a 2-channel unit with four bands of parametric EQ per channel. There are 32 non-volatile memories for storage of frequency, bandwidth, boost/cut, high and low pass filters and input level. The two channels can be tracked in stereo and up to 28 channels can be ganged and controlled simultaneously. There is provision for interface with an external computer for remote/auto control and/or increased memory capacity. Information on this unit is still in the preliminary stages and it may be six months before it is fully available.

The Auditronics *PPEQ-1* programmable parametric equaliser is a somewhat different unit. Each equaliser is a self-contained unit and four can be held in a standard 19 in rack. They have three EQ bands with variable frequency, boost/cut, Q, peak/shelf selection on high and low bands together with a separate in/out switch for each band. All variable parameters are displayed at all times and each unit has 32 non-volatile memories and the ability to interrogate them without affecting programme material. There is also provision for interfacing with external equipment as well as remote control.

The new equaliser system from Neumann—or to give it its full title, *Autonomous Micro Processor Controlled Equaliser System AME591*—may prove a more acceptable approach to equaliser automation than other systems as it remains channel orientated. The system is intended as a replacement equaliser system within a mixing console and comprises a central operating panel, a memory unit and an equaliser unit for each channel. The equaliser is based on the Neumann *W491* unit and consists of three bands with adjustable high and low pass filters. Pressing one of the buttons on the equaliser connects it to the central operating panel as the equaliser has no controls although it does have three LEDs per band and two for the filters that will give an approximation of the selected EQ on that channel. The COP allows the user to set the EQ required for each channel and allows facilities such as A/B comparison between old and new EQ settings. The memory unit permits storage of up to 10 console EQ settings of up to 48 channels. The memory unit is also equipped with a data cassette recorder for further off-line storage. **Keith Spencer-Allen**



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AES 74th Convention Summary of papers

Keith Spencer-Allen

THE 74th AES Convention was held at the New York Hilton Hotel from October 8 to 12. This was the first time the Convention had been held at this location and the general response of those exhibiting and attending papers was that the venue had proved quite satisfactory. There were, however, some worries expressed in the hours before the exhibition opened that a backlog at the loading bay might delay the opening but in the event everything was in order. Aside from the wide range of exhibits, demonstration rooms and other technical presentations there was the usual wide range of papers presented. The 80-odd papers were presented under 10 topic headings: Digital Recording & Broadcasting; Signal Processing-analogue; Studio Design; Sound Reinforcement; Loudspeakers: Transducers & Low-Frequency System Alignments; Loudspeakers: Network Considerations; Signal Processing—digital; Psychoacoustics & Subjective Testing; Test & Measurement; Disc Recording & Multichannel Sound.

In this report, a selection of the more relevant papers are covered and where available the AES preprint number is given so that further information can be obtained if desired.

Digital Recording & Broadcasting

This was the first of the paper sessions of the convention and was in fact an extra day, a day before the exhibits opened. The reason for this was to allow exhibitors to attend papers.

The first paper presented was 'Digital Recording and Reproducing Techniques Using a Thin-Film Head' by Yasuharu Shimeki, Misao Kato, Shiro Tsuji and Hiroshi Matsushima of Matsushita Electric. The paper outlined the development of a digital multitrack tape machine capable of recording 16 audio channel on a ¼ in tape. The advantages of using narrow band tape are tape economy, simplified tape handling, reduced equipment size and the possibility of using the same machine for mastering on. The paper then detailed the development of a thin-film head containing 20 poles in ¼ in to meet the design concept and the measured results they achieved with this head design. The recorder employs a one track/channel system and follows the AES standard of 16-bit, 48 kHz sampling rate. The actual recorder was on show in their demonstration room with a master version and an editing system and the compactness of the system was obvious. Matsushita have, however, made it clear that they will not be putting this system into production. (Preprint 2027)

In a paper entitled 'Digital Audio modulation in the PAL and NTSC Laser-Vision Video Disk Coding Formats', K A Schouhamer Immink, A H Hoogendijk and J A Kahlman from the Philips research laboratories described an extension to the *LaserVision* video disc format to include a digital audio signal. This research was prompted by attempts to improve the signal-to-noise ratio obtained from the analogue tracks of NTSC format discs which are some 10 dB inferior to PAL and SECAM systems. They showed the feasibility of a combined digital and audio signal in the NTSC format with the digital format following the Compact Disc standard but resulting in a fully compatible disc. Although the same system is possible with PAL and SECAM discs it cannot be combined with the analogue audio carriers and would therefore not be compatible. (Preprint 1997)

Dr Roger Lagadec of Studer and G McNally from the BBC Research Department presented a paper entitled 'Labels and Their Formatting in Digital Audio Recording and Transmission'. The paper proposed the use of labels as a means of conveying operational and commercial information about their host digital audio programme. This information might be operational data such as serial number, take number, date of origin, duration, etc; technical data such as pre-emphasis, signal description, signal compression data, balance settings, auto EQ, etc; commercial data for copyright, etc; and other information such as names of musicians, musical score details, comment, etc. The paper covered these topics and framework for the development of labels for diverse requirements and the consideration that this information must be recorded, transmitted and recovered. The proposal is compatible with the AES/EBU serial digital audio interface. This is a very detailed paper and we have only been able to touch on some of the points in this report and if you are interested you should obtain Preprint 2003.

Craig C Todd and Kenneth Gundry of Dolby Laboratories presented a paper 'Optimizing the Delta Modulator' covering the theory and performance of the delta modulation system A/D conversion. They also covered the parameters which can be varied to obtain the maximum performance as well as the outline of a system for broadcast applications. (Preprint not available)

'The DASH Format: An Overview' was the title of a paper from T Kogure of Matsushita, T Doi of the Sony Corporation and R Lagadec of Studer. This covered the Digital Audio Stationary Head format proposed by several manufacturers of professional digital recording equipment. This is an update on the previously published format and is intended to show the format in its final version after changes have been made to allow for 2-channel recording requirements, and future increased track densities. The manufacturers are, as before, hoping to promote the acceptance of this format to increase compatibility between future digitally equipped studios. (Preprint 2038)

Signal Processing—Analogue

Karl D Lahm of Jules Cohen and Associates presented a paper entitled 'A Unified Approach to Audio Level Metering' in which he described a unified metering technique combining the functions of a VU meter and a PPM in a single unit. The proposed system presents the user with amplitude information that is most appropriate to the system in which the meter is installed. To convey this information in a clear and non-distracting manner, the requirements have been met by using a VU type meter with a driver circuit that displays average VU levels or normalised peak levels, whichever are the greater. The paper then gives further details and some practical examples. (Preprint 2015)

'Audio-optics—The Next Technology' was the topic of a paper presented by George Bowley of Audio-optic Technologies Inc in which he summarised general audio system deficiences as they stand at present and then introduced fibre-optic technology, outlining the advantages that optical signal systems will have, the medium, the communication systems and audio-optic applications in areas such as signal processing, etc. The paper achieves what the author aimed at—a concept catalyst—and contains some worthwhile ideas. (Preprint 2013)

Studio Design

This session was opened with a paper by Manfred Schroeder of the University of Gottingen and Bell Laboratories. 'Progress in Architectural Acoustics and Artificial Reverberation' was a general overview of recent developments in architectural acoustics and his current areas of work such as why classical reverberation-time formulae often give wrong results, why reverb measurements are best made with pseudo-random maximum length sequences, the use of phase gratings to achieve equal energy scatter of incident sound, etc. (Preprint not available)

Peter D'Antonio and John Konnert were the authors of a paper entitled 'The Schroeder Quadratic-Residue Diffusor: Design Theory and Application' presented by the first mentioned author. It is not so widely known that polycylindrical diffusers are far from uniform in their dispersion patterns with regard to frequency. This paper went into greater detail about a point mentioned by Manfred Schroeder himself in his paper, covering the first-order diffraction theory of the Schroeder Quadratic-residue diffusor. This system will provide a broad frequency band dispersion with a uniform coverage if certain parameters are

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AES papers

correctly calculated. The paper covers a computer program which will calculate the diffraction pattern based on this theory together with design parameters, practical examples and some studio applications. One obvious application of this theory is with LEDE control rooms although there are many other possibilities. This is an area we hope to cover in more depth in the near future particularly as there should soon be some practical experience of actual installations. (Preprint 1999)

'Surround Sound in the Eighties —Design Principles for Surround Monitoring Environments' by Greg Badger of Audiometric Labs/ Soundfield Systems and Chips Davis of LEDE Designs was unfortunately not presented by them. The paper, however, principally outlined the problems of monitoring surround sound within more recent design type control rooms with particular reference to the LEDE control room where the rear monitor speakers would obviously find themselves in a quite unsuitable environment. The paper outlined methods of dealing with this. (Preprint 2046)

Bruce Bartlett of Crown International presented a paper entitled 'The Tonal Effects of Classical Music Microphone Placement'. This paper covered an observation that recordings in standard miking positions for classical music are invariably closer than the normal seating positions of the audience and can be proved to be brighter than the audience hear. Microphones placed in audience positions are, however, often dull in comparison. The paper presented details of measurements, the suggested mic positions and the use of HF cuts, etc. In the presentation a tape was played to demonstrate these effects but the problems associated with filling a large ballroom with critical sound material is very difficult and I am not sure that in some areas of the hall the effect the author was trying to achieve was too much like the impression he actually achieved. (Preprint 1994)

In 'The Use of Boundary Layer Effect Microphones in Traditional Stereo Miking Techniques', John Lehmann and Michael Lamm of Dove & Note Recording Co, described the use of boundary layer effect mics in classical miking techniques such as Blumlein, Lauridsen or X-Y, M-S, ORTF, etc. They also demonstrated a system for boundary mic placement consisting of two pairs of hinged plates forming front and rear Vs. These are separated by a third pair of hinged plates. The Vs open and close in unison and determine the spacing between capsules mounted on them, while the side plates adjust to alter the angle of acceptance to the capsules. This device is known as the *L²MicArray* and will allow the use of boundary mics in the classical techniques already mentioned with full diagrams being provided in the paper. (Preprint 2025)

Sound Reinforcement

The first paper in this session was presented by Don Davis of Syn-Aud-Con and was a description of one user's approach to *TEF* measurement of sound systems. 'Basic TEF Techniques' includes examples of what are becoming standard TEF measurements and some new views of old measurements and the value of some new measurements. (Preprint 2036)

The increasing use of small computers for sound reinforcement applications was highlighted in a paper entitled 'A Microcomputer Program. For Central Loudspeaker Array Design' by David Albertz, John Eargle, D B Keele and Ronald Means of JBL Inc. The paper describes a program which solves inverse-square losses as seen through a loudspeaker's distribution pattern at points on a seating area. For fairly normal distribution of seating, the program will allow calculation of up to 230 points. The program can also deal with multiple speaker arrays taking into account phase and amplitude relationships. With the addition of acoustic data on the room, direct-to-reverberant values can be calculated and intelligibility estimates can be made. (Preprint 2028)

Loudspeakers: Transducers and Low-Frequency System Alignments

Laurie Fincham of KEF presented a paper entitled 'Refinements in the Impulse Testing of Loudspeakers'. The paper discusses the problems and solutions that have been encountered in 10 years of testing phase and amplitude frequency response characteristics using impulse response measurement. There was particular emphasis on reducing errors in LF measurements as well as new methods for determining the free field response of a loudspeaker system down to low frequencies. (Preprint 2055)

Marshall Buck of Psychotechnology Inc and Cerwin Vega Inc delivered a paper under the title of 'Measurement of Spatial Reproduction Quality by Interaural Cross-correlation Techniques' in which he described the development of a mechanised technique for the assessment of spatial imaging qualities which would be easier and perhaps more effective than the use of listening panels. The paper details investigations into the interaural cross-correlation coefficient using such techniques as dummy head mic pick-up and fast Fourier techniques. (Preprint 2044)

Signal Processing—Digital

This particular session was started with a paper entitled 'Single-Stage Sampling Frequency Conversion' by Messrs Lagadec, Pelloni and Koch from Studer. Early designs for digital universal sampling frequency converters were based on four stage digital filters supplemented by an asynchronous buffer. In this paper a prototype design for a single stage sampling frequency converter was described and its feasibility shown. (Preprint 2039)

Dr Martin Jones of Neve Electronics covered the time-division multiplexing methods for transmission of digital audio signals. In 'Signal Transmission and Routing in the Digital Audio Studio' he described how it was now cost effective to transmit 128 digital audio signals on one 20-pair ribbon cable with signals of 20-bit and 48 kHz sample rate. The system is flexible in use for the studio operator and is successful over copper cable and optical fibre. (Preprint not available)

Messrs Niimi, Fujino and Shimizu of Nippon Gakki Co (Yamaha) gave details of the development of a new digital reverberation system such as the new model that was shown for the first time in the exhibits area. The paper 'A New Digital Reverberator with Excellent Control Capability of Early Reflections' covered the design concept, hardware structure and some application examples. (Preprint 2026)

The amount of R&D work being done by Studer in the digital field is demonstrated by yet another paper presented by R Lagadec and D Pelloni. Entitled 'Signal Enhancement via Digital Signal Processing' it covers research and attempts at recreating original music signals from less than perfect sources. The process involves the splitting of the digital signal into linear phase narrow band signals for individual processing before being recombined. Some practical examples were described but as the paper says, it is very difficult to access results in words although early results are promising. Aside from the difficulties of telling the computer which noises are just noises and which noises are musical such as the breath of a flautist. It follows from this that some of the problems encountered are not just technical but with such a potentially powerful system musical and psychoacoustic considerations have to be included. The value of a fully operative system such as this could have real application in processing of analogue material for CD release although this appears to be a little way off as yet. (Preprint 2037)

John Snell of Lucasfilm presented a paper entitled 'The Lucasfilm Digital Audio Console' which gave details of the system they have developed for their film postproduction studios. The console has real-time programmable controls offering equalisation, reverb, synthesis, vocoding, Dopler-shifting and other effects. Alphanumeric displays are used to label controls and the continuous movement of controls is stored on multiple computers with storage capacity up to 16 Mbytes each (Preprint not available)

Psychoacoustics and Subjective Testing

Most of the papers in this section were covering measurement and assessment techniques for manufacturers but two papers were worth special mention. 'Surround Sound in the Eighties—Localisation and Psychoacoustics' by Martin Willcocks of Willcocks Research Consultants and Greg Badger of Audiometrics Labs, gave an account of their recent developments in surround sound and an overview of the current possibilities together with a set of conclusions based on their observations with listening positions and head rotation, etc. (Preprint 2029)

Worth a mention is the paper from Wieslaw Woszczyk of McGill University and Floyd Toole of the National Research Council, Canada for their paper 'A Subjective Comparison of Five Analogue and Digital Tape Recorders'. In this they detail experimental methods to evaluate these machines and the results. All the machines were unnamed and as such conclusions must be somewhat vague but it was noticeable from a study of the results that the listeners actually found it difficult to distinguish the digital machines. There were also found to be large similarities and small differences between machines and most of the machines found listeners with preferences. The authors conclude with the statement that their use of freshly-made first generation recordings gave one story but a quite different pattern could be possible if further generations of recordings were equally evaluated. (Preprint 2033)

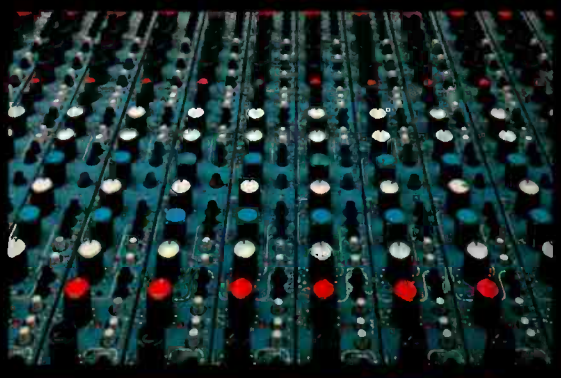
Disk Recording and Multichannel Sound

'Ambisonics in Multichannel Broadcasting and Video' by Michael Gerzon, consultant to the NRDC, considered the most effective methods of using three or four related audio channels and then covered the Ambisonics UHJ hierarchy for transmitting total sound fields. All the options were covered from monaural to 4-channel periphony and a range of user soundfield adjustments in stereo and psychoacoustically optimised surround sound reproduction modes. (Preprint 2034)

The second paper was almost a continuation of the earlier 'Surround Sound in the Eighties' but with the subtitle of 'Advances in Decoder Design' by Martin Willcocks of Willcocks Research Consultants. He covered briefly the history of decoding systems for both discrete and matrix systems. There was then reference to current Ambisonics and USQ systems and comparisons are drawn together with subjective decoder performance tests. (Preprint 2017) ■

ANGELA 28/24 at Sanwa Video, Osaka, Japan

ANGELA 28/24 at Castle Audio, Dallas, Texas



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Studio Miraval Recording a la Provencale

Terry Nelson

Miraval is situated in the heart of southern Provence—'the countryside of Cézanne' proclaim the signs on the autoroute—about 12 km from Brignoles and just over an hour from either Nice or Marseille (depending on how fast you drive!). Access to the studio is quite straightforward, simply leave the autoroute at Brignoles and turn up towards Le Val, carry on through in the direction of Barjols and after 4 or 5 km there was a sign on the left pointing into the woods with the legend 'Studio Miraval'. From there on the ride gets progressively bumpier until you arrive in front of the group of buildings that form the estate of Miraval.

Discovered by Jacques Loussier in 1969 while visiting friends in the region, Miraval was bought first of all as a home. Though in a state of general disrepair, the atmosphere of the place was such that M Loussier felt compelled to buy it—at what can be described as 'a reasonable price'. Today, one shudders to think! In the beginning, thoughts of a studio were far away and it was not until five years later that the first ideas for a studio germinated.

As a musician, Jacques Loussier needs no introduction. A concert pianist, known primarily for his interpretations of Bach—with or without a rhythm section—he is also well known as a composer for film and television as well as concerts. (Commercials are no stranger to him, either. Remember the ad for Wills *Whiffs* with the cigar coming out of the mummy case?) Finding the atmosphere of Miraval more conducive to the flow of ideas than the pressures and stress of Paris, M Loussier felt it would be to his advantage to build a small 8-track demo studio where ideas could be put down on tape and developed in peace. This would make working easier and not oblige him to go to Paris to spend money on studio time with all the attendant hassles. The music could be written at Miraval and recorded in the Paris studios (usually Decca) when ready.

At this point his secretary Dany, came into the picture. Her view of the situation was why stop at a demo studio? It would still be necessary to go to Paris to make the final recording so why not record at Miraval instead and be rid of the pressures for once and for all?

From early on Jacques Loussier had never liked the pressures of working in city-based studios. "How can you work if the musicians are more worried about finding a place to park—and, having found it, worry more about feeding the parking meter, than concentrating on the music in hand?" Then while Pierre is away at the meter, Louis says he'll go for a quick drink, with everyone else following suit and they all end up in the bistro. Then it's 'why not eat something now and...' Well, you get the picture. The problem is that the studio clock is still ticking away, expensively, and adding further to the pressures. In fact it is the 'working against the clock element' that has always disturbed M Loussier the most, making him feel caught up in a vicious spiral.

The idea of having a studio of his own where he could work when he wanted and for as long as he wanted, without having to worry about running into overtime started to become very appealing—as well as being logical. Patrice Quef, Dany's husband, Loussier's longtime friend and one of the leading engineers in Paris, was brought in as someone who would be able to install the studio and run it.

Finding a place to build the studio was the least of the worries. Miraval has many barns and farm buildings and one in particular lent itself admirably to the installation of a studio. One of the previous owners of Miraval was a certain Joseph Lambot whose claim to fame is that he was the inventor of reinforced concrete. To test his theories he built a large barn with ground and first floors, the exterior walls being traditionally built with large stones and therefore nice and thick with a concrete interior. The roof is beamed and joisted as it would be with timber with the exception that all is done with reinforced concrete. However, you have to look pretty closely to realise that those joists are not wood! The exterior aspect of the building, some 50 × 30 m, retains the same look as the surrounding buildings and does not jar with the 'old world atmosphere' of Miraval.

The first Studio Miraval emerged in 1976 with a Neve console and 16-track Studer *A80*. The total studio floor area was 450 m^2 including three large isolation booths. The acoustics were 'in-house' and were designed by Messieurs Loussier et Quef. Though empiric, the treatment worked out quite well, consisting of fabric covered Rockwool against the walls and sloping ceiling with hardwood panels randomly placed to give a mixture of dispersion and absorption. The control room was placed at one end of the barn with a large window providing natural light from the rear.

The moment the studio became operational a familiar story unfolded. Word got round and Miraval was soon under pressure to become more commercially orientated as a result of people starting to discover the, then relatively new, concept of the studio 'away from it all'. A phrase that has unfortunately become time-worn over comparatively few years.

It was clear that if the studio was going to be run as a fully commercial operation it would have to be comparable to the top Paris studios, especially in view of its isolated position and the following year saw the control room equipped with a Neve 32/24 console and an *A80* 24-track recorder. However, as studio owners and their bank managers are painfully aware, recording studios are restless beasts and are committed to constant evolution if they are not to be doomed to extinction. Thus September, 1982, saw the present version of Studio Miraval emerge.

Part of the decision to update the studio involved building a new control room at the other end of the studio, further down the barn, where there was more than enough space to build as big a room as necessary. The various design propositions ranged from the serious to the...well let us say, not serious at all and be nice about it! The design that was retained was that proposed by Rabit Studios of London, who seem to have strong connections with France.

The first impression is one of space and even the SSL console looks a little lost sitting in the middle! Well, with over 60 m^2 to play in I suppose it would really. With this size the room must rank as one of the largest in Europe at the moment and provide easy, unhindered working conditions. The design can be said to have very definite Hidley influences—not that that is a negative remark by any means—with a mixture of traps and reflective/dispersive surfaces. The ceiling is completely trapped with two reflective hardwood surfaces, one over the monitors at the front of the room and the other to the rear of the console, with two fabric covered openings, one over the console—presumably to reduce splash—and the other to the rear of the room up to the rear wall. The back half of the room has full traps all the way round with the back wall trap being ported at the front of the platform providing seating at the rear of the room. The side walls—and thus the traps—are faced with a mixture of fabric and slanting wood panels. These latter increase in spacing as they get towards to the back, a style termed 'progressive absorption' by the designers. The rear wall is carpet covered as is the floor area of the control room, with the exception of a parquet rectangle behind the console. This could be for acoustic

reasons or just to let the chairs roll more easily! Or both. I have mentioned a seating platform at the rear. This is quite deep and has sofas all the way across with seating for about 12 people. If one takes into account the front of the dais then two dozen listeners can be comfortably installed without breathing down the engineer's neck! This can be termed 'very convenient'.

The front half of the control room is a mixture of brickwork and hardwood, with the studio window effectively divided into three sections, viz: one under each monitor and the sliding glass double doors that give access to the studio. To the far side of each window is a large carpet lined soffit designed to take a Studer 24-track *A80*. At present one is occupied by its intended occupant and the other serves as a very handy tape store! Moving round the room each side from the soffits are flush wall racks which contribute further to the clean lines of the room. The main entrance lock—or door—is to the left of the room, after which there is a soffit for two *A80* master recorders, wall rack for the computer electronics and power supply for the console and a lighting control panel for adjusting the atmosphere in both control room and studio.

As the original control room had daylight, this aspect has not been forgotten either and there is a large window to the right of the console in the wall. Due to the thickness of the outside walls this is almost a bay window and a substantial window sill has been made. When I was at Miraval in September there was sufficient daylight to work in the room without any extra lighting and this I find very agreeable. The area around the window is trapped over and under, as are the monitor mounts, soffits and door areas. The overall impression is pleasantly low-key and airy, with more of a 'drawing room' feel to it than anything else.

With the installation of the new control room a new desk had to be bought. In keeping with the policy that the studio must offer at least as much as the Paris ones, automation was now an essential and after much heart searching—they had been very satisfied with the Neve—an SSL *4000E* was chosen. This is a 48-inch input mainframe fitted with 36 I/O modules and eight VCA groups and so far they are very pleased with it—even if it does take a bit of getting used to.

Recording is with a Studer *A80* 24-track running at 15 and 30 in/s with Dolby should it be needed. Mastering and copying is on two *A80* stereo machines, one running at 7½ and 15 in/s and the other at 15 and 30 in/s. The latter machine also has ½ in tape conversion for very high quality mastering. There was a slight problem here in that there was no cutting suite in France at the time of installation for ½ in masters! Well, at least the studio was at the forefront of progress. Two channels of Dolby are also available for mastering if needs dictate.

Monitoring is UREI throughout, with *815 Time Aligned* speakers driven by a *6500* amplifier and equalised with *539* attenuate-only ⅓-octave graphics. Contrary to some control rooms the EQ curve is nothing drastic—no more than 2 dB down at any one place—so the final sound stays reasonably unmangled. Secondary monitoring, well, what else but desk mounted Auratones driven by a Quad *303*.

Other listening/copying equipment includes two wall rack mounted *A77*s (Revoxes for our younger readers), Sony cassette deck, Quad *33* preamp and Eagle turntable. The remainder of the outboard gear is mounted into two low profile desktop racks that have been built as extension wings to each side of the console. As well as looking attractive they are very handy to use. The goodies mounted therein include two Klark Teknik *DN 27* graphics and an Orban *622B* stereo parametric equaliser, MXR DDL, AMS *DM2-20* tape phase simulator and *DMX15-80* DDL, Orban *41BA* stereo limiter, Eventide *H910 Harmonizer,* Klark Teknik *DN70/71* delay unit and last, but not least. two UREI *LA-4* comp/limiters. Later during my visit one of the original Aphex units was found hiding in the studio and Patrice wasn't sure about the last time that it was used! However, as they say, it's there if you need it.

Reverberation is provided by an EMT *240* and AKG *BX20*. An attempt has also been made to use the disused wine cellars as a natural reverberation chamber. The sound was a great success (as may be imagined) but there were slight problems with Radio Monte Carlo coming through rather prominently at times! This was due to the combined conditions of the transmitter being nearby and the lines up from the cellars acting as good aerials. However, the problem is going to be re-thought and we can look forward to some vintage reverb!

The studio will also be trying out a selection of digital reverberation units from AMS, Quantec, Sony, Lexicon and Eventide, with the lucky winner being installed soon after trials. A Publison *DHM 89* DDL/pitch changer is also on order —as are additional microphones.

Now that the technical list is done with, what does the control room sound like? Answer, control rooms should be neutral. Well, OK, what sort of sound is the studio capable of producing, is that better? Seriously, though, this is always a difficult question to answer as it is very much dependent on personal taste but here I go anyway. I first had a listen to a demo that had been done by a local group called Step Ahead and was straightaway impressed by a very clean and uncluttered sound. A nice broad spectrum and just the right amount of attack—or 'punch' if you prefer—where necessary. Actually, for a French group this is saying something as they always want to get the kitchen sink in there if they can! Naturally, we also listened to some of the latest Jacques Loussier productions and all I can say is that there is a very definite feeling of 'space' in the recordings—wide, with a pronounced naturalness that is very appealing.

In spite of the fact that the monitors are quite close together, the stereo imaging in the room is very good. I did a 'blindfold' test along the length of the console and the image was rock steady. The listening area at the rear of the room is very good, as well, which should stop people from wanting to go up to the console.

Moving on to the studio itself, there are two points of access: either from the control room through the double glass doors or from the far end directly from the entrance lounge.

Moving the control room to the other end of the studio obviously left a space and this has been converted to a live area, the acoustic design also having been done by Rabit Studios. This gives an area large enough for full sized string sections, brass, drums, what you will, without any space problems whatsoever. In fact, if there is one thing that Miraval is not short of, it is room to move in! The far wall is the exposed stone wall of the barn complete with mullioned window for lots of daylight. The spacing of the stones as well as their unevenness makes for excellent dispersion as well as looking attractive. The ceiling and two walls are faced with unvarnished pine panelling with the walls being made in staggered sections to cut down symmetry and reduce standing waves and flutter echoes. The panelling has also been installed with alternate inclined angles in each section and contrasts nicely with the parquet floor that makes up the remaining surface in the live area. Absorption is in the form of traps in the ceiling and side walls, with the entrance ports at floor level and either end of the ceiling. The sound is as one would expect it to be—that is, bright and diffuse and ideally suited to accoustic instruments. Drums have also been recorded in this area with great success due to the presence or 'live quality' of the sound. The drummers feel part of the group this way which makes for a better feeling in the music, a point not to be ignored.

The rest of the studio is as previously described with a moquette covered floor and a mixture of absorptive materials on the walls and ceiling with random hardwood panels. There are also two smaller windows in the outside wall which can be opened or closed with shutters as desired. As well as the extensive floor space there are three large isolation rooms averaging between 20 and 25 m² each. These date from the original building of the studio and are treated internally in the same fashion as the studio, with the exception of less wood panels. Their main use at present is for loud guitar amplifiers or the *Leslie* for the organ. However, it is planned to improve their utility in the fairly near future by rebuilding the acoustics in each room differently, thus giving the choice of different acoustic environments for those times when it is needed; effects need not necessarily be confined to electronics!

Microphones in evidence at the time of my visit were Neumann *U87* and *KM86*, Sennheiser *MD441* and *MKH 416* rifle microphones with a fair sprinkling of AKG *D202*s. Feeling the selection rather restrained, the studio is in the process of working out the shopping list for additional models. I was quite struck by the profusion of rifle microphones—a fairly uncommon sight in most recording studios. For fairly close applications these are used for vocals, cymbals, hi-hat and, on occasion, for bass drum. More distant applications include string sections and ensembles. Experimentation is obviously the key word here, coupled with the demands of the situation and result desired.

Instruments available in the studio are Bosendorfer grand (what else?), clavichord (a delightful little instrument), new model Rhodes, ARP *Pro-Soloist* and Synergy synthesisers, Eminent organ with *760 Leslie,* tack piano, four timpani, Roland electronic drum computer and Marshall stack for would-be and confirmed rockers.

Foldback to the studio is via monitors and headphones, the latter being Koss and lightweight Sennheisers. Speakers available are a selection of the original control room *Lockwoods,* JBL *4301B* (installed on the roof of the isolation booths) monitors or small Eagle wedge speakers that can be put on stands next to the music! Power for foldback is provided by Quad *303* amplifiers.

The studio itself has an interesting acoustic—not really resonant but very 'present' and natural. There is an impression of volume without needing to play loudly and this was confirmed by my having a plonk on the piano and clavichord. Discussing this with Patrice and Jacques, I was told that acoustic ensembles can hear each other very well in the studio and that foldback is very rarely required. At the same time there is a good separation between the instruments with any leakage adding a natural enhancement to the overall sound. However, should further separation be required, there are acoustic screens of different heights available, together with high screens that butt up against the ceiling to make booths. Since the building of the 'live' end, the studio area totals around 500 m² and the record so far is the Marseille Symphony Orchestra at full strength—150 musicians and no complaints of overcrowding!

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Miraval

The barn housing the studio has a circular tower on one corner. This contains the entrance lounge, offices and kitchen. Over these the space is taken up by the reserve water tank and the tower is topped by bedrooms and a terrace, the rooms being reached via a picturesque outside bridge/staircase. The studio at present takes up just over half of the length of the building—together with the control room—and approximately two thirds of the width. The remaining third gives direct access to the control room and has the toilet facilities. However, as can be imagined, this still leaves a lot of space and with the concrete walls this area is often put to good use when a 'lively' acoustic is called for but the instruments need to be separate from the main studio. At present there are no plans for the area the other side of the control room but one can be sure that the space will eventually be put to good use. A rehearsal studio, perhaps?

Miraval's philosophy as laid down by Jacques Loussier, is one of 'no surprises!' The studio charges a flat daily rate depending on the length of stay and the client is free to work the hours he wants without worrying whether he is running into overtime or not. All that is in the studio is available to the customer, eliminating the need to pay for extras. I was told that an average day is around 15 hours (how many unions would wear that!) so the atmosphere must be conducive towards work!

In fact, one of the best bargains of Miraval must be the overall feeling and atmosphere of the place. The actual domain dates back to the 13th century (and probably before) vestiges of which are still apparent in some of the buildings, and as such, the place has a very lived in air about it that seems to get through to people. Besides working, people can generally unwind with swimming and tennis in the summer and strolling around the estate with its woods and vineyards. For those who want to laze around in the off-hours there is TV in front of the open hearth fire in the entrance hall (a VCR is shortly to be installed) or there are various games tables. However, if you feel the need to dive into the fleshpots, Nice and Marseille are not that far away and Brignoles is just down the road. In spite of this, I was told that most people who come to Miraval soon get caught up with strange impulsions to work which, after all, is what they are there for. One is there to work, to create, call it what you will, and being away from the everyday distractions allows the ideas to flow, meaning that in three days it is quite possible that what would normally have taken a week to do will have been done—and better!

Since the 'public opening' of the studio, several of the old farm buildings have been renovated and turned into small apartments and rooms for clients. This follows the policy of constant improvement and further accommodation will eventually be available. I was very privileged during my stay, I was in the curé's house! (The curé was the resident chaplain in times past.) This is a charming one up/one down built on to the chapel next to the château. The chapel itself has a marvellous acoustic (I'm becoming to wonder about high vaulting, hmmm) and there are plans afoot to incorporate it into the studio complex as a small studio. It would be ideal for chamber music, for instance.

Being in France, it would be unusual if the gastronomical side of things was neglected (or the grub, in plain English!) and this is most certainly not the case. The cooking is taken care of by the charming young housekeeper, Silvie, and she kept us going with a selection of excellent menus, the pleasure of which was refined by being able to eat outside on the terrace most of the time. Due to the climate in Provence, meals are often taken outside even in November and December! I will warn the weight-watchers among you, however, that the clients leave on average with three kilos extra and that the record is ten! (Don't they feed you at home, then?) Liquid refreshment is not forgotten either—how many studios do you know of that have their own cellar? That is, produce their own wines. Being a farming estate Miraval also has its own vineyards and I can tell you that the wines are excellent; light with a fruity perfume and highly recommended!

Business at Miraval is very much a 'family affair' with the company being the partnership of Jacques Loussier and Patrice Quef. Madame Quef—or Dany—is the secretary with Sylvie in charge of housekeeping.

Patrice Quef started his recording career in 1967 as an assistant engineer at Phonogram Studios in Paris and stayed there until joining Jacques Loussier in Miraval. However, this is not to say that he led a cloistered life at Phonogram and indeed he says that he found it a very good training ground as he was often in contact with English engineers and thus able to learn from them. One of the difficulties in France, he feels, is that engineers are very jealous of their 'secrets' and there is not the exchange of ideas there is amongst the Anglo-Saxon community. In the light of recent *Studio Sound* editorials, we talked about the opportunities for the training of engineers in France and the outlook would appear to be pretty dismal there as well. M Quef feels that there are two factors contributing to the situation: with fewer record companies maintaining their own studios in France, there is less and less opportunity to receive a good 'in-house' training, also, more and more engineers are turning freelance and are consequently less and less inclined to share their knowledge, ending up in a vicious circle. Not that Patrice was not 'loaned out' occasionally during his tenure with Phonogram, but this enabled him to work with top producers and arrangers and thus gain experience in a way that is far more difficult today. Once again, it can be voiced that this is going to create a potentially dangerous situation as, after all, we don't live for ever and *somebody* has to ensure continuity!

Since Miraval opened its doors there has been a steady flow of well known names, both from France and abroad, with the French market including Alan Stivell, Maxime LeForestier, Charles Aznavour, Chantal Goya, Marseille Opera and, naturally, Jacques Loussier! International artists include Mama Bea from Italy, John Lewis of MJQ fame and Pink Floyd, who came to do vocals for *The Wall* as well as some instrumental overdubs. Patrice said that working with the Floyd was a very interesting experience and remembers how impressed he was by their calm professionalism and complete lack of histrionics! The latter a change from groups who feel that smashing the place up is the way to assert oneself as a rock musician — a strange idea.

Towards the end of my stay I asked Jacques Loussier what Miraval had brought him personally. The first point was that it has allowed him to experiment and take his time over projects that previously would not have been financially feasible. The freedom from money worries has given a definite boost to his creativity and ideas for composing and has allowed him to take more chances. The studio has also provided a central anchor for his affairs which led to the creation of his own publishing company in 1980 providing a financial back-up for the studio. He has noticed a change in attitude with the musicians as well. Most of the session players come from Marseille and would appear to be genuinely interested in the music they are to play than in the clock (at least they don't have to feed the meter!). As a composer, M. Loussier finds this an enormous aid in bringing his music to life and thus making for recordings that are more exciting than just mere renditions. I suppose that the 'community atmosphere' must rub off on everybody. At present Jacques Loussier is engaged in a project with Jon Anderson of Yes and in December should be getting together with another Jon, this time Jon Lord.

Studio Miraval very definitely has its sights on the international market and is in a constant state of renovation, be it technical or the installation of an indoor swimming pool underneath the studio or the opening up of further accommodation. There is still plenty of space for enlarging the studio when necessary and future re-investment plans include upgrading the 24-track recorder—or getting two for 46-track recording. As it stands, the studio is fully capable of first class results and if you feel like mixing work with gastronomy, peaceful surroundings and friendly people, then Studio Miraval could be the place for you—you won't even have to mortgage the house to pay for it, rates are very reasonable.

I would like to thank Jacques Loussier, Patrice Quef, Dany and Sylvie for their most warm welcome and hospitality during my visit. I look forward to seeing them again soon and until then, au revoir.

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Festival Hall sound

It *can* be done. When Mel Tormé, Carmen McCrae and George Shearing appeared 'for the first time together on a London stage' at the Royal Festival Hall they used a high powered rock sound system. It was run at tactfully low level to give just a little lift where needed. And when jazz singer Carmen McCrae did the old Anthony Newley song *What Kind of Fool Am I* she started off using a hand mic but halfway through dropped it to waist level and then right down to her side, so that it wasn't giving her any help at all. She just belted it out, over her backing trio, to fill the hall with real live sound.

It says a lot for Carmen McCrae's voice that it lost very little in level. It says a lot for the sound system that there wasn't much of a change of quality. I'll lay good money it's the first time a jazz singer has sung even one song in the Festival Hall without amplification. Who will be first to try a full set? After all opera singers do it at Covent Garden, with a full orchestra, every night.

Entente almost cordiale

There's a happy ending to my previous story of the sound at the annual Nice Jazz Festival. As reported, this 10-year-old, 10 day event, arguably the longest and largest jazz and jazz-rock binge in the world, was originally marred by very poor sound in the three open air stages. A local French firm was employed to provide what was really nothing more than a PA system, with horns of the type you normally find behind a cinema screen. Then last year the promoters, George Wein and Simone Ginibre, brought in British firm Star Hire to install a pop-style sound system on one of the stages. For the first time ever this major musical event had decent speaker stacks, a main mixing desk at centre front out in the audience and an on-stage monitor mix.

But that was on only one of the three stages. The other two still had stacks of ageing horns, no on-stage monitor mix facility and a main mixing desk so far off to the side of the stage that the engineers couldn't hope to hit a decent balance. The political fur flew when some of the major acts, especially the jazz-rock and electric blues bands, played once with the French sound system and then insisted on playing their remaining sets on the stage with the British sound system. This created a hot potato because the Nice Festival relies heavily on support from the Nice town council which is in turn obliged to support local enterprise, like the PA firm.

In last year's report I wondered how the festival organisers, who must keep musicians and audiences happy with a decent sound system, could also manage to keep the local council happy. They did it with a delightfully diplomatic compromise. This year Star Hire was contracted to handle the sound for the two main stages, with the local French company contracted to handle the smaller stage. By carefully juggling the music, so that none of the major acts or electric groups played on the French stage, the organisers preserved French honour but kept the musicians and audience happy. And this year even the mixing message finally got through. Every stage had a mixing desk out in the audience at the centre front.

It's ironic that it's taken until the 80s for the oldest pop music of all, jazz, to take sound reproduction seriously. Equally ironic, it's only been made possible by spin-offs from the development of new technology for the pop music industry which jazzers love to hate. But even the biggest names in jazz still travel without a sound engineer and rely on whoever or whatever is available at the gig. Only the jazz-rock group of Jaco Pastorius brought their own sound engineer this year, while the big bands of Woody Herman and Lionel Hampton didn't. Nor did the blues band of Buddy Guy or even the Fats Domino rock and roll show. In fact the first show by Domino was an audio disaster because curiously, for someone so long in the business, he seemed to understand nothing about stage sound.

Domino insisted on singing a couple of feet off the microphone, but he wanted the drum kit hard by his piano and wouldn't use a boom stand for the mic. He also listened to advice shouted from ignorant camp followers around the stage. At one point he insisted that the sound engineers boost his vocal sound by pointing all the monitor speakers out into the audience. Then of course he couldn't hear himself sing. In fact, with eight 1 kW Turbosound stacks, four at each side of the stage, there was more than enough power in reserve. But how do you explain to a featured musician halfway through his act, that the side-of-stage stacks are designed to fire out into the audience, which is why you need onstage monitors, and if you sing two feet from a mic near a drum kit there's a limit to how much gain you can get without feedback and drum noise! The puzzle is, how can an artist like Domino be on the road for 20 years and still have learned so little about stage sound? Perhaps we've all grown so accustomed to big name pop groups touring with an entourage that includes a full sound crew, that we forget that there are still middle-name pop artists who are touring with the bare bones of a band waggon.

Many jazz artists still treat anything to do with stage sound with mistrust. "You guys will always get it wrong anyway," said one drummer at Nice, removing all the carefully positioned mics from his kit, and leaving just one *C451* slung over the top. Needless to say his kit sounded far too toppy. Perhaps this is why some jazz groups become big box office draws while others, who are equally competent, keep making only bread and butter money. The successful group is the one that has learned how to use stage sound. In short, they regard amplification and electronic aids as an ally, not as an enemy.

There was one useful lesson to be learned for the future, and one sour note. The lesson for the future is on component stability. When you put a black mixing desk out in the hot Mediterranean sun, the temperature inside gets like an oven. Components, especially integrated circuits, aren't rated for oven heat. This year Nice in July was hotter than ever, and once in a while an IC would go into hf oscillation, or expire altogether. In future, designers of mobile sound systems may have to look again at the temperature rating of the components they use.

The sour note was musical. Promoter George Wein used to be a jazz pianist of adequate ability. He even made a few records before making his name as a promoter of jazz concerts. Over recent years he's joined in the occasional jam session at the end of his promotions for a bit of fun. "Good old George," says the band and audience, applauding him as we applaud a vicar when he proves that he can just about ride a motor bike. But this year George Wein appeared repeatedly, playing with awesome inadequacy and putting a damper on the performance of some normally fine musicians. If George Wein has a best friend, he should tell him. After all, Harvey Goldsmith hasn't joined in with Dylan, The Who, Simon and Garfunkel. Or at least not *yet*. Heaven forbid that George Wein should start a trend.

Abbey Road

Apparently 20,000 Beatles fans turned up at Abbey Road and paid £4.50 each to worship in Studio 2 because the Beatles once made music there. There's a story, possibly apocryphal, going round about PM Margaret Thatcher's reaction when told of the Abbey Road experiment. "Why turn a recording studio into a shrine," she is reputed to have said at a music biz function. "Can't they get enough work for the place?"

Custom sheik

Video is already big business in the Middle East because off-air television is even more boring than Channel 4 in Britain. There's also a natural national aversion to film. In that part of the world there's all manner of family feuding, with frequent killings for vengeance or ambition. When you show a film you have to put out the light and sit in rows watching a screen. So most people have someone behind them in the dark. It is very hard to enjoy a film when you are worrying about a knife in the back. But you can watch video with the room lights on, sitting in semi-circles round several monitor sets.

How long, I wonder, before the Middle East spawns a recording industry? If it ever happens then you had better know what you are taking on if you take a job there.

All the unsavoury publicity for social life in Jeddah created by the Helen Smith inquest reminded me of a friend who went out there to work. He was a part-time musician and jazz buff. To retain their sanity, he and his friends formed a jazz band. But all their instruments had to be smuggled in. A trombone got through the Customs as plumbing, a drum kit came in as hat boxes and a piano was cleared as spare parts for a Boeing 747. Some Janet Reger underwear for his wife, ordered by mail, had to be labelled 'handkerchiefs'.

The band didn't last long because the houseboy stole the trumpet player's gold lacquered instrument and, thinking it was pure gold, tried to melt it down. So my friend then took a part-time job on the local radio station, playing jazz records. That didn't last long either. First he was censored for playing what he announced as soul music, because only the sheiks can have soul. Then he played a record by the Joe Harriot Quintet, featuring poet and trumpeter Shake Keane. He was lucky. They didn't stone him to death, they just fired him. ■

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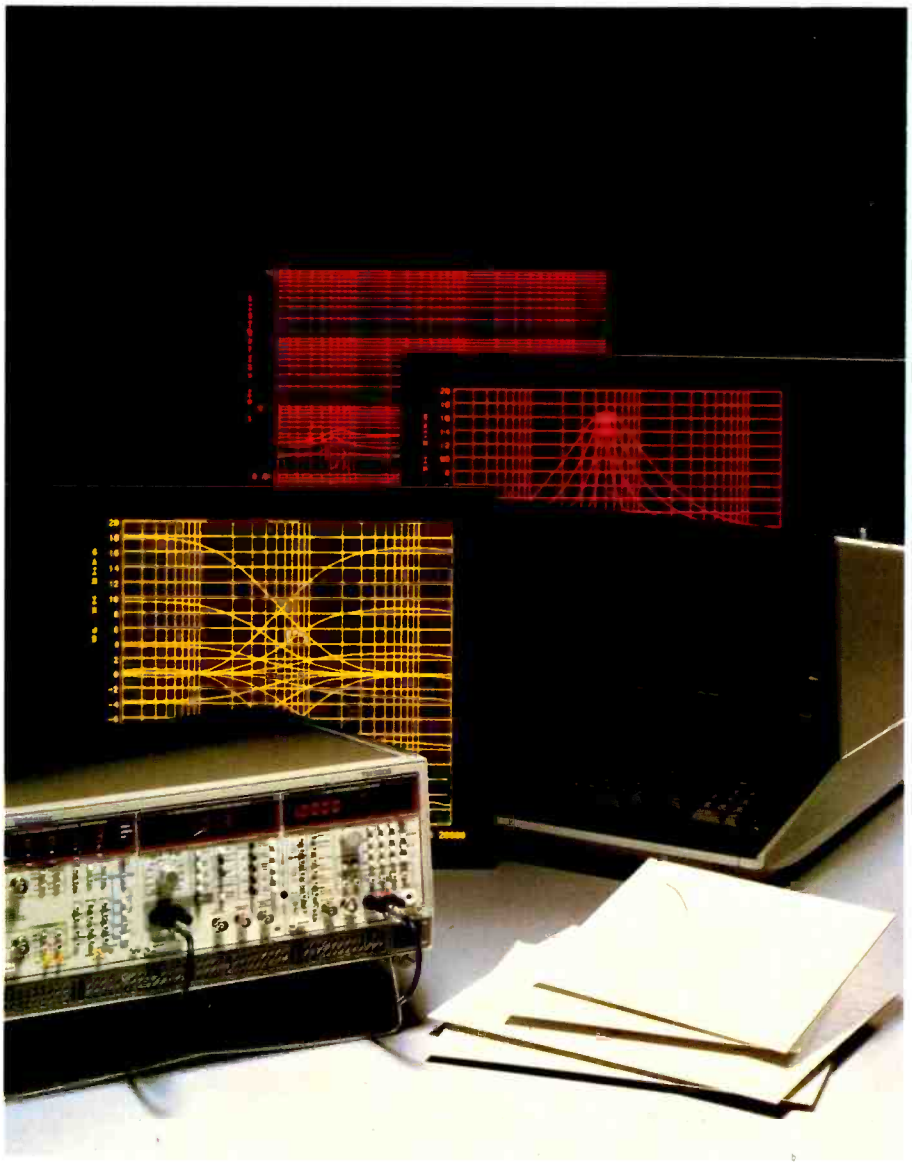

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The manipulated man

Graham paul Wayne

Gerard Paul Hodgkinson's article in the September 1983 issue, which applied conventional industrial psychology to the training of studio balance engineers, aroused a fair amount of comment, much of which centred around possible dangers of applying objective, 'scientific' techniques to training for what was essentially an artistic, 'subjective' field. The high technology of the recording studio represents a set of tools, it has been argued, which are used in the same way as any other craftsperson uses tools—to create an individual, artistic work. From this point of view the integration of the 'operator' with the 'machine' represents a dangerous dehumanising influence which, if adopted, would seriously damage the act of creation of recorded sound. In this critique, Graham paul Wayne discusses the requirement for individuality in the recording studio, and a system of training which, while not disregarding the need for formalised tuition where applicable, rests primarily on 'training by example', the traditional method by which craftspeople learn, and the traditional method by which studio engineers have learned their art.

THE 20th century system of education serving the industrial nations is founded on the need for manpower to serve the factory-state. The objectives of this institution are to impart knowledge that will enable its wards to fulfil this destiny, and no more; educational value in the current climate is measured by its cost-effectiveness. To achieve this end, the subject matter is slanted like bad journalism in an attempt to manipulate the human potential, readying it for annexation to the mighty machine. Human values are ascribed the functions of the very machine meant to serve mankind; robot reliability, clockwork performance, routines without deviation—those aspects of machine operation that it cannot do for itself. Like the educational system that engenders it, society corrupts its youth by demeaning the human spirit on which it is founded.

Industrial Psychology, as purported by Gerald Paul Hodgkinson in his article of the same title (*Studio Sound*, September, 1983) is a tool for the furtherance of that spiritless process.

As with many experts, the sub-language of their discipline gives rise to wordy confusion in which the simple is defined by the complex. From this 'complex complex' the psychologist draws generalities by which he makes his knowledge useful, but in this case those rules form the bristle of a broom with which he would sweep under the mat those undesirable vagaries of human nature he seeks to suppress in the name of efficiency, the aspect of human nature I call Individuality.

We all learn, work and play at different times and in different ways. Some people display talents that are unique to them and vary greatly in the way they apply their expertise. The studio engineer is a case in point. Only this individuality makes the meld between a certain engineer and producer, or engineer and artist, possible.

By regarding man and machine as one unit, as in GPH's man-machine model, mechanical thinking must prevail, for a study is only possible given certain constants; those of machine predictability. To make sense of the model, man must be subject to the same rules of function as the machine in order that a generality may be inferred: given a machine of x design and a man of y design (?), z will occur. Man is asked, if he's lucky, to sacrifice his human-ness to the abstractions of industrial psychologists.

To assess an individual applying for a post demands the insight of another individual of equal experience at least, who will make a decision based on subjective reactions to the person, data about that person and knowledge of the job entailed. To lose faith in one's judgement and substitute a set of models as the new criteria for selection is a foolish move when the models will only serve to screen out the wheat in favour of the chaff. Neither can training be subject to this formalisation for people learn at dissimilar speeds and in differing orders, given the opportunity. That they learn well should be the main objective. No amount of training will ensure the recipient becomes a good sound engineer, or anything else for that matter. Heart, love of the work, care and respect for ones own standard of performance; these are the deciding factors in the acquisition of high standards. Many prospectives have their illusions rudely shattered when subjected to the unglamorous routines and tediums of studio life.

Conversely, I have no doubt that a few dodgy-looking lads have become fine and dedicated practitioners of the art. Some far-seeing and presumably mystically-inclined studio manager had to give them a chance. Our psychologist wouldn't.

Of course there are indifferent studio engineers and always will be no matter how they are trained. But if we are to consider improving the method of passing on this information where should our priorities lie; towards efficiency, and consistency between batches like our conveyor-belt schooling of children, or to promote the individual exercising of creativity, choice and responsibility? And is it possible to design a training scheme that will encompass the huge diversity of product and work environments that the trainee may encounter? These problems point towards 'on-site' instruction where the practice of the craft is made relevant by actual product, not hypothetical ones.

However, GPH has little apparent regard for the informal teaching process. He bemoans the 'subjective and unsystematic manner' in which trainees are enlightened. Would he then recommend a thorough restructuring of our subjective, unsystematic industry and its varying, various products towards some mass-produced constant, in order to apply his industrial psychology to it? Some of us like the excitement of a fast-developing industry where change and growth are the constants to which we are committed. New technology is the catalyst for much of this growth. If GPH and his fellows seek to restrain this energy by generalising it into immobility and analysing it into impotence they have failed to see that the key figure in this revolution is the one to whom the rules are inapplicable: the Individual. Mr Normal may use new technology but Mr Individual will be the one to discover new ways to apply it to the life he leads in order that it be more satisfying to him, more creative. One is conditioned to suppress his individuality, the other trained to exploit it. GPH's 'training regimes' produce what his terminology so tellingly implies: regimented results, regimented minds, regimented progress, if indeed there could be such a thing. Here by way of example, the typical BBC sound engineer of 10 years ago springs to mind; all technique and no compassion for change.

My final comments concern the teaching of skills by craftsmen as opposed to professional pedagogues following a formal syllabus.

'Skills are handed down by way of tradition,' sneers GPH. In doing so he seems to discount the value of all our 'pre-psychology' culture, produced by artists and craftsmen trained exclusively in this fashion. They followed their traditions; reliable guides, for they evolve by continually sifting together the grain of method and the seed of invention from generation to generation.

Traditions are the fruit of common sense, but they require another ingredient to sustain them and keep them alive: love of the craft. This is the vital force that a trainee can assimilate by close observation of a craftsman at his work.

No contradictions exist. The bored teacher sets a boring example, no matter how much theory he may belabour us with. The student must have the vital example. He must witness change, growth and enquiry on the part of his teacher in order to understand fully that the learning of a man does not stop at school-leaving age but continues through the course of his life, a part of the process of growing as a person. The engineer who learns these lessons is one whose work we will probably respect, setting the standard for those who follow. Those who have already blazed a trail may tell you of an engineer's work that they admire, who they emulated or learned from. They learned, as do all the monkey family—by example. One way then to improve the skills of young engineers would be to expose them to examples of differing styles of approach, in different studios, making the range of possibility clear and consequently exciting. Perhaps an exchange scheme, of tape-ops between studios, would promote a better standard and lead to improved communication between trainees and those with valuable information to pass on.

Without stepping into the quicksand which surrounds those arguments concerning any measurement of an abstract product such as recorded music, I hope this article may offer some pertinent observations about the characteristics of a good studio engineer, and some comments regarding how *not* to train one. To care for one's craft is the most important attribute a trainee can display, for that is the essential motivation to learn and improve. Caring is not subject to the definitions or measurements of industrial psychology and it seems unlikely that such tools can ever serve the recording studio industry to any great extent. ■

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Digital misconceptions

Everything in professional audio has its problems, and digital audio is no exception. Because digital audio is new and imperfectly understood by many of us, and because some manufacturers and users feel threatened by this new technology, a lot of nonsense is being said and written about it. This article is intended to help clear away some of the mystery and misconception which still surround this new, exciting and challenging technology.

Keith Armstrong was a section leader of Neve's Digital Console Project, responsible for Analogue and Conversion systems, until he left to join Audix in March 1983.

Keith Armstrong
C. Eng, BSc (Hons) MIEE, MAES

SEVERAL references are given at the end of this article for those who are inclined to pursue the subjects raised in more detail. Barry Blesser's masterwork (Ref 1) published in 1978 is still the major contribution. Re-reading it recently, I find little to amend even in the light of all the R & D that has taken place since its publication. However, I do not intend to go into detail here. Having experienced most of the problems of digital audio at first or second hand (courtesy of my colleagues) I shall leave maths to one side and discuss them in practical terms, concentrating on how they sound and measure.

Converters

Converting the real-world (analogue) signals we all know and love into strings of digital numbers and back again is a tricky business. Despite the best efforts of many major companies professional quality analogue to digital converters (ADCs) and digital to analogue converters (DACs) are few and far between. Burr-Brown may be recommended for ADCs and DACs, Hybrid Systems for DACs, with one or two companies such as Analog Devices and Analogic coming on well. I reserve judgement on the converters used in the various *Compact Disc* systems until I can evaluate them properly.

Converters are a real rat's nest of problems but these are solvable at reasonable cost, as described below.

Quantisation and dither

Digital coding systems have limited resolution. In the linear-coded system that is the current professional standard the number of digital bits used is a figure of merit, resolution increasing by 6 dB for each bit added, giving us a theoretical resolution of 96 dB (below maximum level) for a 16-bit system. The implications of this for the dynamic range are discussed later.

Looking at a waveform which has been through a basic ADC-DAC combination we see that it looks as if it is made of Lego bricks, as everthing is a series of staircases of varying slope. Of course, the post-conversion ('reconstruction' or 'anti-imaging') filter smooths away these tiny transitions but that does not stop them from being a nuisance at low signal levels, where the relative sizes of the quantisation levels (the Lego bricks) are more significant. In practical terms this means that distortion is very good for maximum level signals (0 dB) but increases rapidly as the level approaches −96 dB at which point the signal is turned into a perfect square-wave with a terrible sound. Below −96 dB the signal is ignored completely!

Luckily, the 7th Cavalry is on hand, in the shape of 'dither', to rescue us. Dithering means adding a small uncertainty, and in audio we use about 0.5 of an LSB of white noise or of a square-wave at half the sampling clock rate, or a mixture of both added to the signal before the ADC or its preceding filter. The popular misconception here is that dither is only a masking effect, but this is not true. Dither, properly applied, allows the system to handle signals which are even below −96 dB with low harmonic distortion. Blesser (Ref 1) describes this on pages 744-745 and an even more comprehensive analysis can be found in reference 2.

An undithered conversion system used on a piano recording with the program level attenuated to −70 dB below system maximum sounds as if it has had a combined fuzz-box and axe job done on it! With white noise dithering it is transformed into real piano, complete with all the nuances of ambience which must be somewhere around −110 dB, plus white noise. With half-clock dither the background noise level is diminished, but is less white-sounding, and once again the very quiet piano is reproduced in full detail.

Dither is essential in any

professional ADC, whether it is part of a tape recorder or a console. Use the attenuated piano test to check it out.

Quantisation errors

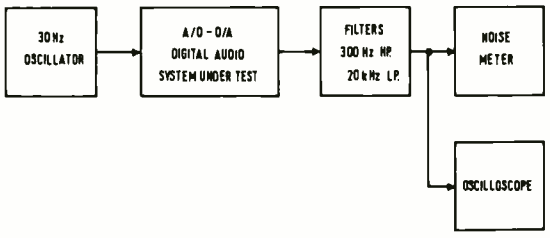
If all the Lego bricks we used to build our 'digital waveforms' were equal in height then dither would work perfectly. Unfortunately, quantising levels can vary dramatically in their steps: a converter is called monotonic when its differential non-linearity is no more than ±0.5 of an LSB. This is considered a good converter, even though some of its steps are close to being no steps at all, and others are almost twice as big as they should be! A favourite converter trick is to introduce quite major quantising errors when the digital codes exhibit dramatic changes, for example the change 011111111111111 to 100000000000000 is about as major as can be and often puts a large error in the quantisation right where it is least wanted, at 0 V. A popular technique to overcome this particular error is to offset the analogue audio into the converter by a small DC voltage so the error is only excited by largish signals and is thus less obtrusive (Ref 1, p.757).

Purchasing good converters is the only answer. The dither level is then set to that which gives the best compromise overall. The relevant figure of merit in this case is usually the monotonicity of the converter expressed as an equivalent number of bits and should be found on the manufacturer's data sheet. A good 16-bit linear converter would be monotonic to 16 bits. Some manufacturers actually quote quantising error in terms of differential non-linearity but beware of those who merely quote non-linearity because this is a different animal (see below) and sounds different too.

The quantisation errors of the converters in a system show up in the sound as modulation noise, which is of course only there when the signal is and is not measured by the usual no-signal tests. This noise is not as subjectively annoying as steady background noise because it tends to be masked more by the signal producing it. Mr Gilchrist and his colleagues at the BBC have been measuring modulation noise in converters using a modified 30 Hz distortion test setup (see **Fig 1**) in which virtually all the distortion products are removed by a 300 Hz high-pass filter leaving only the modulation noise to be measured (Ref 3). This straightforward technique is effective and correlates well with normal noise measurements. Using this method I have measured a supposedly pro-quality system to have a modulated signal to noise ratio of only 40 dB! Although there were some reservations about the sound, it was a lot better than one would expect from that measurement.

So use this test to check that the converters you are purchasing are as good as they are supposed to be as far as quantisation errors are concerned.

FIG 1
TEST SETUP FOR MODULATION NOISE (See Ref 3)



Linearity

Like monotonicity, this is a slope problem but is less concerned with the control of Lego brick sizes than it is with the overall transfer characteristic on the large-scale. Thus it is comparable with analogue linearity and produces the same sort of distortions. There is nothing for it but to use quality converters, but at least any resulting distortions sound like analogue and are more pleasant than most digital errors.

Sampling

Part of every successful ADC will be a sampling circuit known as a sample/hold or a track/hold, and DACs will use a similar circuit for different reasons often referred to as a deglitcher.

It is no understatement to say that these circuits are probably the most critical to the performance of the overall system, and their design is an art in itself. In order to achieve professional quality the sampling clock (48 kHz or 44.1 kHz) should have a jitter of less than 5 ns (Ref 1, pp. 759-60) and the actual sampling time should not vary (due to signal level, for example) from this by more than 200 ps. Dielectric absorption in the capacitors used is so critical that only polypropylenes or polystyrenes will do.

Poor sampling-circuit performances result in distortion figures which rise dramatically above 1 kHz, and excessive HF garbage. Careful design is all that is necessary to overcome this.

Layout

The physical layout and arrangement of earths, decoupling and screens make all the difference between a conversion system that works and one that doesn't. The solutions to all the problems mentioned so far can be set to zero by lack of attention to the PCB itself. With analogue circuits, high speed samplers and digital electronics all intimately connected, there are endless possibilities for errors to be introduced that show up as excessive noise or distortion.

As with all things, the proof of the pudding is in the eating. In this case listen to and measure the system carefully even if it is chock-full of expensive modules or ICs.

Dynamic range

Of converters

It is a popular misconception that 16-bit digital audio using linear PCM has a dynamic range of 96 dB. In practice, taking into account all the sources of noise and employing dither, a 90 dB range (noise to clip) is the state-of-the-art (although the system will resolve signals below the least significant bit due to the dither). This is better than analogue tape or disc and accounts for the sales of digital mastering machines and *Compact Disc*, however it is insufficient for the uncontrolled signals present at the inputs of an audio console. We need to be able to convert signals with a dynamic range of 110 dB or so (noise to clip) in order to have digital audio consoles which perform as well as current analogue ones. The problem is all in the ADCs: 16-bit ones with pro quality are not exactly cheap at present; 18- or 20-bit ones are a dream. Once into digits we can add as many bits as we like to preserve headroom, noise floor, etc, in our equalisers, mixers and other processors, but that first conversion is a stumbling block—90 dB is not a very good specification for the output range of a console, either.

The solution of this is left, as they say, as an exercise for the reader. There are a few schemes available, of varying merit, whether and how they are implemented and licensed will really sort out the men from the boys in the digital audio console business.

Of digital processing

Another popular misconception is that digital consoles are stuck with their '96 dB' when they come to do processing such as mixing, equalising or whatever. In fact (mentioned above) once we have the signal in digital form all the techniques of the computer designer become available, and we can add as many bits as is necessary to preserve our signals. An analogue mixing circuit of any merit will have a dynamic range of 110 dB at least, so a 20-bit digital mixing bus will give us comparable performance (approximately 120 dB). But in the digital domain we can add more bits to our mix bus: if we add another 4 bits we can mix 256 coherent maximum-level 16-bit signals together without clipping. Try designing an analogue mix circuit with this sort of capability at a reasonable price! Similar arguments apply to other types of processing although the cost of all the extra bits may mount up considerably.

So it is clear that digital audio processing promises to enhance quality and ease of use, rather than restrict it. This is especially clear when the fact that the trend in digital signal processing ICs is towards 32-bit arithmetic (...192 dB!) is taken into account.

Wraparound and clipping

Digital circuits try harder than analogue ones: when you drive them past their maximum level (ie number) they go back to the beginning and start again, unlike analogue circuits which just clip and go no further. Unfortunately, going back to the beginning and starting again, known as wraparound, sounds much, much worse than clipping! Wraparound is thus to be prevented if at all reasonable.

In the ADC

There are two areas susceptible to wraparound: the ADC and any processing. The ADC can be dealt with by previously clipping the analogue signal, but this must be done prior to the pre-conversion filter (or 'anti-alias' filter) if interesting non-harmonic 'aliases' are not to be introduced by the high frequencies produced by the clipping (Ref 1, p.757). In turn this means that the clipping level must be somewhat less than the ADC wraparound level to allow for the overshoot in the pre-conversion filter (see Filter squarewave response) and this helps to reduce the available dynamic range (noise to clip) of a 16-bit ADC to 90 dB. However, there is no real problem here.

In the processing

In the digital domain, processing often results in signals having a lot more than 16 bits, but if enough bits have not been allowed for all combinations of signals and controls, then wraparound is a possibility — and it sounds awful. One solution is to employ a clever routine that clamps the digital numbers and stops them from wrapping around. This is fine, except that because it occurs within the sampled system, it produces aliases which are not harmonically related to the signal. In practice, all this means is that clipping sounds rougher when it occurs in the digits than one is used to in analogue, but there is not a lot in it and it is far preferable to wrap-around.

Filters

Every pro-audio digital system will involve two amazing filters: a pre-conversion one (often called 'anti-alias') before the ADC; and a post-

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Digital misconceptions

conversion one ('reconstruction' or 'anti-image') after the DAC. These filters will almost always be elliptical types of around 10th order with 'brick wall' responses, and their design and implementation is not trivial at all!

Frequency response

The first common complaint that relates specifically to the filters is that a 20 kHz audio bandwidth is just not enough! My experience so far (and Ref 4) indicates that it is, in fact, good enough providing that the problems described below are dealt with as indicated. I suspect that many of the experiments that have been performed have not used optimal filters and so what was found objectionable was not, in fact, related to the frequency response but was caused by time smearing and/or distortion. I have yet to find anyone who could hear whether optimal 20 kHz filters were in or out of circuit.

In-band frequency response ripple is another phenomenon of steep filters. I have little reliable data on the audible effects of this and suggest that +/−0.2 dB overall ripple for a system is inaudible and readily achieved.

FIG 2
DISTORTION PERFORMANCE OF INCORRECTLY DESIGNED BRICKWALL FILTER

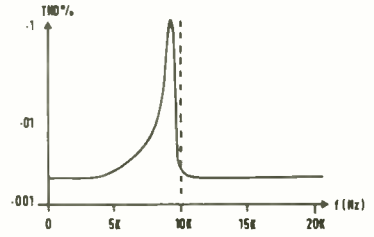
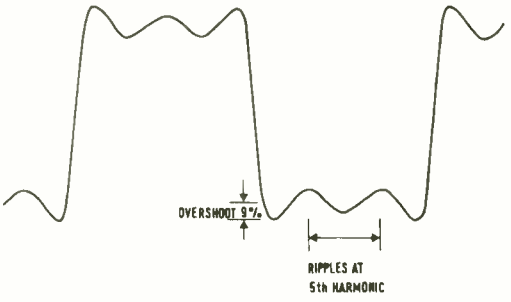


FIG 3
TYPICAL APPEARANCE OF 3 kHz SQUAREWAVE AFTER PASSING THROUGH AN IDEAL LINEAR-PHASE 20 kHz BRICKWALL FILTER



Time-smear

The phase response of a straightforward brickwall filter suitable for digital pro-audio is a fearsome thing! It is easier to get a handle on (and related more closely to subjective effects) if it is described in terms of group delay. A fixed group delay of, say, 50 μs is neither here nor there —you just have to wait that much longer to hear the sound. What is important is any group delay distortion. If the delay varies with frequency (or any other parameter) then some frequency components of the sound will appear at your ears earlier or later than the other frequencies that they started off in synchronism with: this is called time-smear (Ref. 1, pp 758-759) for obvious reasons. It is not unknown for the kind of filters required for digital audio to have group delay distortions approaching 1 ms on a nominal delay of 50 μs. This produces time-smearing which is sometimes noticeable on such sounds as piano transients. The cure for this is to fit a delay-equalising circuit, which is an all-pass phase shift network tailored to the filter in question so that the response of the combined circuit exhibits trivial delay distortion. Of course it costs more and takes longer to design, but there is no physical reason why time-smear should be audible.

Once this is taken care of the measured phase shifts will indeed be large, but will be consistent with a distortionless group delay (i.e. linear phase) and therefore of no importance as long as they are within certain relative interchannel tolerances, long familiar to designers and specifiers of consoles with many transformers in the signal path.

Distortion

Elliptical filters of high order required for brickwall response consist of multiple poles and zeros all arranged in a very clever way. Unless care is taken too many zeros could be upstream of the poles and cause premature clipping of the higher audio frequencies. The trouble is that the following poles roll the clipped products off so sharply that they never reach the output of the filter! Thus we may have a filter which meets its THD specification but sounds lousy—if we had invested in IMD equipment we would have spotted it right away because the distortion products falling in band (just as they do with real program) would not be filtered out in the same way.

Fixing this potential problem is generally quite easy for the filter designer, once he realises what the problem is. If you are only equipped (as so many of us are) for THD measurements alone you could spend a long time wondering why your system measures well but sounds bad. However, there is a way of using THD analysers to point to this problem. A plot of THD against frequency in the region 2 kHz to 12 kHz, in 1 kHz steps, will reveal all. Tell-tale signs are THD that rises steeply as 10 kHz is approached, falling sharply thereafter (see **Fig 2**). Conventional quality assurance tests at 1 kHz, 5 kHz, 10 kHz, 15 kHz for example may totally fail to notice this phenomenon, which usually peaks sharply at just below 10 kHz.

Having identified the problem, the solution is fairly straightforward and need never cause trouble again.

Squarewave response

Don't bother to run a squarewave test on a piece of digital audio equipment—it won't tell you much at all. Likewise, don't reject the equipment because its squarewave response is much worse than you would consider tolerable for analogue equipment.

The problem here is known as Gibbs phenomenon and is caused by the brickwall steepness of the conversion filters stripping off the harmonics necessary for good squarewaves with unprecedented success. A perfect brickwall filter with linear phase (ie no group delay problems) and bags of stability in hand will exhibit 9% overshoot. It is easy to demonstrate to yourself how this occurs if you take some graph paper and 10 minutes to try and reconstruct a 3 kHz square wave from the well-known formula: fo + 1/3(3fo) + 1/5(5fo) + 1/7(7fo) + etc—but cutting off the expression after 20 kHz (ie 5fo in this case) because our filter will not allow 21 kHz or higher to pass. The result will be a pretty poor-looking squarewave with over-shoot and ripple (see **Fig 3**) and this corresponds exactly with what you can expect from a pro-quality digital audio system which sounds perfect.

We audio engineers are very much the poor cousins of telecommunications engineers who, although they may be dealing with 200 Hz—3 kHz and several percent distortion nevertheless use analysis equipment worth tens of thousands of pounds. Consequently we have had to make do as best we can with simple tuned-notch THD sets and short-cuts such as squarewave tests. Squarewaves can tell us a lot about transformers and their phase correction, and a lot about amplifiers slew-rate and stability margins, providing that the overall system is tolerably wide band with gentle roll-off rates. They tell us very little about digital audio systems and it is probably best if you don't look at them, because you may be tempted to reject the system even though it is, in fact, performing with perfect fidelity (in its passband) and sounds excellent.

The squarewave test is not a suitable shortcut to obtain an idea of the overall quality of a digital audio system.

Conclusion

I have shown that although digital audio has its problems they are mainly headaches for the design and the acceptance engineers. Properly engineered digital audio products sound and measure as good or better than analogue audio products in most respects, with the exception of input and output dynamic range where digits give more than analogue tape or disc but less than analogue consoles. The decision on which to buy will thus centre on considerations of cost, reliability, ease of maintenance, facilities, existing equipment and charisma. It is inevitable that some early digital audio products will have their sound problems; the careful manufacturer and the careful customer will hopefully take account of this article and will never have problems of audio quality. ■

Author's Note

The references referred to in the text and listed below all have extensive lists of references of their own.

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Acknowledgements

Much of the work leading to the opinions and results expressed here was performed by my colleagues, to whom I am most grateful not only for their engineering skills but also their good company: David Cox (Technical Design Services), John Lancaster (Neve), Bill Whitehead (Neve), Steve Douglas (Neve), Charles Walker (Neve).

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studiofile

The Garden, London

John Foxx was the original vocalist with Ultravox and stayed with the band for two years until March 1979, when he left and was succeeded by Midge Ure. His first subsequent solo album was entitled *The Garden* on Metal Beat—Virgin Records.

His recording studio, also called The Garden, is now run on a commercial basis, though the whole set up grew out of his own requirement for recording facilities after leaving Ultravox. Corinne Sumcock has managed the studio and been resident engineer since May '82 when John started hiring out on a commercial basis. Corinne and John generate the kind of atmosphere that purely commercially oriented studios seldom offer. Clients are made to feel that the studio is there primarily for them to make good recordings and secondly, to make money. When Corinne is not engineering at the studio, she delegates from a shortlist of highly regarded freelancers. In essence, The Garden is a very nice place to spend time. John's own personal tastes have been reflected in the general design and décor as the place has a light and warm feel.

In fact, the studio has proved so popular (users include Depeche Mode, BEF, Robin Lumley, Mick Glossop, Mike Hedges and Daniel Miller) that John has now been forced to buy a barn 'somewhere in England' to set up a new 24-track for his own use. He likes to record all year round, recording almost twice the material he releases. Also, he tends to arrange for musicians to record somewhat spontaneously, only to find that his studio is booked solid.

John set up the studio in conjunction with freelance engineer Gareth Jones and studio designer/acoustical consultant Andy Munro.

You may recall the mention of John Foxx's name in the Studiofile on Jacob's studio in Surrey in last October's issue. At the time John had all the equipment he needed for a multitrack studio but lacked somewhere to house it all. He was undecided about his future plans, though he had acquired the present Garden premises. The studio is in the basement of a very individual building situated on the corner of Holywell Lane and Shoreditch High Street, to the east of the City of London. Just look out for the distinctive corner clock tower.

John acquired the building in partnership with three friends and together they have turned the one-time toy factory into a veritable creative hothouse. Chris Gabrin has a complete filming set-up which he uses to make promo videos for bands including Culture Club, Peter Mackertich takes photographs, while environmental sculptor Dennis Masi sculpts rooms and John makes music and has his studio in the basement.

When I visited The Garden, John explained how initially, his equipment found its way down to Andy Fernbach's Jacob's Studio down in Surrey.

"I'd bought all this equipment and I was waiting for this place to be finished. I wasn't sure if I was going to start a studio or keep the equipment for my own use and use this as a rehearsal space or whatever. It was all kind of fluid at the time.

"I heard about Andy's studio and he didn't have very good equipment in there at the time. So I thought I could leave my equipment down there, use it when I wanted to and let Andy use it for his studio if he wanted. So we did that for a while and it worked out fine. I did an album there."

John decided that in the long term he would want studio facilities nearer to London merely from the geographic point of view. Travelling up and down to Surrey would eat up too much time, though he regards the arrangement with Fernbach to have been a good one for all concerned on a transitory basis.

"It helped Andy because at that time he couldn't really afford to re-equip his studio so he was in a funny situation. He really couldn't charge the kind of rates that he needed to to enable him to re-invest in better equipment."

To begin with *The Garden* was the title track on his album. And the reason for using it for the studio was well? "I couldn't think of another name at the time."

Prior to being a toy factory for 60 years or so, the building had been a fashion house and showroom since the end of the 19th century. As you might expect, the building is solid and well constructed. When John started the conversion work he was loath to do away with the original white tiled walls and parquet flooring: "Some of the reflections were beautiful and we were reluctant to change it."

However, he decided that some control of the acoustical reflection was desirable and so he brought in Andy Munro to do the business. John decided to ask Andy to treat the main studio room in such a way as to retain much of the original acoustics while at the same time, making them controllable. They were able to consider the acoustic treatment first and foremost because the building stands on a bed of clay, resulting in virtually complete isolation from outside noise.

Much of the control in the main room is achieved with Munro-designed floor-to-ceiling acoustic screens. John has been almost stunned by their effectiveness in allowing the creation of isolation booths as well as particular acoustical effects as and when required.

The complex consists of the control room, the main studio room, a second room which houses the Yamaha grand piano and the 'crypt', which is the name they've given to what is essentially a tiled tunnel under the pavement. This area is, of course, highly reflective and can be used for natural acoustical effects. Also the stairway down to the studio has its own acoustics, so The Garden can offer four different sets of natural acoustical settings, each with variations. They find these acoustics are ideal for a multitude of recording techniques, including playing a dry or direct injection recording through speakers to add natural ambient effects. This can add a whole new dimension to drum machines and there are many other applications, most obviously with synths.

The control room is relatively large with a floor area of 40 m^2 while the main studio's area adds up to 65 m^2.

On the subject of the design of the studio, John enthuses about the input from Gareth Jones and Andy Munro. He holds the opinion that as far as studio designers and consultants are concerned, Andy is the best. He also voiced his appreciation of Gareth's efforts, particularly with regard to the importance of choosing the most suitable equipment and general layout.

At an early stage in the planning of the studio John had wanted to equip the studio entirely with English equipment but regrets that that was not possible because of unavailability or unsatisfactory specs. "I wanted to buy English equipment for the whole studio but couldn't because it didn't compare. I was pleased to be able to buy the Amek desk which is English."

They did, however, manage to set up as a transformerless studio. On this subject John said, "I think it (The Garden) was probably the first *completely* transformerless studio in the UK. Air went transformerless about six months after we got started but it was part of our design concept right from the start. And it does seem to work, it makes the sound a bit more punchy."

The monitor system consists of Eastlake cabinets housing JBL bass units and TAD horn drivers powered by HH *V800* MOSFET power amps through a Lindsey graphic EQ. Sound processing equipment includes dbx *160* compressors, Eventide *949* and *910 Harmonizers*, a DeltaLab *Acousticomputer*, an ADR *Scamp* rack with two compressors and four expander-noise gates, two Roger Mayer noise gates, MXR flanger, Drawmer *DS 201* noise gates, ADR *Panscan*, Roland *Chorus Echo*, Lexicon *224* digital reverb, one EMT *140* stereo plate and an AMS DDL and pitch changer.

The console is an Amek *2500*. Tape machines are MCI *JH24* multitrack, an MCI *JH10* stereo machine and two Revox *B77*s for the usual echo and other support functions.

The studio piano is a Yamaha 6 ft 6 in grand.

The range of microphones includes Crown *PZM* models, Neumann *U47*s and *U89*s, AKG *C451* and *414*, *D12* and *D202* models, Shure *SM7* models, Electro-Voice *RE20* and Beyer *201* models.

Ralph Denyer

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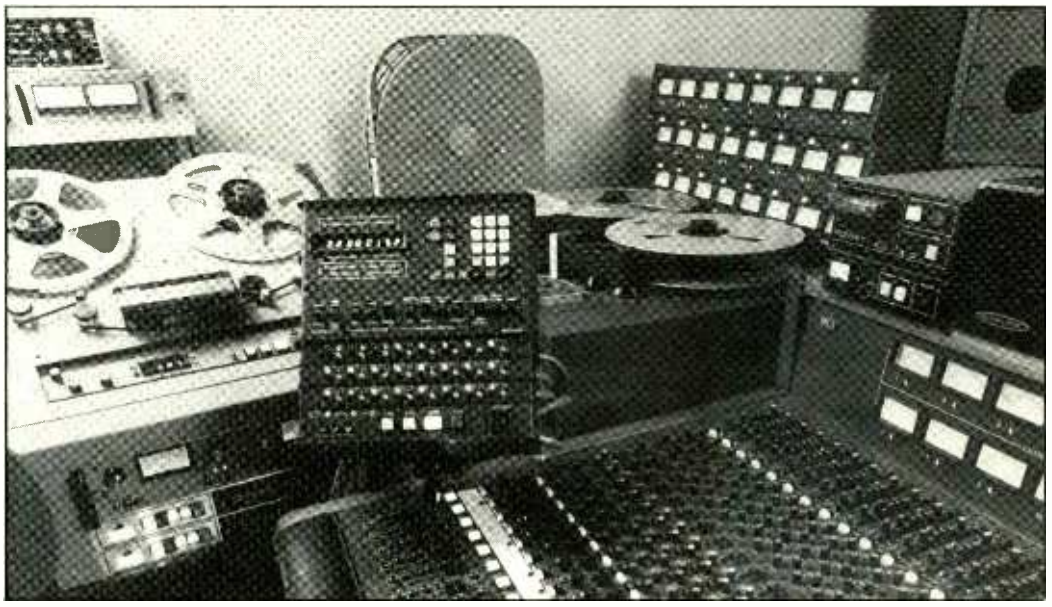
EVTR, London



You'll find video facilities in more and more studios these days, in case you hadn't already noticed. Channel 4, home video, video promos—these and other areas have opened up new work in sound production.

EVTR, on the other hand, is different. It's a video production facility that's now getting into audio—perhaps offering a new slant on the term 'recording studio'.

EVTR started off round the back of Wells Street, where the company helped to edit news stories for neighbouring ITN. The 'E' in the name stood for 'Express'. EVTR has since moved to Great Titchfield Street and into a mixture of on-line and off-line video editing and sound sweetening work for TV, though compiling news, features and sport for the syndicated television news agency UPITN forms a regular part of the business.

As we started chatting about post-production sound dubbing, my mind wandered back to distant memories of sprocketed 35 mm tape and revolving prisms. It was then that I noticed the MCI 24-track and a *JH45* synchroniser from the same company.

"We use a 24-track synchronised to a *BVU* video machine for post-producing TV soundtracks," engineer Chris Barton explained. "I prefer to use 10½ in spools because you get a much quicker servo lock on the smaller and less massive spools. Compared to the 14 in diameter jumbo reels, that is."

So who needs 24 tracks for programmes like Channel 4's *Kill or Cure*, training videos for Avon Cosmetics and a number of commercials, all recent assignments for EVTR? The answer is that the tracks are used for two parts of the programme on the one reel.

"Twelve of the tracks are used for the first 30 min," said Chris, "because after the line-up sequence at the beginning, that's all the 10½ in tapes will take. But then we use the other 12 for the remainder of the programme." This, I thought, sounded like a 'Sledgehammer strikes Nut' type of story.

"There's very little track wastage in practise," said Chris, "because you've got the two original tracks from the edited programme, and don't forget that the timecode needs two tracks—one for signal and one as a guard band. So that's four for a start, before you've begun to add the extra sounds that you've used the system for in the first place—like a voiceover, music, effects, and bounced premixes."

Surely there are enough tracks to work on without needing to bounce, I suggested.

"It's common to double up on effects tracks," replied Chris, "and you might want to bounce these over tricky joins in the original soundtrack caused by the picture editing process." It all comes down to the Ancient Law of the Balance Engineer which states that you've always got less tracks than you think you have, I suppose.

"Then there are multilingual programmes like the training tape we did for Avon" Chris continued, "where several languages can be laid side by side, with timecode and the original English guide track recorded once only. This tape, designed to show sales people how to recruit doorbell-ringing agents, took one day each to record seven European languages in the overdub room next to the control room, and another day to mix the versions.

"We can either lay down separate timecode tracks on the 24-track for each part of a programme, or use an offset on the synchroniser," said Chris, adding that EVTR uses a modified *BVU* video machine to read timecode (recorded on one of the audio tracks) while winding, as well as during normal playback.

"The two machines remain in lock throughout, with the *BVU* driving the system and providing a guide copy of the pictures. If a client wants to do several different things on a job, as with shorts or commercials for instance, different sections can be distributed across the tape, all in sync; which obviously saves time and tape."

Chris Barton's opposite number on the video side, Andy Thompson, described how the sound and vision are correlated both before and after sound sweetening on the multitrack.

"We usually dub the original soundtracks of the edited video programme straight on to multitrack with timecode," he said, "and return it after sweetening. But we could save a tape generation if sound quality was paramount by 'assembling' sound in the same way that a video tape is edited."

For those of you blissfully unaware of the joys of video editing, 'off-line' editing uses cassette copies (typically on ¾ in *BVU* tape) of the original 1 in video tapes for making editing decisions. A timecode related list of edits is compiled on a floppy disc (EVTR use a *CMX 340* editor) which is then taken with the original tapes to the auto-assembling stage.

The auto-assemble equipment ploughs through the list, dropping all the scenes from each reel in turn onto a master tape, and so building up the finished sequence in the style of a jigsaw puzzle. This process is called 'conforming'. As Andy Thompson says, the sweetened sound is dubbed back onto the final edited programme; captions can be added in the VT editing suite.

The advantage is that you only use expensive 1 in machine time (in other words work on-line) when you've perfected the edits, not while you're still trying to make up your mind. By the time you read this EVTR should have two more 'C' format machines to allow tapes to be conformed in-house with the *CMX* editor handling the whole operation.

EVTR has just installed a computer-controlled rostrum camera for adding visuals and artwork through the Grass Valley *1600 1-X* vision mixer. This complements a sophisticated Chyron caption generator on the graphics side, which like most things these days uses a floppy-disk, in this case to create typeface captions indistinguishable from artwork.

Back in the studio control room, a colour video monitor hangs above the MCI *JH600* desk, with another smaller screen installed in the overdub/voiceover room. But apart from that, we're back in the world of sound, with a curtained view of the 56 m^3 studio. It's a fairly standard line-up with the 24-track taking pride of place, alongside another MCI ¼ in.

Effects and processors include a UREI compressor/limiter, Eventide *Harmonizer,* the Valley People's *Dyna-mite,* and an Ursa Major *Space Station*. Monitoring is on the ubiquitous Tannoy *Reds*, with the equally ubiquitous and inevitable Auratone 'tranny boxes' perched lovingly on the top beam of the desk.

So far the studio has been used mainly in association with video work. The acclaimed title sequence for *Death of an Expert Witness* for example was shot in the studio, using miniaturised models to dramatic effect. Just recently a 12-part arts series for Channel 4 called *Heads* was picture-edited at EVTR, with all the sound recorded in the studio. Echo and the Bunnymen's Italian tour was also sound-and-picture edited for the Beeb's *Old Grey Whistle Test.*

But it's also a sound studio in its own right. Jazz player Slim Gaillard popped in recently and EVTR is even contemplating pressing an album of music recorded by him with jazz 'giants' Johnny Griffin, Kenny Clarke and Kai Winding. This only awaits the response from broadcast companies to a video recording made at the same time.

Other projects include a classical Brazilian guitarist, though Chris Barton sees no reason why more popular acts should not use the studio. Paul Wells, who was assistant chief engineer at CBS for six years and installed most of the sound studio, still records there.

As Chris pointed out, the video and sound operations in TV production follow each other rather than happen at the same time—and this gives EVTR a lot of scope for much more work recording music.

Richard Dean

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Sony CY-24 digital tape splicer

Hugh Ford

SPLICE editing digital tape recordings has its own particular hazards for not only is the digital tape very thin but good splices *must* be made if disaster is to be avoided. The tape is difficult to handle when using normal tape splicing blocks in addition to which some splicing blocks easily cause edge damage with thin tape. Further, whilst with analogue recordings at professional tape speeds a relatively large gap even in a 90° splice will remain inaudible, this is not the case with digital recordings.

Using digital machines, the tolerance to poor splices depends to a large extent upon the particular machine in use and its error correction system. For instance, I have found the ¼ in Telefunken/Mitsubishi stereo machine far more tolerant than the Sony *PCM3224* 24-track machine with its ½ in tape being very difficult to handle using normal splicing blocks.

It is because of these difficulties that Sony have developed the *CY-24* digital tape splicer which of course may be used equally well for analogue tapes. The block handles both ¼ in and ½ in tape in the same way with separate parallel slots for each tape width.

The splicer is based on an alloy plate 200 mm long and 70 mm deep with 40 μm deep slots across its width to accommodate the tape. The profile of these slots is such that they do not retain the tape like the *'Editall'* block, in fact the edges are slightly angled to form a 'V', the bottom width of which is just on the nominal maximum tape widths of 0.248 and 0.498 in.

Across the centre of the splicer is a 305 μm slot for cutting at 90° with a single edged razor blade, the common width of which is 250 μm giving a comfortable sliding fit. This is where the similarity to other splicers ends.

A steel rod extends the full width of the rear of the splicer to provide a hinge for four splicer 'arms' which fold across the splicer and locate positively on to spring loaded ball bearings recessed in the front of the splicer. Each of these 'arms' has a hard rubber end stop which rests on the top near side of the block and foam rubber pads which press on the surface of the tape in the slots. The tape is thus positively held in the splicing block at each side of the cutting slot by one or two 'arms'.

The two outer 'arms' are fixed longitudinally whilst the two inner 'arms' can slide on their hinges 10 mm longitudinally. The point of this is that if the tape is located with the inner 'arms' alone the edit point can be very accurately aligned with the cutting slot by sliding the inner 'arms' along the length of the splicer.

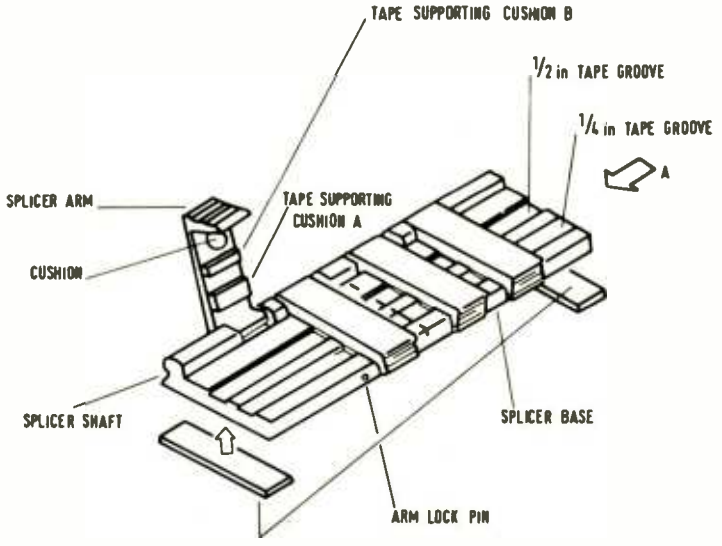
In operation the edit point is located using the analogue track and marked, preferably using a video type alcohol based marker as supplied with the splicer, as this avoids tape damage which can be caused if using a wax type marker. The tape is then rough cut beyond the edit points and placed oxide downwards in the splicing block with the edit points aligned with the cutting slot and the 'arms' closed. If very accurate location of the edit point is desired the tape can be moved using the inner sliding 'arms'.

A single-sided razor blade is then used to simultaneously cut both sections of tape at the edit point after which one or both sets of arms may be raised to remove the unwanted pieces of tape.

The two tape ends are replaced in the splicer and clamped by the sliding 'arms' in their central position. The sliding 'arms' may be used to very accurately butt up the tape ends, preferably clear of the cutting slot. The tape can then be held in place by either or both sets of 'arms' whilst the joint is made with splicing tape. The latter should preferably be 10 μm thick video splicing tape as it is easier to handle than normal splicing tapes, more flexible and less prone to stretching.

In use experiments with 20 μm thick recording tape showed that it was very easy to make excellent splices, so good in fact that the splices were invisible to the naked eye without any chance of accidental overlapping—in fact such a good butt splice that it could only be seen using a x10 magnifying glass.

My only criticisms of this excellent splicer are that sliding the moving 'arms' requires them to be gripped at the front and back such that care is needed not to touch the tape where it enters the edges of the splicer. Whilst stick-on rubber pads are provided as feet it would be nice if it had pre-drilled holes for mounting on to a recorder.



Conclusions

The development of this digital splicer overcomes my criticism of the difficulty of editing when using the Sony *PCM-3224* ½ in 24-track machine.

Splice editing the very thin tape used on digital audio machines can be very difficult with normal splicers and I strongly recommend anyone involved in such editing to try this Sony product—it gives great confidence in a sound splice without the possibility of accidental tape damage. ■

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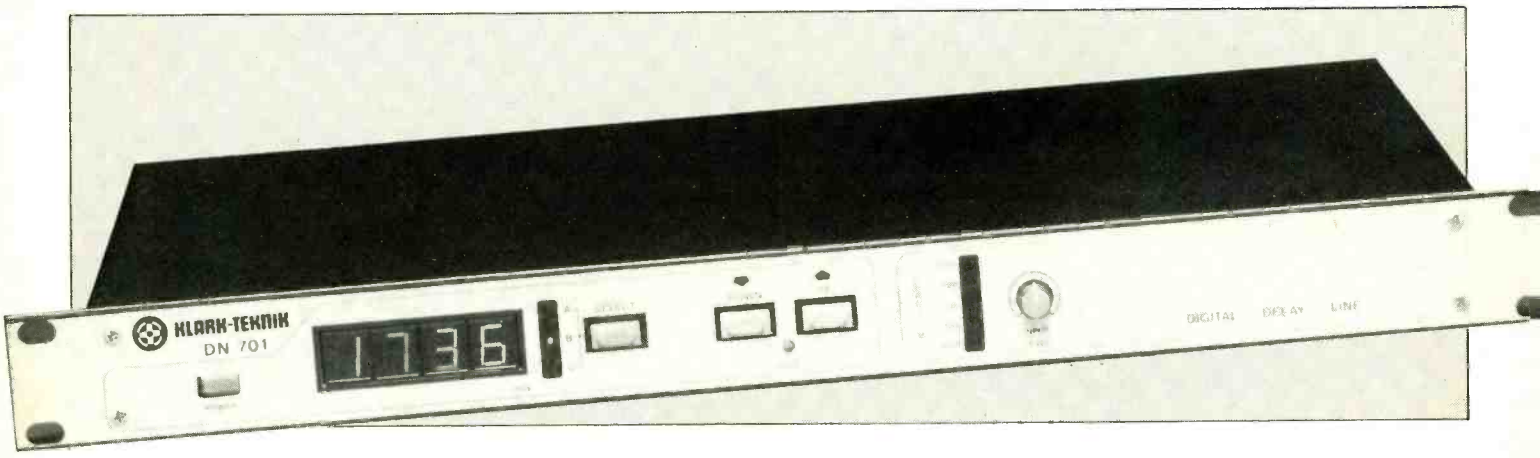
Klark Teknik DN 701 digital delay line

Hugh Ford



THE Klark-Teknik models *700* and *701* digital delay line only differ in their maximum delay capabilities of 434 ms and 1.73 s respectively. The units are general purpose high quality digital delays in a configuration particularly suitable for sound reinforcement in view of the single input and three outputs which can be set to different delay times. As standard, the input is electronically balanced but a balancing transformer is available as an option at the time of ordering. Also as standard, the outputs are unbalanced but transformer outputs can be retrofitted.

A successive approximation type A/D converter is used in the 15-bit digital system, the input and output filters being of the 7-pole elliptical type.

The unit is 19 inch rack mounting and 1 U in height. It has a silver alloy front panel with black legends and the back and sides are formed from a plated steel 'U' section to which are attached the top and bottom alloy covers.

The majority of the electronics are on a single, very good quality, printed circuit board covering the base with the front panel controls with some electronics mounted on a separate board behind the front panel. All components are clearly identified with some integrated circuits being socketed to ease servicing. The general standard of construction, component layout and overall quality of all components is excellent.

At the front panel the power on/off push-button is on the left followed by a 4-digit gas discharge display of the delay time in milliseconds. Successive pressing of a 'select' button switches the delay display between the three outputs with three LEDs showing which delay is currently selected.

Delay time setting is by means of two push-buttons which increment the currently displayed channel up or down in 26.5 μs steps with the display having 1 ms resolution. The 26.5 μs steps are speeded up if a button is held depressed for more than 7 s.

Between the increment buttons a push switch operated by poking an instrument through a hole in the front panel inhibits the increment controls. When this has been done the delay display suecessively switches through the three delay times with A, B and C LEDs showing which is the current output being displayed.

Proceeding to the right, five LEDs in vertical array form the headroom indicator with calibrations at 0, −3, −5, −11 and −16 dB. To the right of this is the input sensitivity potentiometer.

To the rear of the unit locking *XLR* connectors form the audio input and three outputs with mains power being applied via an IEC connector with a nearby clearly identified power fuse and voltage selector, the transformer secondaries being protected by internal fuses.

The only remaining rear panel feature is a slide switch for isolating the electronics from the chassis ground to eliminate hum loops.

MANUFACTURER'S SPECIFICATION
Delay range: 1,736 ms.
Minimal increment: 26.5 μs.
Readout (ms): 4-digit.
Frequency response: 20 Hz to 15 kHz, +0.5 −1 dB.
Dynamic range: >85 dB (20 Hz to 20 kHz unweighted).
Distortion: <0.01% total harmonic distortion (1 kHz at +10 dBm).
Input: electronically balanced.
Impedance: >47 kΩ.
Level: 0 dBu to +18 dBu.
Outputs: unbalanced (3 outputs).
Impedance: <70 Ω (minimum load impedance 600 Ω).
Level: 0 dBu to +18 dBu (internally adjustable).
Power requirements: 110/120/220/240 V, 50/60 Hz.
Consumption: <20 VA.
Weight: 2.8 kg nett 3.5 kg shipping.
Dimensions (whd): 482 × 44 × 262 mm (19 × 1¾ × 10½ in).
Terminations: audio 3-pin *XLR* power 3-pin CEE.
Options: transformer balanced input/outputs.
Manufacturer: Klark-Teknik Research Ltd, Coppice Trading Estate, Kidderminster, Worcs DY11 7HS, UK.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, NY 11735.

Input and outputs
The balanced input had an impedance of 93.5 kΩ when operating balanced, or 46.5 kΩ when operating unbalanced with a signal handling capability of >+22 dBm and a maximum sensitivity of 500 mV for 0 dB level indication.

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TABLE 1

	– 1 dB	– 3 dB
Without transformers	15.9 kz	16.6 kHz
With transformers	15.6 kHz	16.2 kHz

TABLE 2 Maximum output to noise

Measurement method	Min gain	Max gain	With signal
22 Hz to 22 kHz RMS	– 95.2 dB	– 92.5 dB	– 87.6 dB
A-weighted RMS	– 98.8 dB	– 94.3 dB	– 89.0 dB
CCIR-weighted RMS	– 91.8 dB	– 86.0 dB	– 80.8 dB
CCIR-weighted quasi-peak	– 86.5 dB	– 80.0 dB	– 76.0 dB
CCIR-weighted ARM	– 99.0 dB	– 94.0 dB	– 87.8 dB

Common mode rejection was satisfactory as shown in **Fig 1**.

With or without the optional output transformers the maximum available output level was +13 dB.7V into a high impedance with the source impedance being 69 Ω without transformers or 17 Ω with transformers, the internal output level control being of the full range type.

As supplied an output level of +13.0 dB.7V corresponded to full scale in the digital department which was 2.5 dB above the onset of illumination of 0 dB level indication with the lower indications being within 1 dB of their nominal levels. The rise time of the level indicator was on the slow side at 350 ms with a fall time of 400 ms with the indication corresponding to peak levels. I would certainly have liked to find far faster indication of peaks.

Frequency response and noise

No pre-emphasis being used, the frequency response is independent of level with the response with and without output transformers being shown in **Fig 2** for 3 dB below maximum level.

Table 1 shows how the high frequency roll off varied slightly in the presence of output transformers.

Noise in the output depended upon the input gain setting under no signal conditions with quantising noise appearing with low level signals. This was measured in the presence of a 10 Hz signal outside the noise measurement bandwidth to ascertain the degree of noise increase in the presence of signals.

There was mild breakthrough of tones from the delay time display depending upon which output it was monitoring. In the control inhibit mode this led to cycling of the tone frequency. Whilst these signals made no measurable difference in output noise they were audible at very low levels (see **Table 2**).

Bearing in mind that this is a 15-bit non-companded system the noise performance is good as might be predicted.

Distortion

Second and third harmonic distortion was measured at 3 dB below clipping with and without the optional transformer outputs and found to be consistent at 0.01% in either case rising to 0.03% 40 dB lower in level.

Similarly intermodulation distortion to the CCIF twin tone method was good as shown in **Fig 3** for peaks 3 dB below clipping rising to 0.03% 40 dB lower in level.

Squarewave reproduction was as shown in **Fig 4** for a 1 kHz squarewave, the ringing arising partly from the input anti-aliasing filter.

Delay time

None of the aforementioned measurement were affected in any way by the amount of delay in the system. The actual delay time steps in 26.5 μs increments with the display resolution being 1 ms, just prodding the setting buttons achieves the minor increments. The actual measured delay time for the maximum setting of 1,736 ms was 1,737.30 ms controlled by the internal crystal. This has a nominal frequency of 4 MHz with the actual frequency being 3.9991 MHz.

Summary

This Klark-Teknik delay unit is very well made using first quality components with a sound standard of construction. Overall the performance is very good resulting from the lack of any form of companding and pre-emphasis which in other units lead to noise breathing and other effects. ■

FIG 1.
KLARK TEKNIK DN 701
COMMON MODE REJECTION

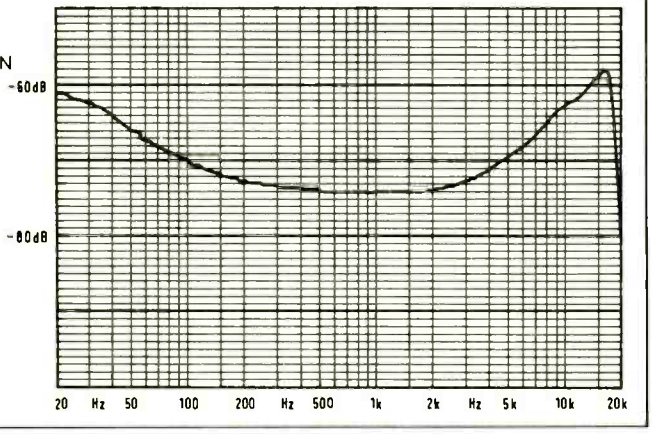


FIG 2.
KLARK TEKNIK DN 701
FREQUENCY RESPONSE WITH & WITHOUT OUTPUT TRANSFORMERS

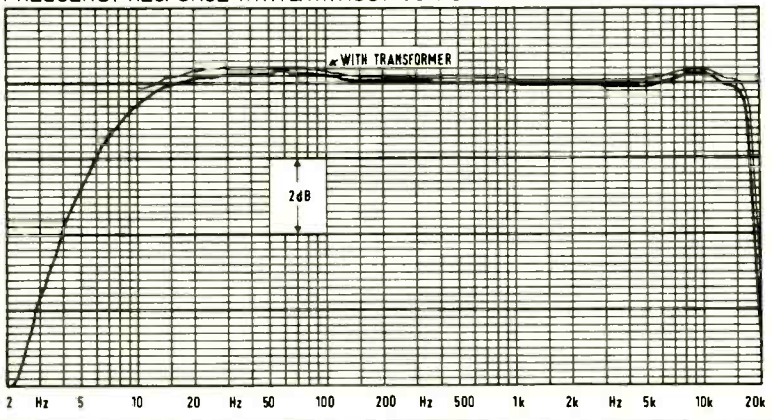
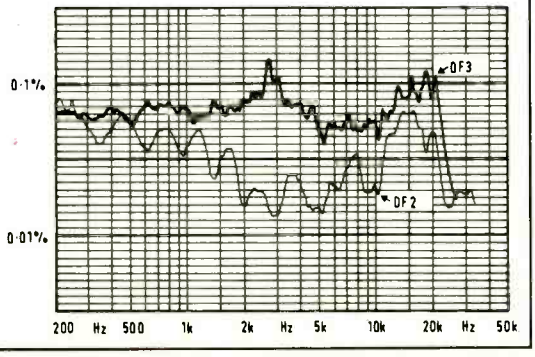


FIG 3.
KLARK TEKNIK DN 701
IM DISTORTION



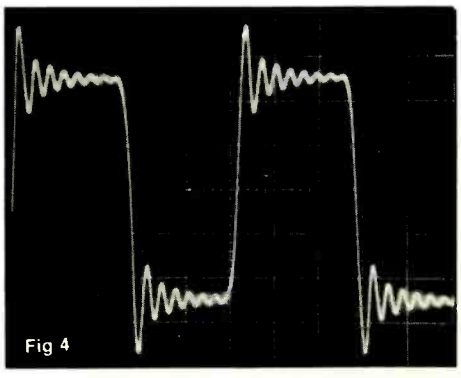
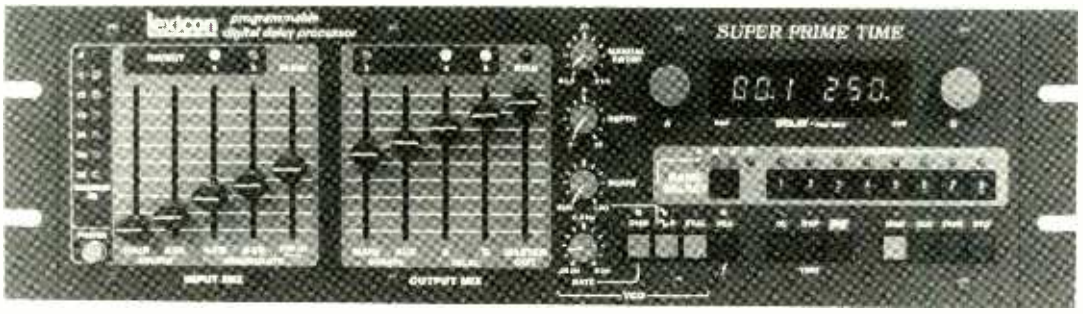


Fig 4



Lexicon model 97 'Super Prime Time'

Hugh Ford

MANUFACTURER'S SPECIFICATION

General performance, controls and indicators
Frequency response: 20 Hz to 15 kHz, ±1 dB at × 1 delay sweep; 20 Hz to 20 kHz, ±3 dB at × 1 delay sweep.
Total harmonic distortion plus noise: 0.03% typical, 0.05% maximum below 5 kHz at × 1 clock (XTAL clock); 0.37% maximum, 20 Hz to 15 kHz at × 0.5 clock.
Dynamic range: better than 85 dB, 20 Hz to 20 kHz noise bandwidth.
Delay range: 200 μs to 320 ms at full 20 kHz bandwidth (standard), option 1 extends maximum to 640 ms, option 2 extends maximum to 1.28 seconds.
Delay modulation: adjustable from none to a 3:1 sweep of delay time. LFO modulation rate is adjustable from 0.05 Hz (20 seconds for full sweep) to 500 Hz in two ranges.
LFO shape: continuous adjustment (blend) is available between sinewave and envelope functions, or between squarewave and envelope functions.
Dynamic recirculation control: VCA control of feedback makes possible long delay time due to large amounts of recirculation without undesirable layering or overlap.
Factory presets: 8 effects programs are permanently stored in memory (ROM)—basic flange, resonant flange, doubling, trebling, chorus, slap echo, moderate echo, echo with recirculation.
User storage: 32 effects programs can be stored in non-volatile solid state memory (ni-cad battery protected RAM), unlimited storage via standard audio tape record/playback of memory contents.
Headroom indicator: 7 level LED display shows input mix level (combined Main, Aux and Recirculation) relative to maximum '0 dB' level in 6 steps.
Input mixer: slide controls for Main Input, Auxiliary Input, 'A' Delay Feedback, 'B' Delay Feedback, as well as a 6 dB/octave low pass filter for feedback, adjustable from 600 Hz to 20 kHz cutoff.
Output mixer: slide controls for Main Input, Auxiliary Input, 'A' Delay, 'B' Delay and Master Output Level.
Overload indicator: red LED indicates Master Output overload (which may be caused by any of the Output Mix sliders).
VCO and LFO controls: rotary controls for Delay Sweep, LFO Depth, Waveform Shape, and LFO Rate; pushbuttons for Rate × 100, Sine/Square Modulation, XTAL/VCO.
Delay selection: rotary controls for Delay Taps 'A' and 'B', with large amber 7-segment digital LED time displays, Display factors both Delay Select and Delay Sweep controls for accurate 3-digit resolution of overall delay time.
Register storage: pushbutton control of store and recall.

Interface information
Input connectors: Main and Auxiliary Inputs: *XLR-3* female connectors in parallel with standard tip-ring-sleeve ¼ in phone jacks.
Input impedance: >50 kΩ in parallel with 300 pF for Main Input, >20 kΩ in parallel with 150 pF for Aux Input. Both inputs may be used balanced or unbalanced.
Input levels: 0 dBV to + 19 dBV (– 20 dBV to 0 dBV with Gain switch on rear panel) for Main Input, 0 dBV to + 19 dBV for Auxiliary Input.
Output connectors: *XLR-3* male in parallel with tip-ring-sleeve ¼ in phone jack for Master Output, ¼ in phone jacks for Input Mix, Delay 'A' and Delay 'B' outputs.
Output impedance: 200 Ω balanced or unbalanced for Master Output, 600 Ω unbalanced for Input Mix, Delay 'A' and Delay 'B' outputs.
Output levels: + 22 dBV for Master Output when driving balanced loads of 600 Ω or greater. + 16 dBV for Master Output when driving unbalanced loads of 600 Ω or greater. + 16 dBV for Input Mix, Delay 'A' and Delay 'B' outputs when driving 2 kΩ loads or greater.
Input/output coupling: direct balanced electronic (Main).
Remote connectors: ¼ in phone jacks on rear panel for Delay Sweep, Modulation Input, Modulation (LFO) Output, Rate, Bypass, Infinite Repeat, Register Step, Tape (store) input and Tape (store) output.
Power: 100/120/220/240 V (switchable inside the chassis), 50/60 Hz, 50 W maximum, Standard IEC power connector and cord, Mains fused (1¼ in domestic, 20 mm export), secondaries fused with European 20 mm fuses; RFI power line filter is standard (and all jack RFI filtered).
Backup power: Ni-Cad 3.6 V automatic recharging (cells intended for continuous trickle charge).
Size: standard 19 in rack mount, 5¼ in high by 13½ in deep (483 mm × 133 mm × 343 mm).
Weight: 17 lb (7.7 kg) net. 20 lb (9.1 kg) shipping.
Manufacturer: Lexicon Incorporated, 60 Turner Street, Waltham, MA 02154, USA.
UK: FWO Bauch Limited, 49 Theobald Street, Boreham Wood, Herts WD6 4RZ.

The Lexicon 'Super Prime Time' is a very complex microprocessor controlled effects unit containing eight dedicated programs. These effects cover basic flanging, resonant flanging, doubling, trebling, chorus effect, slap echo, moderate echo and echo with feedback. In addition to these effects which are stored in read only memory, the unit has four banks of eight registers which can store the users control settings for selected effects. These 32 registers have a battery backup such that their contents are not lost when power is removed.

A further feature is that the contents of all or some of the user's registers can be stored on tape and subsequently recalled to the unit. In addition to this basic feature for storing and recalling data from tape, the stored data may be assigned a file number between 1 and 88 with individual files being recalled by number. Thus a library of 32 × 88 = 2,816 different lots of control settings can be stored on a single tape which can be a cassette or other media.

The internal audio routing within the unit is rather complicated with **Fig 1** showing the main audio chain. Two balanced audio inputs are provided, the main input and the auxiliary input, the former having a switchable 20 dB extra gain for operating with domestic signal levels. Both inputs are fed to computer controlled attenuators before being summed. From here the delayed outputs are produced by first passing the signal through an anti-aliasing filter before digitising the signal, with the capability of adding an analogue recirculation signal. The digitised audio is then stored for a variable time before being converted to analogue to provide two separate delayed audio signals which can have different delay times.

The original summed analogue signal from the two inputs passes to two other routes. In the bypass mode it is switched to the master output

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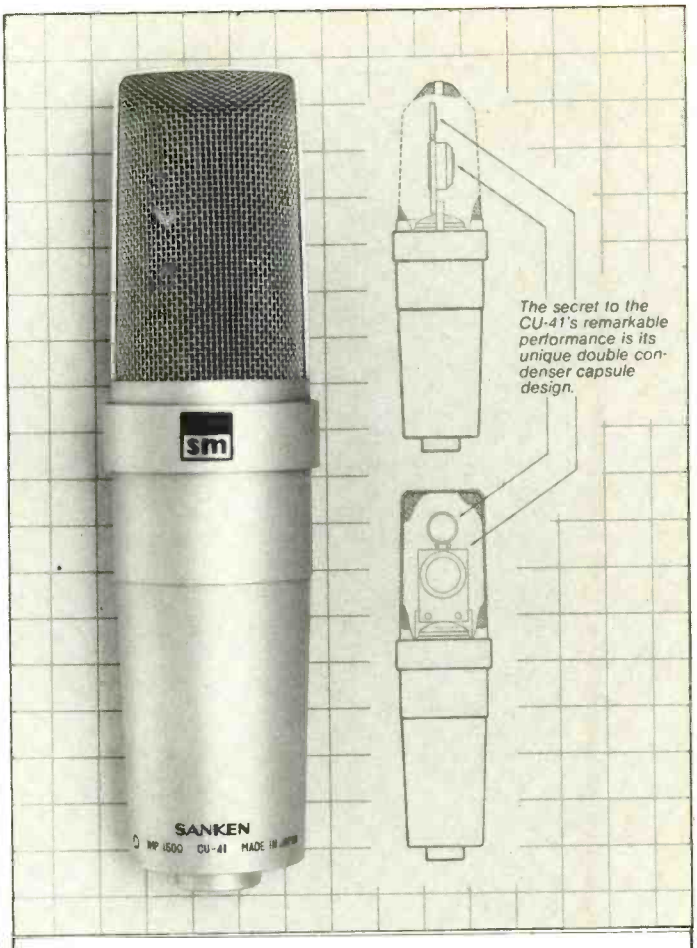

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via a computer controlled attenuator, the master output in the normal mode being derived from a mix signal. The second feed is summed with the delayed signals to form the unbalanced input mix output. Reference to **Fig 2** shows how the delayed signal and the normal master output signal are derived. The main, auxiliary and two delayed signals are fed to computer controlled attenuators with all except the auxiliary input signal then being fed to computer controlled phase inverters.

From these signals the master output is the sum of the auxiliary, main and delayed signals whilst the recirculation signal and the delayed signal to the input mix output are the sum of the two delayed signals after they have passed through a computer controlled low pass filter variable from 700 Hz to 20 kHz.

This is but the basic description of the complex audio routing with all the front panel controls feeding the computer section rather than being conventional analogue controls.

Within the time delay section the 'Super Prime Time' is not just a straightforward delay unit but has various modulation facilities and external control abilities. **Fig 3** shows some of these features which starting at the top left include remote control of the bypass switching. Next there is an infinite repeat function which simply recirculates the stored audio (the stored time depending upon the time settings) until infinity repeat is released. The next function 'register step' is particularly useful as it allows sequential switching of the stored control settings from an external device, be it a switch or a control pulse to give a cue. Thus the unit can step sequentially and automatically through 32 user control settings.

The remainder of **Fig 3** is devoted to modulation effects with the modulation in and modulation out connections allowing two or more units to be ganged for stereo or other purposes.

Timing within the unit relies on the master clock in the form of the voltage controlled oscillator with the switched option of a crystal oscillator. Varying the master clock frequency varies the delay time in a manner different from changing the delay time control—the former altering the overall time stored with the latter changing the delay line 'taps'. Modulation of the delay time can be controlled in a number of manners with computer control of the switching. Firstly manual control of the master clock is possible internally or externally from a 0 to 10 V control signal with the depth control affecting the mix of the manual delay control and the other sources with the maximum range being ×0.5 to ×1.5 with reference to the nominal delay setting.

The second and third sources of modulation are via the modulation in/out jacks. This source is the sum of the output from an envelope follower at the input to the A/D converter and a manually controlled oscillator. The latter has a frequency range from 0.05 Hz to 500 Hz in two ranges with sine- or squarewave modulation.

The embodiment of the unit is a 19 in rack mounting case three rack units in height with the totally alloy case being fitted with removable rack mounting ears. All controls are on the front panel with all connections at the rear.

Conveniently the front panel layout may be divided into four sections, input mix controls, output mix controls, voltage controlled oscillator controls and the digital section. To the far left of

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FIG 1
LEXICON MODEL 97
MAIN AUDIO CHAIN

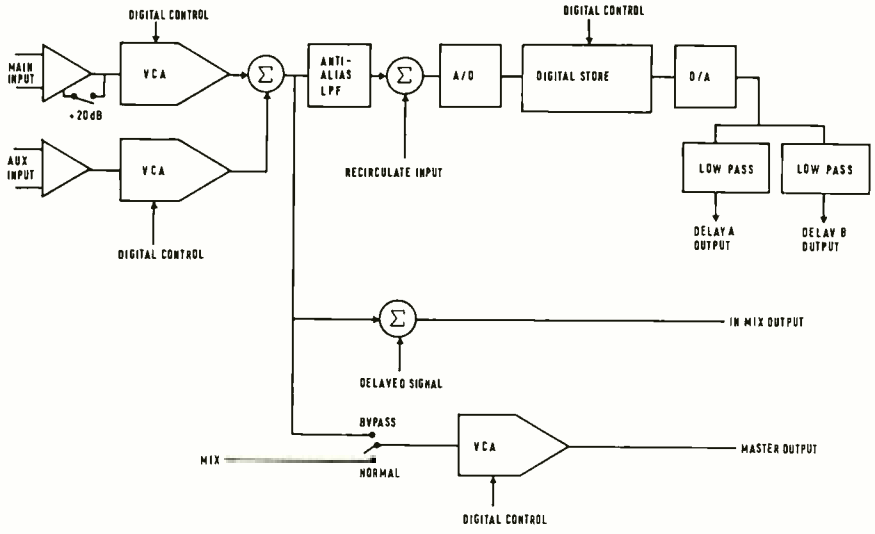


FIG 2.
LEXICON MODEL 97
DELAY AND MASTER
OUTPUT SIGNAL DERIVATION

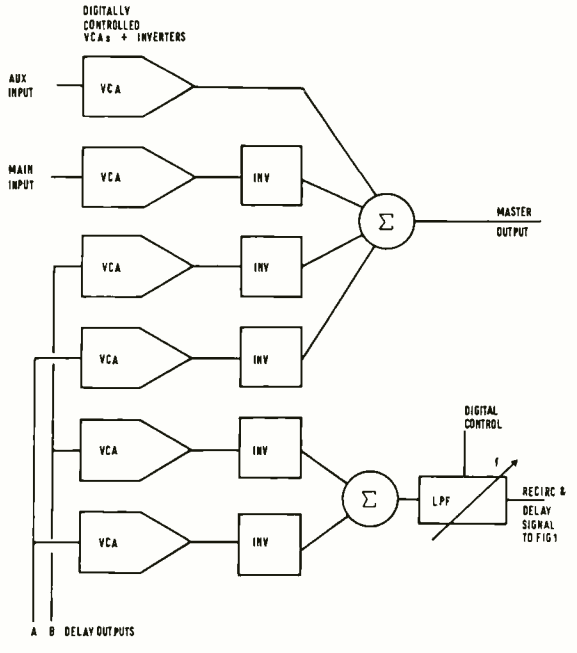
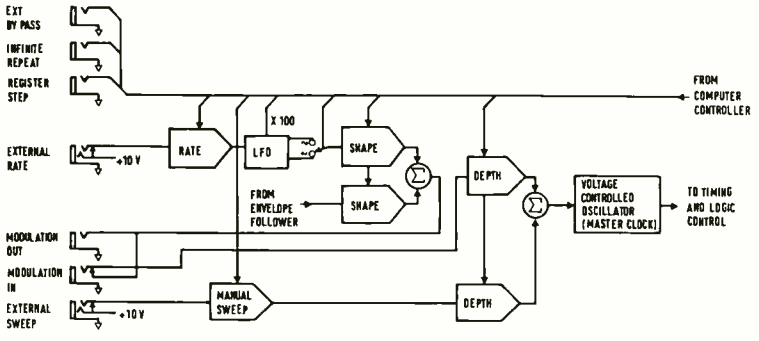
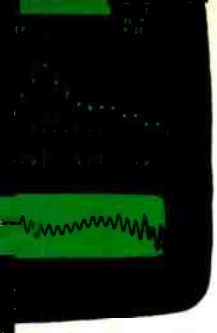


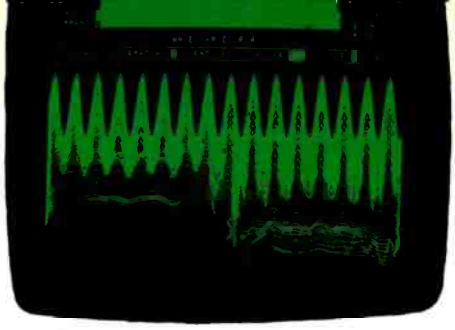
FIG 3.
LEXICON MODEL 97
VCO SECTION BLOCK DIAGRAM





Waveform Generation

Keyboard Sequencer



Waveform Display

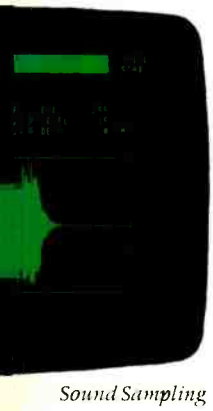
Disk Control

Control Parameters

Real Time Composer

Keyboard Control

Waveform Drawing



Sound Sampling

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Analog Interface

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the panel a vertical array of one red, two yellow and four green LEDs form the headroom indicator with 6 dB increments. The pushbutton power switch below this remotely operates the switch at the rear of the unit.

Within the input mix section are five vertical slider controls with the main input level control being the only one that does not have its setting digitally stored. The second control is the auxiliary input level with two further controls affecting the two delayed recirculation returns. Above each of these a yellow LED is illuminated when the returns are phase inverted. Finally the fifth slider controls the frequency of the low pass filter in the combined return input.

The following section containing a further five vertical sliders control the output mix with a master output level control following level controls for the two delay outputs are direct feeds from the auxiliary and master inputs. The master and the two delay signals may be phase inverted in which case yellow LEDs are illuminated above the controls with the master level control having a red overload LED.

Within the voltage controlled oscillator section are four rotary potentiometers and four momentary pushbutton switches. The potentiometers form the manual sweep, depth, shape and frequency controls shown in **Fig 3** with one momentary switch switching the frequency range to x1 or x100. A second momentary switch changes the oscillator output from a sinewave to a squarewave with a third switch enabling the crystal oscillator rather than the variable oscillator. The fourth switch, all of which have associated LED indicators, switches a voltage controlled attenuator into the delay line return feed enabling a dynamic recirculation mode which effectively attenuates long decays in the presence of an input signal but allows long decays when the input signal ceases.

Within the digital section at the top two three-digit fluorescent displays show the two A and B output delay times in seconds or milliseconds with potentiometers controlling the delay times. The horizontal array of switches below this selects registers from the 40 possible registers which are divided into five banks each of eight registers. Eight of the momentary buttons are identified from one to eight for selecting registers within a bank. A further switch with four associated LED's labelled A, B, C and D selects by successive presses, the four user banks A, B, C or D with the pre-programmed bank being selected by holding the bank button whilst depressing a register button.

In the bottom right hand corner of the unit are seven further momentary pushbuttons in groups of three and four. In the group of three, one starts the infinity hold function, a second puts the unit in the bypass mode and the third inserts the phase inverters in conjunction with the register buttons which are used to select the desired inverter. In the group of four buttons one is a 'manual' button which places all controls in a manual mode and disregards any automated settings, this button also being used to escape from certain conditions. Of the remaining three buttons a 'clear' button is used in conjunction with the register buttons to clear the contents of registers with the remaining buttons being associated with recording the contents of registers on to tape, reloading and verifying tape. The tape inputs and outputs can also be used for cueing register changes. Having loaded the desired effects the unit can output cue tones to tape when stored conditions are changed. Thus if the cue tone output is fed to a multitrack master whilst programme is being replayed, manual changing of registers will record a cue tone. The master tape may then be replayed into the tape input resulting in automatic cueing of register changes.

Various other functions are available as many switches have more than one function including internal test programming but these functions are too complex to describe in this review.

To the rear of the unit *XLR* connectors in parallel with tip, ring and sleeve jack sockets form the main and auxiliary inputs and the master output balanced connections with 13 ¼ in jack sockets providing the other connections, all unbalanced. In addition to these there is the IEC mains power connector with a nearby correctly identified power fuse. Four further fuses within the unit protect the transformer secondaries.

Within the unit there are three sections interconnected by ribbon cables, the front panel control section, the audio section and the computer section, all three being very good quality printed circuit boards with a very neat layout and clear component identifications. Furthermore all integrated circuits are socketed for ease of servicing—overall an unusually well made unit with excellent access for servicing.

Inputs and outputs

The balanced main input had an impedance of 100 kΩ in the balanced mode with a capability of handling in excess of +22 dBm with the maximum gain to the master output in the bypass mode being 17.7 dB or 37.8 dB in the normal and high sensitivity settings respectively. The balanced auxiliary input had a similar performance but with an input impedance of 39.5 kΩ in the balanced mode. The satisfactory common mode rejection of the two inputs is shown in **Fig 4.**

At the balanced master output the unit could deliver +21.5 dB.7 V into a high impedance or +20 dBm loaded into 600 Ω for the onset of clipping from a source impedance of 98 Ω. Similarly the in-mix output could deliver +21.5 dB.7 V from an unbalanced source of 609 Ω with the unbalanced delayed outputs giving +14 dB.7 V from a source impedance of 503 Ω at 1 kHz. At the unbalanced tape connections the tape output is ±1.5 V peak-to-peak at a maximum frequency of 5 kHz from a 600 Ω source with the input having an adequately high impedance in the order of 40 kΩ.

Frequency response

In the bypass mode the response from the main and auxiliary inputs to the master output is

83 ▶

FIG. 4
LEXICON MODEL 97
COMMON MODE
REJECTION RATIO

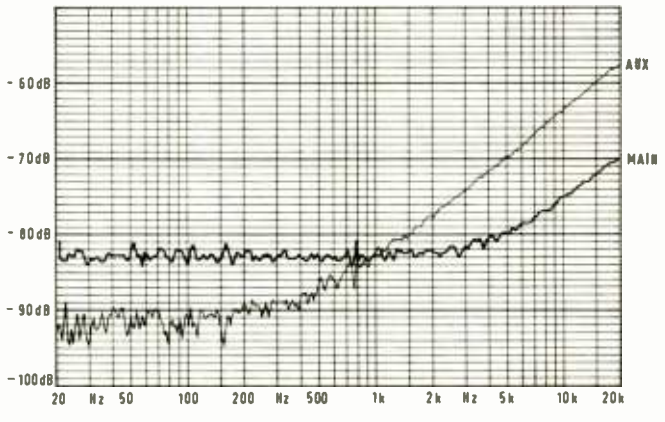
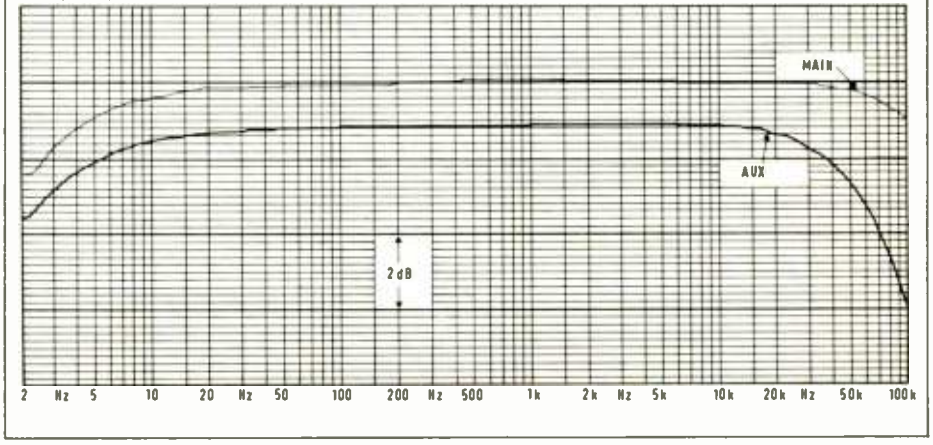
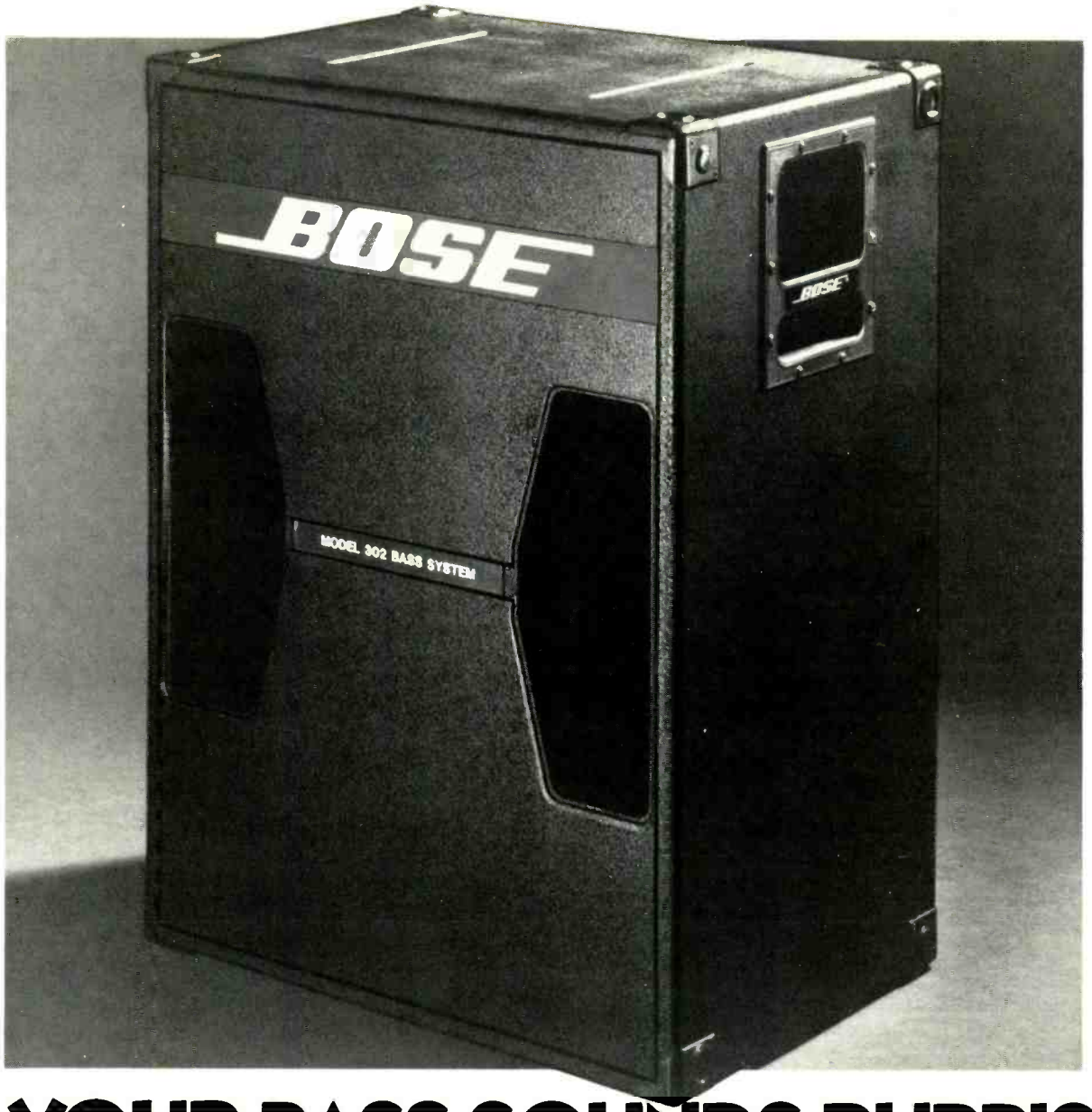


FIG. 5
LEXICON MODEL 97
FREQUENCY RESPONSE 'DIRECT'





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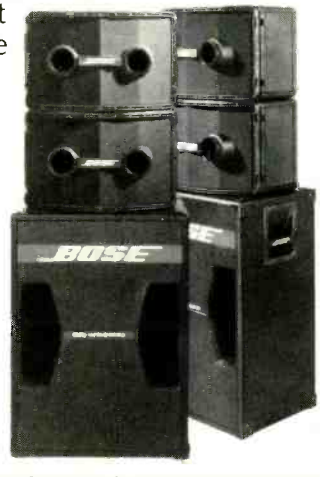

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shown in **Fig 5** to be flat within the audio band, this applying at any gain settings.

The response of the delayed signals was found to be independent of the delay time settings or the effective clock frequency being flat as shown in **Fig 6** at low levels where the filters cut off at 20.22 kHz with a mild boost around 10 kHz. At higher levels it appeared that pre-emphasis is used at the delay section input with an acceptable time constant in the order of 25 μs—thus undue high frequency levels require a reduction in gain, the headroom indicator being post-emphasis.

Distortion

In the bypass mode individual harmonic distortion for the 2nd to 5th harmonics was less than 0.01% at any level below clipping from 20 Hz to 20 kHz. Similarly intermodulation distortion to the twin tone CCIF method was below 0.01% within the audio band rising slightly at higher frequencies and high levels. Harmonic distortion in the delayed channel depended very much upon level, being low at high levels but decidedly high at low levels. These effects are shown in **Figs 7, 8** and **9** for the maximum operating level and 20 dB increments down—note the different vertical scale in **Fig 9.**

A similar situation exists with CCIF twin tone intermodulation distortion which was very low in the bypass mode but in the delayed mode increased with falling operating level. **Fig 10** shows the second and third order products at the maximum operating level with **Fig 11** showing

84 ▶

FIG.6
LEXICON MODEL 97
FREQUENCY RESPONSE DELAY SECTION

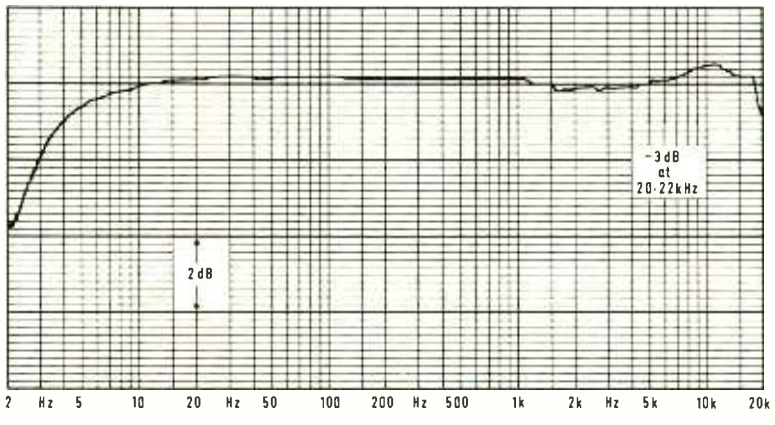


FIG.7
LEXICON MODEL 97
HARMONIC DISTORTION
IN DELAY, MAX IN/OUT

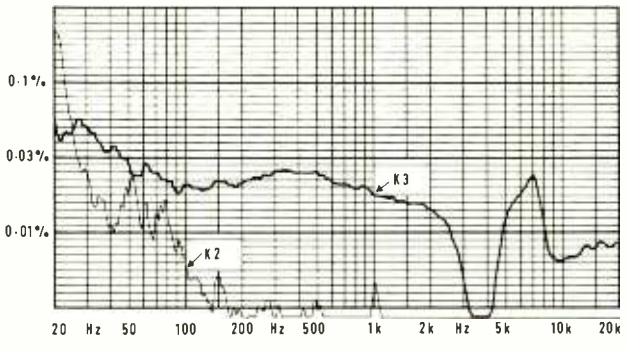
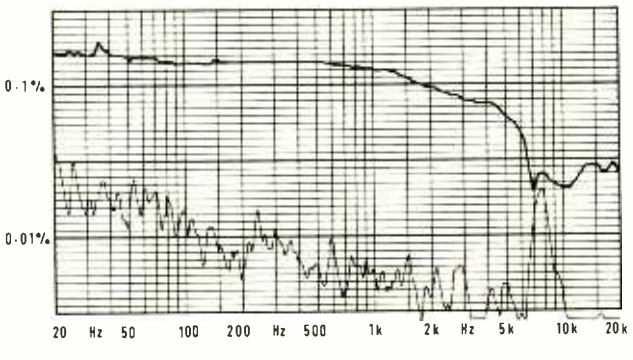


FIG. 8
LEXICON MODEL 97
HARMONIC DISTORTION
IN DELAY -20dB



the situation at 40 dB lower level.

The response to squarewaves was excellent in the direct mode with no overshoot or ringing with the delayed mode producing ringing of a 1 kHz squarewave as shown in **Fig 12.** In either mode the response to tone bursts was without faults with all forms of distortion remaining constant irrespective of the time delay settings.

Noise

No signs of noise breathing or other noise modulation effects were found with the exception of movement of certain controls producing some low level noise in the outputs in the delay modes. **Table 1** relates noise in the master output to the maximum level in the delayed modes. In this mode the excellent performance is of course degraded with the input gain differences being masked and noise depending upon the selection of the crystal clock or the variable clock as shown in **Table 2.** In all conditions clock frequency components and other discrete tones were at less than 300 μV in the outputs with no other extraneous products being present.

Other matters

The LED level metering was placed after pre-emphasis with a very fast peak reading capability giving true peak readings in less than 5 μs with a fall time of 750 ms giving excellent readability.

Delay time indication was found to be accurate to within the resolution of the display with the maximum delay in the review sample of 1.28 s actually being 1.28024 s in the crystal mode. This is the maximum delay version with options being 640 ms or 320 ms. In the manual mode the maximum range of x0.5 to x1.5 was found to be correct with the internal low frequency oscillator having the maximum range.

The levels to and from tape were such that a cassette recorder or professional recorder could be used either for storing data or for storing cues, the stepping through registers working perfectly well with a cassette recorder.

Summary

The 'Super Prime Time' offers many advantages over other delay systems as it does not suffer from the difficulties in using CCD delay systems with their common noise modulation defects. As with any digital system low level signals can suffer degradation, thus it is important to use the maximum drive capabilities of the system—provided this is done the subjective performance is very good.

This unit is extremely well made and given proper servicing data, it should be easy to service with its good clear component layouts and socketed integrated circuits.

Whilst the use of multiple function controls is common practice in calculators and many other digital devices, the operation of this unit requires considerable practice if all the available facilities are to be used. Normal functions are clear from the front panel but there is no clear indication of the many special functions in view of the multiple uses of some controls.

In terms of performance not only do the eight inbuilt programs give a good variety of useful effects but also the unit is very versatile and capable of making and storing infinite number of different effects.

In summary the 'Super Prime Time' offers a superior performance compared with most effects units and an enormous variety of effects. ■

FIG. 9
LEXICON MODEL 97
HARMONIC DISTORTION
IN DELAY -40dB

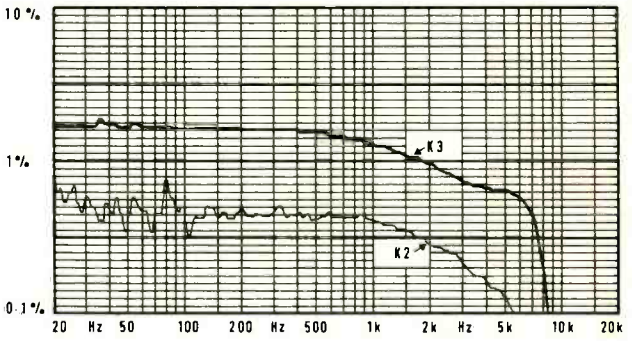


FIG. 10
LEXICON MODEL 97
IM DISTORTION, MAX LEVEL

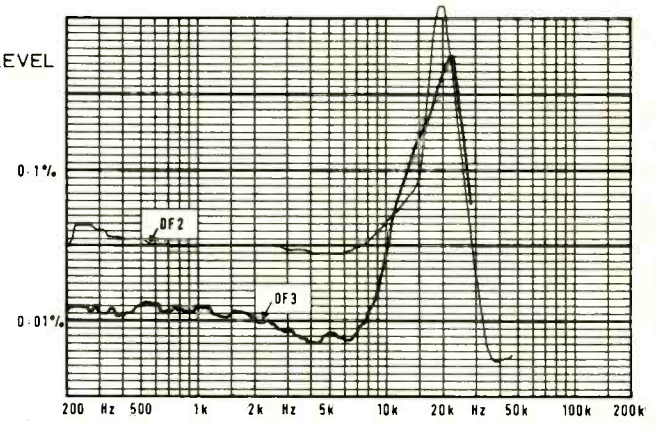


FIG. 11
LEXICON MODEL 97
IM DISTORTION -40dB

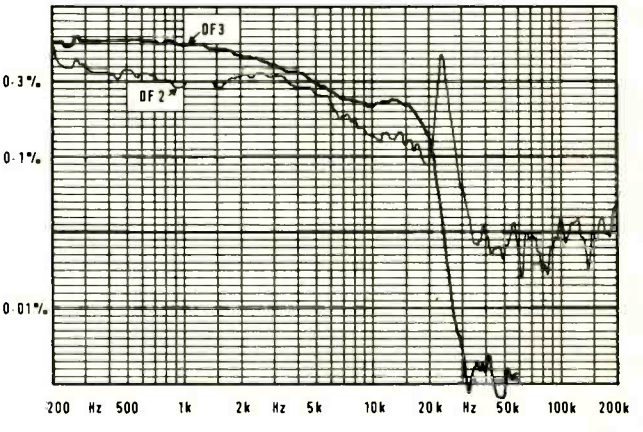


TABLE 1—Noise versus max output bypass

Measurement method	Max	Min input gain
22 Hz to 22 kHz RMS	97.0 dB	103.0 dB
A-weighted RMS	103.5 dB	107.0 dB
CCIR-weighted RMS	96.5 dB	99.0 dB
CCIR-weighted quasi-peak	91.0 dB	94.5 dB
CCIR-weighted ARM (2 kHz)	104.0 dB	106.0 dB

Table 2 — Noise versus max output—delayed

Measurement method	Crystal clock	Variable clock
22 Hz to 22 kHz RMS	84.0 dB	86.0 dB
A-weighted RMS	89.5 dB	87.5 dB
CCIR-weighted RMS	82.5 dB	80.0 dB
CCIR-weighted quasi-peak	77.5 dB	74.0 dB
CCIR-weighted ARM (2 kHz)	89.5 dB	87.0 dB

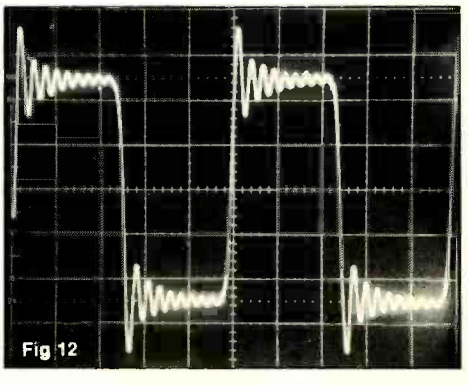


Fig 12

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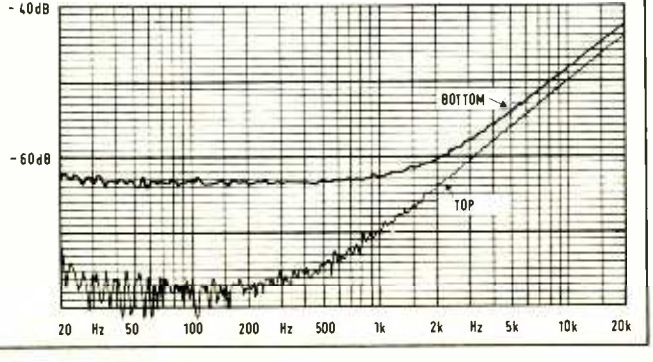
TC Electronic TC 2240 parametric equaliser

Hugh Ford

MANUFACTURER'S SPECIFICATION

Dynamic range (all settings flat, except output level control at 4·5): 116 dB (120 dB bypass).
Signal to noise ratio (at 10 dB headroom): 106 dB (110 dB bypass).
Distortion (THD): 0.015%.
Frequency response at 0 dBm output level (all settings flat except output at maximum): 10 Hz to 100 kHz, + 0/ – 1 dB; 6 Hz to 180 kHz, + 0/ – 3 dB.
Channel separation (at 1 kHz): 100 dB.
Function (equalisation) (measured at + 15 dB. 3 dB peak points, notches exact reciprocal of peaks): + 20 to – 20 dB nom, + 20 to – 50 dB typ.
Bandwidth: 0.1 to 2 octaves.
Q value: 14 to 0.67.
Slope: 6 to 66 dB/octave.
Centre frequency control sweep range: 6.6 octaves.
Frequency ranges: band 1—20 to 2,000 Hz; band 2—50 to 5,000 Hz; band 3—100 to 10,000 Hz; band 4—200 to 20,000 Hz.
Parametric sections overlap: 5.6 octaves.
Overload indicators (headroom at 'glow' point): 6 dB.
Input connections, high level: *XLR* with lock.
Input impedance: 20 kΩ, balanced or unbalanced.
Nominal gain: 0 or – 6 dB.
Total available gain: 20 dB.
Common mode rejection ratio: 50 to 100 Hz, 60 dB min, 80 dB typ; at 1 kHz 40 dB min, 60 dB typ.
Maximum input level: + 22 dBm.
Recommended nominal input level: – 30 to + 10 dBm.
Input connections, low level: jack (phone-plug) unbalanced.
Input impedance: 1 MΩ//68 pF.
Nominal gain: 15 dB.
Maximum input level: + 6 dBm.
Recommended nominal input level: – 45 to – 5 dBm.
XLR output, IEC standard: balanced or unbalanced.
Output impedance (1% match): 50 Ω.
Maximum output level: RL = 10 kΩ, + 27 dBm (50 Vpp); R1 = 600 Ω, + 21 dBm (25 Vpp).
Jack (phone-plug) output: unbalanced.
Output impedance: 50 Ω.
Maximum output level, RL = 10 kΩ: + 21 dBm (25 Vpp).
Mains voltage: 200 to 240 or 100 to 120 V AC.
Power consumption: 5 W.
Dimensions (whd): 482 × 89 × 185 mm (19 × 3.5 × 7.3 in).
Weight: 3.5 kg.
Manufacturer: TC Electronic, Dalbovej 2, DK-8210 Aarhus V, Denmark.
UK: MTR Ltd, Ford House, 58 Cross Road, Bushey, Herts WD1 4DQ.

FIG 1.
TC ELECTRONIC 2240
COMMON MODE
REJECTION RATIO



THE *TC2240* equaliser is one of a series of three equalisers available from TC Electronic which have four overlapping sections of parametric equalisation. The model *2240* reviewed here is a stereo version of the *1140* the channels being identical, whilst the *1120* is a simplified version of the *2240* stereo unit having two channels with ganged controls.

All models are rack mounting units with the *2240* being 2 U in height and the others 1 U. The alloy front panels are finished in black with clear white identifications. Attached to the front panel are plated steel sides and a steel back panel carries all the connections.

Each channel occupies two printed circuit boards, one containing the input/output electronics and the other the equaliser sections. With the exception of the mains power components (including the mains transformer) and the *XLR* connectors all components are soldered directly to the printed circuit boards.

The layout of the controls and connections is such that both channels are identical, one above the other. To the left of the front panel are the input/output sections with a red 'channel on' (not bypass) LED next to the input and output level controls which have a calibrated range of ±20 dB and 0/ + 6 dB respectively. There follow 'EQ match' potentiometers with a ±16 dB nominal range for matching equaliser channels in the non-bypassed mode and a biased spring loaded bypass switch above which there are five LEDs forming a level indicator. The biased bypass switches operate in a press on, press off mode and not as simple switches.

Proceeding to the right are the four equaliser sections each with identical ±20 dB cut/boost and variable bandwidth controls covering 0.1 octave to 2 octaves. The third control in each section sets the frequency of the sections covering 20 Hz to 200 Hz, 50 Hz to 500 Hz, 100 Hz to 10 kHz and 200 Hz to 20 kHz. To the rear all functions are clearly identified with the power input being via an IEC connector with a nearby 20 mm fuseholder.

Normal audio inputs are via balanced *XLR* sockets with ¼ in jack connectors providing a high sensitivity (+ 15 dB) unbalanced input. Similarly the audio outputs are available at balanced *XLR* plugs and unbalanced ¼ in jacks. Further jacks allow remote operation of the bypass function by means of a switch.

No servicing information was provided nor were there component identifications on the boards which had all components soldered into place. Access for servicing was however reasonable.

Inputs and outputs

The balanced inputs at the *XLR* connectors were found to have an impedance of 20 kΩ irrespective of gain settings with a maximum signal handling capability in excess of + 22 dBm. At the unbalanced jack connectors the input impedance was 550 kΩ in parallel with 530 pF with the maximum input level being + 7 dBm.

Gain to the balanced output with the gain controls centred was – 6 dB for the balanced input or + 9.5 dB for the unbalanced input and the common mode rejection for the two channels is shown in **Fig 1.**

Turning to the gain controls, the output level

88 ▶

control was of the full range type with the input gain control having a ±20 dB range about the centre position. Similarly the equalisation match control had a ±16 dB range about the bypass mode.

At the balanced output the unit could deliver +28.7 dB.7 V into a high impedance or +26.6 dBm into 600 Ω with the unbalanced output having an identical performance both with a source impedance of 50 Ω.

Remote switching of equalisation in/out requires a momentary contact with the source being 13.9 V DC from a 49 kΩ impedance.

Frequency response

The frequency response from the balanced input to the balanced output is shown in **Fig 2** for the bypass condition and for the equalisation in condition with all controls in their flat position. The limits of +0/−0.2 dB in the bypass mode are good from 20 Hz to 20 kHz with the limits of

90 ▶

FIG 2.
TC ELECTRONIC 2240
FREQUENCY RESPONSE

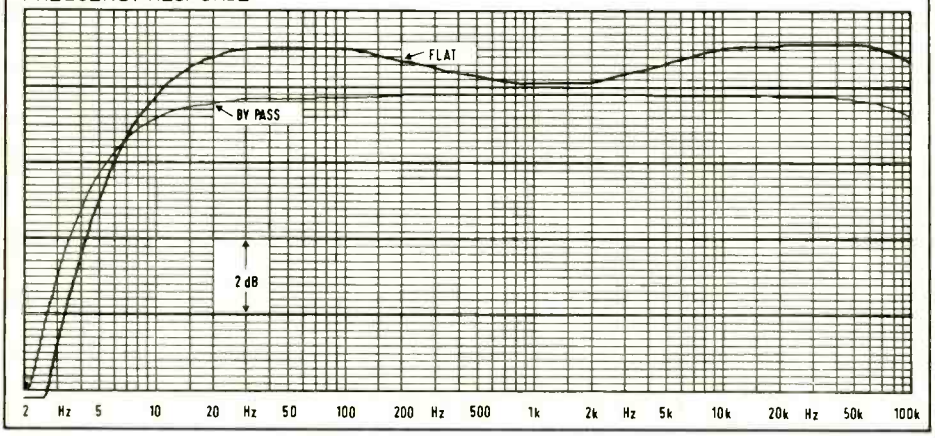


FIG 3.
TC ELECTRONIC 2240
FREQUENCY RANGE, FULL CUT/BOOST 1/2 OCTAVE BANDWIDTH CENTRE 1&3 EQUALISERS

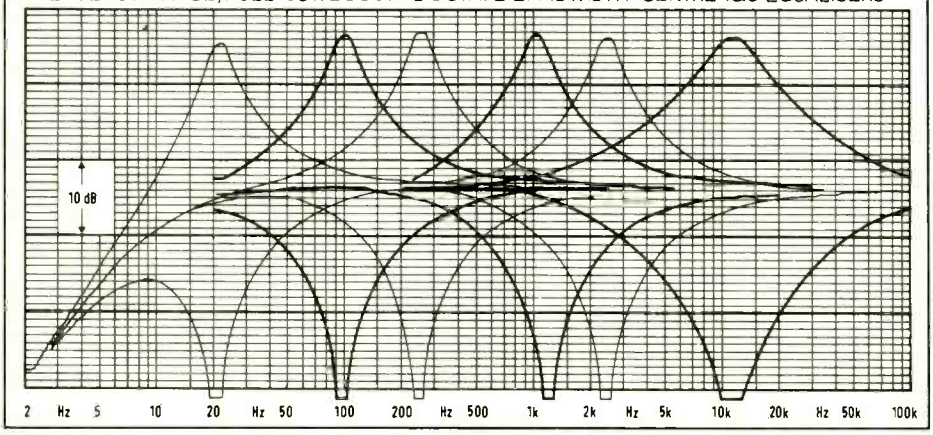
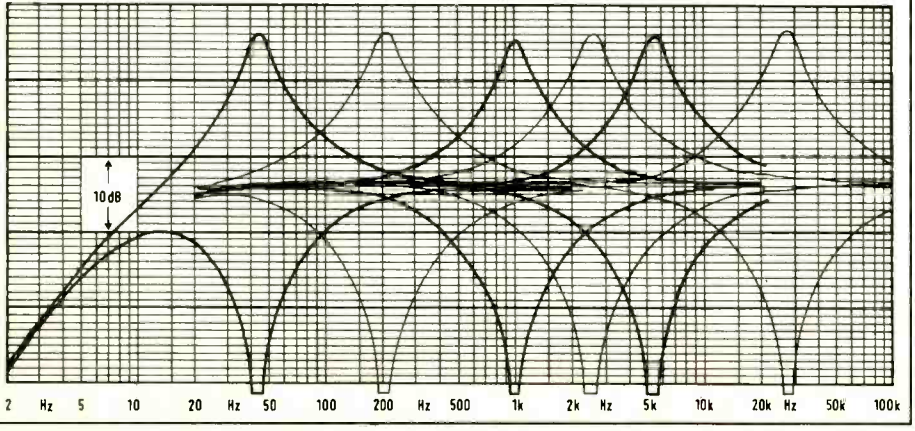


FIG 4.
TC ELECTRONIC 2240
FREQUENCY RANGE, FULL CUT/BOOST 1/2-OCTAVE BANDWIDTH CENTRE 2&4 EQUALISERS



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+1.0/−0 dB in the flat mode being quite acceptable. Changing the gain had no effect upon the high frequency response but led to some fall off at low frequencies extending only to −1 dB at 20 Hz.

Fig 3 and **Fig 4** show the full frequency range of the four equalisers for maximum cut and boost at 0.5 octave bandwidth, it being seen that whilst the range of boost is just over 20 dB the amount of cut exceeds the 20 dB nominal range of the controls. The actual maximum cut was found to interact with the bandwidth control but this is unlikely to have serious consequences.

The effects typical of the bandwidth controls are shown in **Fig 5** for nominal bandwidths of 0.1, 0.5 and 2 octaves. All controls had well contrived laws making setting simple with the calibrations being reasonably accurate.

Cascading two sections could provide very sharp notches, usable for removing hum, etc. Typically two cascaded sections could provide in excess of 65 dB attenuation with the curves for 50 Hz and 100 Hz being shown in **Fig 6**.

FIG 5.
TC ELECTRONIC 2240
EFFECT OF BANDWIDTH CONTROLS

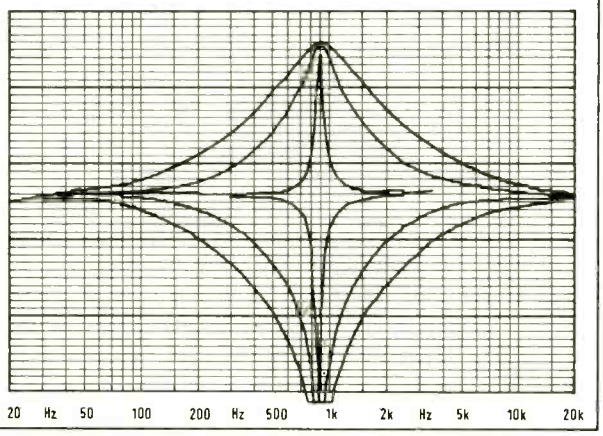


FIG 6.
TC ELECTRONIC 2240
TWO SECTIONS CASCADED

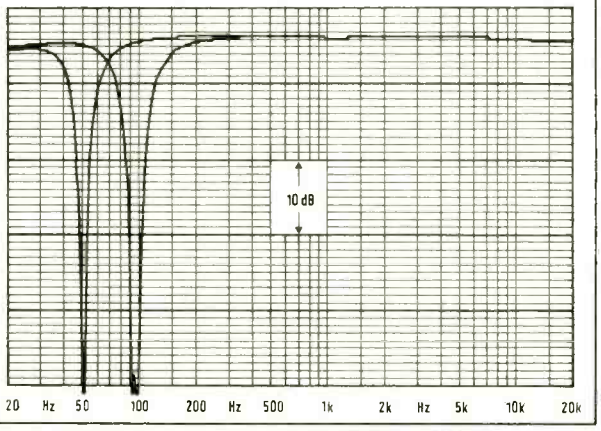


FIG 7.
TC ELECTRONIC 2240
MAX 2ND & 3RD HARMONIC DISTORTION WITH EQUALISERS FLAT

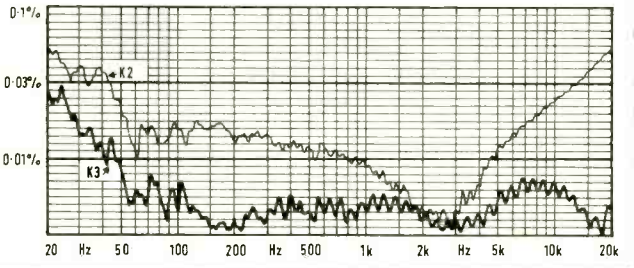
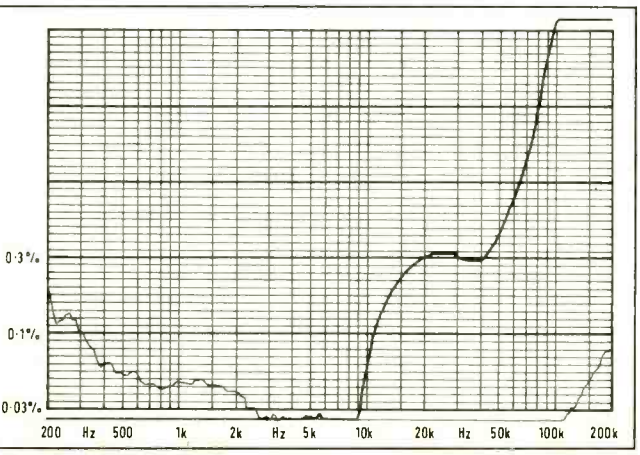


FIG 8.
TC ELECTRONIC 2240
IM DISTORTION
10dB BELOW CLIPPING



Noise

The equaliser was quite remarkably quiet with control settings making little difference to the output noise, except of course, the output level control.

Table 1 shows the noise performance with all controls flat at gains up to unity. With the exception of the two highest frequency equaliser controls which had a small effect on noise, the remaining equalisers amazingly had no significant effect. Bearing in mind the output drive capability these figures show a very large available dynamic range.

TABLE 1

Measurement method	Noise in output (dBm) Bypass	Flat (EQ in)
22 Hz to 22 kHz RMS	− 93.0 dBm	− 93.0 dBm
A-weighted RMS	− 103.0 dBm	− 101.5 dBm
CCIR-weighted RMS	− 88.5 dBm	− 88.0 dBm
CCIR-weighted quasi-peak	− 84.5 dBm	− 84.5 dBm

Distortion

Second and third harmonic distortion in the bypass mode remained below 0.01% from 20 Hz to 10 kHz at maximum gain at any level below clipping rising to 0.03% at 20 kHz at high levels. With the equalisers in circuit in their flat mode the maximum second and third harmonic distortion is shown in **Fig 7** which is typical at high levels with distortion falling at lower levels.

Intermodulation distortion to the CCIF twin tone method was less than 0.03% under 10 kHz at any level below clipping but rose at higher frequencies, the situation at 10 dB below clipping being shown in **Fig 8** where triangulation of the waveform occurs at very high frequencies.

Other matters

Crosstalk between the two channels was at extremely low levels being better than − 107 dB at 20 kHz falling to − 132 dB at mid-frequencies.

The LED level meter was fast in action reaching the steady state indication in 80 μs with a sensible hold time giving a clear indication of peak levels with the calibration of the steps being satisfactory.

Each equaliser section has an overload LED in addition to the level indicator these LEDs being illuminated a little below clipping, however, the action of these was rather slow at about 800 μs with no hold incorporated.

Stabilisation of the internal power supplies allowed a wide range of input voltages with no variation in performance being noted if the input fell from 240 V to 180 V.

Summary

This TC Electronic parametric equaliser gave a really excellent performance in most respects, the only slight hiccup being the high frequency intermodulation distortion. An unusual feature was that not only was noise low but it remained almost constant with variations in equalisation.

Bearing in mind that this is a low cost unit it offers remarkable value for money for domestic or professional use. ■

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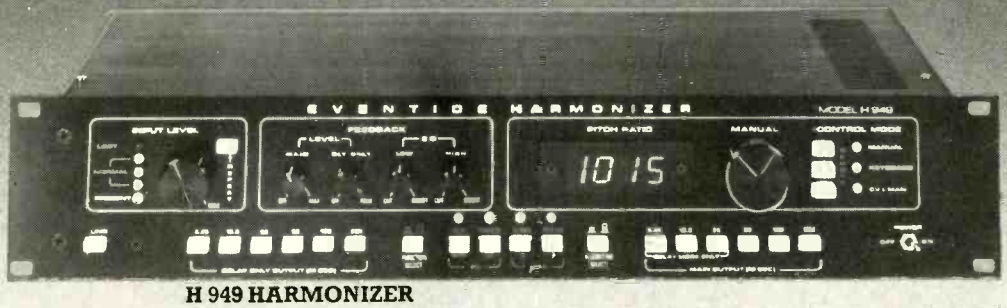
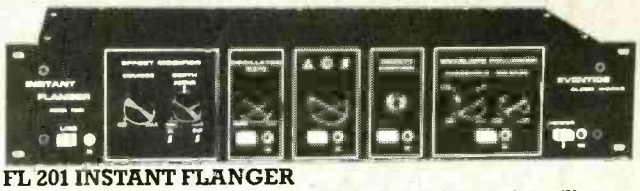


H 949 HARMONIZER
Pitch change: one octave up, two down. Delay: two outputs each 393.75 ms. Micro pitch change. Time reversal. Repeat. Randomized delay. Flanging. High and low feedback E/Q. Two selectable algorithms. Frequency response: 15 khz. Dynamic range 96 dB.

H 910 HARMONIZER
Pitch change: one octave up, one down. Delay: output one, 112.5 ms output two, 82.5 ms. Frequency response 12 kHz. Dynamic range: 90 dB. Feedback control.



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BD 955 BROADCAST DELAY LINE
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2830 OMNIPRESSOR
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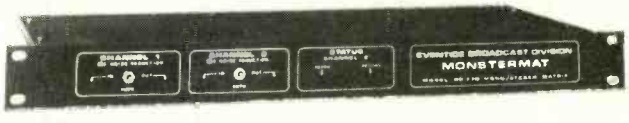
JJ 193 DELAY LINE
Four outputs, each with up to 510 ms of delay, independently switchable in 2 ms steps. Extra delay is optional to a maximum of 1.022 or 2.046 secs. Frequency response: 12 kHz. Dynamic range: 90 dB.



R.D. 770 MONSTERMAT
Mono/Stereo Matrix unit. The Monstermat solves the problem of tape phasing and noise on cartridge machines.

1745M DELAY LINE
Up to five outputs, each with a maximum of 320 ms of delay (640 ms in the double mode) selectable in 20 μ steps. Optional modules available include a pitch changer, and a remote control module which controls the delay line with a micro-computer. Frequency response: 16 kHz (8 kHz in 'double' mode). Dynamic range: 90 dB.

U.K. Distributors
Feldon Audio Ltd.,
126 Great Portland Street. London W1N 5PH Tel: 01-580 4314. Telex: London 28668.

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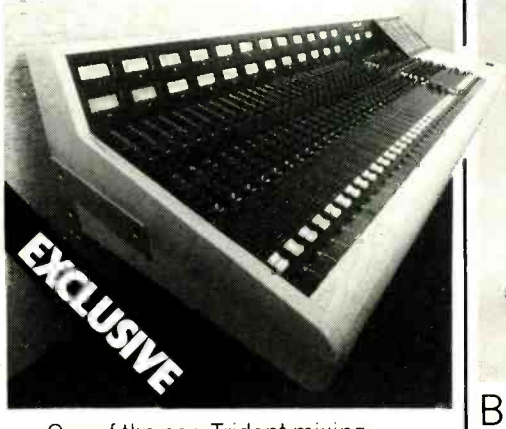
Trident Series 70



One of the new Trident mixing consoles which are exclusive to DLAS. This mixer is the ideal instrument for the producer/engineer having been designed specifically with this beast in mind. It comes in many different configurations – full frame size being 28/16/24 with full patchbay.

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Studio Sound Assistant (ST2)

The ILEA Television Centre, which is equipped to professional colour broadcasting standards requires a Studio Sound Assistant.

Duties include general studio work, rigging, boom operation etc. Some location work will be required.

Experience or knowledge of sound techniques is required.

The current salary range is £7,338-£8,316 plus £1,284 L.W.A.

Application forms and further details are available from: The Education Officer, EO/ESTAB. 1B Room 365, The County Hall, London SE1 7PB.

The closing date for the receipt of complete application forms is 23rd December 1983.

ILEA is an equal opportunities employer

FOR DETAILS OF HOW TO ADVERTISE IN THIS SUCCESSFUL SECTION

CONTACT LINDA GUBBY
01-686 2599 ext. 567

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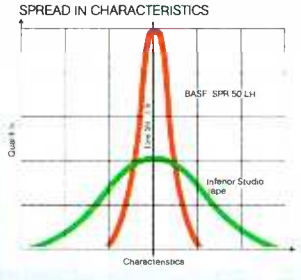
From reel to reel from batch to batch from delivery to delivery quality and consistency.

Compelling characteristics of SPR 50 LH are not only its excellent and well balanced electro-acoustical properties but especially also its consistency. (Narrow tolerances for all characteristics.) You know what it means when things go fast and furious in the studio: No time for adjusting or aligning.



You have to be able to rely on a tape whose characteristics remain; both unchanged and top quality.

BASF's SPR 50 LH gives you this surety.

Reliable in operation
BASF SPR 50 LH.

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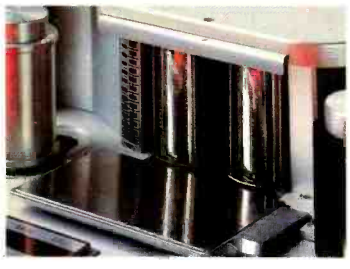
SPOILT FOR CHOICE.



When you come to choose your new multitrack, deciding on a Studer will probably be easy. What will be a little more difficult will be *which* Studer to take - the new A80/VU Mk III or the new A800.

Both machines are superb examples of Studer precision. Both come with the new narrow head block that cuts the travel distance between the erase and record heads to 88 millisecs at 30 ips (now available as a conversion for existing A80/VU models). And both are available in several tape width/channel number configurations.

Whatever your criteria, choosing between the A80/VU Mk III and the A800 won't be easy. But then Studer never have been in the habit of taking the easy way out when it comes to performance.

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