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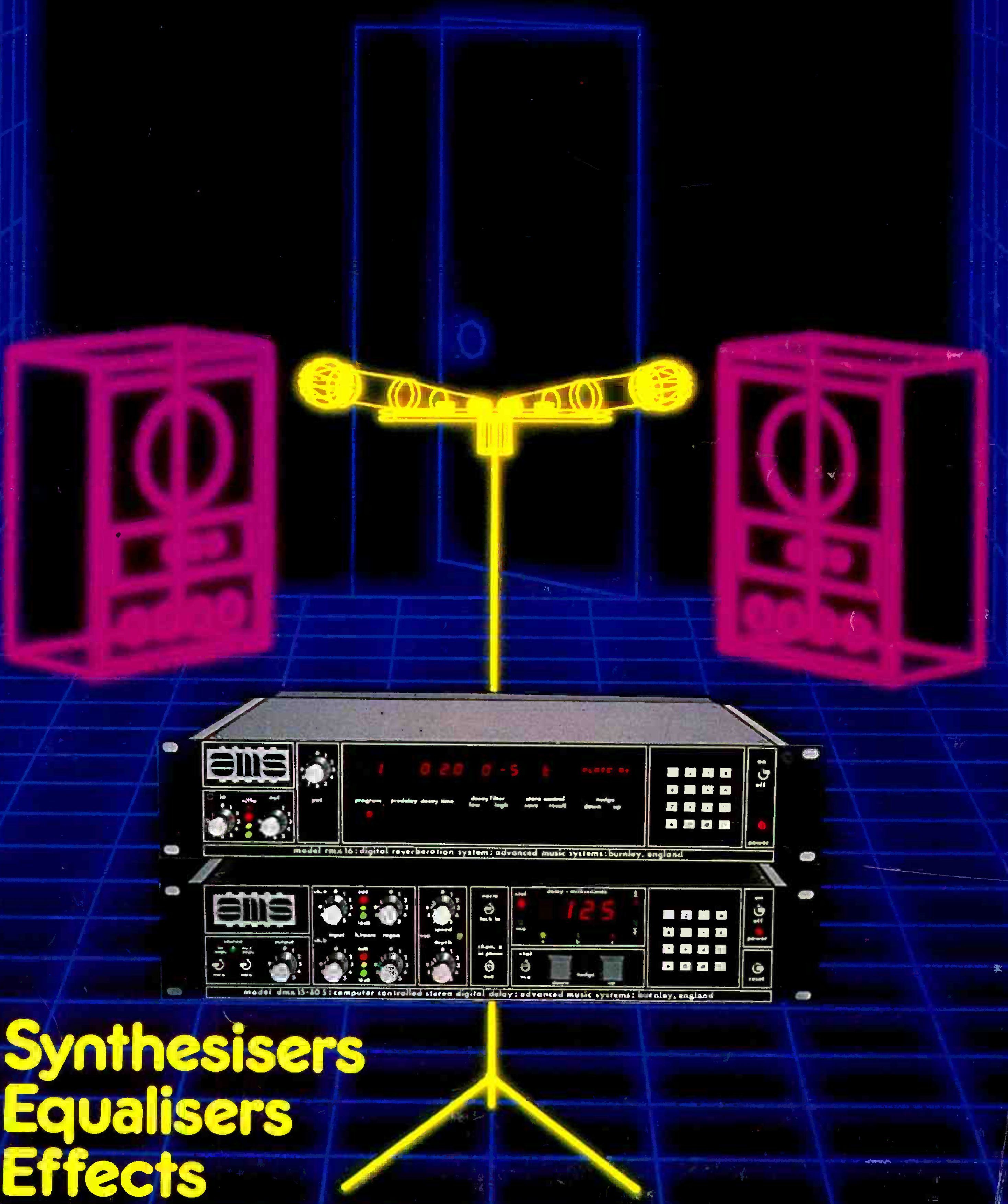
January 1983 £1

A LINK HOUSE PUBLICATION



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AND BROADCAST ENGINEERING

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ABC MEMBER OF THE AUDIT BUREAU OF CIRCULATIONS

Cover: Computer graphics by Electronic Arts Ltd provide the surroundings for Roger Phillips' photograph of AMS effects units

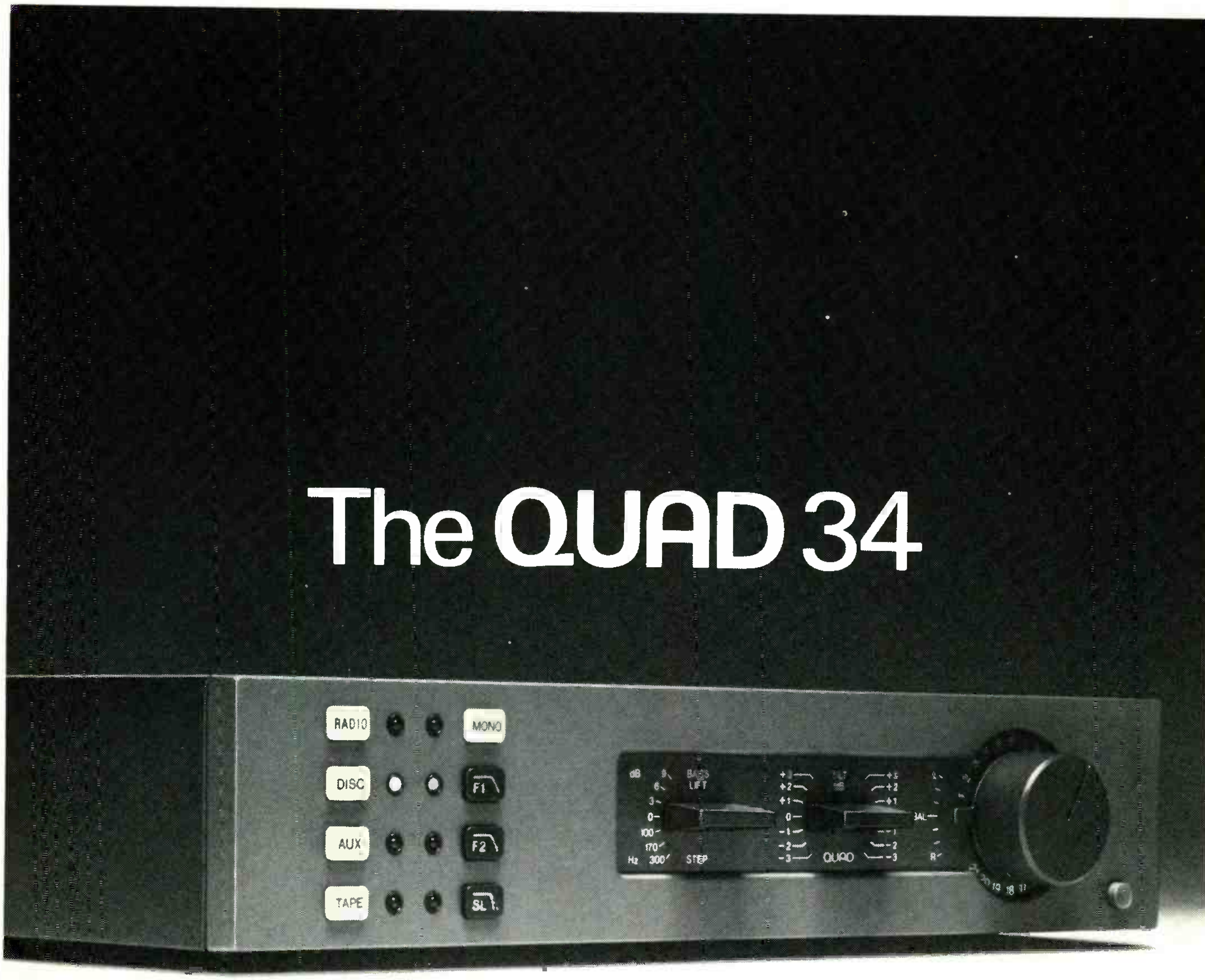
Beati pauperes spiritu, artisque imperiti numerorum

Blessed are the poor in spirit, and those inexperienced in the art of digital. Digits are everywhere: the 'Progressione Dyadica', as Liebnitz called his treatise on binary in 1679, is becoming fundamental to the recording arts. New systems like the dbx *700*, unveiled at Anaheim and prematurely elsewhere, promise to reduce the cost of digital recording, despite non-compatibility with linear PCM methods, although I suspect that Companded Predictive Delta Modulation will really take off when dbx unveil their promised 24-track machine for the price of an analogue recorder. Analogue consoles will be in use for some years to come, and that means, obviously, that it matters not what your multitrack format is as long as it has analogue ins and outs. And given the choice between a CPDM multitrack and an analogue one at the same price, I know which one I would choose. The dbx stereo machine may suffer in the meantime from Japanese competition, in the form of Sansui's *Tricode* and Sony's *PCM-F1* reviewed last month, especially with rumours abounding that it is in fact possible to interface the latter unit with a 16-bit PCM editing system (please tell us if you know how!). *Compact Disc* players are lurking ready to be released upon the world and, believe it or not, there might even be some software to go with them when they finally emerge. The CD seems to have won outright against all the other systems for consumer DADs: I haven't seen any of the others at a show for ages, and it would be a nice change if we had one international compatible system instead of the hundreds that usually seem to emerge whenever the engineering community comes up with a new idea. I hope in vain that compatibility will be the trend for the future.

Digits have been around in music synthesisers for some time: after the Fairlight and the *Synclavier*, we now have at least half a dozen sophisticated machines which may be discovered in the pages of our Product Guide in this issue. One of the most interesting here is the *McLeyvier*: interesting because unlike most of the others it is a sophisticated *analogue* synth under digital control. Another machine worthy of a mention is the beautiful Yamaha *GS-1*, with its FM equation generators and magnetic card loading system. Apart from *looking* lovely (a bit *too* lovely, actually: I would prefer a cheaper one in a roadworthy cabinet) which will no doubt put it into numerous clubs and hotel bars, and bring lots of cheap grand pianos on to the market—indeed, I can see the day coming when only classical musicians will use real pianos at all—it also *sounds* marvellous. Particularly memorable is the church organ sound on card F-5 (Volume One of your sound books, Ancient and Modern) which turns into a lovely cinema organ with the Ensemble button pressed (and try that button on a piano sound for a marvellous 'underwater piano' effect). You may not be able to afford one at £10,000 a time, but the hire fee of about £150 per day is pretty good.

Being January, one wonders what the coming year will bring, as is customary. Here, once again, Digits raise their binary heads. We have already seen some of the early off-putting problems with quantisation noise and top end harshness laid to rest, and there is the possibility that further advances will make even the most hardened sceptics think again. My view is that there are certainly different problems to be dealt with, and that it is still early days, but that digital is the way to go. Even now, there are enough good experiences with digital systems to stop most people reaching for their crucifixes when you murmur '0110111001010111' at them. Perhaps the advent of a few more bits, higher sampling rates and the like will keep people happy. I can't see the 16-bit linear/48 kHz standard lasting too many years, with the field advancing at its present rate; hopefully the next standard will be worked out by everyone in advance. But that is a forlorn hope, I'm sure (totally).

Richard Elen



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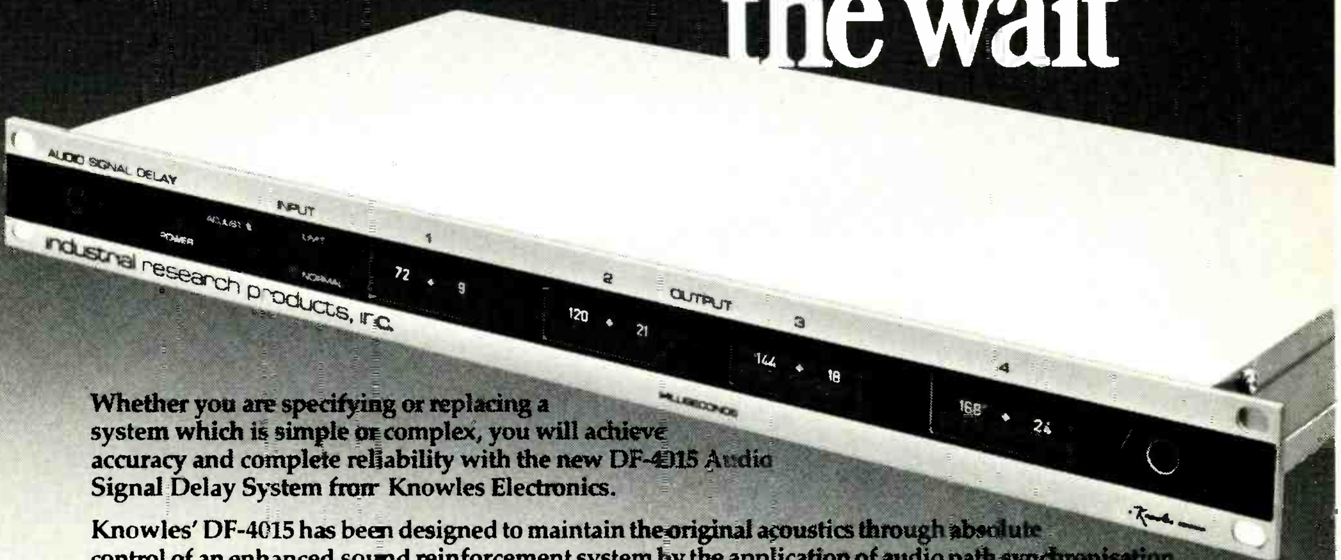

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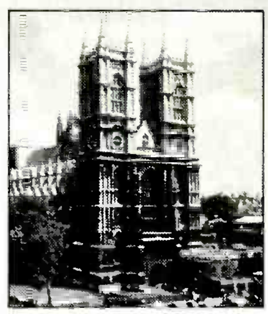
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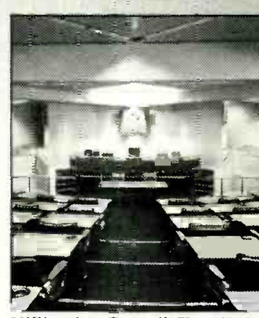
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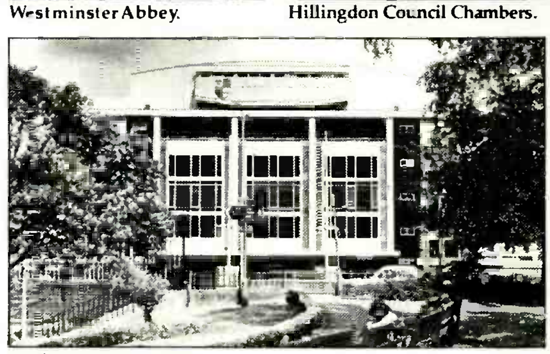
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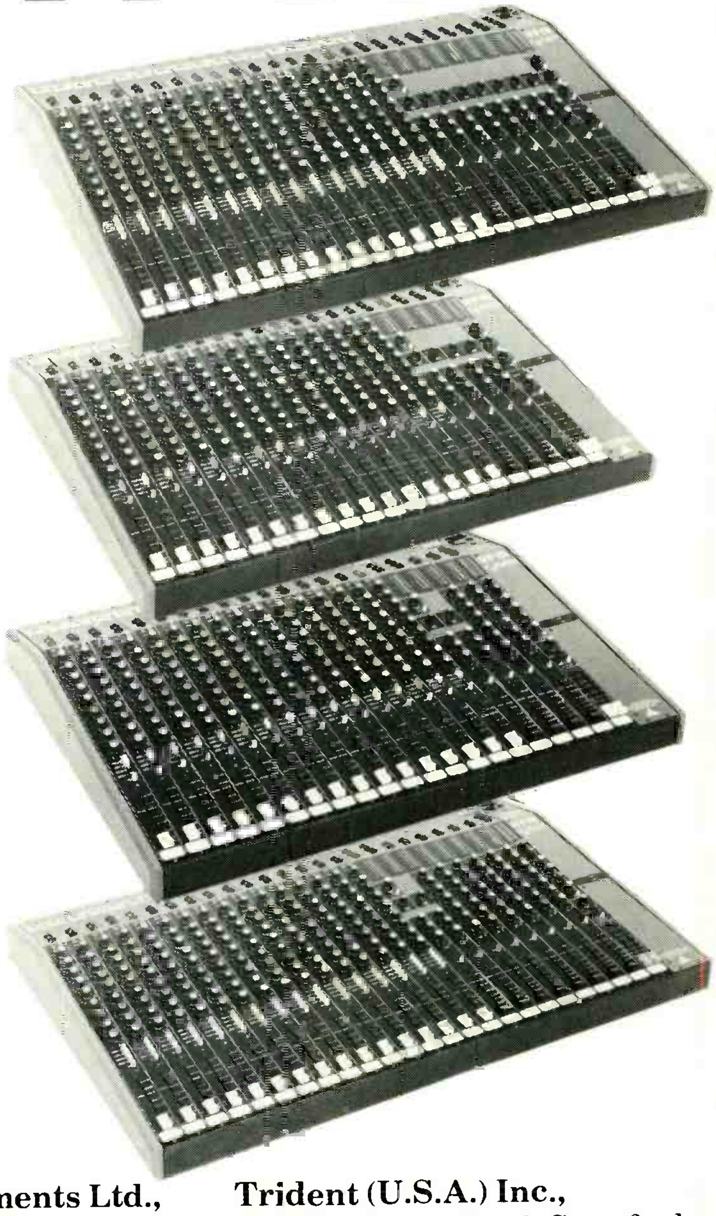

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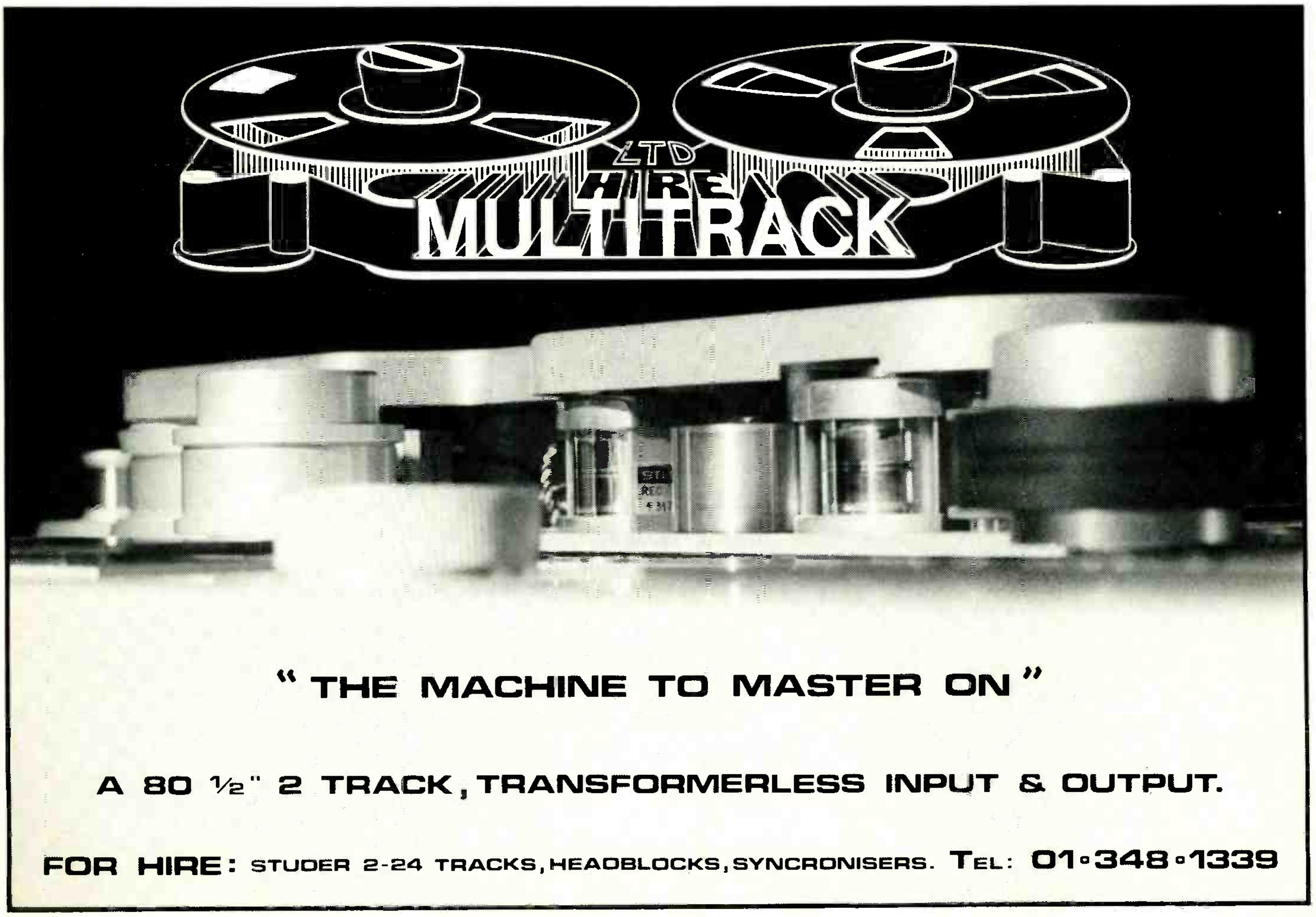
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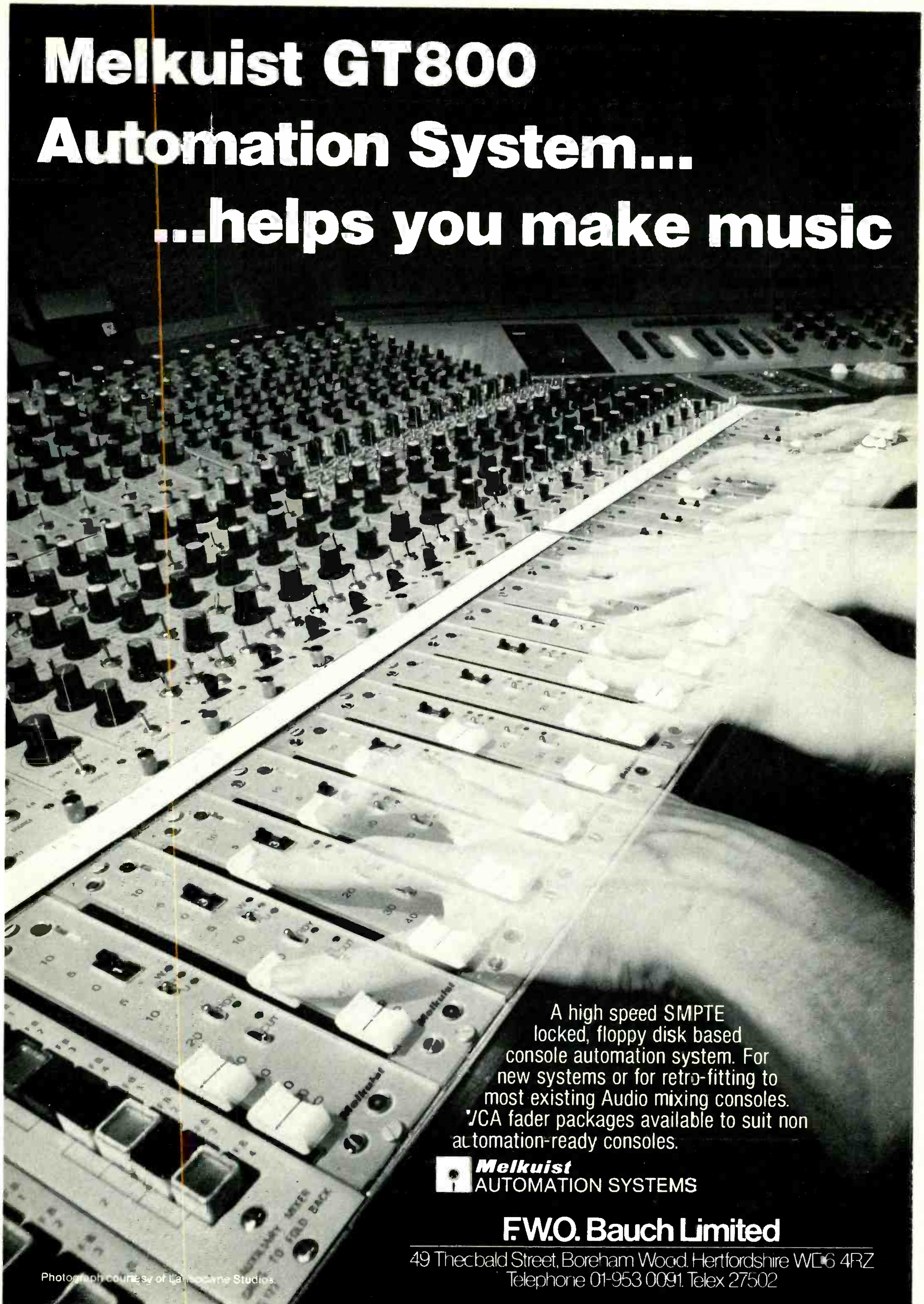

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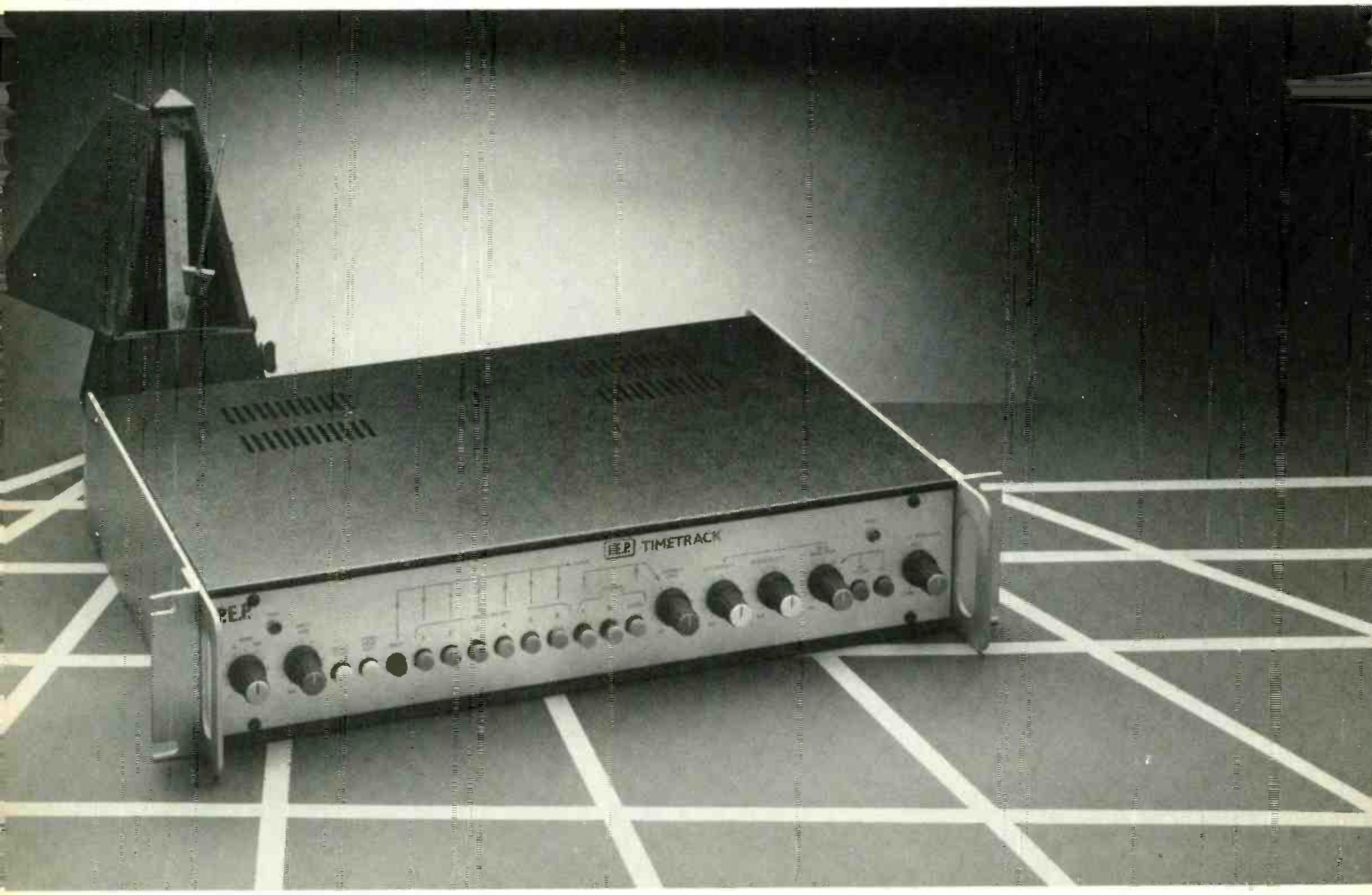

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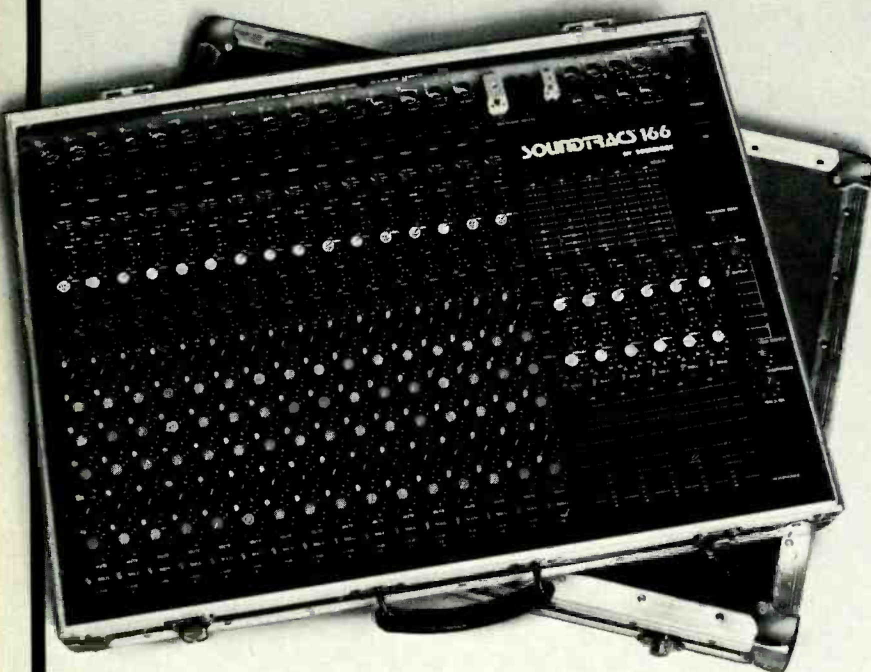

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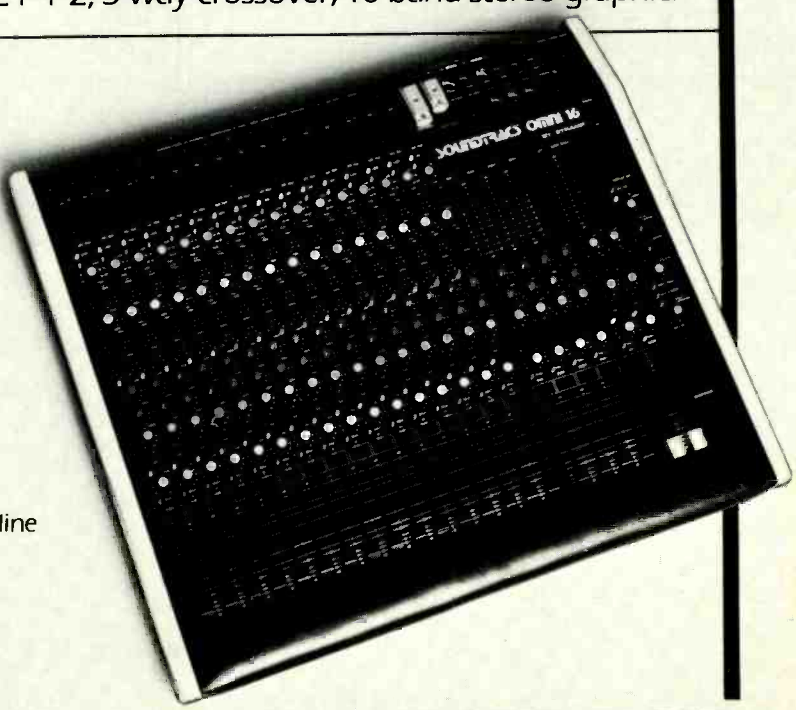

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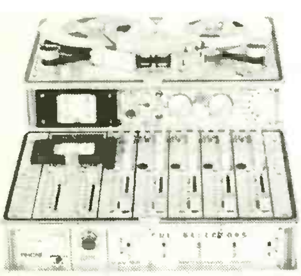

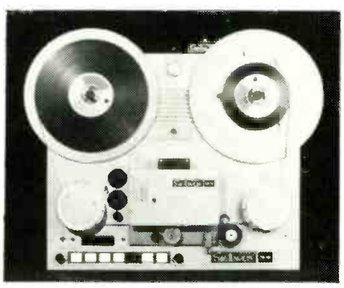

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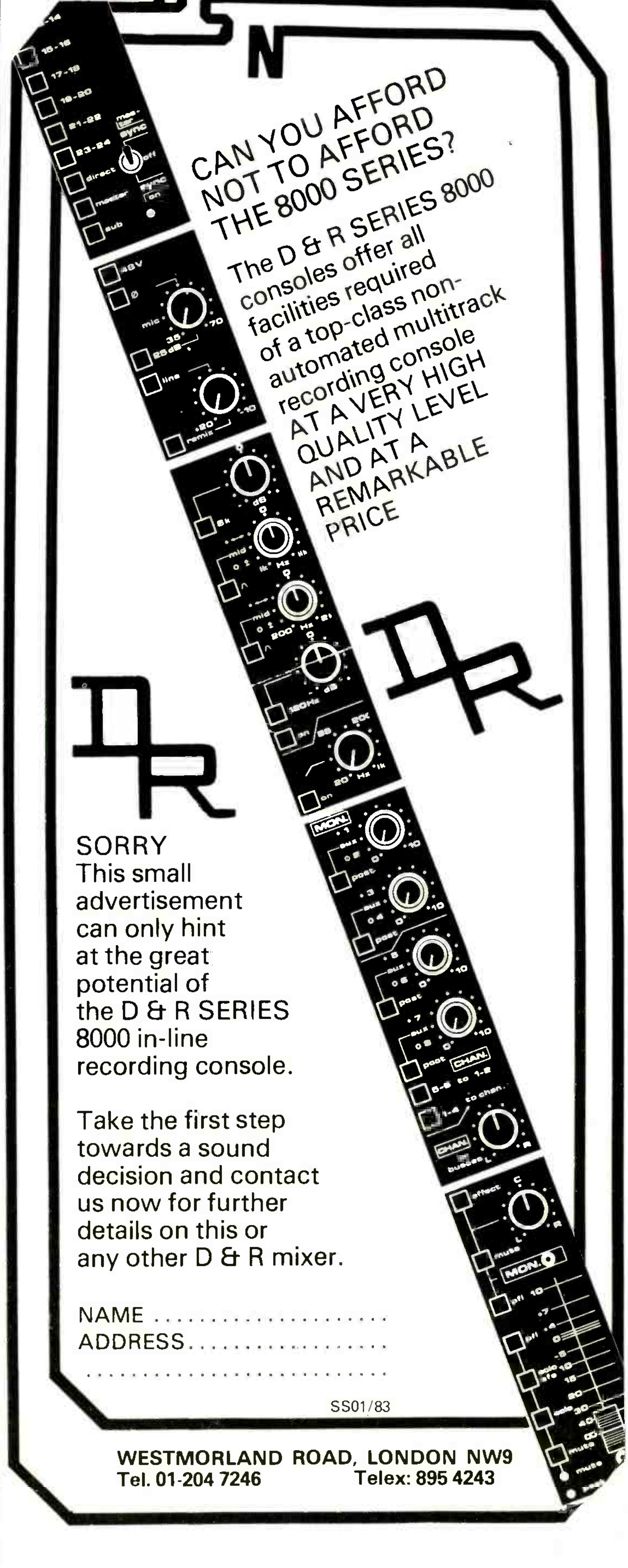

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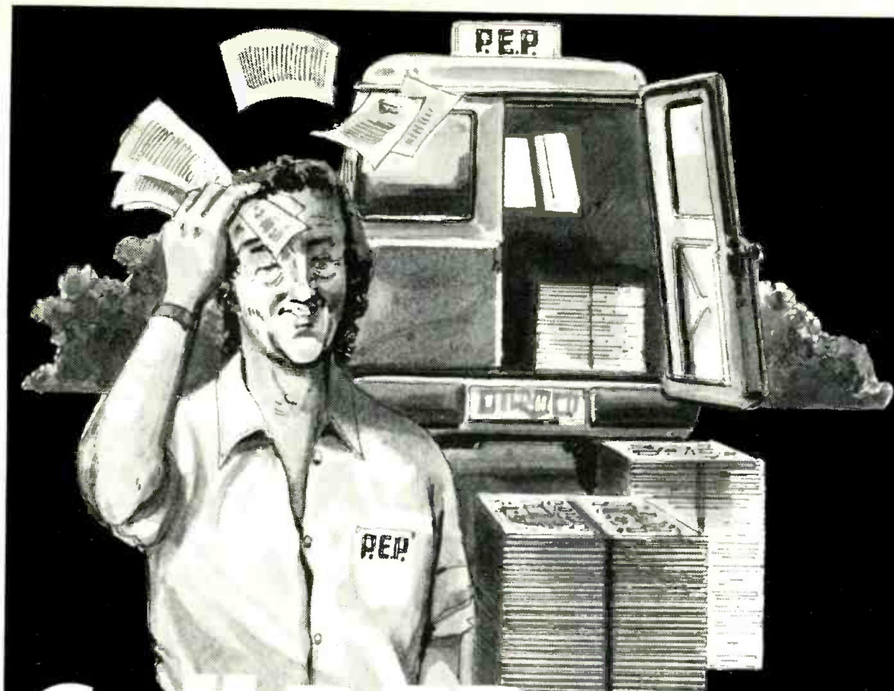

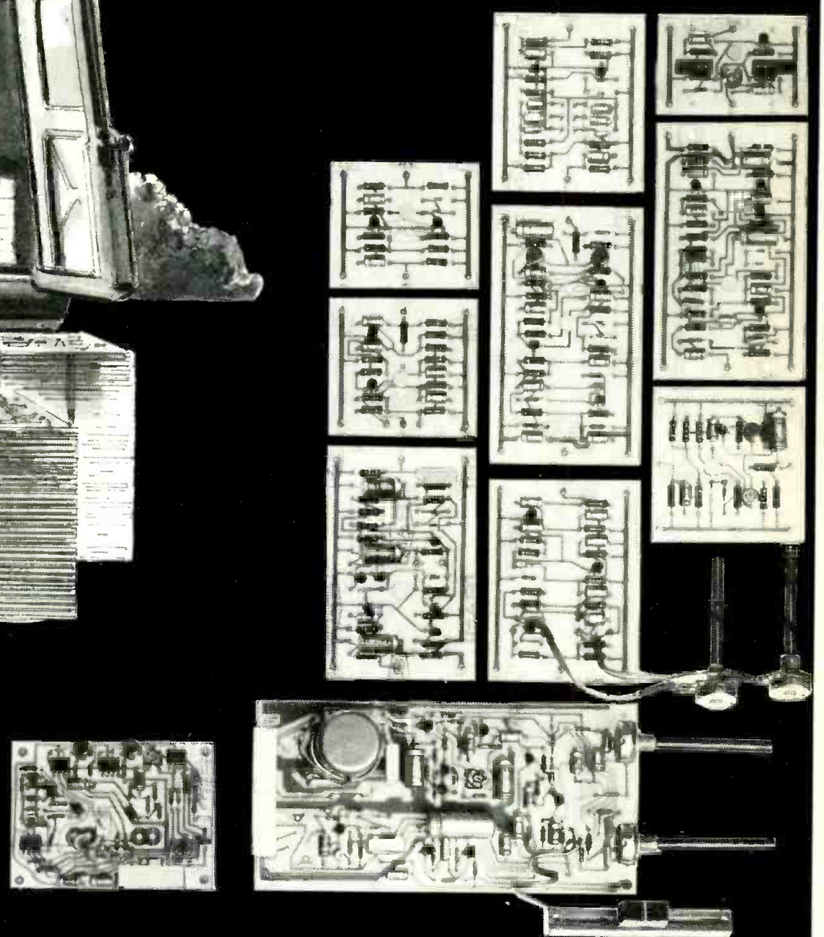

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FREQUENCY RESPONSE REF 1 KHZ 20 KHz	0.5dB	0.5dB	0.5dB	0.5dB	0.5dB	RIAA CURVE ±0.5dB	–0.5dB	0.5dB
FREQUENCY RESPONSE REF 1 KHZ 20 Hz	0.5dB	0.2dB	0.5dB	0.2dB	0.5dB		0.2dB	0.5dB
TOTAL HARMONIC DISTORTION 3 Vrms 1 KHz	LESS THAN .005% at +20dB	LESS THAN .004%	LESS THAN .004% (600 OHM)	LESS THAN .008% (600 OHM)	LESS THAN .005%	LESS THAN .005%	LESS THAN .008%	LESS THAN 0.5%
NOISE (REFERRED TO I/P)	–125dBm USING MT 1	–90 dBm	–110 dBm	–95 dBm	–105 dBm	–72dB REF 2m Vrms 1KHZ AT THE I/P	–125 dBm	–90 dBm
POWER CONSUMPTION (+24V SUPPLY)	10mA	6mA	16mA	12mA	10mA	10mA	20mA	5mA
PHYSICAL DIMENSIONS mm L	115	70	80	130	85	130	160	50
PHYSICAL DIMENSIONS mm W	75	75	75	75	75	75	75	75
PHYSICAL DIMENSIONS mm H	18	15	18	18	15	18	25	15

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diary

Continuing courses

Education at the 'entry level' to recording in Britain is very difficult to come by. The APRS and a few others run admirable courses for the experienced engineer, but there is very little for the home recordist, or the musician interested in recording his or her own material. And unfortunately, some of the courses which *have* been offered from time to time for such people have been vastly overpriced and virtually useless in any sense, practical or otherwise.

A continuing series of courses which do *not* come into the latter category (they are excellent, useful, and cheap for what you get) is offered by Gateway Studio, Battersea. Put together by the studio in conjunction with Bandive, Fostex and Atlantex Music, the series delivers basic instruction on multitrack techniques over four days or evenings for a mere £60 + VAT. Topics covered include sound theory, corrective and creative EQ, reverb and effects, and the application of professional studio techniques to the home recording environment.

Although a good deal of theory and jargon is explained and demystified, the major emphasis is on practical demonstrations of 4-, 8- and 16-track recording techniques and the effects which complement them. The courses represent the start of a series which will ultimately cover all aspects of technical innovations relevant to music and the musician.

Further info may be obtained from: **Dave Ward, Gateway Studio, 1a Salcott Road, London SW11 6DG. Phone: 01-223 8901.**

Memory lane

We at *Studio Sound* are busy compiling a series of articles which, we believe, will present a definitive history of recording in the UK. To prepare the series, which will be published in 1983, we have contacted a number of sources and are amassing a great deal of information. However, there are still gaps in our knowledge. We will be covering the period from the very beginning until about the mid-1970s (after which the scene becomes more or less impossible to follow) and would like to hear from people who can give us information on recording equipment and techniques, plus recordings of the time. Most of all, we would like to trace some of the personnel involved in recording, disc-cutting and production in a number of studios. The particular 'fuzzy' areas which we would like to know more about are listed below, but we would like to hear from anyone who has information on the historical aspects of recording in Britain.

1 Abbey Road and Decca recording activities up to the mid-1950s.

2 Recording personnel involved with IBC/Radio Normandie before 1939, plus information on IBC during the period 1945-55.

3 Jacques and Morris Levy's activities up to the involvement with CBS.

4 General information on Pye's studios going back as far as possible.

5 Background to Sound Techniques (Chelsea) up to the Olympic involvement.

6 Background on Ryemuse.

7 Information on Radio Luxembourg outside of work done at Star Sound Studios.

A good deal of information on the 1960s lurks unseen in back numbers of *Tape Recorder/Studio Sound* but we could do with some pointers to specific dates to know where to look!

Canadian real-time cassette duplication

Comfort Sound, Ontario are offering a real-time cassette duplicating service, having recently installed 10 Sony cassette recorders. Preparation before copying can include facilities such as EQ, compression, expansion and reverberation. Duplication can be either on their standard tape or one of the customer's choice. This facility is in addition to their 16-track studio and mobile.

Comfort Sound Recording Studio Ltd, 2033 Dufferin Street, (between St Clair and Eglinton), Toronto, Ontario M6E 3R3, Canada. Phone: (416) 654-7411.

Synthesists' Union

Reacting to the threatened Musicians' Union ban on synthesisers, the Union of Sound Synthesists (USS) has been formed, not simply to oppose moves damaging to synth players, but also to expand and develop this whole field of music. In the meantime, it seems that the MU is keeping a low profile on the subject of synthesiser usage.

USS is a development of Electronic Synthesizer Sound Projects, the East Molesey, Surrey-based information service which has been offering records, books and information exchange facilities to subscribers all over the world for the past two or three years. More recently, ESSP has begun to supply regular synth music charts to the music weeklies.

While the ESSP information service is available to all those interested in synthesiser and computer-related music (and their mail-order record service is one of the most comprehensive in the world), the Union is aimed specifically at those using computer systems and synthesisers in musical applications, professionally and semi-professionally.

USS insist that the Union has not been set up in opposition to the MU: although they obviously disagree with some MU members on one or two specific points, they see their role as concentrating on a specialised and rapidly growing area, bringing expertise and individuals together to develop a field which is increasing in complexity and importance.

The first phase of the USS campaign to help promote 'the development and education of electronic - computer - synthesiser sound engineering' is now under way, and activities include the preparation of reports on educational facilities, manufacturers, distributors and retailers, plus an examination of radio/TV exposure of synth material and the availability of recordings in retail and other outlets.

Further information on the USS and ESSP may be obtained from **The Sound House, PO Box 37b, East Molesey, Surrey, England,** enclosing an SAE or International Reply Coupons as appropriate.

Swiss computer music

Newly formed is the Schweizer Gesellschaft für Computermusik (Swiss Society for Computer Music), which exists to encourage the development of computer technology in all areas of music, being especially concerned with the artistic exploitation of the vast technical possibilities offered by computer systems.

The Society is interested in contacting people all over the world who have an interest in similar areas. Further details may be obtained from **Bruno Spoerri, Studio für elektronische Musik, Sommerau, CH-8618, Oetwil am See, Switzerland. Phone: (41/1) 929 25 24.**

Sony donates digital gear

The Sony Corporation of America has donated 26 *PCM 10* digital audio processors to the Society of Professional Audio Recording Studios (SPARS) for placement with key recording studios. The donation will enable more professional recording engineers and facilities to gain experience of digital recording systems and will assist in the development and future of digital recording technology. The donations will also enable more recording studios to prepare product suitable for release in *Compact Disc* format when the system becomes available in the USA.

Music clearances made easy

Associated Music Services, the music clearance company set up by ex-MCPs veteran Martin Couche, is changing its name to The Music Clearance Organisation, a name more appropriate to the company's activities. The concept of MCO is to offer to take over negotiations from producers and agencies on the use and clearance of music for commercials, films, A/V presentations and the like, with a view to reducing the cost for such clearances to the end-user. All the phone calls, identification of titles, haggling over licence terms and paperwork can be handled by MCO, with the result that a user simply has to call the company and wait for the results to come back.

On bigger projects, MCO provides personnel to work alongside the production team to make sure that the correct clearances are obtained before the final editing. Video companies will also find the service useful where they need to farm out royalty admin work rather than employ their own staff to do it.

MCO may be contacted at 9 Macklin Street, Covent Garden, London WC2, phone 01-405 7753 or 404 0969.

Otari UK

Otari have set up a UK subsidiary to be known as Otari Electric (UK) Ltd.

The new company will be setting up a dealer network in the UK and providing technical and sales support for their ranges of tape recorders, cassette duplication and audio/video tape winding equipment. Principal staff will be Yoshiaki Shimizu as general manager and Mick Boggis, technical manager.

Otari Electric (UK) Ltd, Unit 2, Herschel Industrial Centre, 22 Church Street, Slough, Berks. Phone: Slough (0753) 38261/2. Telex: 849453.

People

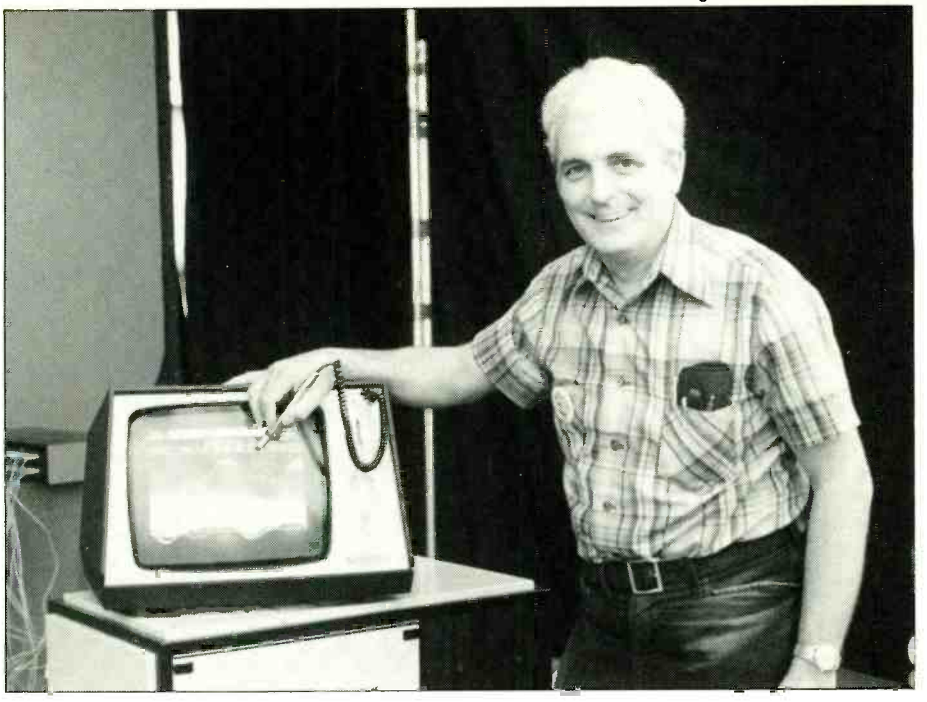
- Criteria Studios, Miami, Florida have appointed Chris Joyce as director of engineering.
- Soundtracs Inc have announced that Robert H Lowig has been appointed to the position of national sales manager at their Farmingdale, NY office.
- Harrison Systems Inc has announced that Dave Purple, former sales manager of Harrison, has rejoined the organisation as sales and marketing manager for broadcast products.
- Paul Headland has been appointed to the board of Molinare Ltd as director of audio.

Sony Broadcast to handle MCI

Following the acquisition of MCI by the Sony Corporation of America, Sony are undertaking a re-organisation of the sales and service support for MCI throughout the world. With regard to Europe, final details of local arrangements have not been completed. Until this is the case, all sales and service for MCI products in the UK, Ireland, Yugoslavia, Hungary, Czechoslovakia, Poland and Bulgaria will be handled by the audio department of Sony Broadcast Ltd. MCI customers requiring sales, service and spares should contact Mike Bennett (audio dept manager) at the address below.

Sony Broadcast Ltd, City Wall House, Basing View, Basingstoke, Hampshire RG21 2LA, UK. Phone: 0256 55011. Telex: 858424. ■

Dr Moog demonstrates the CMI



Moog on music

Barry Fox

BARRY Manilow didn't know what he was starting when he toured Britain in January 1982. He sang for five nights at the Albert Hall, the best seats costing £20 each. But the stage looked very empty: there was no string section, just a small rhythm group and several synthesiser players making the sound of strings. Some people said it was because there wasn't room on stage for a string section, who would all have been knocking each other's arms with their bows. Others said there had been rehearsal hassles with the string section on a previous tour. Some said it was purely a matter of money—by doing without strings the organisers could save £35,000. There were those who said it was a decision taken purely by musical taste. There was even a rumour that when the live album, recorded during the tour, was released it would have strings overdubbed in the studio. I for one sat with pen poised ready to lodge a complaint under the Trade Descriptions Act, but when the album finally was issued only a few extra percussion and vocal sounds had been overdubbed, and all were credited on the sleeve with commendable honesty.

In the meantime all hell had broken loose. The London branch of the Musicians' Union had moved to ban synthesisers from concerts and recording sessions, where other instruments could reasonably expect to be booked. Predictably, musicians started arguing amongst themselves. A rival union for synthesiser players was set up, and correspondence columns buzzed with opposite, and often incoherently argued, points of view.

The eventual outcome of it all, of course, will be that recording producers and concert promoters will go on doing as they have always done, that is, booking whoever and whatever they want for any particular performance.

The real absurdity is that all this is just repeating events which happened 10 years ago in America, because that's where synthesisers really began. The man to thank, or blame, according to your point of view, is Dr Robert Moog. He was in London recently sponsored by Syco Systems to give a series of lectures at the Science Museum and talk about the Fairlight *CMI* computer synthesiser. So we took the opportunity of talking to Dr Moog about synthesisers in general, and his views on the union ban in particular.

Electronic music is much older than Robert Moog, who was born in 1934, but it was Moog who made synthesis, and his own name, a household word. Thirty years before Moog was born, there was an astonishing electronic system in New York called the Telharmonium—a keyboard-controlled bank of motors acting as sound generators, each producing up to 15 kW of power. There was a different generator with a different armature for each pitch. Subscribers to the Telharmonic music service listened to concerts of synthesised music over telephone lines. The equipment was eventually dumped in the Atlantic and no record of its sound is in existence.

In the 1920s a Russian by the name of Leon Terman invented a gadget called the Theremin, a tuned circuit in a box connected to a rod aerial (rather like a car radio aerial) which, as you moved your hand towards and away from it, altered the tuning of the circuit by capacitance thus changing the pitch of notes produced by a loudspeaker. Volume was controlled by approaching a coil with the other hand. The Theremin was a runaway success. The inventor toured Europe and America giving concerts, becoming the darling of Society, but in the late '20s all rights were sold to RCA, who started selling Theremins for the extraordinarily high price (in those days) of nearly $600. Contemporary RCA advertisements describe the Theremin as 'easy to play'—in fact, it's remarkably difficult. Because there is no keyboard, pitching is by ear alone and not surprisingly the Theremin got a bad name and RCA stopped production. Only a few hundred were ever sold. Terman got the rights back from RCA and started making a few one-offs.

For the technically interested, two high frequency oscillators ran at around 200 kHz each. Their beat frequency was audible and changed by capacitance (with the rod) and inductance (with the articulation loop coil). You can hear the Theremin on several film soundtracks—*Lost Weekend, Spellbound* and *The Day the Earth Stood Still*, for example.

Clara Rockmore had a Theremin and continued to play it at concerts around the world—more of Clara Rockmore later. Other than this, the instrument was forgotten.

Meanwhile, in the years following World War II, Robert Moog, about 14 at this time, and his father, a radio amateur, were pottering around in their basement on electrical gadgets. One of the things young Moog built was a Theremin.

In 1954, while still a student of 20, Moog wrote an article for *Radio and TV News* magazine on how to build a Theremin from valve circuits. A few years later he wrote another article explaining how to do it with four transistors. Because it seemed a good idea, he offered to sell kits of all the necessary parts. These kits were offered at $50 each and a thousand were sold making him a profit of $1,500 while still at college. In 1961 that was relatively big money.

Working with the Theremin, which has a five octave range, also gave Moog a trained ear which years of piano lessons had failed to develop. Moog then met up with Herb Deutsch, a musician interested in electronic music, and together they put together a crude synthesiser. In 1964 they showed it to a few friends in Toronto. Word got back to the Audio Engineering Society, then only a new body: "We understand you people are doing something interesting," said the organiser of the following AES convention at the Barbizon Hotel off Central Park South in New York. "Would you like to show it?" they asked. Moog explained that what they were doing was only a hobby and they didn't have any money. Meanwhile CBS had pulled out of the AES convention exhibition and Moog got their space free of charge. He turned up with a couple of tin boxes of circuitry, decorated with paper labels, Surrounded by large stands from the likes of Ampex, Scully and so on, he was, as he put it, 'scared witless'. He gave a paper but was again so scared he didn't know what he was saying. People looked and listened, nevertheless.

In 1966 Walter Carlos, an American composer (later to become Wendy Carlos after a sex change), started to collaborate with Moog. The Carlos recording of *Switched-*

on Bach, put out in 1968, sold better than any previous classical record ever released. Carlos then went on to make *The Well-Tempered Synthesizer* and perform music for the film *A Clockwork Orange* among other works.

In Japan, Isao Tomita began producing extraordinary synthesiser records, some in quadraphonic sound. One of these, Holst's *The Planets*, was banned in Britain for copyright reasons. By now, to the public, the word 'Moog' meant musical synthesis.

There was some interesting Carlos music in *The Shining* and her latest can be heard in the Disney science fiction film *Tron*. But Moog cites the music from *Apocalypse Now*, as heard on the soundtrack album, not the film soundtrack, as one of the best examples of synthesised music. "You heard how the music grows out of the helicopter sound," he says, "but when they came to mix the soundtrack for the film release the effects people took over".

Carlos has now written the Moog music for a 20 minute film made by Dolby Labs. Bob Moog notes ruefully that all his local, rural cinemas 'sound like a sleeping bag'.

Fine line

Moog is obviously irritated by the union's stance. "It all blew up in America 10 years ago," he says. "The union branch, Local 802, tried to ban synthesisers. As late as last year they tried to stop advertisements for synthesisers going in the union magazine, but now they've been outvoted because there are so many synthesiser players in the union."

The union problem prompts Moog to point out the need to stand up to competition from Japan. His work as a consultant for Fairlight and others has shown him that the failure rate for synthesisers and electronic musical instruments made in Italy is 80%. American equipment has a 20% failure rate but the Japanese equipment has a 2% failure rate. Could this be anything to do with the union objection?

"If people want a synthesised sound and they want to book a synthesiser and player, who are the union to ban them?" asks Moog. Is the answer really to do as the *Pirates of Penzance* theatre management have been doing in Drury Lane? Because there are synthesisers on stage, a sad quartet of live musicians has to play in the theatre foyer to satisfy union regulations. For films like *Tron, A Clockwork Orange, The Shining* or *Apocalypse Now*, it's obvious that a synthesiser sound is what the composer intended. The unions say that the ban is on synthesisers mimicking conventional instruments. "But," asks Moog, "how do you draw the line?"

This is a pointed question, because Moog is now a consultant to Fairlight. This £18,500 piece of electronic gadgetry, the *CMI*, is a keyboard instrument hitched to a computer which can, among other things, mimic virtually any known sound and then let it be played from a 6-octave keyboard. The Fairlight can either mimic sounds accurately or alter the waveform. By playing one note from a swanee whistle into a microphone hitched to the Fairlight computer, you can generate a whole scale of swanee whistle notes. Alter the waveform and the swanee whistle starts to sound more like a swanee flute. But no such instruments exist. Is that now a synthesiser sound mimicking a sound that could reasonably be made by a live musician, or is it a synthesised sound of the type that could only be produced by a synthesiser? As Moog says, how do you draw the line?

Musicians like Stevie Wonder, Peter Gabriel and Kate Bush all use Fairlights. Is that breaking a union rule? The Fairlight has been used to produce the sound for the space age Fiat commercials and make the sound of a bumble bee turn into music on the Bulmers Cider radio adverts. It's unlikely that any live musician could have produced those sounds.

On the other hand, it has taken 15 years for orchestral string players to learn to play in tune while wearing headphones. If too much string work goes to synthesiser players, studio string players will opt out of the music business through lack of work. Then what will happen in a few years' time when concert promoters and record producers want a live string sound? There aren't any simple answers.

The Fairlight *CMI* is also used on the new Paul McCartney album, *Tug of War*, made with George Martin. This raises another interesting point. Moog describes a synthesiser as a creative tool, not an electronic alternative to creation. In many ways you can compare the relationship of George Martin and the Beatles with the relationship between a composer and a synthesiser. The Beatles' 'input' of creative ideas into George Martin produced an 'output' of orchestrated musical sound. Without George Martin the Beatles would never have been what they were. Should 'George Martins' perhaps be the subject of the Musicians' Union ban?

Development

Dr Moog—who could give a few D.Phil doctors in the electronic industry a lesson in modesty (clearly far happier being addressed as Bob), —finally owned up, at a gathering of around 30 Fairlight-user musicians, to the real reason for his visit to London. He's been working on a completely new kind of touch-sensitive keyboard. This will enable keyboard players to express themselves on a synthesiser, as if it were a piano. Until now there has been a missing link between the performer and the electronic circuits of a synthesiser. The circuits have to be programmed with nuances in advance and then triggered by playing the keyboard. So there's an awful lot of forethought. What some synthesiser players miss is the opportunity to think on their feet, and that's what Moog will soon be offering. His new keyboard will give synthesiser players the opportunity to be expressive and also oblige them to learn their technique all over again.

Moog's keyboard is built up from keys of ceramic material which are touch-sensitive in five different ways. By moving your finger in one direction you control one function, move it sideways and you control another. Press it down and the speed of depression, sensed by a photo cell, controls another function. Another sensor at the end of travel detects force. Rock the key sideways and you've registered another control. The control signals can be assigned to any function of the synthesiser circuits. Stroking the keys can open up filters, rocking them can bend pitch, speed of attack can be controlled and so on. Moog has been working on the new keyboard with university back-up, and when it's available, probably next year, it will turn synthesiser playing on its head. It will also make the Musicians' Union stance even more difficult to maintain. Can a musician who has learned to play such an extraordinary instrument really be banned simply because it's an extraordinary instrument?

Robert Moog has never, to use musicians' parlance, had a 'real job'. The nearest he came to it was in 1971 when he sold his own company, Moog Music Inc, to Norlin Industries, the musical giant that owns Gibson and Les Paul guitars. It made him financially secure, but although he had the title 'President' of Moog Music he had no authority and obviously hated corporate politics.

In 1977 his contract expired and he and his wife started up their own business again. "We drew the line. I'm too old for corporate politics," says Moog. "In fact I've always been too old." What about academic work? "Politics in academics are as rough as in corporate industry," says Moog. He is his own man, and intends staying that way. Interestingly he speaks with respectful affection of the situation they have in Bell Laboratories where there's no hierarchy to create time-wasting political situations. Everyone is a member of the technical staff or MTS, and gets on with the job of research.

Moog and his wife are obviously very active. They have made a record of Clara Rockmore, the original Theremin virtuoso, which is on the Delos record label. It's available, but not easily obtainable. They even dreamed up a ballet, with Clara Rockmore dancing round a giant Theremin on stage.

Big Briar Inc, the firm which Moog and his wife Shirleigh run together, will make more or less anything you want in the field of electronic musical instruments. They'll even make you a Theremin, at a cost of $1,800, but it will take even a trained musician months, if not years, to learn to play it properly. The irony is that the kind of ban imposed by the London branch of the Musicians' Union would probably outlaw the Theremin as well. So your time learning it would be wasted. ■

Opus 3 and, in the background, the Liberation allows the musician greater freedom of movement

new products

Symetrix low-power power amp

Symetrix have added the *Model A-220* stereo power amplifier to their range. They claim that it was produced to meet a need for a high performance, low-power stereo amplifier for applications such as headphone distribution systems and small monitor loudspeakers. The amplifier is said to develop more than 20 W/channel into 8 Ω with distortion at full rated output of less than 0.02% at 1 kHz.

Features include balanced and unbalanced inputs, a mono bridge mode producing greater than 40 W into 8 Ω, high temperature thermal shutdown and output short circuit protection. The *A-220* requires just 1¾ in of standard 19 in rack mount space.

Symetrix Inc, 109 Bell Street, Seattle, Washington 98121, USA. Phone: (206) 624-5012.

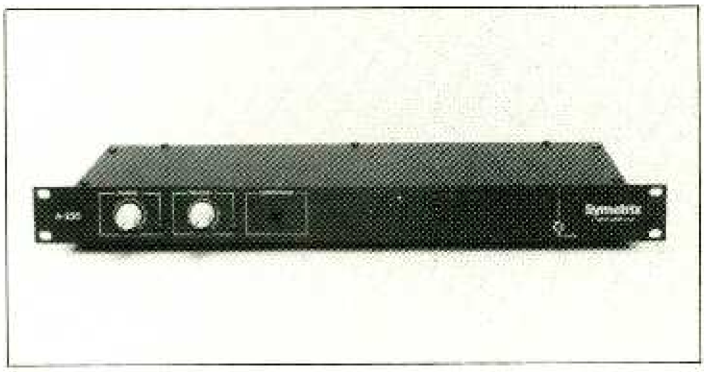
Fairlight CMI developments

Fairlight Instruments have announced two new options for the Fairlight CMI digital synthesiser, the *Rhythm Sequencer* and the *Analog Interface.*

The *Rhythm Sequencer* allows real time composition of complex rhythmic phrases, up to 250 of which may be linked together to form a complete song. After specifying a time signature and bar length, eight separate channels of sound may be added while the sequence is looping. All pitch and dynamic information is recorded from the CMI's 6-octave keyboard and an adjustable time correction facility may be used to correct any playing inaccuracies. Being interactive, all notes are displayed on the screen as they are played. The *Rhythm Sequencer* may also be programmed outside real time by using a combination of light pen and the music keyboard.

The *Analog Interface* is a hardware/software option that allows the CMI to control or be controlled by any voltage controlled synthesiser or effects device. It will permit the user, for example, to harness the CMI's compositional facilities (*MCL, Rhythm Sequencer, Real Time Sequencer*) to control eight 1 V/octave synthesisers simultaneously, or to play the CMI from a guitar or *Lyricon* etc. The package features 16 user assignable inputs and outputs and all patching may be done with software. These patches can then be stored on disk for instant recall at a later date.

Fairlight Instruments Pty Ltd, 15 Boundary Street, Rushcutters Bay, Sydney, NSW 2011, Australia. Phone: (02) 331 6333. Telex: 27998. UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-724 2451. Telex: 22278.



Symetrix Model A-220

USA: Fairlight Distributors LA, 1616 Butler Avenue, West Los Angeles, California 90025. Phone: (213) 478-8222. Telex: 910 3426481. **USA:** Fairlight Distributors NY, 10th Level, 575 Madison Avenue, New York, NY 10022. Phone: (212) 605-0296.

Yamaha's digital synthesisers

To the uninitiated, the idea of digital keyboards may sound as abhorrent as the 'electronic piano' did to the acoustic pianist in the late '60s. Digital isn't a dirty word when it comes to sound generation. We all know of the benefits this technology brings us in recording hardware, but now digital synthesisers are with us and people are beginning to realise that these instruments really do offer the musician something new.

Yamaha recently launched their prestigious *GS* range of keyboards which utilise a hybrid digital technology based around devices known as FM digital equation generators. This system, which was developed in America and 'bought up' by Yamaha's licensing division, derives complex waveforms from the sidebands set up when two (or more) high frequency oscillators cross-modulate one another. The trick is to be able to control the effect, as mathematically it is quite a complex puzzle. Naturally, these devices are computer-controlled; however, such are the intricacies of this system that the *GS* instruments are non-user-programmable. There is of course a programmer designed to preset the sounds but, as Yamaha revealed when they put the device on show at last year's Japanese Music Fair, its cost is comparable to that of the instruments themselves and as such it isn't considered worthwhile marketing them (except to very special order).

So far there are two *GS* keyboards in Yamaha's catalogue—the *GS 1* with a recommended retail price of just £9,999.00 (nice of them to give you a whole pound change) or the *GS 2* at £4,795. However there is good news for those of you with a somewhat smaller budget, in the shape of the *CE 20*—more of this 'treasure of a keyboard' in a minute.

The *GS 1* really is the 'Rolls Royce' job—it certainly looks smart and would be most at home snuggled down in the thick pile carpet of a recording studio. The casework is rather reminiscent of a minigrand piano, but don't try to lift the lid! If there's a smarter looking electronic musical instrument then I've yet to scratch it (oops!). This is a real nice piece of furniture. The rationale behind the rather simplistic looking control panel is 'sound'—Yamaha believe that these instruments should let you concentrate on being a musician rather than a programmer. Personally, I think that they've gone a bit too far as the only adjustable parameters are vibrato, tremolo, ensemble circuit (on/off), touch response, and the equaliser (bass, mid and treble). Three foot pedals are also provided to switch the tremolo and vibrato, and to act as a damper/sustain pedal.

The two main qualities that make the *GS 1* stand out, however, are the voicings, and the touch responsive keyboard. The latter is an 88-key job A1 to C7 and can be used to play up to 16 notes at a time (no jokes about having 16 fingers!). Whereas a conventional instrument will have a touch keyboard that allows you to control just the dynamics of the note played, the *GS 1* offers much much more. For example if you have loaded in a piano voicing, not only will the sound become louder the harder you hit the key, but also extra harmonics and overtones are introduced simulating the character of the acoustic piano far more faithfully than any other electronic instrument.

Now the actual sounds are fed into the instrument by means of a small magnetic strip. The instrument's memory has room for 16 different presets so, by sticking these (rather fiddly) strips into the machine you can load up an arsenal of your favourite voicings. Yamaha 'give' you a rather nice wallet containing a whole host of exciting sounds with which to programme your *GS* machine, and they are constantly adding new titles to this library. The sounds are wonderful; I don't think I've come across such nice a nice collection of usable preset sounds before. However, I suppose at the price that's the least you can expect. Yamaha are right, this is a musician's instrument, but more user variables wouldn't have gone amiss, and perhaps some split keyboard facilities (it's a long enough keyboard, that's for sure).

Whereas the *GS 1* has four FM equation generators, the *GS 2* has but two. This means that the actual definition of the sounds isn't quite as good as that of the *GS 1* (it is still excellent though) and the *GS 2* still retains the 16-note polyphony. The *GS 2*s casework is designed to make it more of a gigging instrument—the legs fold up into the lid, there are convenient handles and the casework is primarily *Tolex* covered. Otherwise the *GS 2* is pretty similar in performance to the *GS 1*, save that here we have a less ostentatious 73-note (E0 to E6) keyboard.

Now the good news is that Yamaha have taken the technology of the *GS* instruments and incorporated it in a superb little keyboard that goes under the designation *CE 20*. The *CE 20*, it has to be admitted, doesn't look too spectacular—in fact it looks downright plain, but switch it on, and have a tinkle (pardon me) and you'll see that for £999.00 (or less) the *CE 20* is a real treasure.

This marvellous machine offers 14 monophonic presets, and six 8-note polyphonic sounds. There is, I understand, a single FM generator on board which gives nice polyphonic sounds but, when operated in monophonic mode with all its power assigned to one note, the quality of the sound matches that of the *GS 1* and the results are quite astounding (in particular the trombone preset). The only dark cloud cast over this machine is that some units have been known to suffer a bit from background noise.

The keyboard is touch sensitive, both velocity and force; it encompasses four octaves (C2 to C6) and has a nice pleasing action to it. The *CE 20* offers a wide range of imitative voicings, which can be further modified using a set of manual override sliders. This is a fine machine, though for some reason it doesn't seem to be attracting the publicity it deserves; it must be the drab casework —an all-important consideration in the image-conscious business.

So, technology never ceases its relentless advance, and in the world of electronic keyboards there is always something new to 'invest' one's money on. The *GS* machines are a bit on the pricey side, it has to be said; still the instrument hire companies probably won't complain, it is this kind of instrument that provides their bread and butter (I think the *GS 1* goes out for around £150 a day). But if you have a grand sitting around, burning a hole in your pocket (as if) take a close look at the *CE 20*. **David Crombie**

Bandpass filter design

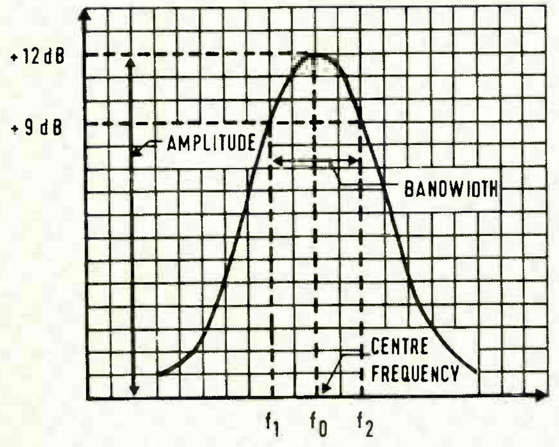
Dennis Bohn

Analogue and digital audio designers are often confronted with every imaginable aspect of active and passive filters. Mostly, these are bandpass filters partially characterised by a quality factor, Q, and a bandwidth, BW. There exist enough books on active and passive filter design to fill a modest office library; however, there are certain aspects of the relationship between bandwidth as expressed in octaves and Q that are poorly documented — if at all.

The purpose of this article is to document the mathematical relationships between these two variables and provide a useful 'look-up' table for each, as well as calculator programs applicable to the Hewlett-Packard handheld *Model 11C* family.

A BANDPASS filter is characterised by three major parameters: centre frequency, amplitude response (gain), and bandwidth. Centre frequency is the frequency at which the amplitude is maximum; the gain of the filter is the maximum amplitude response occurring at the centre frequency; and bandwidth (or passband) is the frequency range between the −3 dB points located on either side of the centre frequency. Bandwidth can be expressed in several ways: in frequency as being so many Hz wide, in octaves, or in decades. Far and away, the most common usage in audio is to express bandwidth in octaves. It is here that the literature falls short of giving sufficient mathematical relationships to allow answers to be expressed easily in 'octaves'.

FIG.1 BANDPASS FILTER PARAMETERS



When first designing a filter, normally the required BW in octaves is known and the associated Q needs to be calculated; once the filter has been designed then Q is easily found by measuring the −3 dB frequency points and the BW in octaves can then be calculated.

Sometimes only Q is known and the BW in octaves is desired. This calculation is not obvious — nor easy. The next section will present the necessary closed solutions for each of these calculations. **Fig 1** shows a bandpass filter with its associated parameters labelled for clarity and will be used for derivation purposes.

Given the −3 dB points, to find BW and Q

If the −3 dB points are known, then calculating the BW in octaves is straightforward:

Let $f_2 = y f_1$, where y is any positive, real number

Define N as the number of octaves of BW, ie:

N octaves means that $y = 2^N$

then: $$f_2 = 2^N f_1 \quad (1)$$

solving for N gives: $$N = \frac{\log y}{\log 2} \quad (2)$$

and by definition: $$Q = \frac{f_0}{f_2 - f_1} \quad (3)$$

Given the BW in octaves, to find Q

If the BW in octaves is known but not the actual −3 dB frequencies, and Q is to be calculated, then the following development will lead to the required formula.

In general, it can be assumed that f_0 is the geometric mean of the skirt frequencies, f_1 and f_2:

$$\therefore \quad f_0 = \sqrt{f_1 f_2}$$

from (1): $$f_0 = \sqrt{f_1(2^N f_1)}$$

or: $$f_0 = \sqrt{2^N} f_1$$

from (3): $$Q = \frac{\sqrt{2^N} f_1}{2^N f_1 - f_1}$$

or: $$Q = \frac{\sqrt{2^N}}{2^N - 1} \quad (4)$$

Table 1 shows several examples of equation (4) for BWs commonly used in audio design work.

Given Q, to find BW in octaves

If only Q is known and the BW in octaves is desired then equation (4) must be turned around and re-expressed in terms of Q.

from (4): $$Q = \frac{\sqrt{2^N}}{2^N - 1}$$

or: $$\sqrt{2^N} = Q(2^N - 1)$$

let: $$y = 2^N$$

then: $$\sqrt{y} = Q(y - 1)$$

squaring: $$y = Q^2(y^2 - 2y + 1)$$

re-arranging: $$y^2 - \left(\frac{2Q^2 + 1}{Q^2}\right)y + 1 = 0$$

applying the quadratic solution:

$$y = \frac{2Q^2 + 1}{2Q^2} \pm \sqrt{\frac{\left(\frac{2Q^2+1}{Q^2}\right)^2}{4} - 1} \qquad (5)$$

N is now found from equation (2).

The squaring operation in equation (5) introduces an extraneous root which appears as the 'minus' square root term. The 'plus' square root term gives the correct answer and the 'minus' the reciprocal answer, ie, the 'plus' answer leads to y, while the 'minus' answer is $1/y$.

Some more helpful information is given in the forms of **Table 1** and **Table 2.**

TABLE 1 Q v BW

BW (octaves)	Q
2	0.667
1	1.414
⅔	2.145
½	2.871
⅓	4.318
⅙	8.651

TABLE 2 BW v Q

Q	BW (Oct)	Q	BW (Oct)	Q	BW (Oct)
0.50	2.54	1.50	0.945	6.50	0.222
0.55	2.35	1.60	0.888	7.00	0.206
0.60	2.19	1.70	0.837	7.50	0.192
0.65	2.04	1.80	0.792	8.00	0.180
0.667	2.00	1.90	0.751	8.50	0.170
0.70	1.92	2.00	0.714	8.651	⅙
0.75	1.80	2.145	⅔	9.00	0.160
0.80	1.70	2.50	0.573	9.50	0.152
0.85	1.61	2.871	½	10.0	0.144
0.90	1.53	3.00	0.479	15.0	0.096
0.95	1.46	3.50	0.411	20.0	0.072
1.00	1.39	4.00	0.360	25.0	0.058
1.10	1.27	4.318	⅓	30.0	0.048
1.20	1.17	4.50	0.320	35.0	0.041
1.30	1.08	5.00	0.288	40.0	0.036
1.40	1.01	5.50	0.262	45.0	0.032
1.414	1.00	6.00	0.240	50.0	0.029

TABLE 3 Programs for handheld HP-11C type calculators

To solve for Q given BW in octaves

```
g P/R
f LBL C
2
x≷y
y^x
STO 1
√
ENTER
RCL 1
1
−
÷
g RTN
g P/R
```

To use, enter BW and press f C.

To solve for BW in octaves, given Q

```
g P/R
f LBL D
g x²
STO 1
2
X
1
+
RCL 1
÷
2
÷
STO 1
g x²
1
−
√
RCL 1
+
g log
ENTER
3.3219
X
g RTN
g P/R
```

To use, enter Q and press f D.

Doublespeak?

For me the high spot of the summer Consumer Electronics Show, held in Chicago in June, wasn't the plethora of incompatible video discs, the miniature VHS recorder that is still too large to count as a truly portable point-and-shoot home video system, or the welter of TV games cartridges that let you hunt, maim or kill videographic blobs from the comfort of an armchair. It was the Variable Speech Control, or VSC, technology buried away in a small demonstration room off the main thoroughfare of the giant exhibition hall.

VSC is a San Francisco firm which has developed a system of replaying recordings at high speed, but with the pitch reduced to normal. So the sound output is rapid speech, or music, clipped in time but still intelligible because there is no Donald Duck effect. VSC circuitry is already incorporated into some VHS domestic video recorders, under licence. These enable you to run a video tape at twice normal speed, while hearing the soundtrack at double speed, but normal pitch. The snag is that double speed is often too fast for intelligibility, especially if the original speech is brisk. Also, there are plenty of times when it would be useful to replay an audio tape, rather than a video tape, at high speed.

VSC has now built speech compression circuitry into a portable tape recorder which is selling in the USA for around $200. I brought one back and now couldn't live without it. Messages on my telephone answering machine, taped interviews, taped broadcasts and taped letters, can all be replayed at anything up to twice normal speed, with continuous speech and pitch variation. You'd be surprised how much time can be saved this way. The latest news is that VSC has licensed the circuit to Integrated Sound Systems of Long Island, New York to produce a broadcast quality unit costing around $8,000. The portable $200 player is of course of relatively poor audio quality, but it's perfectly adequate for speed speech listening if nothing else.

The broadcast unit can be used to tighten up interviews, for instance, with non-professional speakers who waffle. When linked to film or video equipment, it can be used to 'time-justify' any kind of programme to fit a pre-determined time slot. If you've got an interview that lasts one minute five seconds, you can run it a little faster to fit a one minute slot. If you've got a one hour five minute film, you play the same trick. [*A similar job to that done by devices from Lexicon and Eventide—Ed.*] The chances are no-one will notice. After all, feature films are already running 4% fast on European television. Cinema films are shot at 24 frames per second and on television they are screened at 25 frames per second. No-one notices the change of pitch, let alone the faster action.

The VSC idea was patented back in the mid '60s by Sanford Greenberg of Buffalo. Greenberg was a graduate student at Harvard and in the early '60s he found that he was losing his sight. So he began a study of speech compression and, with Murray Schiffmann, finally came up with a workable circuit. But it's taken until now for large scale integration to bring the cost down to a reasonable level. The theory is really quite simple, once you've read the original patents.

The accelerated sound, from tape or disc, is chopped up into segments, and some segments thrown away. This leaves a train of sound segments and blank spaces. The sound segments are now fed into a Bucket Brigade Delay line and as each segment passes down the line the delay is gradually increased. So the segment stretches in time and its frequency drops. If all the variable parameters for chopping and stretching are correctly matched, then the output of the delay line is a continuous train of sound segments, all joined at zero crossing points, and together providing an accelerated replica of the original sound signal. The technique can also be used (although not in budget domestic equipment) to raise the pitch of a slowed signal. In this case the VSC circuit has to fill in gaps created by compression of the segments. It does so by reducing the delay in the line as the segments pass through. The resultant gaps are filled by using corrected segments more than once.

Of course, as the editor notes, studios and post production film and video suites already use pitch correctors, but to the best of my knowledge VSC is the first company to put custom-designed speed-pitch control over a wide range, into a single package of broadcast quality with a slave unit for stereo. It's made by Integrated Sound Systems Inc of Long Island, a subsidiary of VSC and handles speeds over the range of 1 to 2.55 × with a frequency response of 20 to 15 kHz at the higher speeds and 20 to 20 kHz over the lower speed range. Dynamic range is 96 dB and ISS claims there is no intermodulation between voice and ambient noise or musical accompaniment.

Heaven forbid that broadcasters should start using VSC circuits to 're-edit' films which have been deliberately directed at slow pace. But you'd be surprised, once you start listening to speeded up interviews, how tedious they sound when then heard at normal speed. The more I use VSC circuitry the more I'm convinced that humans suffer from a basic design fault. We are equipped to listen intelligently at a much higher speed than we are equipped to talk intelligently.

Note, incidentally, the careful way I've avoided saying, without qualification, that the VSC circuits are the first and only ones of their kind. I've learned the hard way never to use words like that.

Many thanks, for instance, to the several readers who've written in pointing out that the Rank-Wharfedale phase checker, which gives a direct readout of loudspeaker phase, isn't the first or only one of its kind. It seems that quite a few firms have been selling similar units over recent years. How easily they are available to anyone wanting to buy one off-the-shelf without hassles, is another matter.

Industrial noise

An interesting point raised by a correspondent: how long will it be before a producer, engineer or tape op sues a studio for loss of hearing caused by exposure to excessive sound levels in their control room over a long period of time? As he points out, 120 dB is now regarded as a base line by some engineers.

If you listen at those levels for long sessions over a period of years, you are virtually certain to take the edge off your hearing. As he also points out it's unlikely that any studios have thought to insure themselves against claims for damages, eg from a tape op who has no choice but to listen at the levels set by the engineer and producer calling the shots.

Although it's a long time ago, some people still have memories of Studios 2 and 3 at Abbey Road in the late '50s. A reference microphone was slung over the desk at the engineer's listening position and connected direct to a meter with red markings. If anyone monitored loud enough to hit the red, they were fired. It was EMI's way of ensuring that nobody sued them for damages. Sir Joseph Lockwood, it seems, knew about legal problems in the weaving and sheet metal industry, where workers routinely end up deaf after a lifetime of high level noise into unprotected ears. Can anyone with a long memory of Abbey Road remember at what level those safety meters were set to go into the red? It would be a very interesting barometer on monitoring sound levels today. My bet is that the Abbey Road red level was well below 100 dB.

I almost had the opportunity to find out the answers first hand, by talking to some of the people who have worked at Abbey Road over the years. Many of them gathered at the studio to celebrate publication of a book on the history of Abbey Road. But, like a lot of other journalists who would have been interested, I never heard about the event until afterwards.

Pity; it might also have been an interesting book to tell you about.

Exploding musicians

Every year at the Albert Hall there's a pop classics concert which features the *1812* with cannon and mortar effects. The cannon is electronically triggered and there's been a running dispute over the years between the Musicians' and Electricians' Trade Union about who should be employed to 'play' it.

After lengthy negotiation, a brotherly compromise has been arrived at. An ETU man pushes the cannon trigger button but he doesn't read music; an MU percussion player cues the ETU man by tapping him on the shoulder. The snag, of course, is that the MU man has to take the ETU man's reaction time into account. And this can vary from shot to shot. Not surprisingly the musical timing isn't always too precise. There's also an added problem. The cannon doesn't always fire and at £3 per shot the promoter refuses to pay for duds. A minion stands by the MU and ETU men to count the shots that work.

Nevertheless, and despite all this, one night, one year, everything went right. Sixteen times the MU man tapped the ETU man on the shoulder at just the right moment in advance, sixteen times the ETU man pressed the trigger with the same reaction time and sixteen times the cannon went off on cue exactly where Tchaikovsky intended. As the applause died away the MU man congratulated the ETU man by slapping him on the shoulder...Bang.

Agony

There is a producer and arranger who is very fussy over monitor loudspeakers. One studio in which he works has two sets and he doesn't like either. So he brings in a third set of his own. For one reggae session he recorded alternate beats of the basic reggae rhythm separately, and then replayed them with alternate beats on left and right sepakers. Then he tried it with alternate beats on right and left speakers. Then he tried it on the next set of monitors, left and right, right and left. Then he tried it on the third set and between sets. In the final mix you couldn't hear the guitar clearly anyway. But for a full day in the studio the band chatted, played chess and totted up all the lovely overtime due to them. We are reliably informed that this story does not feature in any of the record industry's submissions to the Government in support of a tax on blank tape to boost its ailing profits. ■

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Ground Star Laboratory, Nashville

In the classified pages of the Nashville telephone directory, under the heading 'Recording Service—Sound and Video', there are some 190 separate listings. That works out to one for every 200 people in this city on the banks of the Cumberland River that calls itself 'The Athens of America'.

Which is an odd appellation, in that there isn't even one Greek restaurant in town. What there are, of course, are a lot of Bar-B-Q pits and country-music clubs in between the souvenir shops, wax museums and Western clothing stores, but Nashville is still a city of charm and interest. One of the better spots, not listed in the phone book, is a unique and attractive recording studio right in the heart of the 12-block area known as Music Row.

Unlike the dozens of other studios and music-business offices here, Ground Star Laboratory keeps a low profile. The name of the studio appears nowhere on the outside of the building—the sign says 'U.S. Recording'—which was what the former owner, Roy Orbison, called the place before he sold it in 1978 to the present owner, the blind singer-pianist-songwriter, Ronnie Milsap.

"Everything we do here has to do with Ronnie's organisation," explains Ben Harris, the Kentucky-born chief engineer, "whether it's his own records, or artists that he's developing." The studio shares the building with four publishing companies, a management firm and a fan club, which all together comprise Ronnie Milsap Enterprises. "Ronnie's been asked many times to lease the studio to outside people, but he's refused every offer. He may do it someday, but it would have to be somewhat more expensive than the going Nashville rate for 24-track." (Currently about $125 an hour.)

Harris isn't bragging. Though there are dozens of excellent facilities within a stone's throw of his studio, he has a special sense of pride about Ground Star, and justifiably so. The heart of the operation is a *Necam*-automated 40 × 32 Neve *8078* console (according to Harris, the last such board manufactured) that feeds two transformerless Studer *A80* 24-tracks, one an updated Mark II, the other a brand-new Mark III. Locked together with a BTX *Shadow* SMPTE synchroniser, the two machines are both used routinely on Milsap's own sessions: a typical basic session consists of two electric guitars, two acoustic guitars, steel guitar, bass, drums, two keyboards, vibes and Milsap's live lead vocal.

In Milsap's way of recording, this last often turns out to be a keeper. "He's well accomplished at doing his vocals while the tracks are going down," says Harris, "and it certainly sounds better that way." The basics are mixed down to two tracks on the second tape machine, and then the overdubs go on: background vocals, strings, horns, percussion, guitar leads and sometimes more keyboards.

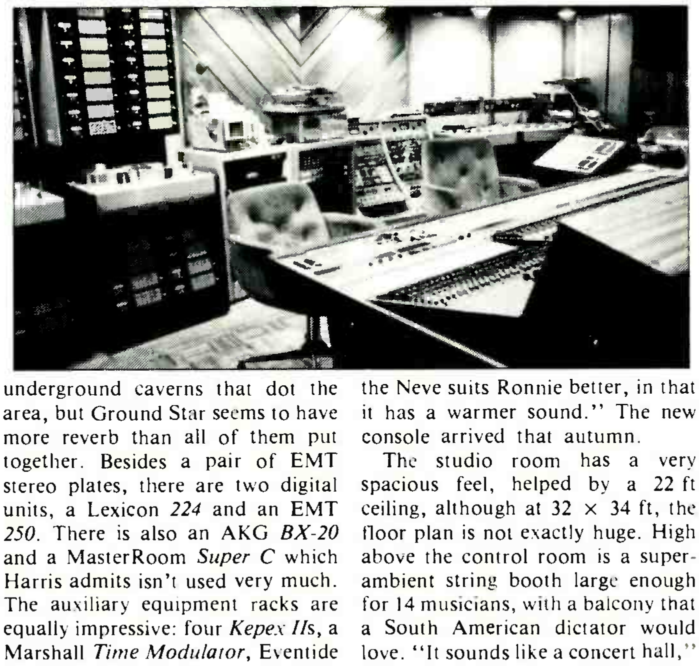
There are two other Studer *A80*s in the control room, a ¼ in deck and a new stereo ½ in. "Working with the ½ in is like taking a blanket off the sound," says Harris. "If we record tone at the same level on both machines and play it back through the board, the meters will say the same thing, but the ½ in will *sound* about 3 or 4 dB louder. Of course, there's also a noticeable difference in signal-to-noise. We do everything on the ½ in now, and we just use the ¼ in for safety copies." There is also a stereo Ampex *ATR-100* for dubs.

Dolby mainframes with both Dolby and dbx cards are in place for all of the tape machines, and Milsap has been using Dolby on his recent projects, running the tape at 30 in/s, but Harris is thinking of changing that. "Whenever I do outside sessions, I run 30 in/s with no noise reduction, and it makes level-matching much easier when I come back here. We're considering doing it here, too—it sounds a little more open that way." The tape choice is Agfa *PEM 468*, in all widths.

One of the great tourist attractions of Western Tennessee is the huge underground caverns that dot the area, but Ground Star seems to have more reverb than all of them put together. Besides a pair of EMT stereo plates, there are two digital units, a Lexicon *224* and an EMT *250*. There is also an AKG *BX-20* and a MasterRoom *Super C* which Harris admits isn't used very much. The auxiliary equipment racks are equally impressive: four *Kepex II*s, a Marshall *Time Modulator*, Eventide flanger and *910 Harmonizer*, Lexicon delay, Orban parametric equalisers and sibilance controllers, UREI *LA-4* and *1176* limiters as well as a *545* parametric EQ, a pair of Teletronix tube *LA-2A*s, ADR *Vocal Stressors*, dbx *160* limiters, and hidden under the console an EXR *Exciter*. Not to mention a UREI *VidiGraf* generator feeding a video screen between the monitors.

Said monitors are Sierra Audio designs with TAD tweeters and JBL *2235H* woofers. They are biamped with a McIntosh *2500* on the bottom and a *2200* on the top, through White *4320 passive* ⅓-octave equalisers. "We tried everything," says Harris. "UREI graphics, even Orban parametrics. The White active units sounded good, but when we found the passive ones, we stopped looking. They're cut-only devices, but because we're using them in only a couple of bands, the power loss is negligible—about 3 dB or so." Other monitors are around, including *MDM 4*s, JBL *4311*s, and *Auratones*, and a pair of time-aligned UREI *813*s lives in the studio room.

When Milsap purchased the studio he completely rebuilt it, with Valley Audio handling the control room design work, and Rudi Breuer in charge of construction. "Ronnie wanted a room that was super-analytical," says Harris, "but it came out *too* dry. It was just what he had specified, but it wasn't really what he wanted." So in the spring of 1980, Harris and Valley's Bob Todrank pulled out all of the sound-absorbing material from the wall surfaces and replaced it with hardwoods. They also built bass traps and slot absorbers into the back wall and ceiling. At the same time, the Neve was ordered to replace the Sphere *Eclipse* board then in use. "It was a beautiful board," recalls Harris, "with 40 inputs and Allison automation, but it didn't sound the way Ronnie wanted. It's a bright board, with really sharp transients—it sounded really good on rock. But the Neve suits Ronnie better, in that it has a warmer sound." The new console arrived that autumn.



The studio room has a very spacious feel, helped by a 22 ft ceiling, although at 32 × 34 ft, the floor plan is not exactly huge. High above the control room is a super-ambient string booth large enough for 14 musicians, with a balcony that a South American dictator would love. "It sounds like a concert hall," says Harris, "but some of the players don't like it much because when they're sitting, there's no line of sight with the floor. Even when I put a video monitor up, they say they feel like they're in prison." Harris also uses the area as a live chamber when mixing.

On the main floor is a five-sided drum booth with removable windows, and a square booth in which resides a set of Musser vibes. Between them is a room which contains all but the keyboard of a 9 ft Yamaha grand piano. The keyboard sticks out into the main room, and access to the rest of the instrument is through a crawlspace from the vibes booth. "We don't change the mike setup very often," laughs Harris.

The studio's mike collection is, as you might expect, comprehensive: Neumann *U47* FET, *U48*, *M49* and *U67* tubes, *U87*s, *KM84*s and *KM86*s, AKG *414*s, *451*s, *421*s and *C12*s, Sony *ECM50*s, Electro-Voice *RE-20*s, and 'everything Shure makes'. There are a dozen or so DI boxes, both active and passive.

Besides the acoustic grand, the studio owns a Yamaha electric grand, two stereo Rhodes', two Wurlitzer electrics, an *OB-X*, a Roland vocoder, ARP *Omni*, *PRO-DGX* and *2600* synths and assorted Fender and Ampeg guitar and bass amps. Headphones are fed with a set of Valley People boxes that each contain a stereo 4-channel mixer with full panning and an 8-watt per channel amplifier. "They're available to anyone," says Harris, "but we're the only studio we know of that has actually taken delivery."

Sitting on the piano, at the sides of the music rack, are a pair of *Auratones*. "Ronnie likes to play when he does vocal overdubs," Harris explains, "and these make it much easier for him. We run them out of phase, so there's very little leakage into his vocal mike. We started the design for the studio with the piano booth. Ronnie specified where he wanted everything else located so he could have a good working feel with the other players."

Except for the fact that the piano faces away from the control room, there is little to indicate that the owner and principal client of Ground Star Laboratory has been blind since birth. "Ronnie isn't one of those people who has to get led around by someone, or use a cane," says Harris. "He just charges right ahead, full bore. Even in live performance, he gets up and runs around the stage. The only consideration we have to make is that there's a clear path between the piano and the control room, or at least if I move a baffle or something in the way, I have to be sure and tell him."

"He's very technically oriented and has a phenomenal set of ears, and he is quite capable of running the console by himself. The Sphere was easier for him, in that it had the in-line graphic equalisers, while the Neve has circular knobs for EQ, but he still sometimes sits down and works the board."

The city of Nashville is something of a testament to those country artists who have turned artistic and financial success into trivial excess—everywhere you go there are solid gold Cadillacs, silly museums, clothing shops, souvenir stands or photo-developing stores, all with the names of famous country stars figuring prominently in the neon signs above them. So it's refreshing to come across one artist who is channelling his money back into the music business, and in large measure, to make better-sounding records. Even a blind man can see the value in that.

Ground Star Laboratory, 12 Music Circle South, Nashville, Tennessee 37203, USA. Phone: (615) 244-4861.

Paul D Lehrman

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product Effects guide

ACES (UK)
AC Electronic Services, Broad Oak, Albrighton, Shrewsbury, Shropshire SY4 3AG. Phone: 0939 290574.

ACSLR: low cost rack mounting mechanical spring line reverb system with in/output level controls, response and depth controls.

A/DA (USA)
Analog/Digital Associates, 2316 Fourth Street, Berkeley, California 94710. Phone: (415) 458-1311.

STD-1: stereo tapped analogue delay line producing six simultaneous delays. Delay time varies from 1.3 to 55.5ms and integral LFO will allow large number of time related effects with delays being asignable to the stereo outputs as desired.

ADVANCED AUDIO DESIGNS (USA)
Advanced Audio Designs, 3890 Steward Road, Eugene, Oregon 98402. Phone: (503) 485-4251.
UK: Turnkey, 8 East Barnet Road, New Barnet, Herts. EN4 8RW. Phone: 01-440 9221. Telex: 25769.

Model D-250: digital delay line, 0 to 250ms delay range in 1ms increments.

AKG (Austria)
AKG GmbH, Brunhildengasse 1, A-1150 Wien. Phone: 0222 92.16.47. Telex: 11839.
UK: AKG Acoustics Ltd, 191 The Vale, London W3 7QS. Phone: 01-749 2042. Telex: 28938.
USA: AKG Acoustics Inc, 77 Selleck Street, Stamford, Connecticut 06902. Phone: (203) 348-2121. Telex: 84451121.

BX5: stereo reverb unit, 1, 2, or 3s reverb decay time.
BX10: 2-channel mechanical reverb unit, 1.5, 2.5 or 3.5s reverb decay time.
BX15: 2-channel mechanical reverb unit, reverb time variable in 0.5s steps from 1.5 to 3.5s.
BX20: 2-channel mechanical reverb unit with remote control, 2 to 4.5s continuously variable reverb decay time.
BX22: 2-channel mechanical reverb unit, reverb decay time 2 to 4.5s continuously adjustable.
BX25: 2-channel mechanical reverb unit with remote control, 1.5 to 3.5s adjustable reverb decay time.
TDU 7000: modular digital time delay unit using 12+2 bit system. *M710* input, *M720* output, *M730* delay extension, *M740* remote control, *M750* effects modules and 8-bay *N700* mainframe unit. *M720* features 399ms max delay, *M730* extends delay by 200, 400, 600 or 800ms. *M750* provides time base modulation plus VCO facilities and adjustment of nominal delay time.

ALTEC (USA)
Altec Corp, 1515 South Manchester, Anaheim, California 92803. Phone: (714) 774-2900. Telex: 655415.
UK: Rank Strand Sound, PO Box 51, Great West Road, Brentford, Middlesex TW8 9HR. Phone: 01-568 9222. Telex: 27976.

Model 1640 Time Delay System: sound reinforcement delay unit for stage usage. Six 20ms delayed outputs, max delay 120ms. Units may be cascaded to a max of 600ms.
Model 1660/1661 Time Delay System: digital sound reinforcement delay unit for stage usage. Six delayed outputs continuously variable from 0 to 510ms. *Model 1661* tamper-proof version.

AMI (UK)
Third Floor (C71), Temple Buildings, Dale Street, Liverpool L2 5RL. Phone: 051-236 7542.

System 3700 Control Unit: multi-function rack mount unit containing pre-reverb delay section with up to 50ms delay with modulation, reverb section with RT60 2 to 3.5s MF and 5.5 to 7s HF. Unit also includes noise gate, limiter and octave equaliser.

AMS (UK)
Advanced Music Systems, 1 & 3 Wallstreams Lane, Worsthorne Village, Nr. Burnley, Lancs. Phone: 0282 36943. Telex: 63108.
USA: Quintek Distributors Inc, 4721 Laurel Canyon Blvd, Suite 209, North Hollywood, California 91607. Phone: (213) 980-5717. Telex: 194871.

DM 2-20: flanger/vibrato/delay generator. Features twin delay path and manual or automatic flanging. Max delay 20ms, 80ms with optional memory module.
DM 2-28: as *DM 2-20* but with 80ms max delay.
DMX 15-80: modular programmable 15-bit DDL/pitch-shifter. 18kHz delay bandwidth. Accepts 102 or 400ms delay cards, 16s max delay.
DMX 15-80S: stereo version of *DMX 15-80* Facility to incorporate two pitch change modules for multiple pitch-shifting.
DMX 15-80SB: broadcast stereo version of the *DMX 15-80*, max delay 2s per channel.
Digital Loop Editing System: loop editing system for the *DMX 15-80 Series*.
DM-DDS: stereo disc mastering DDL, max delay 1.6s expandable 10s. Two bandwidth versions available.
DMX 15R: programmable digital reverb system for interface with any of the *DMX Series* DDLs.
RMX 16: digital reverb system stand alone version of the *DMX 15R* offering the same number of programs and facilities for user modification of programs. LED display for program number, program name and other details.

APE (France)
SCEPA, 23 bis rue Emile Duclaux, 92150 Suresnes. Phone: 506.38.78.

PSH 6T: phase-shifter to fit the APE rack system with modulation by internal oscillator, envelope follower or manual control.
DDL 12T: modular digital delay line to fit APE rack system.

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

Aural Exciter 602B: phase-shift and delay unit (broadcast model and leasing available).
Aphex II: studio *Aural Exciter*, phase-shift and delay unit (broadcast model available).

AUDICON (USA)
Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900. Telex: 554494.
UK: Trad Electronic Sales Ltd, 149b St Albans Road, Watford, Herts WD2 5BB. Phone: 0923 47988. Telex: 262741.

The Plate II Reverb System: second generation plate reverb system with remote control unit. Adjustable reverb time 1 to 4s.

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

S24 Time Shape Module: ADT/flanger and time domain processor from *Scamp* range. 1.2 to 45ms delay range.

AUDI-ENCE (USA)
Audi-Ence Inc, 3325 Vista Oaks, Garland, Texas 75043. Phone: (214) 226-2189.

RFS-1: stereo plate reverb system with reverb time of 4s for nominal octave centred around 500Hz. Adjustable HF and LF response.

AUDIO ENVELOPE SYSTEMS (USA)
Audio Envelope Systems Inc, 2109 West Campbell Avenue, Phoenix, Arizona 85015. Phone: (602) 279-3613.

axrac: modular rack mount signal processing system comprising 8-position powered frame (which will also power a slave frame) and a range of modules including stereo synthesiser, analogue delay line, autopanner, as well as other modules such as preamp, equaliser, crossover etc.

BANDIVE (UK)
Bandive Ltd, 8 East Barnet Road, New Barnet, Herts EN4 8RW. Phone: 01-440 9221. Telex: 25769.
USA: Omnisound Ltd, PO Box 366, Elmont, New York 11003. Phone: (516) 437-7947.

Accessit Compact Chamber: miniature spring reverb, 3.5s decay time.
Great British Spring: tubular spring reverb using custom dual element spring and 3.5s decay time.
Accessit Reverb Processor: expansion device for reducing the decay time of a reverb unit by up to 60%.

BARTH (West Germany)
R. Barth KG, Grillparzerstrasse 6a, D-2000 Hamburg 76. Phone: 040 229 8883. Telex: 0212095.
USA: Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900. Telex: 554494.

Audios: sound storage memory, transposer and time delay unit. Delay section 0.3 to 100ms delay per channel, expandable to 410ms.

BEIGEL (USA)
Beigel Sound Lab, 24 Main Street, Warwick, New York 10990. Phone: (914) 986-1699.

Envelope Control Filter: unit combining a parametric envelope follower, a parametric voltage controlled filter, an external effects loop and envelope control voltage output.

BEL (UK)
UK: Don Larking Audio Sales, 29 Guildford Street, Luton, Beds LU1 2NQ. Phone: 0582 450066.
USA: Omnisound Ltd, PO Box 366, Elmont, New York 11003. Phone: (516) 437-7925.

BF20 Mk2 Stereo Flanger: switchable mono. Delay variable 0.25 to 10ms.
BA40 Delay Line/Flanger: mono analogue delay line/flanger with pseudo stereo outputs. Delay 0.5 to 20ms (flange mode) 4 to 160ms (delay mode).

BIAMP (USA)
Biamp Systems Inc, 9600 SW Barnes Road, Portland, Oregon 97225. Phone: (503) 297-1555.

SR/240: stereo reverb system. 2.5s decay time.

CASTLE (USA)
Castle Instruments, 2 Carteret Court, Madison, New Jersey 07940. Phone: (201) 377-8185.

Phaser III: comprehensive phase-shifter with controls for rate, width, centre, emphasis, blend and switching for the number of notch filters used.
Dual Phaser III: 2-channel version of the above but with the addition of facilities to cross patch between the two channels.

DATABASE (UK)
Database, 1 Vale View Place, Claremont Road, Bath BA1 6QW. Phone: 0225 316102.

Ruby: noise reduction unit for stage effects.

DATATON (Swedon)
Dataton AB, PO Box 257, S-58102 Linkoping. Phone: 013 10.07.11.

System 3000: series of 13 modules including–microprocessor-based 'program' sequencer'; *3002* VC sound generators: *3103* 4-channel filter; *3104* 4-channel envelope shaper; *3107* quad equaliser/preamp; *3203* 2/4-channel joystick module; *3205* 4-channel mixer module; and *3314* quad signal analyser.

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

Model 906 Flanger: from *900 Series* of modules. Delay range 20 to 100ms flange mode, 4 to 40ms delay mode.

DELTALAB (USA)
Deltalab Research Inc, 27 Industrial Avenue, Chelmsford, Massachusetts 01824. Phone: (617) 256-9034.
UK: Scenic Sounds Equipment, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

DL1: DDL with three independent outputs, 5 to 160ms delay.
DL2 Acousticomputer: DDL/effects processor. 16 reverb programmes, 0.25 to 240ms delay.
DL3: DDL, 0 to 120ms delay.
DL4 Time Line: DDL/effects processor, 1 to 512ms delay range.
Memory Module: rack-mounting memory for *DL2* and *DL4*, expanding delay range to 2.5s.
DL5 Harmonicomputer: digital effects/pitch transposing unit. May be used with an external delay unit such as the DeltaLab *DL4*.
DL6: Harmonicontroller: unit for control of the *DL5* and shifts the pitch diatonically giving proper harmony.
Effectron ADM256: digital delay processor with up

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The most advanced studio/broadcast master recorder

OTARI
MTR-10

OTARI

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

Bridging the gap between the new technology and old reliability, it's the logical extension of the innovative technology built into our multichannel MTR-90. The new MTR-10 gives the professional unprecedented control:

- Measurable and audibly discernible performance improvements. More than chips and buzz words, a balanced engineering approach: adjustable phase compensation with internal square/sine wave generator; electronically balanced I/O with direct coupled outputs.
- Full servo, D.C.PLL transport governed by an on-board microprocessor — an industry first.
- Unmatched production features — exclusive multiple edit modes, reverse play, standard alignment level presets, and dual-mode varispeed. Other features nclude controlled wind, preset master bias switching, three speeds and IEC, AES and NAB selectable. Also it includes return-to-zero and offers an optional tape locator w th ten position memory and tape shuttle.
- Comprehensive servicing and support. Fully modular power supply, audio and transport electronics card frame. A highly dedicated distributor service network with factory trained and supported technicians.
- Available in full-track 1/4", 2-track 1/4" and 4-track 1/2". Comes with optional DIN-head and overbridge versions.

product Effects guide

to 256ms of delay and facilities for flanging, chorus, doubling and echo effects.
Effectron ADM1024: identical to *ADM256* but with 1024ms of delay time.
Effectron ADM64: similar to *ADM256* with up to 64ms of delay.

DOD (USA)

DOD Electronics Corporation, 2953 South 300 West, Salt Lake City, Utah 84115. Phone: (801) 485-8534.
UK:Strings & Things Ltd, Unit 2, Chapel Road, Portslade, Brighton, Sussex BN4 1PF. Phone: 0273 420704.

R-870: analogue flanger/doubler with stereo output and a maximum delay time of 75ms in two ranges with internal LFO.

DRAWMER (UK)

Drawmer Electronics, 183 Cobden View Road, Sheffield S10 1HT. Phone: 0742 668520.
UK: Studio Equipment Services Ltd, Braemar Cottages, 6 Manor Road, Teddington. Middlesex TW11 8BG. Phone: 01-943 1368.

DMT 1080 Multi-tracker: analogue delay line based system able to create various stereo effects from a mono input. Variable delay range from 0.3 to 80ms.

DYNACORD (West Germany)

Dynacord Electronic GmbH, Siemenstrasse 41 – 43, D-8440 Strubing. Phone: 09421 3103.
USA: Dynacord Electronics Inc, PO Box 26038, Philadelphia, 19128. Phone: (215) 482-4992.

DRS78: digital echo/reverb unit. Delay range 0 to 320ms.
TAM19: time axis manipulation device.
TAM21: 2-channel analogue phaser/flanger unit with remote control facility.
SRS56: stereo echo/reverb system. Delay range 30 to 560ms, reverb time 30ms to 20s.

EMT (West Germany)

EMT-Franz GmbH, Postfach 1520, D-7630 Lahr. Phone: 07825 512. Telex: 754319.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts, WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Gotham Audio Corp, 741 Washington Street, New York, NY 10014. Phone: (212) 741-7411. Telex: 129269.

EMT 140TS: stereo reverb plate with adjustable 1 to 4s reverb period. Optional remote control.
EMT 140Q: quadraphonic version of *EMT 140TS*.
EMT 240: stereo reverb plate, adjustable 1 to 4s reverb period.
EMT 245: digital reverb decay range 0.4 to 4.5s in 16 steps (frequency dependent).
EMT 251: digital reverb. Four programmable outputs, decay time variable 0.2 to 4.5s dependent on frequency. Initial delay 0 to 80ms or 40 to 120ms.
EMT 444: digital delay unit, delay time 1 to 255ms in 1ms steps.
EMT 446: digital signature tune repetitor available in three versions with either 5s, 6s or 12s store.

EVENTIDE (USA)

Eventide Clockworks Inc, 265 W 54th Street, New York, NY 10019. Phone: (212) 581-9290.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.

Model 1745M Digital Delay: delay time 0 to 320ms continuously variable.
Model H910 Harmonizer: delay time 0.3 to 60ms pitch-shift mode, 0 to 112ms in 7.5ms steps delay mode.
Model 2830 Omnipressor: dynamic modifier combining compressor/expander/noise gate/limiter characteristics.
Model FL201 Instant Flanger: delay time 200µs to 10ms, 50ms max. May be altered to an *Instant Phaser* by inserting the *BPC 101* phaser card.
BD955: broadcast delay line with max 6.4s delay. Versions available with 1.6 and 3.2s delay.
H949 Harmonizer: delay time 0 to 300ms in 50ms steps pitch change mode, 0 to 393.75ms in 6.25ms steps delay mode.
Model JJ193: DDL, delay time 0 to 510ms in 2ms steps. Optional 1.022 and 2.046s.

Model CD254: DDL, delay time 0 to 254ms in 2ms steps. Two outputs.
HM80 Harmonizer: delay range 0 to 270ms.
Model SP2016: digital reverb/programmable effects processor. Non-volatile memory stores up to 32 programs. Delay range 0 to 1.6s in 25µs steps (16kHz bandwidth) or 0 to 3.2s in 50µs steps (8kHz bandwidth).

EXR (USA)

EXR Corp, 11523 Dexter-Pinckney Road, Pinckney, Michigan 48169. Phone: (313) 878-9445.
UK: Turnkey, 8 East Barnet Road, New Barnet, Herts. EN4 8RW. Phone: 01-440 9221. Telex: 25769.

EXR Exciter Model EXIII: signal clarification and boosting unit.
EXR Exciter Model SP-1: 2-channel unit for small studios and live usage.
EXR Exciter Model IV: features include continuously variable notch positioning, dual level operation, adjustable process noise gating and adjustable process limiter.

FOSTEX (Japan)

Fostex Corporation, 512 Miyazawacho, Akishima, Tokyo 196. Phone: 0425 456111. Telex: 2842203.
UK: Bandive Ltd, 8 East Barnet Road, New Barnet, Herts EN4 8RW. Phone: 01-440 9221. Telex: 25769.
USA: Fostex Corporation of America, 15431 Blackburn Avenue, Norwalk, California 90650. Phone: (213) 921-1112.

3050: single channel delay line with a maximum of 270ms delay time with integral LFO.

FURMAN (USA)

Furman Sound Inc, 616 Canal Street, San Rafael, California 94901. Phone: (415) 456-6766.
UK: Atlantex Music Ltd, 1 Wallace Way, Hitchin, Herts. SG5 0SE. Phone: 0462 31511. Telex: 826967.

RV-1: mechanical reverb system. Decay time 1.8s with 30 to 40ms initial delay.

GELF (UK)

Gelf Electronics Ltd, Unit 5 Mount Avenue, Bletchley, Milton Keynes MK1 1LS. Phone: 0908 77503/64762.

Auto Phasing Unit GP14: voltage-controlled phasing unit, decay range 30ms to 3s.

HH ELECTRONIC (UK)

HH Electronic, Viking Way, Bar Hill, Cambridge CB3 8EL. Phone: 0954 81140. Telex: 817585.

Digital Multi Echo Unit: mechanical reverb, analogue electronic CCD delay. Delay range 21.5 to 312ms.
Digital Echo Unit: analogue electronic CCD delay. Delay range 34 to 208ms plus repeat effects.

INDUSTRIAL RESEARCH PRODUCTS (USA)

Industrial Research Products Inc, 321 Bond Street, Elk Grove Village, Illinois 60007. Phone: (312) 439-3600.
UK: Knowles Electronics Ltd, Victoria Road, Burgess Hill, Sussex RH15 9LP. Phone: 04446 5432. Telex: 87460.

DA-4006, DA-4007 Audio Signal Delay: electronic digital, rack-mounting delay units.
DA-4008 Audio Program Delay: electronic digital delay with 240ms max delay.
DC-4011 Audio Program Delay: digital CCD delay. Delay range 96ms in 4ms steps.
DD-4012 Sound Delay Module: digital CCD delay. Delay range 64ms in 4ms steps. Optional 96, 128 and 160ms delay range.

IVIE (USA)

Ivie Electronics Inc, 500 West 1200 South, Orem, Utah 84057. Phone: (801) 224-1800. Telex: 910-971 5884.
UK: FWO Bauch Ltd, 49 Theobald Road, Boreham Wood, Herts WD6 4RZ. Phone: 01-953 0091. Telex 27502.

5404: modular digital delay, part of the *5000* system. Maximum delay is 127ms adjustable in 1ms steps.

KLARK-TEKNIK (UK)

Klark-Teknik Research Ltd, Walter Nash Road West, Coppice Trading Estate, Kidderminster, Worcs. DY11 7HS. Phone: 0562 741515. Telex: 339821.
UK: Autograph Sales Ltd, Stable 11, British Rail Camden Depot, Chalk Farm Road, London, NW1 8AH. Phone: 01-267 6677.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, NY 11735. Phone: (516) 249-3600.

DN34: analogue time processor, delay time 0.38 to 53ms continuously variable.
DN36: analogue time processor, delay time 0.5 to 50ms continuously variable.
DN70: digital time processor. Combines three separate digital delay times, delay range 163ms, or optionally 326 or 652ms.
DN71: add-on control unit for *DN70*, offering delay setting in 20ms steps, plus pitch control, time sweep effects and freeze control.
DN72: add-on memory bank for *DN70*, giving 18 pre-selected non-volatile memories.

LAWSON (USA)

Lawson Inc, 842 Reeves Road, Antioch, Tennessee 27013. Phone: (615) 834-8614.

LP-1 Plate: plate with adjustable 1 to 4s reverb time.

LEXICON (USA)

Lexicon Inc, 60 Turner Street, Waltham, Massachusetts 02154. Phone: (617) 891-6790.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts. WD6 4RZ. Phone: 01-953 0091. Telex; 27502.

Model 91, Model 92 Digital Delays: single-channel DDLs. *Model 92* has two outputs. Delay time 0 to 120ms in 7.5ms steps.
Varispeech Model 27: real time pitch-shifting unit.
Model 93 Prime Time: digital delay/processor/ mixer. Delay capacity 0 to 128ms, with add on *Delay Module Memory* 256ms.
PCM 41: digital delay processor. Delay time 0 to 400ms or 800ms with 6kHz bandwidth.
Model 122 Digital Delay: DDL with mono or stereo versions. Delay time 40 to 320ms in 5ms steps (mono), 40 to 160ms in 2.5ms steps (stereo).
Model 1200 Audio Time Compressor: allows recorded material to be played back at different speeds without pitch change.
Model 97 Super Prime time: developed from the *Prime Time* and offers storage of complete panel settings which may be dumped to tape. There are also eight presets. Full bandwidth response on complete range of delay based effects. Maximum delay time of 1.92s.

LEXICON MODEL 224:
UK: Scenic Sounds Equipment. 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Model 224 Reverberation System: electronic digital reverb system with interchangeable programs. Pre-delay up to 256ms, decays from 600ms to 70s.
Model 224X: further development of the *224* offering full 15 kHz bandwidth variable bandwidth control, dynamic decay facility, system of pages to allow sliders to be assigned to control differing parameters, non volatile register storage and ROM storage for 36 user presets and 32 basic programs.

LOFT (USA)

Phoenix Audio Laboratory Inc, 91 Elm Street, Manchester, Connecticut 06040. Phone: (203) 646-7806.

Model 440 Analogue Delay Line/Flanger: wide variety of effects with noise reduction. Delay time 0.5 to 150ms in four ranges.
Model 450 Analogue Delay Line/Flanger: with maximum delay in standard form of 160ms and up to 320ms with *EM-450* extender module. 18kHz bandwidth.

MARSHALL (USA)

Marshall Electronic, 1205 York Road, Suite 14, Lutherville, Maryland 21093. Phone: (301) 484-2220.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.

Minimodulator: digitally programmed time modulator.
Model 5402 Time Modulator: development of the *Model 5002* time domain modifier. Delay continuously variable up to 400ms. Two delay lines with 3 taps and 72.1 sweep range.

MICMIX (USA)

MicMix Audio Products Inc, 2995 Ladybird Lane, Dallas, Texas 75220. Phone: (214) 352-3811.
UK: Scenic Sounds Equipment, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Dynaflanger 265: versatile amplitude/frequency enabled/sweep flanger.
Master-Room XL-305: stereo rack-mounting mechanical reverb system, decay time 3.5s.

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

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

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

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

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

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

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

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

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

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

PZM FROM HHB

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

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

product Effects guide

Master-Room XL-404: also known as the *Plate Synthesiser* and consists of two channels of mechanical reverb with variable decay control, 4-band EQ, LED metering and mix control.
Master-Room XL-210: mechanical reverb system, stereo/mono switchable. Decay time 3s.
Master-Room XL-121: reverb system using custom spring elements, single channel with a decay time of 2.5 at 1kHz.
Master-Room XL-515: stereo mechanical reverberation system with three operational modes to give live chamber, plate and concert hall. Control parameters include decay 1 to 6s with digital display of time, high low, and two mid-range sweep controls.

MXR (USA)
MXR Innovations Inc, 740 Driving Park Avenue, Rochester, NY 14613. Phone: (716) 254-2910. Telex: 978451.
UK: Atlantex Music Ltd, 1 Wallace Way, Hitchin, Herts. SG4 0SE. Phone: 0462 31511. Telex: 826967.

Model 151 Delay System II: digital time delay unit with effects processing. Delay capacity 1.6s expandable to 3.2s.
Flanger/Doubler: signal processing unit providing variety of time delay effects. Delay range 17.5 to 70ms (doubling), 0.25 to 5ms (flanging).
Pitch Transposer: real time pitch-shifter.
Model 174: compact pitch-shift doubler for chorus, ADT, flanging and pitch shifting.
Model 175: digital delay line with 0.31ms to 328ms range.

NEPTUNE (USA)
Neptune Electronics Inc, 934 NE 25th Avenue, Portland, Oregon 97232. Phone: (503) 232-4445.
UK: Theatre Projects Services Ltd, 11 Marshalsea Road, London SE1 1EP. Phone: 01-403 3838. Telex: 885659.

Model 351: mechanical spring line reverb unit with five bands of EQ.

NEUTRIK (Liechtenstein)
Neutrik Ag, Oberglass 16, FL-9494 Schaan. Phone; 075 2.63.83. Telex: 77771.
UK: Eardley Electronics Ltd, Eardley House, 182–184 Campden Hill, London W8 7AS. Phone: 01-221 0606. Telex: 299574.
USA: Philips Audio Video Systems Corp, 91 McKee drive, Mahwah, New Jersey 07430. Phone: (201) 529-3800.

AD4: analogue delay line. Four separate adjustable outputs, delay time 12.5 to 50ms, 25 to 100ms, 37.5 to 150ms and 50 to 200ms.

ORBAN (USA)
Orban Associates Inc, 645 Bryant Street, San Francisco, California 94107. Phone: (415) 957-1063. Telex: 171480.
UK: Scenic Sounds Equipment, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

111B Reverb: 2-channel spring reverb unit. Decay time 2s, delay time 30ms between direct/reverb.
245E Stereo Synthesiser: generation of simulated stereo from mono sources.

PSE (UK)
Production Studio Equipment Ltd, 72–74 Eversholt Street, London NW1. Phone: 01-388 5392.
Worldwide distribution: Connectronics Ltd, 20 Victoria Road, New Barnet, Herts EN4 9PF, UK. Phone: 01-449 3663/4044. Telex: 8955127.

Studio Reverb: mono inpu· dual output mechanical spring reverb. Multiple 33, 37 and 41ms delay, decay time 2.5 to 3s.

PUBLISON (France)
Publison Audio, 5–11 Rue Crespin du Gast, F-75011 Paris. Phone: (1) 375.64.07.
UK: Scenic Sounds Equipment Ltd, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

DHM89B2: stereo, digital manipulation unit of memorised sounds. Dual digital delay, max 1.2s.
DHM83B: similar to *DHM89B2* but lower delay 600ms max, quasi stereo delay.
Fullmost: relief enhancer to increase brightness of music or speech.

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

System 5: digital programmable electronic reverb system. Reverb time 0.5 to 6s in 10 steps. Various programs, 16 EQ settings, memory facility to tape or disc.

QUANTEC (West Germany)
Quantec GmbH, Postfach 152, D-8016 Feldkirchen bei Munchen. Phone: 089 903.67.25.
UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-723 3844.

Room-Simulator (QRS): rack mounted digital reverb system. User can 'dial up' actual room volumes in cubic metres as well as adjusting parameters such as reverb time, pre-delay, equalisation and first reflection delay. Stereo input and four outputs.

QUANTUM (USA)
Quantum Audio Labs Inc, 200 Park Avenue South, New York, NY 10003. Phone: (212) 260-2300.

QA-201 Reverb: stereo chamber using two *Accutronics* reverb units, each channel having its own input level and HF controls.

REBIS (UK)
Rebis Audio, Kinver Street, Stourbridge, West Midlands DY8 6A. Phone: 0384 71865.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, NY 11735. Phone: (516) 249-3600.

RA200 Series Delay System: delay system using two *RA205* ADT/delay modules, *RA208* modulator and *RA209* mixer module. *RA208* can control *RA205* and *RA209* to create various effects. *RA205* has internal feedback and mix controls, variable delay time in two ranges ·2 to 40ms and 4 to 80ms.

ROLAND (Japan)
UK: Roland (UK) Ltd, Unit 6, Great West Trading Estate, 938 Great West Road, Brentford Middlesex TW8 9DN. Phone: 01-568 4578. Telex: 888941.
USA: Roland Corp US, 2401 Saybrook Avenue, Los Angeles, California 90040. Phone: (213) 685-5141.

Roland Rack System: includes *SRE-555* echo unit, *SDD-320* dimension D chorus effect, *SBF-325* stereo flanger.
SDE-2000: single channel digital delay with up to 640ms in two ranges, full modulation controls with variable waveform, delay time readout, several remotable functions and feedback loop insert.

SEQUENTIAL CIRCUITS (USA)
Sequential Circuits Inc, 3051 North First Street, San Jose, California 95134. Phone: (408) 846-5240.
Europe: Sequential Circuits, PO Box 16, 3640 AA, Mijdrecht, Netherlands.

Model 500 PRO-FX: rack system with the basic mainframe containing a system controller and 6-module space. Three expansion frames can be added giving a total of 30 effects. The system controller allows programming of all operating modes of the *PRO-FX* and up to 64 programs may be stored as well as a tape program storage dump and retrieve facility. The *500 series* modules at present contain phase-shifter, distortion, reverb, frequency divider/sync, as well as parametric EQ and 4×2 mixer.

SONY (Japan)
UK: Sony UK Ltd, Pyrene House, Sunbury-on-Thames, Middlesex TW16 7AT. Phone: 09327 89581/876441. Telex: 266371.
USA: Sony Corporation of America, 9 W 57th Street, New York, NY 10019. Phone: (212) 371-5800. Telex: 424595,

DRE-2000: 2-channel programmable digital reverb unit with digital and analogue I/O capability. Four modes of pre-programmed reveb can be selected with up to 50 programmable stores.

SOUND WORKSHOP (USA)
Sound Workshop Professional Audio Products Inc, 1324 Motor Parkway, Hauppauge, NY 11787. Phone: (516) 582-6210. Telex: 649230.

262 Stereo Reverberation System: mechanical stereo reverb system with extended LF and HF response, two channels of EQ with ±15dB sweepable from 50Hz to 1kHz and 500Hz to 10kHz.

STOCKTRONICS (Sweden)
Stocktronics Electronik, Grevgatan 49, S-11458, Stockholm. Phone; 08.60.01.11.

RX4000 Reverberation Plate: portable stereo reverb plate with decay time of 4s at 500Hz (option for 2s).

STRAMP (West Germany)
Peter Struven GmbH, Bornheide 19, D-2000 Hamburg 53. Phone: 040 801028.

Echo-700: combined stereo echo (plus reverb), phaser and vibrato unit. Delay and effects sections can be linked to create other special effects.

STUDIO TECHNOLOGIES (USA)
Studio Technologies, 6666 North Lincoln Avenue, Lincolnwood, Illinois 60645. Phone: (312) 676-9400.
UK: Turnkey, 8 East Barnet Road, New Barnet, Herts EN4 8RW. Phone: 01-440 9221. Telex: 25769.

Ecoplate: reverb plate with adjustable reverb time from 1 to 7s, horizontal or vertical mounting.
Ecoplate II: smaller version *Ecoplate* with reverb time from 1 to 6s.

SYMETRIX (USA)
Symetrix Inc, 109 Bell Street, Seattle, Washington 98121. Phone: (206) 682-3076.

Phase filter: phaser using frequency notching techniques.

SYNTON (Netherlands)
Synton Electronics BV, Zandpad 46, Postbus 83, NL-3620 Breukelen. Phone: 034.62.34.99.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.
USA: Big Briar Inc, Leicester, North Carolina 28748. Phone: (704) 683-9085.

Phaser 203: analogue phase-shifter.

TAPCO (USA)
EV-Tapco, 3810 148th Avenue NE, Redmond, Washington 98052. Phone: (206) 883-3510. Telex: 910-449 2594.
UK: Shuttlesound Ltd, 200 New Kings Road, London SW6 Phone: 01-736 0907.

4400A: 2-channel spring line reverb system with graphic EQ and 2.1s decay time.

TECNICOBEL(France)
Tecnicobel, 8 rue de la Croix-Matres, BP 26, F-91122 Palaiseau Cedex, Paris. Phone: (1) 920.80.39. Telex: 6925423.

CRA60 echo chamber: rack-mounting mechanical reverb system with adjustable reverb time. Features stereo operation, remote control of reverb and EQ adjustable initial delay 0 to 33ms.

TIME TUNNEL (USA)
Wang Voice Communications Inc, Executive Plaza, Hudson, New Hampshire 03052. Phone: (617) 459-5000.

Model 150: broadcast DDL with 6s fixed delay.
Model TDG-1: modular digital delay system converted to stereo by adding modules.

UREI (USA)
United Recording Electronics Industries, 8460 San Fernando Road, Sun Valley, California 91352. Phone: (213) 767-1000. Telex: 651389.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts. WD6 4RZ. Phone: 01-953 0091. Telex: 27502.

Model 927: DDL with delay capacity of 0 to 127ms, four outputs individually adjustable in 1ms steps.

URSA MAJOR (USA)
Ursa Major Inc, Box 18, Belmont, Massachusetts 02178. Phone: (617) 489-0303. Telex: 921405.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.

SST-282 Space Station: digital electronic processor for reverb, multitap delay line, feedback delay and echo, 16 programs of eight delay tap times, taps can be mixed with direct sound and reverb/echo added. Delay capacity of 255ms, 0 to 2.5s decay time.
8X32 Digital Reverb: programmable digital reverb unit featuring HF and LF decay, pushbutton control of decay time, LEDs show settings in use with readout of decay time and peak signal level, up to 19.9s decay time.

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Super Prime Time lets you put your personal stamp on audio effects.

The Super Prime Time is the first micro-processor-based special effects device and audio processor which enables you to create, store and recall effects for any piece of music live or in the studio.

Eight conventional sound effects are built-in (flanging, tripling etc), and there are 32 additional memories that can be used to modify, create and store your own personal effects.

There is up to 1.92 seconds of full bandwidth delay (from 20Hz to 20kHz), which increases versatility.

Each effect can be recalled at any time through the panel controls or at the touch of a footswitch.

Complete effects programs can be stored on tape, and re-loaded on to any Super Prime Time processor when needed.

Super Prime Time. The effects are out of this world. For full details, contact F.W.O. Bauch Limited.

lexicon

Lexicon Inc, 60 Turner Street, Waltham, MA 02154

F.W.O. Bauch Limited

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

product Effects guide

WMS (USA)
Wasatch Music Systems, 805 East 3300 South, No 4, Salt Lake City, Utah 84106. Phone: (801) 467-4722.

900-A: digital delay producing flanging, doppler, vibrato and chorus, pitch shifting, ADT, 'Leslie' and cardboard tube echo. Variable delay to 20ms max.

YAMAHA (Japan)
UK: Yamaha Musical Instruments, Mount Avenue, Bletchley, Milton Keynes, MK1 1JE. Phone: 0908 71771.
USA: Yamaha International Corp, PO Box 6600, Buena Park, California 90620. Phone: (714) 522-9105.

Model E1010: analogue delay line with bass and treble EQ, feedback and mixing controls with frequency/depth modulation controls, 10 to 300ms delay. ■

product Equalisers guide

ACCESSIT (UK)
Bandive Ltd, 8 East Barnet Road, New Barnet, Hertfordshire EN4 8RW. Phone: 01-440 9221. Telex: 25769.
USA: Omnisound Ltd, PO Box 366, Elmont, New York 11003. Phone: (516) 437-7947.

Parametric equaliser: free standing, single channel 2-band unit ±12dB, 50 to 300Hz and 3 to 14kHz.

ACES (UK)
AC Electronic Services, Broad Oak, Albrighton, Shrewsbury, Shropshire SY4 3AG. Phone: 0939 290574.

ACGE/10SJ: 2-channel, 10-band graphic equaliser 25Hz to 12.8kHz.
ACGE/15SJ: 2-channel, 15-band graphic equaliser 25Hz to 17.3kHz.
ACGE/27MJ: single channel 27-band graphic equaliser 40Hz to 16kHz.

ADC (USA)
Audio Dynamics Corp, Pickett District Road, New Milford, Connecticut 06776. Phone: (203) 355-2671.
UK: BSR Ltd, Powke Lane, Cradley Heath, Warley, West Midlands B64 5QH. Phone: 0384 65191.

SS-2 Mark II: 2-channel 12-band graphic equaliser with ±12dB range.
SS-3: 2-channel 12-band graphic equaliser with each band centre frequency switchable to one of three frequencies covering 26Hz to 21kHz. ±12dB range.

AEC (West Germany)
Audio International Vertribs GmbH, Box 560229, Gonzenenheimestrasse 2B, D-6000 Frankfurt 56. Phone: 0611 504733. Telex: 413039.

C-41: 2-channel graphic equaliser with 10-bands per channel from 31.5Hz to 16kHz.

ALTEC (USA)
Altec Lansing Corp, 1515 South Manchester, Anaheim, California 92803. Phone: (714) 774-2900. Telex: 655415.
UK: Rank Strand Sound, PO Box 51, Great West Road, Brentford, Middlesex TW8 9HR. Phone: 01-568 9222. Telex: 27976.

1650: single channel graphic equaliser with 15dB attenuation in 28, ⅓-octave bands between 31.5Hz and 16kHz.

APE (France)
SCEPA, 23 Bis rue Emile Duclaux, 92150 Suresnes. Phone: 506.38.78.

SWEQ 12T: single channel modular 4-band parametric equaliser covering 20Hz to 20kHz with ±20dB control range and variable Q.
PE4 6T: 4-band single channel parametric equaliser in modular format with range from 20Hz to 20kHz ±15dB.
GE9 6T: modular single channel 9-band equaliser covering 60Hz to 16kHz ±12dB.

APSI (USA)
Audio Processing Systems Inc, 40 Landsdown Street, Cambridge, Massachusetts 02139. Phone: (617) 354-1144.
UK: Scenic Sounds Equipment, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Model 5590: single channel EQ module with 9, ⅓-octave bands 35Hz to 16kHz up to +15dB, −12dB.
Model 5620: 4-band parametric EQ module.

ASHLY (USA)
Ashly Audio Inc, 100 Fernwood Avenue, Rochester, New York 14621. Phone: (716) 544-5191.
UK: Atlantex Music Ltd, 1 Wallace Way, Hitchin, Herts SG4 0SE. Phone: 0462 31511. Telex: 826967.

SC63: 3-band single channel parametric equaliser ±15dB over maximum range 16Hz to 24kHz.
SC66A: similar to *SC63* but has two channels and an additional equalisation band.
SC-68: parametric notch filter with eight filters covering 16Hz to 24kHz, 30dB attenuation.

AUDIOARTS (USA)
Audioarts Engineering, 286 Downs Road, Bethany. Connecticut 06525. Phone: (203) 393-0887.

4200A: stereo parametric equaliser with four bands covering 22Hz to 21kHz ±32dB.
4100: similar to *4200A* but mono with instrument preamp.

AUDIO CONTROL (USA)
Audio Control, 6520 212th SW, Lynwood, Washington 98036. Phone: (206) 775-8461.

D10: 2-channel graphic equaliser with slider controls ganged, 10 single octave bands 31.5Hz to 16kHz ±12dB.
D11: 2-channel graphic with same facilities as *D10* but with addition of analyser section.
C22: 2-channel graphic equaliser with 10 bands 32Hz to 15.5kHz, ±15dB control range.
C101: 2-channel graphic equaliser with same facilities as *C22* but with analyser section and internal pink noise generator.
Richter Scale: 2-channel graphic equaliser covering low frequencies only from 31.5Hz to 125Hz including analyser section and other test features.

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

SO3: single channel parametric equaliser module from the *Scamp* range with three bands covering 20Hz to 20kHz, ±20dB.
SO4: single channel parametric/shelving equaliser from *Scamp* range with bands and control range as *SO3* but with variable Q.
SO7: single channel system or room equaliser module in *Scamp* range with 10 single octave bands between 31.35Hz and 16kHz ±12dB.
E900RS: stereo parametric equaliser with four bands covering 40Hz to 16kHz ±20dB, fixed Q.
E500/560: 2-channel 'dynamic' equaliser with high and low pass filters over 100Hz to 10kHz, parametric notch filter over 20Hz to 20kHz, input and output limiters/expanders. Has the ability to alter dynamic range over selected sections of the frequency spectrum.
E950: single channel (ganged 12-band) or 2-channel (6-band) *Paragraphic* equaliser covering 31.25Hz to 16kHz with variable centre frequencies, Q and a control range of ±28dB.

AUDIO DEVELOPMENTS (UK)
Audio Developments, Hall Lane, Walsall Wood, West Midlands WS9 9AU. Phone: 0543 375351. Telex: 338212.
USA: Audio Developments, 1640 Fifth Street, Suite 224, Santa Monica, California 90401.

AD070 Prographic: programmable graphic equaliser with 15, ½-octave bands, ±14dB range and 16 stores.

AUDIX (UK)
Audix Ltd, Station Road, Wendon, Saffron Walden, Essex CB11 4LG. Phone: 0799 40888. Telex: 817444.

Model 902: single channel graphic equaliser with 11 bands, 45Hz to 14kHz ±12dB.
Model 908: single-channel graphic equaliser with 27, ⅓-octave bands ±12dB 45Hz to 16kHz.

BANDRIDGE (UK)
Bandridge Ltd, 1 York Road, London SW19. Phone: 01-543 3633.

FE5: 2-channel 5-band graphic equaliser with control range of ±12dB over 60Hz to 10kHz.

BARTH (West Germany)
R. Barth KG, Grillparzerstrasse 6a, D-2000, Hamburg 76. Phone: 040 229 8883. Telex: 0212095.
USA: Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900. Telex: 554494.

W308: single channel parametric equaliser with three bands covering 40Hz to 16kHz with ±22dB range and switchable Q.

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

EQF-2: single channel parametric equaliser/filter with control range of ±12dB in three bands from 25Hz to 20kHz and adjustable high and low pass filters.

BIAMP (USA)
Biamp Systems Inc, 9600 SW Barnes Road, Portland, Oregon 97225. Phone: (503) 297-1555.

EQ/110R: 10-band single-channel graphic equaliser ±15dB from 32Hz to 16kHz.
EQ/210: stereo version of *EQ/110R* but with separate controls for each channel.
EQ/270A: 27-band single-channel graphic equaliser with ⅓-octave bands from 40Hz to 16kHz ±12dB.

CARLSBRO (UK)
Carlsbro Sales Ltd, Cross Drive, Low Moor Road Industrial Estate, Kirkby-in-Ashfield, Notts NG17 7LD. Phone: 0623 753902.

D10: 2-channel 10-band graphic equaliser with single octave centres. Control range ±15dB.

CATHEDRAL (UK)
Cathedral Sounds Ltd, Fourways, Morris Lane, Halsall, Ormskirk, Lancashire L39 8SX. Phone: 0704 840328.

SGE20: 2-channel graphic equaliser in 10 bands ±12dB from 30Hz to 16kHz.

CERWIN-VEGA (USA)
Cerwin-Vega Inc, 12250 Montague Street, Arleta, California 91331. Phone: (213) 896-0777. Telex: 910-496 1589.

GE-2: 2-channel graphic equaliser with 13 bands 31.5Hz to 16kHz ±12dB.

COURT (UK)
UK: Theatre Projects Services Ltd, 11 Marshalsea Road, London SE1 1EP. Phone: 01-403 3838. Telex: 885659.
USA: Quintek Inc, 4712 Laurel Canyon Blvd, Suite 209, North Hollywood, California 91607. Phone: (213) 980-5717. Telex: 194781.

GE-60: 2-channel graphic equaliser with 30, ⅓-octave bands control range of +20dB and −10dB 25Hz to 20kHz.

CROWN/AMCRON (USA)
Crown International Inc, 1718 West Mishawaka Road, Elkhart, Indiana 46514. Phone: (219) 294-5571. Telex: 810-295 2160.
UK: HHB Hire & Sales, Unit F, New Crescent Works, Nicoll Road, London NW10. Phone: 01-961 3295. Telex: 923393.

EQ-2: 2-channel graphic equaliser covering 20Hz to 20kHz in 11 bands ±15dB with Q adjustable up to ±½ octave.

DAN DUGAN (USA)
Dan Dugan Sound Design, 833 14th Street, San Fransisco, California 94114. Phone: (415) 621-0781.

Multi-channel parametric: 3-band parametric equalisers with preset adjustments over 23Hz to 21kHz with −7dB control range and variable Q. Available in 8- or 4-channel units and the intended use is for flattening the head response of tape recorders.

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Eventide Clockworks

The Eventide SP 2016 programmable effects processor:

The most versatile audio processing instrument ever developed is now available: EVENTIDE, with over a decade of leadership in digital audio effects again advances the state-of-the-art with the SP2016 Programmable Effects Processor. In a single 3½" rack mounted device, EVENTIDE has engineered the most powerfully versatile digital processing system ever employed in an audio component.

FEATURES OF THE SP2016:

REVERB
At your fingertips is a wide variety of reverb programs with operator control of all parameters plus superb audio spec. performance and reverb quality. The SP2016 accepts EVENTIDE's new software "Reverb Library" Roms, a growing collection of plug-in programs.

DIGIPLEX® ECHO
The SP2016 provides EVENTIDE's Digiplex echo, our digital version of multiple-head tape echo. Exclusive features include incredibly stable operation, giving literally hours of decay time with no noise build-up.

CHORUS EFFECTS AS YOU'VE NEVER HEARD THEM BEFORE
ADT takes on a whole new meaning. "D" can now stand for dozens, not just double! Each voice can vary randomly in time, amplitude and space.

FULL BANDWIDTH DELAY
16kHz: 0 to 1.6 sec. delay in 25 microsecond steps.
8kHz: 0 to 3.2 sec. delay in 30 microsecond steps.

SELECTIVE BAND DELAY
The first in a series of dramatic new effects exclusive to the SP2016. You can separate the signal into a number of frequency bands and independently delay each band up to 3.2 seconds.

FLANGING AND PHASING
Quality and control features far surpassing existing devices.

CLASSICAL DIGITAL LINEAR PHASE FILTERS
For PA, crossovers and EQ. Design filters to your specifications using the IEEE-compatible remote controller.

FULL STEREO OPERATION
2 in, 2 out.

PROGRAM SOFTWARE SUBSCRIPTION SERVICE
The SP2016's digital circuitry is so powerful, we've yet barely tapped the ultimate capabilities inherant in the Programmable Effects Processor. EVENTIDE continues to develop new and exclusive effects for the SP2016. Not just updated and refined reverb programs (although we will offer these, too) but totally new and unique effects. Because the SP2016 is a fully programmable system, we can supply these new effects (as well as revisions) as they are developed via convenient plug-in modules. They will be available individually or through our program software subscription service. The SP2016 is obsolescence proof.

SELF-TEST FEATURE
The most extensive self-test capabilities of any pro-audio product ever! Should a problem develop, the SP2016 will spot it and even pinpoint the part number of the suspect I.C.

U.K. Distributors

FELDON AUD

126 Great Portland Street, London W1N 5PH Tel: 01-580 4314. Telex: London 28668.

product Equalisers guide

dbx (USA)

dbx Inc, 71 Chapel Street, Newton, Massachusetts 02195. Phone: (617) 964-3210. Telex: 922522.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-743 2812. Telex: 27939.

905: 3-band single channel modular parametric equaliser covering 20Hz to 20kHz ±15dB.
20/20: single channel 10-band graphic equaliser with LED representation rather than sliders. Range of ±15dB 31.5Hz to 16kHz with auto EQ and integral RTA.

DOD (USA)

DOD Electronics Corporation, 2953 South 300 West, Salt Lake City, Utah 84115.
UK: Strings & Things Ltd, Unit 2, Chapel Road, Portslade, Brighton, Sussex BN4 1PF. Phone: 0273 420704.

R-815: 15-band single channel graphic equaliser with ⅔-octave frequency centres between 25Hz to 16kHz ±12dB.
R-830: 2-channel version of *R-815*.
R-831: 31-band single channel graphic equaliser with ⅓-octave centres between 20Hz and 20kHz and ±12dB control range.

D & R (Netherlands)

D & R Electronica BV, Chassestraat 26, 1057 JE Amsterdam. Phone: 020-18 35 56.
UK: DSN Marketing Ltd, Westmorland Road, London NW9 9RJ. Phone: 01-204 7246. Telex: 895 4243.

Parametric equaliser: modular single-channel parametric equaliser with two bands 60Hz to 10kHz ±18dB with variable Q.
High-low pass filters: 2-channel high and low pass filters in modular form.

DYNACORD (West Germany)

Dynacord Electronic GmbH, Siemenstrasse 41 – 43, D-8440 Strubing. Phone: 09421 3103.
USA: Dynacord Electronics Inc, PO Box 26038, Philadelphia, Pennsylvania 19128. Phone: (215) 482-4992.

EQ 270: single channel 27-band graphic equaliser with ±12dB control range.
EQ 210: 2-channel 10-band graphic equaliser ±12dB at single octave centres.
EQ 1400: 2-channel 10-band graphic equaliser for PA use.

FORMULA SOUND (UK)

Formula Sound Ltd, 3 Waterloo Road, Stockport, Greater Manchester, SK1 3DB. Phone: 061-480 3681.

S19G: 2-channel graphic equaliser with 19 bands per channel ±12dB 31Hz to 16kHz.
SG19A: similar to *S19G* but with analyser section and dual LED display.

FOSTEX (Japan)

Fostex Corporation, 512 Miyazawzcho, Akishima, Tokyo 196. Phone: 0425 456111. Telex: 2842203.
UK: Bandive Ltd, 8 East Barnet Road, New Barnet, Hertfordshire EN4 8RW. Phone: 01-440 9221. Telex: 25769.
USA: Fostex Corporation of America, 15431 Blackburn Avenue, Norwalk, California 90650. Phone: (213) 921-1112.

3030: stereo graphic equaliser with 10 bands per channel 31.5Hz to 16kHz with range of ±12dB.

FURMAN (USA)

Furman Sound Inc, 616 Canal Street, San Rafael, California 94901. Phone: (415) 456-6766.
UK: Atlantex Music Ltd, 1 Wallace Way, Hitchin, Herts SG4 0SE. Phone: 0462 31511. Telex: 826967.

PQ-3: 3-band single channel parametric equaliser covering 25Hz to 10kHz, 20dB boost and infinite attenuation with adjustable Q.
PQ-6: 2-channel version of the *PQ-3*.

IEM (USA)

International Electro-Magnetics Inc, Eric Drive and Cornell Avenue, Palatine, Illinois 60067. Phone: (312) 358-4622.

Model 213: single channel graphic equaliser covering 10 octaves 32Hz to 16kHz with ±15dB control range.
Model 231: 31-band single channel graphic equaliser covering 20Hz to 20kHz in ⅓-octave bands ±12dB.

IVIE (USA)

Ivie Electronics Inc 500 West 1200 South, Orem, Utah 84057. Phone: (801) 224-1800. Telex: 910-971 5884.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts WD6 4RZ. Phone: 01-953 0091. Telex: 27502.

Model 5303: modular 27-band equaliser with 27, ⅓-octave bands 40Hz to 16kHz and ±10dB range.
Model 5306: notch filter with six tunable ⅒-octave filters 50Hz to 3.2kHz with 0 to 12dB notch depth.

K + H (West Germany)

Klein + Hummel, Zeppelinstrasse 12, D-7302 Ostfildern/Kemnat. Phone: 0711 455026. Telex: 723398.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts WD6 4RZ. Phone: 01-953 0091. Telex: 27502.
USA: Gotham Audio Corp, 741 Washington Street, New York NY 10014. Phone: (212) 741-7411. Telex: 129269.

UE400: 2-channel parametric equaliser with three bands covering 15Hz to 20kHz with control range of ±12dB and high and low pass filters.
UE 200: single channel version of *UE400*.

KLARK-TEKNIK (UK)

Klark-Teknik Research Ltd, Walter Nash Road West, Kidderminster, Hereford & Worcester DY11 7HS. Phone: 0562 741515. Telex: 339821.
UK: Autograph Sales Ltd, Stable 11, British Rail Camden Depot, Chalk Farm Road, London NW1 8AH. Phone: 01-267 6677.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, New York 11735. Phone: (516) 249-3600.

DN15: stereo graphic equaliser with 11 bands from 50Hz to 16kHz including preamplifier and source selection inputs, etc.
DN22: 2-channel graphic equaliser with 11 bands ±12dB from 50Hz to 16kHz.
DN27A: single channel 27-band graphic equaliser with frequencies from 40Hz to 16kHz and ±12dB range.
DN30/30: 2-channel 30-band graphic equaliser with ±12dB range from 25Hz to 20kHz.

LEUNIG (West Germany)

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

PARAM: computer assisted parametric equaliser, six bands and up to 128 channels. (See Console Automation product guide *Studio Sound* December 1982).

LINDSAY (UK)

Lindsay Electronics Ltd, Unit 5, Salome Works, Prospect Place, Trowbridge, Wiltshire BA14 8QA. Phone: 02214 64282.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Model 7606: single channel graphic equaliser with 27, ⅓-octave bands 40Hz to 16kHz ±12dB.

LOFT (USA)

Phoenix Audio Laboratory Inc, 91 Elm Street, Manchester, Connecticut 06040. Phone: (203) 649-1199.

Model 401: single channel parametric ±18dB in four bands with adjustable Q 1.6 to 3 octaves.

LOG (France)

Log Audio Equipment, 44 sente aux ânes, Le Boulay, F-78950, Gambais. Phone: 16(3) 487.10.97.

EQ4A: single channel modular parametric equaliser with four bands covering 40Hz to 20kHz with control range of ±15dB and variable Q.
EQ4B: 2-channel equaliser with similar features to the *EQ4A*.

M-JAY (UK)

M-Jay Electronics Ltd, 90 Kingsdale Gardens, Drighlington, Bradford BD11 1EZ. Phone: 0532 852075.

GE9-2: 2-channel graphic equaliser with nine single octave bands 50Hz to 12.8kHz ±12dB.
EF2 electronic filter: 2-channel high and low pass filter with 12 or 24dB/octave slopes.

MOOG (USA)

Moog Music Inc, 2500 Walden Avenue, Buffalo, New York 14225.
UK: Moog Music, 11 Forth Wynd, Port Seton, East Lothian, Scotland. Phone: 0875 812033.
USA: Norlin Co, 7373 North Cicero Avenue, Lincoln Wood, Illinois 60646. Phone: (312) 675-2000.

Graphic equaliser: single channel graphic equaliser with nine single octave bands 31Hz to 8kHz ±15dB and shelving filter at 16kHz.
Parametric equaliser: 3-band single channel parametric equaliser 31Hz to 16kHz, with control range ±12dB or ±20dB dependent on Q setting.

MXR (USA)

MXR Innovations Inc, 740 Driving Park Avenue, Rochester, NY 14613. Phone: (716) 254-2910. Telex: 978451.
UK: Atlantex Music Ltd, 1 Wallace Way, Hitchin, Herts SG4 0SE. Phone: 0462 31511. Telex: 826967.

Dual 15-band Equaliser: 2-channel graphic equaliser with 15, ⅔-octave bands from 25Hz to 16kHz on ⅓-octave centres and control range of ±12dB.
31-band Equaliser: single-channel graphic equaliser with 31, ⅓-octave bands from 20Hz to 20kHz, control range ±12dB.

NEPTUNE (USA)

Neptune Electronics Inc, 934 NE 25th Avenue, Portland, Oregon 97232. Phone: (503) 232-4445.

Model 910: single channel graphic equaliser with nine single octave bands and ±12dB range.
Model 1020: 2-channel 10-band single octave graphic equaliser with ±12dB range.
Model 1021: revised version of the *Model 1020*.
Model 2710: 12-band single channel graphic equaliser with ⅓-octave bands 40Hz to 16kHz, ±12dB.
Model 2711: ⅓-octave graphic equaliser at standard ISO frequencies ±12dB.
Model 342: 2-channel parametric equaliser with four bands covering 16Hz to 24kHz with ±15dB control range and variable Q.

NTP (Denmark)

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

Type 182-100: 3-band modular equaliser covering 60Hz to 10kHz with max range ±12dB.
Type 182-200: modular high and low pass filter with 18 frequencies and 18dB/octave slope.
Type 582-100: programmable graphic equaliser with 14, ⅔-octave bands and ±14dB range. For automation details see Console Automation product guide in *Studio Sound* December 1982.

ORANGE COUNTY (Canada)

Orange County Electronics Corp, 534 Berry Street, Winnipeg, Manitoba RH3 045. Phone: (204) 774-3413.
USA: Parasound Inc, 680 Beach Street, San Francisco, California 94109. Phone: (415) 673-4544.

DEQ: single channel parametric equaliser module with four bands covering 20Hz to 20kHz with +20dB and −60dB control range.
PEQ: single channel parametric equaliser similar to *DEQ* but with ±20dB range.
SEQ: single channel parametric equaliser with four bands 35Hz to 18kHz ±20dB range.

ORBAN (USA)

Orban Associates Inc, 645 Bryant Street, San Francisco, California 94107. Phone: (415) 957-1063. Telex: 171480.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Model 622B: 2-channel parametric equaliser with four bands 20Hz to 20kHz, variable Q and control range +16dB and infinite attenuation.
672A paragraphic: 8-band parametric equaliser covering 20Hz to 21kHz ±16dB range.
Model 674A: stereo version of *672A*.

PEAVEY (USA)

Peavey Electronics Corp, 711 A Street, Meridan, Mississippi 39301. Phone: (601) 483-3565.
UK: Peavey Electronics (UK) Ltd, Unit 8, New Road, Ridgewood, Uckfield, Sussex TN22 5SX. Phone: 0825 5566. Telex: 957098.

EQ27: 27-band single channel graphic equaliser from 40Hz to 16kHz ±15dB.

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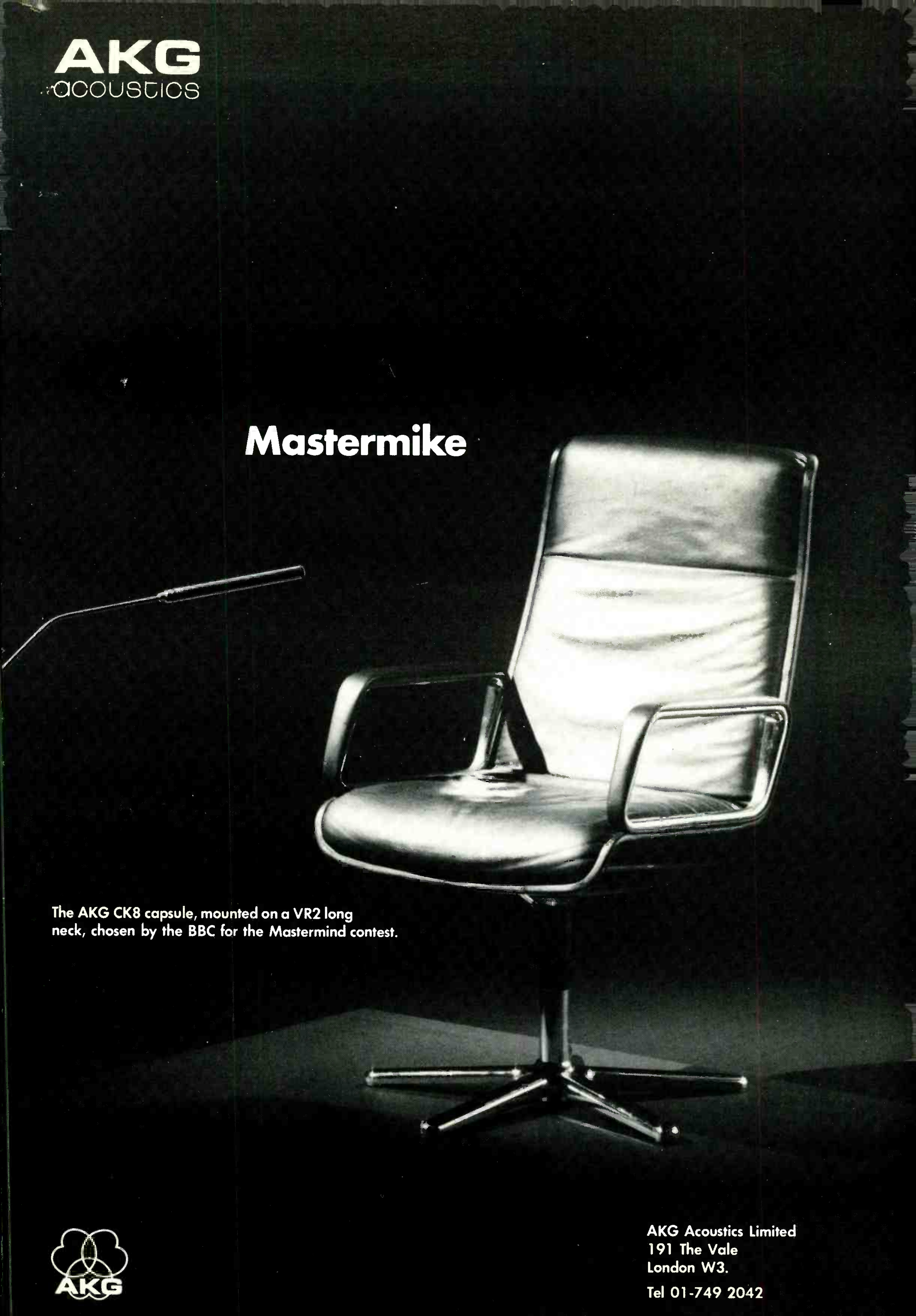
AKG
acoustics
Mastermike
The AKG CK8 capsule, mounted on a VR2 long
neck, chosen by the BBC for the Mastermind contest.
AKG
AKG Acoustics Limited
191 The Vale
London W3.
Tel 01-749 2042

product Equalisers guide

Stereo graphic: 2-channel graphic equaliser with 10 bands ±15dB 30Hz to 16kHz.

PRO AUDIO (UK)
Pro Audio Ltd, 30 Wolsey Drive, Walton-on-Thames, Surrey KT12 3AZ. Phone: 09322 21078.
USA: Eastern Acoustic Works Inc, 59 Fountain Street, Box 111, Framingham, Massachusetts 01701. Phone: (617) 620-1478.

PA20: 2-channel graphic equaliser with 10 single octave bands and control range of ±12dB from 50Hz to 12kHz.
PA27: single channel graphic equaliser with 27, ⅓-octave bands with a control range of ±12dB between 40Hz and 16kHz.

PULTEC (USA)
Pulse Techniques Inc, 1411 Palisade Avenue, Teaneck, New Jersey 07666. Phone: (201) 837-2575.

EQP-1A3: single channel equaliser with variable band frequencies, control range and Q.
MEQ-5: 3-band single channel equaliser covering 200Hz to 5kHz with cut or boost variable with band.
HLF-3C: high and low pass filters with variable cut frequencies.
HLF-23C: stereo version of *HLF-3C*.

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

EQ-333: 3-band single channel parametric equaliser with switched frequencies from 50Hz to 15kHz and ±12dB range.
EQ-444: 4-band single channel parametric equaliser with switched frequencies between 50Hz and 18kHz ±12dB.

REBIS (UK)
Rebis Audio, Kinver Street, Stourbridge, West Midlands DY8 6A. Phone: 0384 71865.
UK: Scenic Sounds Equipment Ltd, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.
USA: Klark-Teknik Electronics Inc, 262A Eastern Parkway, Farmingdale, New York 11735. Phone: (516) 249-3600.

RA402: 2-channel parametric equaliser with four bands 20Hz to 18kHz ±21dB and variable Q.
RA204: modular parametric equaliser with one band switchable 20Hz to 2kHz or 200Hz to 20kHz, variable Q and ±21dB control range.

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

SEQ-315: 2-channel 15-band graphic equaliser with ±12dB range.
SEQ-331: single channel graphic equaliser with 31 bands and ±12dB range.

SAE (USA)
Scientific Audio Electronics Inc, 701 East Macy Street, Los Angeles, California 90012. Phone: (213) 489-7600.

1800: 2-channel parametric equaliser with two bands 40Hz to 20kHz, variable Q and ±16dB control range.
2800: 4-band version of *1800* covering 10Hz to 20kHz.

SHURE (USA)
Shure Bros Inc, 222 Hartrey Avenue, Evanston, Illinois 60204. Phone: (312) 866-2200. Telex: 724381.
UK: Shure Electronics Ltd, Eccleston Road, Maidstone, ME15 6AU. Phone: 0622 59881. Telex: 96121.

SR107-2E: single channel, 10 single octave equaliser covering 32Hz to 16kHz.

SOLIDYNE (Argentina)
Solidyne SRL, Tres de Febrero 3254, 1429 Buenos Aires. Phone: 701-8622.

9180: modular single-channel parametric equaliser rack with plug in sections and choice of modules up to nine for full complement.

SONTEC (USA)
Sontec Electronics 10120 Marble Court, Cockeysville, Maryland 21030. Phone: (301) 628-2283.

2 MEP-250A: 2-channel 5-band parametric equaliser covering 8Hz to 25kHz ±12dB range and variable Q.
MES-430B: 4-channel 3-band parametric disc mastering equaliser with programme and preview channels.

SOUNDCRAFTSMEN (USA)
Soundcraftsmen, 2200 S Ritchey, Santa Ana, California 92705. Phone: (714) 556-6191.
UK: REW Professional Audio, 114 Charing Cross Road, London WC2. Phone: 01-836 7851. Telex: 8814193.

AE2420-R: 10-band analyser/equaliser covering 30Hz to 15.36kHz with control range of +22dB and −28dB.
RF2215-R: 2-channel graphic equaliser, 10 bands 30Hz to 15.36kHz ±22dB.
TG3044-R: 2-channel graphic equaliser with 21 bands, ⅓-octave below 1kHz and alternate ⅓-octave above with ±22dB range.

SOUND WORKSHOP (USA)
Sound Workshop Inc, 1324 Motor Parkway, Hauppauge, NY 11787. Phone: (516) 582-6210. Telex: 649230.

Parametric equaliser: modular parametric equaliser with single channel and three bands covering 30Hz to 18kHz and variable Q; ±14dB control range and high pass filter.

SPECTRA SONICS (USA)
Spectra Sonics Inc, 3750 Airport Road, Ogden, Utah 84403. Phone: (801) 392-7531.

Model 500: single channel equaliser with two switched bands ±12dB.
Model 501: single channel with two bands and continuously variable control range ±10dB.
Model 502: single channel equaliser with three switched bands and ±12dB range. High and low filters.
Spectra Sound 1000B: 2-channel graphic equaliser with 10 bands 31Hz to 16kHz and ±8 or ±16dB control range.

SUNN (USA)
Sunn Musical Equipment Co, Amburn Industrial Park, Tualitin, Oregon 97062. Phone: (503) 638-6551.

SPL4120: 2-channel graphic equaliser with 10 bands between 31.5Hz to 16kHz with ±15dB control range.

SYNTON (Netherlands)
Synton Electronics BV, Zandpad 46, Postbus 83, NL-3620 Breukelen. Phone: 034 62.34.99.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.
USA: Synton USA, 269 Locust, Northampton, Massachusetts 01060. Phone: (413) 586-3777.

227 Filter: designed for use in electronic music recording. 27-band graphic equaliser with two sets of sliders allowing two response curves for each signal. Each filter has 48 dB/octave slope.

TAPCO (USA)
EV-Tapco, 3810 148th Avenue NE, Redmond, Washington 98052. Phone: (206) 883-3510. Telex: 910-4492594.

2230: single channel graphic equaliser with 27, ⅓-octave bands between 40Hz and 16kHz with ±12dB range.
2210: 2-channel 10-band graphic equaliser with ±12dB range from 31.5Hz to 16kHz.

TEAC (Japan)
USA: Teac Corp of America, 7733 Telegraph Road, Montebello, California 90640. Phone: (213) 726-0303. Telex: 677014.
UK: Harman (Audio) UK Ltd, Mill Street, Slough SL2 5DD. Phone: 0753 76911. Telex. 849069.

GE-20: 2-channel graphic equaliser with 10 bands 31.5Hz to 16kHz ±12dB.
PE-20: 4-channel parametric equaliser with three bands covering 30Hz to 10kHz ±12dB.

TECNICOBEL (France)
Tecnicobel, 8 rue de la Croix-Matre, BP26, F-91122 Palaiseau Cedex. Phone: (1) 920.80.39. Telex: 692543.

CF 50: single channel modular parametric equaliser with four bands and ±15dB control range.
CFD 60: 2-channel rack mount parametric equaliser with four bands and ±15dB control range.

TOA (Japan)
UK: Toa Electric Company Ltd, PO Box 82, Castle Street, Ongar, Essex. Phone: 0277 364333. Telex: 995554.
USA: Toa Electronics Inc, 1023 Grandview Drive, San Francisco, California 94080. Phone: (415) 588-2583. Telex: 331332.

RE-11: single channel octave spaced 10-band graphic equaliser with each band having switchable Q and variable centre frequency so that entire range from 23Hz to 24kHz has control range of ±10dB.
RE12: stereo version of *RE-11*.
E-2300: ⅓-octave equaliser with 28 bands from 31.5Hz to 16kHz with −15dB range.

TRIDENT (UK)
Trident Audio Developments Ltd, PO Box 38, Studios Road, Shepperton, Middlesex TW17 0QD. Phone: 09328 60241. Telex: 88139832.
USA: Trident (USA) Inc, 652 Glenbrook Road, Stamford, Connecticut 06906. Phone: (203) 357-8337.

CB 9066: single channel parametric equaliser with three overlapping bands and 16dB range with variable Q, 2dB to 18dB/octave including variable high and low pass filters.

UREI (USA)
United Recording Electronics Industries, 8460 San Fernando Road, Sun Valley, California 91352. Phone: (213) 767-1000. Telex: 651389.
UK: FWO Bauch Ltd, 49 Theobald Street, Boreham Wood, Herts WD6 4RZ. Phone: 01-953 0091. Telex: 27502.

Model 535: 2-channel graphic equaliser with 10 bands between 31.5Hz to 16kHz and a range of ±12dB.
Model 537A: single-channel graphic equaliser with 27 bands ±12dB from 40Hz to 16kHz 0 to −15dB range, high and low pass filters, screwdriver adjustable.
Model 539A: single channel room EQ filter set with 27 bands from 40Hz to 16kHz, 0 to −15dB range, high and low pass filters, screwdriver adjustable.
Model 545: single channel parametric equaliser with four overlapping bands, Q adjustable ¼ to 2 octaves, low and high cuts and ±15dB range.
Model 546: 2-channel parametric equaliser version of *Model 545* but in mono mode all filters may be cascaded.
Model 555: single channel high and low pass filters adjustable frequencies with 18dB/octave slopes.

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

Maxi Q: single channel modular parametric equaliser with three bands covering 31Hz to 16kHz, −40dB and +14dB control range with variable Q.

WESTREX (USA)
Westrex Co, 2629 West Olive Avenue, Burbank, California 91505. Phone: (213) 846-3394. Telex: 698254.
UK: Westrex Co Ltd, Bilton Fairway Estate, Long Drive, Greenford, Middlesex. Phone: 01-578 0957. Telex: 923003.

ST3015: single-channel equaliser with 15 bands from 50Hz to 12kHz ±14dB.

WHITE (USA)
White Instruments Inc, PO Box 598, Austin, Texas. Phone: (512) 892-0752.
UK: Scenic Sounds Equipment Ltd, 97–99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.

Series 4000: single channel 27-band equaliser with ±10dB range 40Hz to 16kHz. Includes 12dB/octave high pass filter and several options.
Model 4004: single-channel passive cut only equaliser, with 24, ⅓-octave bands from 63Hz to 12.5kHz, 0 to −15dB range and variable low and high pass filters.

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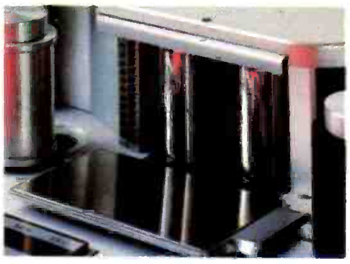
SPOILT FOR CHOICE.



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.

STUDER
INTERNATIONAL AG
CH-8105 Regensdorf Telephone (01) 840 29 60

F.W.O. Bauch Limited

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

STUDER REVOX AMERICA INC Nashville Telephone (615) 254-5651 STUDER REVOX S A R L Paris Telephone 533 5858 STUDER REVOX CANADA LTD Toronto Telephone (416) 423-2831

product guide Equalisers

Model 4100: 10-band 2-channel equaliser with frequency centres at 31.5Hz to 16kHz ±10dB range with high pass filter.
Model 4300: single channel monitor equaliser with 41 bands—28 at ⅙-octave between 40Hz and 894Hz, and 13 bands at ⅓-octave between 1kHz and 16kHz, all ±10dB range.
Model 4310 and 4311: single channel equaliser with 41 bands – seven ⅓-octave between 40Hz and 160Hz, 29 ⅙-octave between 180Hz and 4.5kHz, four ⅓-octave bands 5kHz to 10kHz and one octave band at 12.5kHz all ±10dB. *4311* is balanced and *4310* unbalanced.
Model 4240: single channel equaliser with 27 bands in ranges and spacings that are applicable to voice programme equalisation. ±10dB control range.

YAMAHA (Japan)

USA: Yamaha International Corp, PO Box 6600, Buena Park, California 90620. Phone: (714) 522-9105.
UK: Yamaha Musical Instruments, Mount Avenue, Bletchley, Milton Keynes, Buckinghamshire. Phone: 0908 71771.

Q1027: single channel graphic equaliser with 27 bands ±12dB. ■

product guide Synthesisers and Vocoders

Note: There is such a wide diversity of electronic music devices available today that we have had to be selective in this Guide to avoid filling the entire magazine. We have therefore omitted a large number of instruments, primarily those aimed at the 'performance' market; information on these can be readily obtained from the regular musicians' magazines. Instead, we have concentrated specifically on those systems which are aimed more at the 'studio' market, particularly the increasingly popular digital and digitally-controlled systems which, due to their expense and complexity, will often be purchased by studios rather than individuals, and will not generally be given coverage in musicians' reference sources.

ADAPTIVE SYSTEMS (USA)

Adaptive Systems Inc, PO Box 9356, Newark, Delaware 19711. Phone: (302) 366-0478.

Synthia: computer based synthesiser comprising 61-key, velocity and pressure sensitive keyboard with eight pre-select buttons each fully programmable, six pedal inputs assignable to any voice or effects parameter, eight fader or four joystick expression panels. Touch sensitive plasma display with two 256KByte digital tape drives mounted on top of the keyboard. This unit is the player's link to the computer and all options are presented in English in menu fashion and the player makes choice by touching the display.

The system can have from eight to 64 voices with each voice generating eight harmonics which are completely controllable. Full range of effects can be assigned to the voices including Time-Slice Voicing where it is possible to change character of the note in segments as short as 1/100th second.

The *Synthia* uses a rack mount computer with 4.8MHz CPU, 80KByte ROM and 64KByte RAM and can handle up to 16 keyboards, and 64 voices. There are a full range of accessories available including printers and 16-track digital recorder.

ARAK (UK)

Arak Sound Ltd, Preston House, High Street, Crowthorne, Berks. Phone: 03446 2550.

Polycontroller: polyphonic keyboard controller which can be connected to any standard synthesisers. CV and gate outputs for up to 10 voices and separate ADSR for each with one set of master controls. 'Bend Bar' for bending the pitch of notes. Computer interface which will allow the control of any synthesiser attached to the Polycontroller by the computer. Vibrato, portamento and V/octave adjustments.

ATOM UK (UK)

Theatrescene Special Effects, Suite 4, 12 – 13 Henrietta Street, London WC2E 8LH. Phone: 01-240 2116.

ATOM SP400: 20-channel vocoder with twin mike and line inputs.

Features of the *SP400* include go/no-go LED level indicators; a harmonic boost facility for increasing the harmonic content of the excitation input signal; a synthetic S facility allowing insertion of a noise burst into the excitation when a sibilant is sounded (LED indicators shows whether a voiced or unvoiced sound is being uttered); an excitation follow mode to synthesise a voice with varying pitch; an output mixer to mix articulation, excitation and vocoder outputs; and remote switch facilities allowing preset mixes to be switched into live performances.

The optional *SPX400* special effects unit which derives its power from the *SP400* adds the following facilities: channel patching, allowing formant shifting, formant inversion and many others; freeze facility (with LED indication and optional remote foot switching) allowing a vocoded output to be frozen on any selected syllable; and realtime spectrum analysis via an LED display. The latter function can also be frozen via the aforementioned freeze facility. Both units are 19in rack mount.

BARTH (West Germany)

R Barth KG, Grillparzerstrasse 6a, D-2000 Hamburg 76. Phone: 040 229 88 83. Telex: 0212095.
USA: Audicon Inc, 1200 Beechwood Avenue, Nashville, Tennessee 37212. Phone: (615) 256-6900. Telex: 554494.

Musicoder: compact rack mounting vocoder with 16 filter channels and mono/stereo output. Input level controls for program and control signals, hi-boost switch to add mid and high frequency emphasis to the control signal to increase the harmonic content prior to vocoding; bypass matching controls; auto-bypass to route programme to output if no control signal is present and this may be done in a fast or slow mode; threshold and level controls for noise generator; output level controls.

BIG BRIAR (USA)

Big Briar Inc, Leicester, North Carolina 28748. Phone: (704) 683-9085.
Europe: Synton Electronics BV, Zandpad 46, Postbus 83, 3620 AB Breukelen, Holland. Phone: 03462-3499. Telex: 40541.

100 Series Keyboard Controllers: keyboard using standard size wood action keys with four independent modes of touch sensitivity available for all keys: X left-to-right finger position. Y front-to-back finger position, Z up-and-down key motion and V is key velocity. The keys are all scanned digitally and the data formatted for entry into computer or digital programmer. Wide variety of options and keyboards of up to six octaves.
300 Series: electrically sensitive plate that responds to touch front-to-back and left-to-right. The user places his finger on the plate and moves it and the generated outputs are proportional to the finger to plate edge distance. Also option with third output based on finger pressure on plate. Analogue output as standard with digital option.
500 Series: a *Therim*-type controller that reproduces the configuration of the original *Therim* without the signal generating facilities. Pitch control is by movement of the hand towards the vertical pitch antenna and volume by the hand approaching the horizontal loop antenna. Analogue output with digital option.

BUCHLA (USA)

Buchla Associates, PO Box 5051, Berkeley, California 94705.

Two series of modular electronic music components. The *200 Series* includes 12 modules with various cabinet and power supply options. The modules are *Model 207* mixer/preamp, *Model 208* programmable sound source. *Model 221* kinesthetic input port, *Model 227* system interface, *Model 230* triple envelope detector, *Model 257* dual control voltage processor, *Model 259* programmable complex waveform generator, *Model 266* 'source of uncertainty', *Model 281* quad envelope generator, *Model 285* frequency shifter/balanced modulator, *Model 292* quad voltage controlled low pass gate and *Model 296* programmable spectral processor. The *300 Series* enables digital control of the *200 Series* and includes *Model 300A* processor, *Model 329* patchbay and *Model 364* multiple arbitrary function generator.
Touche: digital/analogue keyboard controlled synthesiser. Integral 16-bit computer for user communication and data processing using FOIL language. Sound generation by multiplexed digital signal generator with crystal derived pitch. 24 digital oscillators combined into eight voices assignable to variety of polyphonic split keyboard and multi-instrument modes. Fully programmable with up to 64 labelled instrument definitions storable with additional programs storable on tape for later retrieval.
Buchla 400: multi-functional electronic musical instrument system. Score editor that functions in real time. Six orchestrally differentiated voices can be simultaneously displayed, auditioned and edited. A graphic display employs a linear time notation to display more musical data than is possible with conventional notation. All functions such as instrument definition, dynamics, tempo, registration and tunings are programmable. Can decode, display and track SMPTE timecode enabling uses for film and video, etc. touch sensitive keyboard which can be organised in traditional or specialised fashion and tuned to any scale. Other features include pressure sensitive joysticks, control voltage interfacing and analogue modifiers. System uses three computers and an operational language known as MIDAS and is programmed in music FORTH. All circuitry on plug-in cards enabling easy servicing and future expansion.
Buchla 404A: similar to *400* but with expanded memory of 126Kbytes and the ability to run MIDAS and PATCH V musical languages, both of which are appropriate for composition and performance. The *404A* can be supplied with either or both languages.

CASHEAB (USA)

Casheab, 5737 Avenida Sanchez, San Diego, California 92124. Phone: (714) 277-2547.

Casheab Music System: consists of a microcomputer, a terminal, 32-channel synthesiser and a 61-note keyboard. The synthesiser consists of two cards – a synthesiser card and a controller card. The *S-100* host processor programs the waveforms into the synthesiser. 16 waveforms can be stored and any channel can use any one of them. The internal processor can also control the signal envelopes as well frequency modulation.

Software is provided on a CP/M compatible disk and includes waveform creation, music compiling and a real-time operating program.

CASIO (Japan)

UK: Casio Electronics Co Ltd, 28 Scrutton Street, London EC2A 4TY. Phone: 01-377 9087.

Range of performance orientated and compact personal synths.

CON BRIO (USA)

975 San Pasqual Street, Suite 313, Pasadena, California 91106. Phone: (213) 795-2192.

ADS 200: digital synthesiser with two manual keyboard, video display and disk drive. Contains five microprocessors, 64 multiwaveform digital oscillators and dual 16-bit stereo output channels. Instant recall from floppy disk and assignment to left or right side of keyboard – split position variable. Tuning storable on disk with fine tuning on panel. Ensemble position allows grouping of voice assignments, transpositions, volume balancing, channel assigns and cued playback materials for recall as complete groups which can be selected by footswitch. 'Unlimited' storage space for polyphonic tracks on disk and replay of up to four tracks at once. Video screen can be used to display in conventional musical notation what has just been performed which can then by printed by interfacing with a line printer and the necessary software package. Six waveforms are available: sine, triangle, sawtooth, square, pulse and noise together with conventional ADSR or 16-segment envelopes. Synthesis modes include additive synthesis, phase modulation, frequency modulation etc. All parameters can be floppy disk stored. Comprehensive interface capability and software available includes an operating system, text editor and macro-assembler. Can be interfaced with *PDP 11, PDP 10, LSI 11*, Nova or IBM computers.
ADS 200-R: a portable and more economical version of the *ADS 200*. It is available in a variety of flexible system configurations in either 32- or 64-voice versions.

DIGITAL KEYBOARDS (USA)

Digital Keyboards Inc, 105 5th Avenue, Garden City Park, NY 11040, USA. Phone: (516) 747-7890.
UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-724 2451.

Crumar GDS: digital computer controlled synthesiser with 61-note velocity sensitive keyboard. System consists of keyboard, video display, alpha-numeric keyboard, *Z-80* microprocessor, and two disk drives. The system can use up to eight independent voices and each voice

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product guide
Synthesisers and Vocoders

may use up to 16 of the 32 available oscillators. Video display for program parameters; 16-bit D/A converter; 65 input devices. 8-track digital event recorder with editor, speed, transposition and rhythm controls and allows the superimposition of improvised materials. Complete system under software control allowing contiual update.

The Synergy: a 48-voice totally digital polyphonic synthesiser, 6-octave dynamic keyboard which controls 48 voices, 24 of which are preset while a further 24 voices are available on an interchangeable cartridge. Features of the synthesiser include assignable performance characteristics for each voice; key sensitivity with respect to volume, timbre and modulation; vibrato rate, depth and delay; and speed of portamento (three modes – smooth with retrigger, smooth with no retrigger of envelopes, and semitone quantisation). Other features include a sustain pedal and a sustenato pedal which functions like the centre pedal on a piano. Normally, eight notes are playable simultaneously, although this varies according to the number of oscillators required for each note of the active timbres (there are 32 digital oscillators, and most timbres require four oscillators).

DIGITAL MUSIC SYSTEMS (USA)

Digital Music Systems Inc, 137 Pearl Street, Boston, Massachusetts 02110. Phone: (617) 542-3042.

DMX-1010: programmable digital audio signal processor designed for music synthesis and music signal processing. In signal processing modes, delay, reverb, phasing, flanging, equalisation, compression and limiting are available. All synthesis and processing is performed digitally using 16-bit standard. Programming language for all functions has been derived from MUSIC V. Controlled by two computers, a *PDP-11/03* microcomputer for for operator interface and processing at rates of up to 100 Hz and a *DMX-1000* ultra-high speed microprogrammable audio processor. All software and user programmes are stored on daul 8in floppy disk drives. Unit also includes CRT terminal and real time control panel.

EMS (UK)

Datanomics Ltd/EMS, Westminster Road, Wareham, Dorset BH20 4SP. Phone: 09295 6311. Telex: 418480.

Vocoder 2000: compact vocoder system with 16 filters; 14, 6th order active bandpass and low and high pass filters, 16 envelope followers, dynamic range of 55dB, slew/freeze controls, internal oscillator and noise generator.

Synthi 100: contains 12 identical oscillators controlled by two 5-octave keyboards with individual outputs of sine, sawtooth, pulse or triangle waveforms. Other features include two noise generators, random generator, three envelope shapers, 16 filters, internal 256 sequencer under microprocessor control, separate microprocessor for keyboard scanning, analogue delay line, spring reverb system, joysticks, full patching facilities and double beam oscilloscope, digital frequency meter/timer/counter and optional digital voltmeter and sequencer free store display.

EMU SYSTEMS (USA)

E-mu Systems Inc, 417 Broadway, Santa Cruz, California 95060. Phone: (408) 429-9147.
UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-724 2451. Telex: 22278.

Emulator: not a synthesiser in the strictest sense, but a computer-controlled polyphonic keyboard. The *Emulator* has the capacity to 'record' sound into its digital memory and then allow that sound to be played at any pitch on the 4-octave keyboard with up to 8-note capability. Inputs for the digital memory can be from microphone, tape, record, another synthesiser or from a 5¼ in floppy disk, an integral part of the instrument. Each disk has a storage capacity of two sounds and sounds in the *Emulator* memory can be dumped on to it. Split keyboard capability allows two sounds to be played simultaneously. Modulation wheels for vibrato and pitch bend in addition to controls to sustain sound regardless of original length by defining a loop within the sound that is sustained as long as the note is held down. Multitrack polyphonic sequencer included.

FAIRLIGHT (Australia)

Fairlight Instruments, 15 Boundary Street, Rushcutters Bay, Sydney 2011. Phone: (02) 33 5222.
UK: Syco Systems, 20 Conduit Place, London W2. Phone: 01-724 2451. Telex: 22278.
USA: Fairlight Distributors LA, 1610 Butler Avenue, West Los Angeles, California 90025. Phone: (213) 478 8222. Telex: 9103426481.

Fairlight CMI: digital computer based synthesiser consisting of keyboard, video display, alphanumeric keyboard, CPU and two disk drives. The keyboard is 6-octave, velocity sensitive with 8-note polyphone capability with its own central processor. A second slave keyboard is available. The tuning of the keyboard is referenced to a master crystal 'clock' and the scale is normally equally tempered although it is easily changed. Controls and foot pedals, etc, assigned to a function after programming. Synthesis includes Fourier arbitary waveform, and video display with light pen. A waveform displayed on the screen can be modified using the light pen or originated on the screen. The alpha-numeric keyboard may also be used to modify the waveform. A 3-dimensional waveform of any sound can be displayed, showing change over time and tuning drift. Capability to sample signal from audio input – mike or tape, digitise, manipulate and pitch it to the keyboard to enable playing of natural sounds which then may be modified using the CMI facilities. Real time programmable sequencer which also stores key velocity information on to floppy disk with a maximum of 30 min of music. Merge capability between sequences, click-track generation and the ability to replay up to seven sequences while recording another. Full music composition language software (MCL). Eight separate channels of audio output are available for mixdown or multitracking. SMPTE coding available. Disk drives use two disks for system and the other for library.

HAZELCOM (Canada)

Hazelcom Industries Ltd, 39 Hazelton Avenue, Toronto, M5R 2E3. Phone: (416) 961-7090.

McLeyvier: digitally controlled analogue synthesiser. The basic system components include a 61- or 88-note piano-type keyboard, a disk storage unit, multiple digitally controlled analogue synthesisers, video display terminal which displays musical notes and notation as well as accepting commands in English language, integral typewriter keyboard, pen plotter with the ability to set down a musical score on paper complete with all notation.

LINN (USA)

Linn Electronics Inc, 18720 Oxnard Street, Tarzana, California 91356. Phone: (213) 708-8131.
UK: Scenic Sounds Equipment Ltd, 97 – 99 Dean Street, London W1V 5RA. Phone: 01-734 2812. Telex: 27939.
UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-724 2451.

Linndrum: programmable drum machine containing digital recordings of a wide range of drum and percussion sounds stored in computer memory and capable of storing as many as 49 different rhythm patterns which are user programmable in real time with adjustable error correction and complete editing functions. Snare, toms and congas are all tunable by front panel controls, or control-voltage inputs. Allows pre-programmed patterns to be sequenced for playback and dynamics, odd time signatures and 'human rhythm feel' are all programmable. All patterns remain in memory even when the unit is unpowered. Tape storage functions enable programmed data to be kept on cassette for later reloading. The *Linndrum* will sync to a wide variety of synthesisers and sequencers and can overdub to tape.

A useful feature is the provision of a front panel stereo mixer section with volume and pan controls which augments the unit's separate outputs for all sounds.

MOVEMENT COMPUTER SYSTEMS (UK)

Movement Audio Visual, 61 Taunton Road, Bridgewater, Somerset TA6 3LP. Phone: 0278 424560.

Percussion computer: computer-controlled percussion synth with VDU capability. Enables composition using real and synthesised sounds with fills, verses, time changes, etc, which can be assembled on screen. Visual edit capability. Programmes may be dumped on cassette. Uses 16k computer for which other programmes are available.

MOOG (USA)

Moog Music Inc, 2500 Walden Avenue, Buffalo, New York 14225.
USA: Norlin, 7373 North Cicero Avenue, Lincolnwood, Illinois 60646. Phone: (312) 675-2000.
UK: Moog Music, 11 Forth Wynd, Port Seton, East Lothian. Phone: 0875 812033.

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

John Donne, 1571-1631.

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

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

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

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

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

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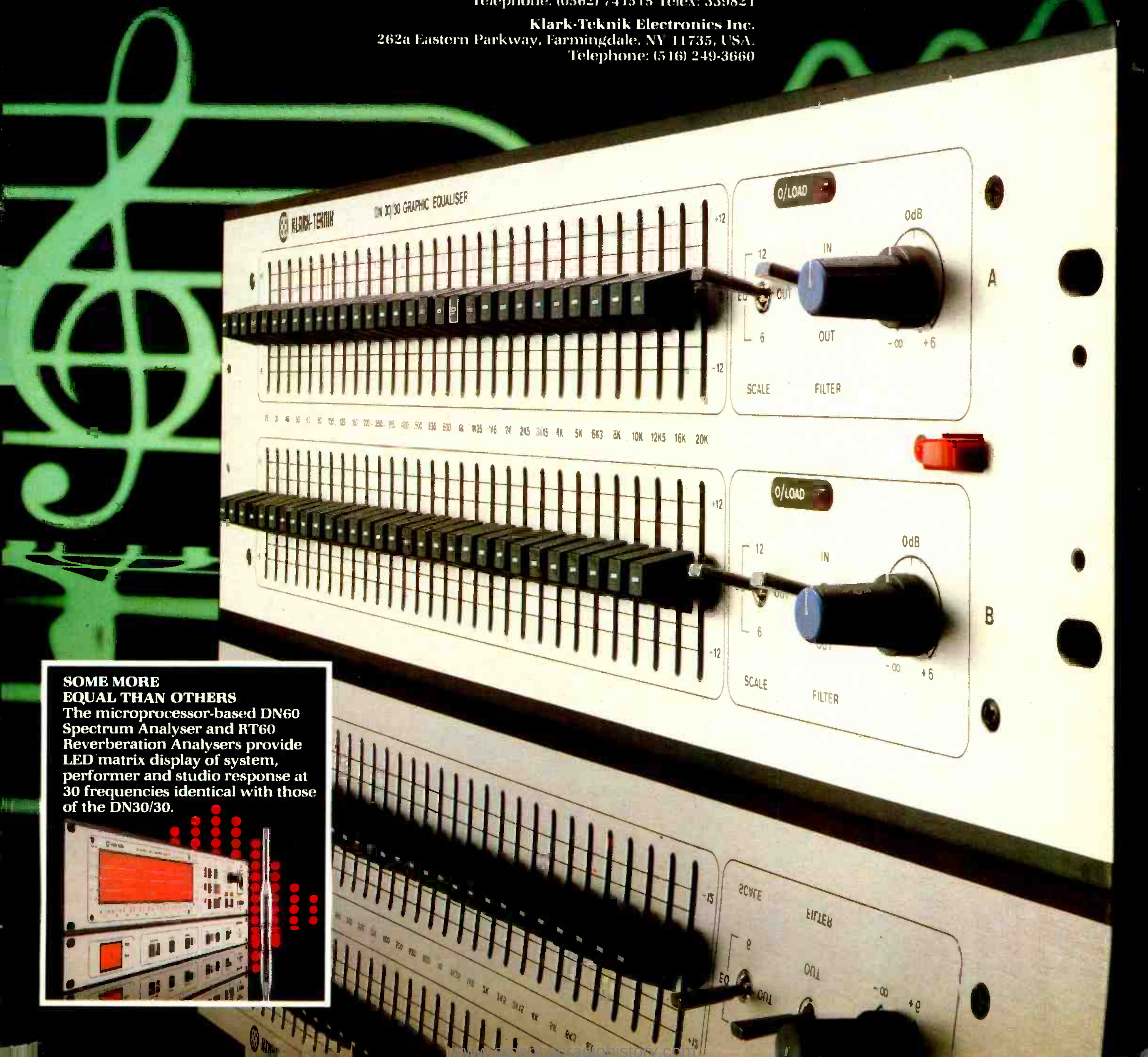
sound science

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

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



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

product guide
Synthesisers and Vocoders

Moog Studio Systems: three models – *15*, *35*, and *55* all modular for permanent installation.
Polymoog: polyphonic keyboard controlled synthesiser.
Moog vocoder: 16 channels with external patching between analyser and synth sections. Line and mike inputs 0dBm at 20kΩ and 40dBm at 30kΩ respectively. Carrier input 0dBm at 100kΩ. Outputs +10dBm and 100kΩ. 60dB S/N. Sample/hold facility to provide expanded control of momentary program input; foot switch jacks for remote switching of sample hold, patch select and bypass.

MULTIVOX (USA)

Sorkin Music Co Inc, 370 Motor Parkway, Hauppage, New York 11787. Phone: (516) 895-6300. Telex: 2279866.
UK: London Synthesiser Centre, 22 Charlton Street, London NW1 Phone: 01-387 7449.

Range of performance orientated synthesisers and a digital sequencer.

NEW ENGLAND DIGITAL (USA)

New England Digital, 48 N Main Street, White River Junction, Vermont 05001. Phone: (802) 295-5800.
UK: Turnkey, 8 East Barnet Road, New Barnet, Herts EN4 8RW. Phone: 01-440 9221. Telex: 25769.

Synclavier II: compact computer controlled digital synthesiser, available with eight, 16, 24, 32 voices or to special order 128. 5-octave, 61-note keyboard with split, delay function on low notes, auto arpeggio, repeat, frequency modulation, chorus and adjustment of the scale from equal tempered to whole tone scales, etc. or individual pitch adjustment of single notes. Automatic *A-440* tuning. Internal metronome. All functions controllable from front panel and no computer access necessary. Synthesis uses 'Partial Timbre' system with each partial timbre consisting of 24 adjustable harmonics, volume envelope generator, harmonic envelope generator, adjustable vibrato, adjustable portamento rate and special effects. Separate envelope generator for each voice. All programming is by multiple selection buttons and master control knob with digital readout. The disk drive is located under the keyboard with front access and each disk can be used to store up to 64 programmes or 15,000 notes. All recalled programmes are modifiable. Controllers include two foot pedals, six foot switches, ribbon controller and an optional velocity sensitive keyboard. 16-track digital memory designed to operate in a manner analogous to a tape recorder. Central processor is 16 bit, with 128KByte memory and can be stored in a flight case of less than 20sq in. Full interface facilities.

Analysis/synthesis option: based on a Winchester hard disk storage unit, the sampling system allows real sounds to be captured at a sampling rate of 50kHz (ie full audio bandwidth) for between 100s and 54min, depending on the number of disk units attached. Analysis of the waveform in both the time and frequency domain is possible, and software is provided to allow spectral analysis via Fast Fourier Transform functions on any sampled sound. Components of the sound may be digitally filtered for modification and reconstruction of the sound. On the A/D conversion side, 14- and 16-bit converters are available. Further sound modification and graphic editing of the sound is possible on the *VT640* graphics terminal and up to 24 sounds may be triggered from the Winchester drive. The sampled or modified sound may be performed on up to two octaves on the *Synclavier's* keyboard.

OBERHEIM (USA)

Oberheim Electronics Inc, 2250 South Barrington Avenue, Los Angeles, California 90064. Phone: (213) 473-6574.
UK: London Synthesiser Centre, 22 Charlton Street, London NW1. Phone: 01-387 7449.

OB-SX: pre-programmed polyphonic with integral keyboard. 24 presents on plug-in memory chip. Each preset can be modified by front panel controls but available as an option is a further 24 programme chip giving a range of 48 presets. Custom chips to order. the *OB-SX* is available with four or six voices; a 4-octave keyboard; portamento; LFO; detunable oscillators; filters; ADR; the 'Hold' position maintains the output of the keyboard for as long as desired with the note of chord being played when the button is selected. The 'Chord' position will transpose any note or chord held in the 'Hold' setting by playing a single note on the keyboard. Pitch bend and modulation levers. Oberheim computer interface and provision for external voltage control connections.
OB-Xa: programmable polyphonic keyboard-controlled synthesiser developed from the *OB-X*. Memory for 120 programmes. 5-octave keyboard range with split double combinations, transpose, split location and balance setting; auto-tune, continuous edit, polyphonic portamento; sample and hold, noise generator, pitch bend and modulation levers; two VCO, 2- and 4-pole filter and two ADSR per voice; two separate LFO. Cassette interface for recording and entering programmes with check facility for confirming accuracy of recently copied programmes. Computer interface.
DSX: digital polyphonic sequencer with a capacity for 3,000 notes polyphonically and in real time expandable to 6,000. Eight independently controllable CV and gate outputs with Oberheim and Moog triggers; editing and overdubbing facilities, alpha-numberic display, sequence merge facility, battery back-up, cassette interface, records all programme changes as well as notes, can be 'synchronised' to tape for multitracking, step by step mode, can be connected to other *DSX* for more elaborate systems involving more synthesisers.
DXM: programmable digital drum machine using real drum sounds stored in memory. Two programming modes, real time and single-step and can be used to create song structure, time signature, sequence length, dynamics and tempo. Digital alpha-numeric display; separate output for each voice as well as stereo output; external sync facilities; trigger and CV inputs allow external modulation of pitch and volume; battery back-up; cassette interface; 50 prerecorded sequences and 24 different drum and percussion sounds.

PAIA (USA)

PAIA Electronics Inc, 1020 West Wilshire Boulevard, Oklahoma City, Oklahoma 73116. Phone: (405) 843-9626.

P4700: computer synthesiser including 37-note encoded keyboard, D/A converters, control oscillators/noise source, balanced ring modulator/VCA, reverb springline, VCO, VCF, envelope generator. Includes *8700* computer, (*6503* microprocessor with random access memory and cassette interface) with software to control monophonic/polyphonic playing and sequencing.
Modular systems: *4700 Series* of modules includes VCAs, mixers, reverb, VCOs, VCFs, ADSR, control oscillators, power modules, joystick controllers, D/A converters, digital sample and hold cabinets and keyboards.

PASSPORT DESIGNS (USA)

Passport Designs Inc, 785 Main Street, Suite E, Half Moon Bay, California 94019. Phone: (415) 726-0280.

All *Soundchaser* systems require a 48K *Apple II* computer with disk drive.
Sounchaser Analog: polyphonic synthesiser consisting of 49-key 4-octave keyboard, interface card, *Analog Voice Cards* and the *Music Operating System* software. Features 3-voice analogue card with oscillator, filter and amplifier under computer control. The sequencer section is fully polyphonic with a 3,000 note plus capacity.
Soundchaser Digital: the basic comprises the standard 4-octave keyboard with interface card, the *Music System* digital synthesiser cards and the *Turbo-Traks* digital performance software. The complete system is a polyphonic synth with 16 digital oscillators, with 100 preset sounds and unlimited user facilities. Includes multitrack sequencer with 4,000 notes plus looping capable sequencer. The full *Soundchaser* system also includes *Notewriter* and *Music Tutor* software.

PPG (West Germany)

UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-723 3844.

PPG Waveterm: comprises the *PPG Wave 2.2* keyboard and the *PPG Waveterm* terminal. The *PPG 2.2* is a self contained digital synth. With its 2,000 waveforms it offers over 100 sounds that may be stored in the non-volatile memory. It has eight voices that can be filtered and modulated in real time with the 5-octave keyboard. It also contains an 8-track polyphonic sequencer with facilities for editing after recording as well as mixing the contents and then storing for instant recall.

The *PPG Waveterm* is a high resolution graphics terminal that extends the capabilities of the *Wave 2.2*. Facilities for additive waveform synthesis and natural sound sampling as well as mixing natural and synthesised sound. These sounds may then be loaded into the *Wave 2.2* and processed by its own controls for real time performance.

RHODES (USA)

Rhodes/CBS, 1300 E Valencia Drive, Fullerton, California 92631.
UK: CBS/Fender, Fender House, Centenary Estate, Jeffreys Road, Enfield, Middlesex.

Chroma: 16-voice polyphonic synthesiser with 50 user programmable presets and 100 more programs on cassette tape. 5-octave keyboard with velocity and pressure sensitive keys, cassette interface and additional interfaces for computers such as the *Apple*.

RIVERA (USA)

Rivera Music Services, 48 Brighton Avenue, Suite 11, Boston Massachusetts 02134. Phone: (617) 782-6554.

Rivera produce a comprehensive modification package for the Moog *Minimoog* synthesiser. The modifications include the provision of keyboard control voltage and gate outputs, plus oscillator and filter signal outputs, to allow external processing of the *Minimoog* signals. They also will modify and supply other makes of synthesiser.

ROLAND (Japan)

Roland Corporation, 7/13 Shinkitajima, 3-chome, Suminoe-ku, Osaka 559.
USA: Roland Corp US, 2401 Saybrook Avenue, Los Angeles, California 90040. Phone: (213) 685-5141.
UK: Roland (UK) Ltd, Great West Trading Estate, 983 Great West Road, Brentford, Middlesex TW8 9DN. Phone: 01-568 4578. Telex: 934470.

Digital sequencer CSQ-100: storage capacity 168 notes, 84 per channel. Calibration for matching any voltages differences between synthesiser and sequencer allowing use with any 1V/octave equipment; internal metronome; CV and gate information may be loaded independently; sequence may be programmed for stop and start positions; can be run one or two channel; continuous or one-off play; portamento control; full interface facilities and remoting connections.
Digital sequencer CSQ-600: storage capacity 600 notes, 150 per channel. 4-channel memory with rechargable memory with rechargeable battery back-up; memories may be replayed in any combination, singularly or together or in any order, updating of parts of stored material as well as forward and backstepping through the memory; programmable portamento including all the features of the *CSQ-100*.
TR-808: programmable drum machine. Enables up to 16 patterns to be programmed, 12 main patterns plus four intro/fill patterns which may be inserted into the rhythm manually or automatically. Individual drum sounds may be programmed with the preset buttons or by tapping in the rhythm manually in real time, each sound being selected by a rotary switch, with 12 positions, some 'instruments' being switchselected after the rotary selector. Additionally, pre-programmed patterns may be arranged and 'recorded' on to 12 tracks, each containing up to 64 patterns. If the 'recording' exceeds 64 measures, it runs automatically to the next track. Thus the rhythm for an entire piece may be programmed. Individual and mixed outputs are featured.
System-100M: fully modular analogue synthesiser system with modules including *110* basic VCO/VCF/VCA, *112* dual VCO, *121* dual VCF, *130* dual VCA, *131* output mixer, *132* dual CV/audio mixer and voltage processor, *140* LFO and dual envelope generator, *150* ring modulator/noise generator/LFO, *172* phase shift/audio delay/gate delay, *182* analogue sequencer. All the modules are fully inter-patchable using mini patch leads and the front mounted module sockets. There are three keyboards that may be used with the *System-100M*. The *184* is a 4-voice 49-note keyboard controller with four CV and four gate outputs. It also has an auto arpeggio function and a selectable assign mode. The *180* and *181* are slightly more basic 32- and 49-note keyboards with transpose and portamento control.
MC-4 Microprocessor: digital sequencer that will accept numerical keyboard inputs CV inputs from instrument keyboards and real time inputs. The output has four channels and this allows the control of four synthesisers. There are full editing functions such as copy, insert, delete step forward/back etc. The internal memory is 48KByte on the *MC-4B* and 16KByte on the *MC-4A* with the option of the *OM-4* expander for the *MC-4A* to increase the value to 48K. The *MC-4B* has an 11,500 note capacity and the remaining capacity can be displayed on the readout. Data may be dumped to tape.

RSF (France)

RSF, 19 Rue Claire Cazelles, F-31200 Toulouse.
UK: Syco Systems Ltd, 20 Conduit Place, London W2. Phone: 01-724 2451. Telex: 22278.

Polykobol: polyphonic 8-voice synthesiser individually assignable to two banks of presets.

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product guide
Synthesisers and Vocoders

64-position memory; cassette interface; autotune capability; programmable arpeggio; digital delay line; *6502* microprocessor; 5-octave keyboard; integral sequencer; modulation and pitch bend wheels.

SENNHEISER (West Germany)

Sennheiser Electronics, D-3002 Wedemark 2, Phone: 05130 8011. Telex: 0924623.
UK: Hayden Laboratories Ltd, Hayden House, Chiltern Hill, Chalfont St Peter, Gerrards Cross, Bucks SL9 9UG. Phone: 02813 88447. Telex: 849469.
USA: Sennheiser Electronic Corporation, 10 W 37th Street, New York, NY 10018. Phone: (212) 239-0190. Telex: 421608.

Vocoder VSM 201: comprehensive 20-channel vocoder with an analysis range of 100 to 8kHz. Each channel adjustable for emphasis and 'speech addition/multifilter' with between channel adjustments for silence bridging. Individual access to envelope for each channel. LED metering.

SEQUENTIAL CIRCUITS (USA)

Sequential Circuits Inc, 3051 North 1st Street, San Jose, California 95134. Phone: (408) 946-5240.
Europe: Sequential Circuits, PO Box 16, 3640 AA, Mijdrecht, Netherlands.

Prophet 5: completely programmable polyphonic 5-voice synthesiser. 5-octave keyboard. It comes with 40 pre-programmed patches all of which are modifiable or replaceable. Two oscillators with square and sawtooth on Osc A and sawtooth, triangular and square waveforms on Osc B. Separate pulse width controls on each oscillator. Osc B can be low range of 0.4Hz to 10Hz (in addition to separate 3-waveform LFO) as well as 4-octave tuning range on both. Outputs from B section can be used to modify frequency, pulse width and filter of A section or both sections can be 'hard' synchronised. Mixing section combines the outputs of A and B Osc with a noise source. Filter section is low pass 24dB/octave with resonance control. Two ADSR per voice. Programmed settings stored in five memory banks all recallable or recordable from the programmer section with the operative store being displayed by digital readout. Portamento control. Internal crystal referenced *A440* tuning oscillator. Pitch bend and modulation wheels. Back up battery in memory section. Interface to record memory programmes on to cassette for storage. CV, trigger, gating and sequencer interface jacks.
Prophet 10: 2-manual 10-voice polyphonic synthesiser, five voices for each manual. 64 programme memory, 32 for each manual. The VCO, VCF and VCA control section is similar to the *Prophet 5* with the addition of keyboard mode control, equalisation, voice to manual assignment and pedal control section. There is an optional polyphonic sequencer which can be installed on order or at a later date. Includes micro-cassette recorder for programme and sequencer storage.
Sequencer model 800: storage capacity 256-note with pitch and timing from synthesiser. Memory divided into 16 banks of 16 notes selectable or switchable in any combination in playback. Speed can be varied from ±15 times recorded tempo. Sequences can be stopped, started and stepped as needed. Single notes can be reprogrammed with rest of sequence being effected. Remote control facilities with full interface with *Prophet 5* and *10*.
Programmer model 700: digital programmer intended for use with synthesiser that don't have this facilities internally. It consists of two 5-control envelope generators, three voltage sources and auxiliary functions. The memory is arranged as eight banks with access to the eight programmes in each bank by push button giving a total of 64 programmes. All connections are both jack and mini-phono sockets as well as special 12-pin connector which matches with interfaces provided by other manufacturers or can often be fitted as required.
Polyphonic Sequencer: designed to interface with the *Prophet 5* and offers all the features on the *Prophet 10* sequencer 2,500-note storage, six separate sequences, instant transposition, overdubbing, editing, etc, and tape interface.

SERGE (USA)

Serge Modular Music Systems, 572 Haight Street, San Francisco, California 94117. Phone: (415) 6898.

Exceptionally large range of modular synthesiser components for assembly into a variety of comprehensive formats. Choice of over 40 modules.

SYNTAURI (USA)

Syntauri Corporation, 3506 Waverley Street, Palo Alto, California 94306. Phone: (415) 494-1017.

All Syntauri synthesisers are based around the 48K *Apple II* or *Apple II Plus* computers with language card, one disk drive and a video monitor.
alphaSyntauri Plus 4: programmable keyboard synthesiser using a standard 49-note 4-octave keyboard with a 3,000 note plus sequencer of single track or with the *SuperPlus* option for multichannel recording. 8-voice polyphonic capability using 16 digital oscillators.
alphaSyntauri 5: 61-note 5-octave keyboard with velocity sensitivity. The sequencer has the *SuperPlus* recording facility as standard with an option for *Metatrak* 16-track recording system. The unit includes a software metronome with one to 280 beats/min range. The keyboard has a split facility allowing up to 8 separate instruments to be played within user defined split points. 100 presets as standard with full facilities for waveform synthesis with Fourier, pulse waves and *Draw Waves* program.

SYNTON (Holland)

Synton Electronics BV, PO Box 83-3620 AB, Breukelen. Phone: 03462-3499. Telex: 40541.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1N 5PH. Phone: 01-580 4314. Telex: 28668.
USA: Big Briar Inc, Leicester, North Carolina 28748. Phone: (704) 683-9085. (for synthesisers)
USA: Synton USA, 269 Locust, Northampton, Massachusetts 01060. Phone: (413) 586-3777. (for vocoders)

Syntovox 221: 20-channel vocoder with filters at ¼-octave spacing and 54dB/octave slopes. The 20 control voltage outputs from the analyser section are passed to a 20 x 20 matrix enabling a variety of control permutations. The twenty channel synthesis section has one output and one input with identical filters to the analyser section. Each channel has a modulator with a control voltage input whose sensitivity can be adjusted. Multiway connector for external and computer interface. Fill-in facility and real time LED analysis read-out. Rack mounting.
Syntovox 222: simplified vocoder based on the *221* using separate analysis and synthesis but simplified for stage or studio use without the individual channel controls and matrixing. Rack mounting.
Syntovox 202: even more simplified version of the *222* vocoder intended for musical instrument use. Integral HF synthesis. Rack mounting.
Series 3000: fully modular synthesiser system consisting of 12 modules which can be assembled into a choice of two mainframe sizes. A 3½-octave keyboard is also available. The modules may be used for applications outside of this system and will mount into a 19in rack system. Modules include *3021* voltage controlled oscillator, *3017* voltage controlled state variable filter, *3224* voltage controlled variable band pass filter, *3209* dual VCA, *3211* ring modulator and random noise generator, *3023* LFO, *3015* mixer, *3005* envelope generator, *3223* dual LFO, *3233* sample and hold unit and LFO, *3235* envelope follower/comparator/trigger delay, *3022* binary divider, *3006* multiple, *3010* voltage controlled envelope generator, *3218* dual inverter/slew limiter, *3004* quad attenuator, *3025* mike/line input amp, *3024* line output amp, *3003* keyboard in/out.
Syrinx: versatile monophonic synth with 3½-octave keyboard.

YAMAHA (Japan)

Nippon Gakki Co Ltd, Hamamatsu.
UK: Kemble (Organ Sales) Ltd, Mount Avenue, Bletchley, Milton Keynes MK1 1JE. Phone: 0908 71771.
USA: Yamaha, PO Box 6600, Buena Park, California 90622. Phone: (714) 522-9105.

GS1/GS2: digital synthesisers designed to be musician orientated'. The *GS1* has a 88-note velocity and pressure sensitive keyboard while the *GS2* has a 73-note velocity sensitive keyboard. The *GS1* has a wooden cabinet and is intended mainly for permanent installation while the *GS2* has a more portable design. Both synthesiser have 16 voices with a range of master controls including a 3-band equaliser. The voices are programmed by 'voice cards' on which are recorded magnetically the characteristics of a particular sound. Up to 16 of these cards can be fed into the unit and assigned to the voices. Each keyboard is supplied with a voice library and this is being constantly added to. Copies of the cards may be made with the keyboard itself. Damper, tremelo and vibrato foot controls. ■

QUART INTO A PINT POT
...without spilling a drop!

The TRANSDYNAMIC teams with ADRs Compex, Express, Easyrider or Scamp processors to create a superb tri-band control system.

- **Production mastering — DIR-MIX™ a unique dynamic reduction system that transposes any input dynamic range to suit any dynamic format (disc, cassette, broadcast, film or PA). This means that for the first time very wide dynamic programme (especially digital) can be comfortably processed on to any medium without disturbing side effects. Livens digital dryness by enhancing low level detail and natural reverberant characteristics.**
- **Internal noise generator simplifies set-up procedure. It can be 'set and left' to a house standard, or used as a creative tool.**
- **Selective processing by dynamic tri-band equalisation (i.e. momentary attenuation of high level LF and HF) with absolute level control to maximise broadcast, cutting or recording level with minimal modification to original programme.**
- **Clean-up and restoration of older recorded material. Creating of synthesised 'stereo'.**
- **High energy Rock-box as a tri-band processor giving dynamic control of frequency response and optimising signal definition and average sound pressure level — all without modulation effects.**

letters

Calibration tapes

Dear Sir, We were interested to read Hugh Ford's review of calibration tapes in the 1982 June Studio Sound, *pages 66 to 76. His review is, as always, generally informative and accurate, but it appears that this time there are a number of errors which should be corrected for the benefit of your readers.*

Fig 2, 'Output with azimuth error' referenced on the left column of page 66 has been omitted. The Fig 2 in the middle column of page 66 is the Fig 2 referred to in the lower-right column of page 68, and should be captioned something like 'Low-frequency response of a reproducer'.

At the top of the left column on page 68, the statement 'in terms of nanowebers per millimetre tape width (nWb/mm) . . .' is in error. The correct SI unit sub-multiple is the nanoweber per meter, nWb/m, and this is correctly used throughout the remainder of the text.

Table 1 is untitled, but the text calls it 'Common fluxivities used on calibration tapes'. It is in fact 'common fluxivities used on European calibration tapes, and their values according to the ANSI measurement'. The columns should be labelled 'DIN measurement' and 'ANSI measurement'. As Table 5 shows, these are not values of fluxivity used in the USA. The value 270 in the USA column should read 220.

Table 2a 'CCIR' should not list a value for 30 in/s: the CCIR does not have any standard for program exchange at 30 in/s.

Page 68, right column, first complete paragraph, says '. . . transition frequencies . . . for the theoretical recording chain'. This is incorrect for any but the most simple-minded and useless of theories. I suspect that the author really meant '. . . for the theoretical reproducing chain'. Later it says that 'calibration tapes can even be used at speeds for which they were not intended by using corrections derived from Table 3'. While this is true, it's not as simple as this sentence would let you believe. You have to take into account the fact that the test frequencies and the apparent transition frequencies are both proportional to the reproducing speed. (The apparent time constants are inversely proportional to the reproducing speed.) If you want to check out your understanding of this idea, keep in mind that you can use completely interchangeably the AES 30 in/s, IEC-1 15 in/s and IEC-1 7.5 in/s reproducers and calibration tapes. Any of these calibration tapes will playback 'flat' on any of these reproducers. All *other non-standard combinations give non-flat response.*

The review makes a correct statement ('. . . to define the characteristic of the recording chain . . . fails . . .'), but the phrase 'as used in some standards' puzzles me. I am not aware of any audio standard that has attempted to standardise the response of a magnetic recording system in terms of the recording chain. I can only imagine that the author is referring to the confusing wording used previously in IEC Publication 94, third edition, 1968, in which the relationship between the input voltage and the recorded flux was called 'the recording (sic) characteristic'. This usage caused a number of engineers to believe erroneously that IEC was standardising the recording equalisation. That is why the new standard has abandoned the simple and euphonious expression 'recording characteristic' and replaced it with the much more specific expression 'amplitude/frequency response of the recorded short-circuit flux'.

Page 70, used 300 Ω as the resistance in calculating a capacitor for an RC filter. Since the circuit specified in the previous paragraph places two 600 Ω resistors in series, the correct resistance for the calculation is not 300 Ω but 1200 Ω.

The text, Fig 4, and Useful Formulae (e) all refer to the sin $\pi x/\pi x$ formula for gap loss, without mentioning that this is only a first approximation to the true gap loss formula which was given in 1952 by Westmijze. The error is less than 1% when 1.1 times the actual gap length is used in place of the actual gap length in this approximate formula, and x does not exceed 0.6. Since Fig 4 goes well above $x = 0.6$, it shows the wrong slope for the locus of the maxima (the correct slope is approximately 4 dB/octave, not 6 dB/octave as shown) and the wrong frequencies for the nulls (the null frequencies are not actually harmonically related).

In Fig 5, the frequency scale appears to have been displaced by one decade.

All of the MRL calibration tapes listed in Table 5 have a fluxivity of 200 nWb/m according to the ANSI measurement (220 nWb/m according to the DIN measurement), not 185 nWb/m and 200 nWb/m respectively, as listed here. And, most horrible of sins, you have misspelled our name in this Table—we are a Laboratory, not a Library.

MRL tapes do in fact conform to IEC 94-2, as is mentioned in our literature. Perhaps the author had an early sample which did not so state.

Page 76, right column, second complete paragraph, may be unclear as to the recorded levels. All tapes contain 1000 Hz at the reference fluxivity (variously 200, 250 or 320(DIN) nWb/m) at both ends; the 7.5 in/s tapes have the spot frequencies at a level of —10 dB, while all of the other tapes have the spot frequencies at 0 dB.

Yours faithfully, John G Knight, Magnetic Reference Laboratory, 229 Polaris Avenue, Suite 4, Mountain View, California 94043, USA.

Hugh Ford replies: I am indebted to my friend Jay Knight for detailed comments on the review of calibration tapes in the June 1982 edition of *Studio Sound*.

Somehow the correct Fig 2 has been omitted and the published Fig 2 shows the low frequency effects in a replay head related to pole piece design. Also the top line on page 68 should read nanowebers per metre (nWb/m), not nanowebers per millimetre.

A further sin, my Table 1 has become modified such that 270 should read 220 and the headings

Experts' Errors

Dear Sir, Oh dear. It appears that I and a lot of other people have been under a misapprehension for a long time . . . 0 VU doesn't equal +4 dB! (Experts Errors, page 66 October issue).

If Mr Davis is correct, why do VU meters made by a well-known British company have their scales marked 0 VU = 1.228 V, why does the BBC Engineering Training information sheet, 'Radio Broadcast programmes and their measurement', say the same thing, and why does Hugh Ford make the error in his review on the Tomcat cartridge machine (page 84 in the same issue)?

Perhaps someone should have a look at the original Bell Labs specification.

Yours faithfully, D S Buckley, 46A Pinner Road, Harrow, Middx HA1 4HZ.

Dear Sir, Regarding the Don Davis article 'Experts' Errors' in the October 1982 issue, I wish to comment on his first two examples.

While 0 VU on a true 'Standard Volume Indicator' will equal 0 dBm across a 600 Ω resistive termination, very little, if any, contemporary equipment adheres to this standard. '0 VU' has become a nominal operating level selected to optimise headroom, S/N, and equipment interfacing. Different segments of the audio industry have adopted their own 0 VU references. For example: 0 VU for broadcast is usually +8 dBV (ref 0.775); 0 VU for studio is usually +4 dBV (ref 0.775); 0 VU for hi-fi is usually —4 dBV (ref 0.755); 0 VU for personal multitrack is usually —10 dBV (ref 0.775).

Zero VU on most tape machines refers to a level of flux density on the tape. Input and output level controls make actual interface levels variable. Perhaps the book definition should consider popular usage.

In the second example, the proper usage of decibels is the one describing a power ratio. Use of decibels to describe voltage ratios is allowed only because voltage and power are mathematically related (voltage varies as the square root of power into a given load). Technically decibels should not be used to describe the voltage gain of a transformer as there is no power gain (such measurements should be so qualified).

Yours faithfully, John H Roberts, Loft Professional Audio Products, Phoenix Audio Laboratory Inc, 91 Elm Street, Manchester, Connecticut 06040, USA.

The mistake in *Experts' Errors*, Example 2, was in fact made by our printers after we passed the pages, and this is why we didn't spot it. They transposed the lines referring to voltage and power ratios, so Don Davis is in fact correct. We apologise for our 'Printer's Errors'.—Ed.

Dear Sir, There is probably more confusion about levels than about any other subject in recording, therefore the last thing needed is additional confusion especially when it is paraded as enlightenment. I refer of course to the article 'Experts' Errors' by Don Davis in the October issue.

His Example No 1 is a mixture of pedantry and obscurity. The quotation of the definition may be accurate per se, but a standard VU meter would not be used in such a fashion. The programme signal is rectified by a copper oxide rectifier which is within the meter case. This rectifier has an impedance which changes with applied voltage, a fact which would cause unacceptable distortion of the waveform on the line. Therefore a series resistance of 3,600 Ω is inserted between the line and the meter, and it is this combination which results in an indication of 0 VU for an input voltage of 1.228 V RMS (representing +4 dBm).

A VU meter is a working tool thus any description which deliberately avoids its manner of use is at best worthless.

Yours faithfully, S W Davies, 30 Strutton Ground, London SW1P 2HR.

Dear Sir, We applaud your publication of the article entitled 'Experts' Errors' in the October 1982 edition of Studio Sound. *Mr Davis points out some common mistakes, many of which are considered 'absolute fact' by engineers who are otherwise quite knowledgeable.*

Please keep up the good work with your excellent publication!

Yours faithfully, Richard Cook Jr, Communication Arts, 2526 Twenty-Seventh Avenue South, Minneapolis, Minnesota 55406-1393, USA.

should read European measurement and US measurement—or more correctly DIN measurement and ANSI measurement.

Mr Knight is at pains to point out that the 35 µs time constant at 30 in/s is not a current standard but only used for replaying 'historic' recordings and whilst he says that it was not a CCIR standard I believe that it was a proposal.

The subject of recording or reproducing characteristics has for a long time been an area of confusion in various national and international standards even to the extent of incorrect formulae being 'standardised'. I believe that the current international standards for magnetic tape (but I think not magnetic film) standardise the reproducing characteristic. Certainly the recording characteristic has been used as a standard: I quote from British Standard 1568: Part 1: 1970 which is not alone in its intentions. 'With a constant electromotive force applied to the input of the recording chain the curve which gives a variation in surface induction with respect to frequency . . .' and goes on to state 'The corresponding reproducing characteristic . . . is that which gives a flat response . . . with the relative surface induction stated above'.

In the case of magnetic film British Standard 3154:1959 only mentioned the reproducing characteristic as notes referred to in the text by asterisks!

In the second paragraph of page 70 the capacitor when using a flux loop should be T/1200µ F with a 600Ω source and load—not T/300 as stated and the fluxivities for the MRL tapes in Table 5 should read 200 and 220 nWb/m for the ANSI and DIN reference levels respectively.

The frequency scale in Fig 5 is displaced by a decade and should be the same as that in Fig 4.

Thank you Jay for your help and it wasn't me that called your firm a Library—it should of course be Magnetic Reference Laboratory.

Digital fad

Dear Sir, I'm with Peter Fellgett; it seems to me that I have been trying during my professional career to make things simple, swimming against the current. I have seen tracks proliferated (and degraded), and noise reduction band-aids applied in lieu of improved basic engineering; digital audio recording strikes me as a gigantic technological overkill.

Once the public has gotten over its open-mouthed awe of such scientific magic, it will discover that the 'digital' product, as an analogue pressing, doesn't sound significantly better than discs they could have bought 20 years ago. And when it does sound better, it is because the inflated price has enabled the manufacturer to take more care in the physical production, achieving better electroforming and pressing.

My memory goes back to the time when the magic term was 'hi-fi'. Later came stereo, then Dolby (at one point, one couldn't sell a non-Dolby record, no matter how good it sounded) and now we have 'digital'.

The proponents of this system plan to market a fiendishly complicated audio storage method at hardly low prices to a public which in general seems satisfied with the reproduction from audio cassettes. I know a US record company whose best-selling record, clearly marked stereo, *is in fact completely mono. What's even more interesting is that they apparently didn't know it until I told them. This same public has had great difficulty assimilating stereo (not to speak of those many recording specialists whose preference for good old mono causes them to eliminate as much difference information as possible; my term for such recordings is 'monereo').*

Tony Faulkner has jumped on the bandwagon, and he's probably having a pleasant ride; but most of his technical criticisms of analogue recording (especially ½ in 2-track 30 in/s) are based on factors that are not fixed (ie, bass non-linearity is a playback problem; properly designed play heads can largely eliminate it).

And the actual digital disc may work well in the laboratory, but in the hands of the consumer? Assuming he can be induced to purchase it? Quad was killed by a combination of consumer resistance and corporate stubbornness. Unless digital can compete in price with analogue, it will probably go the way of quad, once the novelty has tarnished. Anyone opting for evolution over revolution is often accused of being anti-progress. I don't oppose 'real' progress.

Over a decade ago I was told, rather belligerently, that if I did not have a ¼ in 4-track quad machine I was not really with it since quad was the wave of the future. 'The clouded crystal ball' indeed!

Yours faithfully, David B Hancock, 127 West 88th Street, New York NY 10024, USA. ■

EVERY summer a caravan of American jazz musicians treks around Europe, playing a series of festivals. There's Nice and Antibes in France, Montreux in Switzerland, Knebworth in Britain and North Sea in Holland. Solo musicians are faced with different backing groups in each country, and only the musicians of featured bands know who they will be playing with. No one knows what kind of sound system they will be encountering from gig to gig.

Contrast this with a travelling rock show, like the Rolling Stones circus, where the band travels with an army of engineers manning a barrage of sound and light technology. Give or take a few problems, the travelling rock band knows what it will sound and look like at each venue. Compared with this happy state of predictability, jazz music is still living in the Middle Ages. But the wind of change may soon start blowing, because audiences who pay up to £10 each to attend are getting restive.

A comparison was planned between the technology backing the Stones circus in Britain and France and the local technology which greeted the summer caravan of jazz musicians festivalling at around the same time in the same countries. But thanks to the Stones' monumentally unhelpful management, PR and concert promotion entourage in both countries, we concentrated on the jazzers instead. In the event, making a comparison between the technology and techniques adopted for the jazz concerts in Nice, France and Knebworth, England proved far more rewarding than trying to cut through the moody arrogance which enveloped the Stones tour. What emerged was an interesting pointer to likely future trends.

The 1981 Nice Jazz Festival relied on native French sound crews on all the three open-air stages which were set up in the gardens of the Cimiez Monastery, high up in the mountains backing the city of Nice (see *Studio Sound* December 1981). But for the 1982 event there were some pretty significant changes. The main Arena stage was once a Roman amphitheatre and is now regularly used for concerts of all types. After the 1982 Nice Jazz Festival there was a spectacular open-air version of Aida in the Cimiez Arena. For the jazz, the feed from the microphones on stage in the Arena was split between the PA system and mobile control trucks of the French radio station, France-Musique, parked outside. France-Musique records everything for future transmission. Wisely the radio engineers record their own mix. Wisely, because, with astonishing lack of sensitivity, the jazz festival organisers built the PA sound mixing booth high up on scaffolding to the extreme side of the stage. So, cursing in French, the PA engineers had to mix the PA sound balance almost entirely by guesswork. As the system used a bank of ancient Altec plywood bass bins and even more ancient *805* horns, it's not surprising that the Arena sound was almost consistently terrible. Too much bass, too little piano, dead vocals, overload distortion. You name it, the paying audience heard it.

Things were almost as bad on the second stage, the Dance stage. Last year the French record company Black and Blue had been taking a split feed from the Dance stage mikes and recording for commercial release. But this year Black and Blue pulled out of the festival and the previous year's creditable Bose system was replaced by some very basic 2-way Cerwin-Vega and Italian Semprini cabinets, four each side of the stage. And, yet again, the festival organisers had built the sound booth right to the side of the stage where the engineers couldn't possibly mix by ear. So on the Dance stage, as on the Arena stage, each musical performance was preceded by a ludicrous mime show. American and French engineers and band roadies waved hand signals from the stage to the audience to the mixing booth and back again to get the on-stage monitor and audience PA balances half-way to reasonable. Apparently, the French hadn't heard of on-stage mixing for the band monitors. And why build a mixing booth to the side of the stage? Apparently no one had told the festival organisers not to. "It's like no one told us not to eat the daisies," moaned one American helper. So why not move the sound mixing booths to a reasonable position in centre front out in the audience once the problems became obvious? The answer to that is in the French *laissez-faire* character. Because the festival lasted 10 days there didn't seem to be any hurry to get on with the job. And by the time the festival was half-way through there didn't seem to be any point in making the change anyway!

Before long most of the groups in search of a decent sound were asking to play on the third, Garden, stage. This is in a beautiful olive grove and for the first time ever the sound system had been rigged, and engineered, by a British firm. Starhire of Baldock, Hertfordshire, had toured with Miles Davis earlier in the year. The promoter for the Miles tour was Simone Ginebrie who, with American promoter George Wein, is now behind the Nice Jazz Festival. Simone Ginebrie wanted exactly the same Miles sound system for Nice. Starhire said yes, provided they got a sound mixing booth centre front out in the audience. On a take it or leave it basis, they got it from Day One. The Starhire system uses a Turbosound desk and amplifiers, with a 4-way speaker stack rated at around 10 kilowatts. In addition, there's around 3 kilowatts of stage monitor power, with on-stage monitor mix.

66 ▶


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Live jazz sound

Capital Jazz Festival: general view and (below) Starhire's Turbosound console at Nice

Benny Goodman live at Knebworth

All pretty routine for a modern rock gig, of course, but light years ahead of the Nice jazz norm. Inevitably there were problems. The power supply had to be tapped off the local street lighting circuit and the single phase supply voltage was often down to well below 200 volts. One night there was a sudden leak to earth which shut down everything with a bang, blowing the output stages of several amplifiers. And although the groups all sent in their stage mike plans ahead of time and promised to turn up in advance for sound checks, the stage plans all turned out to be wrong and often the groups either turned up late for the sound checks, or not at all.

But the real problem was, and will doubtless remain, political. The Nice Festival is part-sponsored by the city of Nice. And, understandably, the city fathers want to see a local French firm booked to handle the sound. This has worked reasonably well for the last eight or nine years of the Festival because it's been a fairly homely and happy-go-lucky sunshine affair. But now, with economic pressures forcing the promoters to book ever-bigger names, and many of them jazz-rock fusion bands which rely on a heavy duty sound system, it's a different story. That's why Simone Ginibrie booked Starhire to repeat the Miles Davis experience. But not surprisingly it didn't go at all well with the French when a bunch of Brits steamed in with a pantechnicon of sophisticated equipment and produced the best sound at the festival. It went down even less well when some groups refused to work on the French stages.

All eyes will now be on Nice in 1983 to see whether the Brits are asked back, and if so, how many of the three stages they are allowed to handle. Audience attendances at Nice this year were clearly down. The city fathers may have to own up to the stark reality that no one these days wants to pay around £7 for a concert ticket to hear a bass that sounds like over-amplified cardboard, a grand piano that sounds like a club upright and a singer's voice deadened through the wrong EQ from a mixer out of earshot. Some of the big names in electric blues and jazz-rock may well refuse to appear unless there's a change in policy.

Capital Radio, main sponsors and organisers of the Knebworth Jazz Festival that followed fast after the Nice shindig, were brought face to face with exactly the same problem. The Knebworth sound system, provided by the London firm Asktam, was every bit as professional as the Starhire system at Nice. And bigger, too. To cope with a fast turn-round between bands, the whole Asktam system was duplicated with two sets of stage mikes, and two Midas desks side by side in the mixing booth out centre front in the audience. On stage there were two stacks of Midas and H/H amplifiers and two matched pairs of 4-way Martin speaker stacks, one for each half of the stage. While one band played on one side of the stage, the next set up on the other. By lining up the mikes on the band setting up through one desk, while mixing the live sound on the other, Asktam could switch instantly between bands.

Often, the time between 'thank you' and 'please welcome' was as little as two minutes flat. With around 30 kilowatts of power, on-stage monitor mixing, and virtually instant switch-over, there was, in theory, no problem about keeping the paying audience of around 10,000 people a day happy. And most of the time the majority of the paying audience were happy. There were some hiccups early on when the EQ settings for an electric rock group (like the Average White Band) obviously and audibly didn't suit a more acoustic jazz group (like Dizzy Gillespie), but for the most part there were no complaints. Until, that is, Benny Goodman appeared on the last night.

Benny Goodman was, 40 years ago, crowned the King of Swing. He made his name with live shows in front of a big band that didn't need much amplification and he made his small group and big band recordings with nice loose miking in cosy studios. These days he's far more interested in Mozart than jazz. Obviously rooted in the past and committed to an acoustic sound, Benny Goodman insisted on loose miking at Knebworth. He'd allow only a single mike on the piano and string bass, with just two mikes high over the drum kit and a few spot mikes out front for the soloists. But when Goodman soloed he played way off mike with his clarinet, the best part of a yard from the nearest pick-up point. Goodman also insisted on playing without any on-stage monitors. In other words, he treated the giant open-air stage at Knebworth, with an audience of nearly 10,000 stretching at least a hundred yards up the hillside, as if it was one of those cosy recording studio dates of the '40s and '50s. The result was utterly predictable. The mikes picked up as much background noise as music and with the risk of feedback, the 30 kilowatt system couldn't do more than keep the first few dozen rows happy. The level of gain being used was all too obvious when Goodman tapped his foot on the wooden stage. It sounded like thunder through the PA system. Before long, while the small Goodman band and a few hundred people in the audience grooved happily to an almost natural acoustic sound, the remaining thousands heard nothing and started to chant "Can't hear, can't hear". Benny Goodman, always famous for his martinet approach, simply growled to the audience (close into the mike so that they could hear): "If you kept quiet you might be able to hear the music." They did, but couldn't. After the performance the Capital compere came on stage and disclaimed all responsibility. A few people subsequently asked for their money back. They'd paid £10 each and had not heard what they paid for. A Capital spokesperson said something irrelevant about Benny Goodman being upset by the London terrorist bombings and complainants got a token refund.

The sad truth is much more basic. To afford artists of the calibre of Benny Goodman, a festival has to be big and sell thousands of tickets. And if the paying thousands are to hear what they've paid for, there has to be a powerful amplification system.

It's true that jazz started out in brothels and small clubs, but economics have forced its growth into large festivals. No technology can communicate the intimacy of a small band in a club to an open-air audience of thousands. The best you can do is rely on a well-tuned sound system, manned by engineers familiar with, and sympathetic to, the music. It's unrealistic to expect success every time with a different sound system and engineers in every city, often with no interest or experience in jazz.

Already, some of the pop-jazz-rock acts are taking their own sound engineers on the European festival circuit. It can't be long now before the whole caravan of jazz artists starts to tour Stones-style with its own sound system and engineers. ■

Soundcraft automation

Last month's article detailed the design parameters of the Soundcraft automation system. In this second part, Graham Blyth describes the operation of the system, hardware and software requirements, and planned future developments

THE operation of the fader and mute mode change controls is best understood by referring to **Fig 3**. A long push of any switch is defined as any time greater than 50 ms, this time being modifiable in software. All local mode changes occur at the moment the switch is released, ie, if you are in UPDATE and wish to return to READ at a precise moment, the mode switch may be pushed down any time more than 500 ms before the desired moment and released exactly on time. The local mute mode switch does not distinguish between short or long pushes because there are only two modes anyway, READ and WRITE. Master mode changes occur as the master switches are pushed. Short or long push is therefore again irrelevant.

MUTE WRITE is entered by pressing MASTER MUTE WRITE or local MTE buttons, if previously in MUTE READ. A red LED indicates the MUTE WRITE mode. Any local or group mute operation causing the green on/off LED to change state will now be recorded by the computer on to tape (the exception being solo mutes if SOL SFE is selected). As mentioned in part one, time delay, both absolute and relative between channels muted simultaneously, is very critical as far as mutes are concerned. Packing the mute information into one byte for eight channels together with an address defining those channels enables a mute change of up to eight channels to be written on less than 2.5 ms of tape, and the entire console in less than 10 ms.

MUTE READ can be entered by pressing MASTER MUTE READ or local MTE buttons, if previously in MUTE WRITE. The red LED will extinguish and the green on/off LED and hence the channel on/off status is entirely controlled by the data coming off the tape. Operation of the local on/off switch is ignored as is any A or B group mute operation. The important exception is that the console Solo In Place system does still mute all channels not soloed and not safe.

FADER WRITE can only be entered by pressing MASTER FADER WRITE (the reasoning behind this decision is explained under REWRITE). It is essential to enter this mode, together with MUTE WRITE, when commencing an automated mix or any part of the recording process with the computer switched on. This is also the only mode in which VCA sub-groups may be set up. The computer now records on to tape the exact position of the faders at any moment in time, ie, if the fader position is at 20 dB attenuation, that is the level written on to tape.

FADER READ is entered by pressing MASTER FADER READ or by a short or long push from WRITE mode or a long push from UPDATE or REWRITE mode. The channel level in the mix is now entirely controlled by data coming off the tape and by any VCA subgroup or grand master fader (unless they are also in READ). Movements of the channel fader will have absolutely no effect on level or on any data in the system.

AUTO READ is essentially a subroutine within the system software. Its purpose is to enable automatic entering of READ from UPDATE or REWRITE, this operation being achieved by holding down the FDR mode switch and moving the fader in the direction indicated by the two green nulling LEDs. At the point that the fader position corresponds to the data coming from the tape, the channel automatically jumps into READ with no change of level. The FDR switch can then be released at any time. Obviously no change to READ should be executed either by the normal long push or using the automatic facility unless the engineer knows that the mix balance from that point on is exactly how he wishes it to be. There is a very good case for leaving a channel permanently in UPDATE, which is really READ plus the ability to make adjustments to previous data, ie, if the balance is right, leave the fader alone, if it isn't, you can do something about it.

FADER UPDATE is reached by pressing MASTER FADER UPDATE or by a short push from READ or REWRITE. UPDATE is an auto-nulled mode, ie, the fader's position as the mode is entered becomes a 'no change' point for previous data. Therefore, by not moving the fader, the data is unmodified and essentially you are in READ. By moving the fader so many dB up or down, you will increase or decrease the level in the mix of a previously recorded pattern. This pattern might be quite complicated such as a stereo shuffle on two channels of Hammond organ as the keyboard player executes a glissando. By entering UPDATE on the two channels concerned, simple movement of the two faders will compensate for over or under exuberance. Having achieved a reasonable mix in WRITE and listened to it in READ, it is advisable to switch the whole console to UPDATE in order to add subtleties, and only if an entirely new pattern is required should any faders be

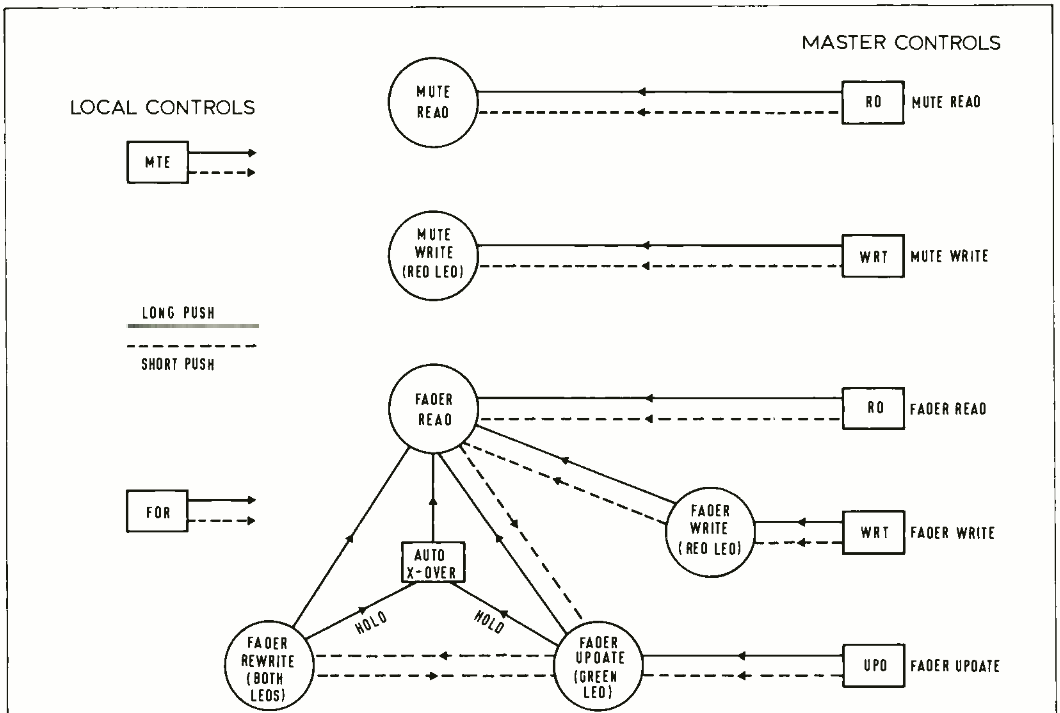

FIG.3 MODE CHANGE STATE DIAGRAM

a design study—part two

Graham Blyth (Soundcraft)

switched to REWRITE. This rule applies to all automation systems that offer UPDATE.

FADER REWRITE can only be entered by a short push from UPDATE. REWRITE is another auto-nulled mode so that there is no change of level at entry point irrespective of where the fader is positioned. The mode may also be entered by two short pushes from READ, going rapidly through UPDATE (auto-nulled so no level change). REWRITE is, to all intents and purposes, exactly the same as WRITE in that entirely new level patterns may be created. The difference is that the data written does not necessarily correspond to the position of the fader knob on the front panel. At the moment of entry, the computer looks at the data going on to tape (say —10 dB), looks at the fader position (say —5 dB) and says 'OK, we'll make these the same by subtracting 5 dB from the fader level from now on'. The engineer may now use the fader in a conventional manner to create a new level pattern. Though the fader can be in any position before entering REWRITE or UPDATE, it is sensible to position it between say 0 dB and —20 dB otherwise control range will be rather limited. Having a continuous fader, such as the one developed by Paul Buff of Allison or the Travis fader from Sphere, is the ultimate solution to all positional problems, but the cost is high. When leaving REWRITE and going to READ or UPDATE, a jump in level can occur since the data from tape, which is what READ and UPDATE respond to, could be very different from that going to tape. If the difference is large then you should not be planning to change mode anyway, but, if you insist, AUTO READ may be invoked or the fader moved as indicated by the nulling LEDs (which are brightness modulated over a plus and minus 5 dB window) before changing mode. Soundcraft are currently investigating two software approaches to letting the computer do all this work automatically.

VCA sub-groups and grand master

All the operations concerned with the VCA grouping system are controlled by the individual channel SET buttons (**Fig 1**) and the SET/CLR button (**Fig 2**). The individual large red LEDs above the SET buttons are used to display which faders are masters, which, if any, is the grand master and which channels are assigned to each master (interrogation mode). The procedures for operating the system are set out below.

Sub-group creation—This can only be achieved if the console is in WRITE mode and the fader chosen to be a master is positioned above the —30 dB mark. While holding down the SET button of the master channel, push in turn the SET buttons of the channels chosen to be the slaves. The master channel LED will blink and the slave LEDs will illuminate permanently. Having completed this process, release the master SET button, at which time all LEDs will extinguish except the master, which will be permanently illuminated. The action of assigning a channel as a slave of a master will not cause any level change of that channel. A channel may be deleted from a group at any time by holding master SET and pressing the relevant channel SET button, at which time the channel will return to its previous level before any master differences were forced upon it.

Grand master creation—The same restrictions apply as before (WRITE mode and fader above —30 dB). While holding down the SET/CLR button in the master control panel (**Fig 2**), press the relevant channel SET button and then release SET/CLR. The channel LED will blink slowly, indicating 'grand master' status. Operation of this fader will now affect all channels, not just those assigned to masters.

Interrogation—Any number of VCA groups may be created, for example, every alternate fader could be the master of its neighbour for stereo channel creation. Each master and the grand master are clearly visible by permanent or blinking LED illumination. It is possible, however, to forget which master controls which slaves. Fortunately the computer never forgets and Soundcraft have set up an interrogation system to answer questions. Holding down any slave or master SET button will extinguish all other LEDs except those belonging to the sub-group in question. Slave LEDs will illuminate permanently and the master LED will blink, the blink speed being noticeably faster than the grand master blink.

Master and grand master isolate—A difficulty associated with using channel faders for two functions, master and actual channel, is that if for example you have set up a drum group with the kick drum as the master and the kick drum is, on afterthought, too loud or soft relative to the rest of the kit, what do you do? Traditionally you gradually adjust the kick drum while simultaneously moving all the other faders in the opposite direction, a very tricky move. Fortunately the friendly computer comes to our aid and makes the whole operation ridiculously simple. A short push on the master SET button causes the steady LED to blink and the fader to revert to controlling the kick drum only. No level change in any of the channels will occur. The kick drum may now be adjusted without affecting the level of the rest of the kit. A second short push restores the master control, the LED illuminating permanently, with again no level change on any of the channels, the whole procedure being as fast as the engineer wishes. This same feature is also available on the grand master fader.

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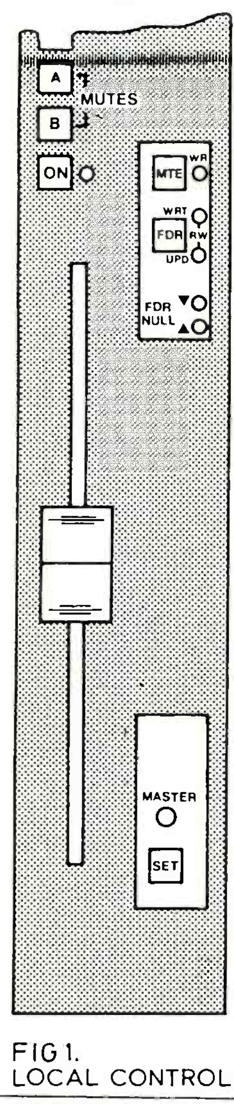

FIG 1.
LOCAL CONTROLS

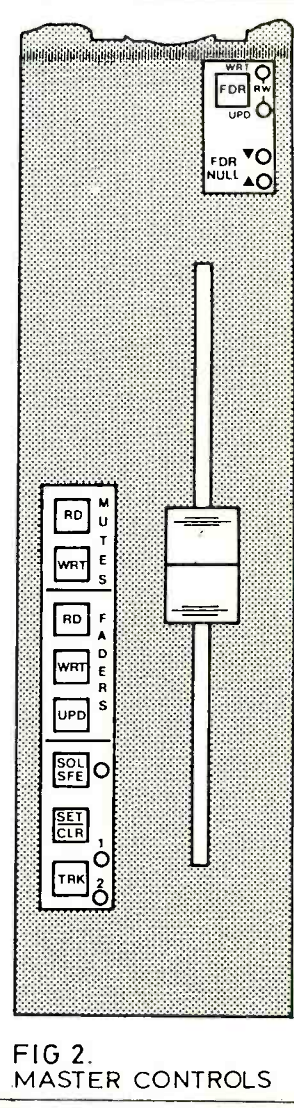

FIG 2.
MASTER CONTROLS

Soundcraft automation

Illegal operations—Any attempt to do something that the current issue of software has ruled out of order will be greeted by a 'beep' from a little transducer fitted to the computer rack underneath the console. No harm is done by attempting such misconduct, the computer just issues an aural slap on the wrist and carries on with its job. One example of an illegal operation would be attempting to assign a slave to a new master without first removing the slave from its previous boss. There are several others.

Data handling—Any master fader movements are written with the same priority as any other fader or mute operation. Movement of a master fader has no effect on any slave channel data being written on to tape. This is a very important point since, if this were not the case, moving the grand master, for example, would cause a huge string of channel priorities for every movement seen by the computer and hence potentially large bounce delays building up with each fresh attempt at a mix. The computer also records all the channel assignments on tape so that following an overnight switch off of the system, the sub-groups may be recovered and the mixing process continued.

Hardware

The hardware is made up of the computer, the data buss and the individual channel interface PCBs, linked by the data buss. The computer consists of three PCBs. The processor board contains the microprocessor, system memory, system software and sufficient I/O porting to interface with the other computer cards and a future master/slave multi-processor system. The analogue I/O board is essentially a high precision, high speed data acquisition and transmit system. It receives and transmits analogue and digital information from and to the data buss. Analogue to digital conversions are done to 12-bit accuracy, the result being anti-dithered, and logarithmically converted in the processor board.

It was in the design of the A/D system that Soundcraft expected to find most of the problems. The situation could and would arise that on two adjacent channels, one fader could be sending 10 V and the next 1 mV. Bear in mind that the data buss is about 8 ft long, serves up to 28 channels, contains high speed digital as well as multiplexed analogue data and that Soundcraft had allowed only 70 μs to sample, convert and send back a modified analogue voltage to the VCA on each channel. To remove any D/A crosstalk problems and any ground noise the fader output voltage was multiplexed differentially. This worked very successfully but some difficulty remained in coping with the 80 dB of voltage range. A combination of careful buss loading and force clamping the sample and hold capacitors to ground as soon as a channel conversion was completed, solved this problem.

The tape I/O board contains the very minimal amount of electronics necessary to send and receive data to and from a tape machine. A phase-locked-loop allows substantial tape speed variation while still correctly decoding data. The code format on tape is self-clocking and uses the time between flux reversals to establish whether data is a 0 or a 1. The system works at an average bit rate of 15 Kbit/s. Since each data word contains address and data it is very important that any error in data and address is picked up so that, for example, the correct channel is muted rather than its neighbour, with disastrous consequences. In addition to conventional odd/even parity a very brutal error-checking system is used. In all the time that this system has been developing incorrect data has never been received on a channel. The detected error rate seems to be about two per minute (ie, two in a million received bits). This was established using a conventional 24-track machine at 15 in/s using edge tracks.

The data buss is a 40-way ribbon cable connecting each channel to the analogue I/O board. It contains the necessary audio and digital power supply voltages and grounds, an 8-bit digital data buss, analogue sends and returns, addressing for up to 64 channels and a function address for up to 64 functions per channel. These functions may be all analogue, all digital or any combination of the two. Each digital function is 8 bits wide and so could be configured as a bank of eight switches. A rough calculation shows a fully equipped 48-track input module to be about 36 functions so the capability of the system should last for a while.

It is intended that a large console would be handled by a number of slave systems controlled by the master central processing unit, which also would supervise all disk activity. The reason for additional slaves is that it could take too long to service 64 × 64 functions with one slave. This, however, might not be the case. As mentioned earlier, a disk system can have a resolution of, at best, 20 ms. It might take 200 ms to scan and prioritise 4,096 functions but if these were handled with a certain 'pecking order', eg, mutes, faders, pan, auxiliaries, EQ, filters, etc, and further to that, any low-priority function could jump up several levels following noticeable activity, then one slave could possibly be man enough for the job. It should also be remembered that even two engineers can only do so much at one time, thus the queue of priorities to be served can never be that long.

Software

The significant and perhaps most difficult feature of a properly working priority encoding automation system is that it is totally asynchronous. There is no way in which the computer can know where the next priority is coming from; off tape, from the faders or from the mutes (or anything else in a more complex console). The total solution is to have a Real Time Multi-tasking Operating System, which takes man years to write or costs vast sums in licence fees. Soundcraft's approach was to design a real time system with a preferred operating sequence and priorities assigned to these operations. The system is still multi-tasking in real time but not in the purest sense. Because, initially, the tape is being used as the data store and there is concern about bounce delays building up, the highest priority is given to data off tape. As soon as the tape I/O board signals to the processor that it has decoded a data word, this word is grabbed and tested to see if it is a priority. If it is, then the computer will make sure that it is the next word put back on to tape. If it isn't and a new priority is not generated from the last scan of the channels, the computer will make a decision as to whether it is more important than the routine word that is next in line as part of the sequential scan process.

Generally 'off tape priority' beats 'new mute priority' beats 'new fader priority'. When there is no priority data, which is the case most of the time, the system writes routine words in the following order: MUTE 1-8, FADER 1, MUTE 9-16, MASTER 1, MUTE 17-24, GROUP DATA 1, MUTE 25-28, FADER 2, etc. Any detected priority will be inserted into this pattern, the intention of which is to guarantee that the full console status be recovered rapidly should the tape be stopped and spooled forward or back. You will notice that mutes predominate the routine scan. This is so that if a drop out occurs and masks a priority mute, that mute will be picked up very rapidly from routine data, certainly within 20 ms.

System speed

Due to the random nature of a multi-tasking system, it is impossible at this stage to say exactly how long is the delay to record, for example, a mute on tape or what the accumulated bounce delay will be after a number of mix attempts. The software has been designed fairly conservatively to accommodate such things as a priority 'log jam' which may not be a critical issue in the real world. The first system has just commenced its useful life at Riverside Recording in Acton.

Issue 2 software should now be blown into PROM and installed with all customers, and that is not the end by any means. All that can be said at this stage is that each data word occupies on average 2.5 ms of tape, bounce delay should average just under 4 ms (extremes have been measured at 2 and 8 ms) and that the total console status will be recovered between 500 and 1,000 ms following any spooling operation. Initial set up is established within 200 ms so tape should be pre-rolled slightly before making the first mix attempt.

Future developments

Apart from the occasional software updates based on user feedback, Soundcraft's next major task is to design the disk based system that can be added to the existing system. This project is scheduled for conclusion by the end of 1983. Very few design decisions have been made so far but it has been decided to use a large solid state memory (at least 64K) for all real time data manipulations, using the disk only for keeping and recovering mixes or sections of mixes.

The reasons are twofold: firstly, RAM is very cheap these days and allows high speed data manipulation; secondly, floppy disks do not enjoy constant reading and writing, they also tend to make a lot of noise when entering a read or write task. These problems are not so great with Winchester hard disks but unfortunately a 5¼ in Winchester may be too costly. At least one additional floppy drive would always be necessary for taking backups to store with the master tape and for loading up software. ■

Author's note: I would like to record my grateful thanks to the following people who have been involved to various degrees in the automation project. Christoff Heidelberger, who joined Soundcraft in 1979, brought the microprocessor into the company. Christoff designed all the hardware, specified all the software and wrote a considerable amount of it himself. Ron Taylor-Lewis wrote all the rest of the software and Dave Dearden helped me define the functions of the system. And last, but by no means least, Mrs Margaret Mason, who typed this script for me!

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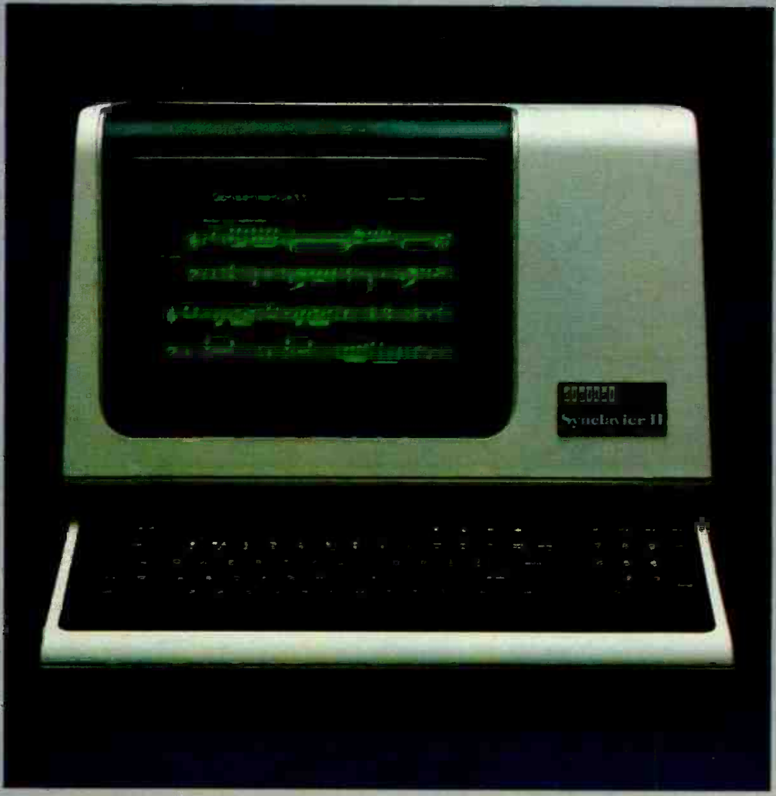

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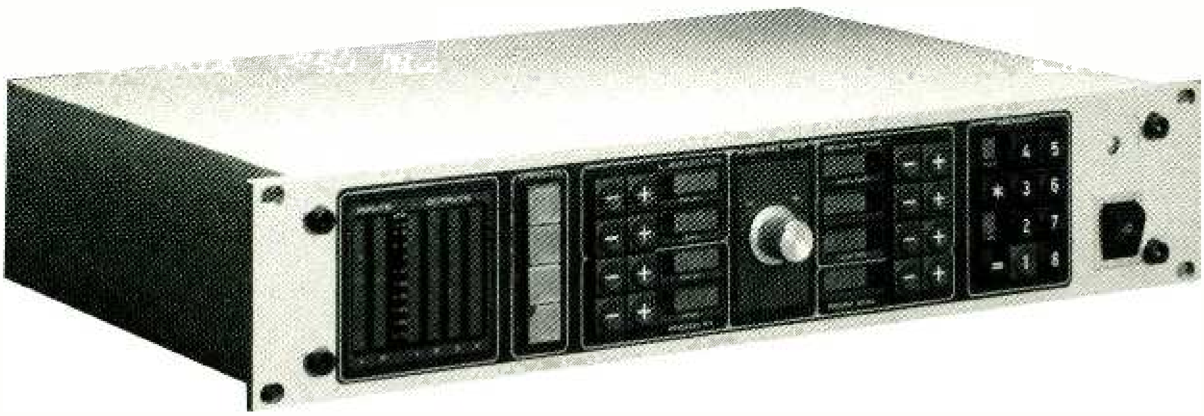
Quantec Room Simulator



MANUFACTURER'S SPECIFICATION
Programs
Reverb
Room size: 1 m^3 to $10^6 m^3$ in seven steps.
Decay time: 0.1 s to 100 s (up to 400 s at 40 Hz).
LF decay times: Coefficient of 0.1 to 10 with 11 steps related to selected decay time.
HF decay times: Coefficient of 0.1 to 2.5 with eight steps related to selected decay time.
Reverb density: more than 10,000/s, average three per 1 Hz of bandwidth depending on room size.
Pre-reverb delay: 1 to 200 ms is steps of 1 ms (optional 800 ms in steps of 4 ms), level −30 dB to 0 dB in steps of 1 dB, plus 'off' function.
1st Reflection: 1 to 200 ms in steps of 1 ms (optional 800 ms in 4 ms steps), level −30 dB to 0 dB in steps of 1 dB, plus 'off'.
Enhance: Simulation of rooms with no perceptible reverberation, seven programs.
Freeze: Special loop program with infinite decay time to add any number of acoustical entries.

Digital
A/D converter: 16-bit, sampling rate 20 kHz, distortion 0.1% typical.
Processor: 26 bit, clocked at 20.48 MHz.
Memory: approx 2 MB of RAM.

Analogue
Inputs: two, balanced, isolated by opto-couplers. Input impedance 13.2 kΩ balanced, 6.8 kΩ unbalanced; level adjustable −20 to +6 dBm. Headroom 12 dB above nominal level; RF filter 18 dB octave beyond 100 kHz.
Outputs: four balanced. Outputs 1 and 2 reverb plus 1st reflection; 3 and 4 for quad use. Output impedance: 100 Ω balanced, 50 Ω unbalanced; minimum load 1 kΩ. Nominal level adjustable −6 to +6 dBm.

General
Dynamics: better than 85 dB unweighted, typical 90 dB, for all decay times.
Frequency response: 20 Hz to 8 kHz, +0/−3 dB.
Power: 220 V, 50/60 Hz, 80 VA (optional 117 V).
Connectors: *XLR-3*.
Dimensions: standard 19 in rack mounting width, height 2 units (260 mm).
Weight: 5.5 kg.
Remote control: identical to front panel, display and operation simultaneously.
Remote cable: twin-screened cable up to 30 ft; optional up to 600 ft.
Protection circuits: prevention of transient clicks; against over-voltage supply; non-volatile memory and settings in case of power failure.
Manufacturer: Quantec GmbH, Postfach 152, D-8016 Feldkirchen bei München, West Germany.
UK: Syco Systems Ltd, 20 Conduit Place, London W2.

THE Quantec *Room Simulator* (let's just call it the Quantec or *QRS*) is a digital reverberation unit designed for studio applications, and is a smart, 2U-high rack-mounting unit whose brushed-aluminium front panel forms an integral heat-sink for the innards. Numeric displays indicate the various settings, which may be adjusted with up/down 'nudge' buttons or with a central rotary incremental control.

Fig 1 is a block diagram of the unit and what it is designed to simulate, namely a room fitted with two loudspeakers (the two inputs) and four microphones (the four outputs). Predelay may be inserted in the inputs, and two of the outputs may have an additional '1st reflection' repeat added if desired.

The unit was evaluated during a week of recording a library album of what might best be described as 'synthesised classical-style' music at Herne Place Studios (featured in *Studio Sound* January 1982). The fact that the Quantec was invariably recorded on the 24-track tape, and that a Lexicon *224* of one sort or another was available throughout the recording and mixing period enabled me to contrast the effect of the

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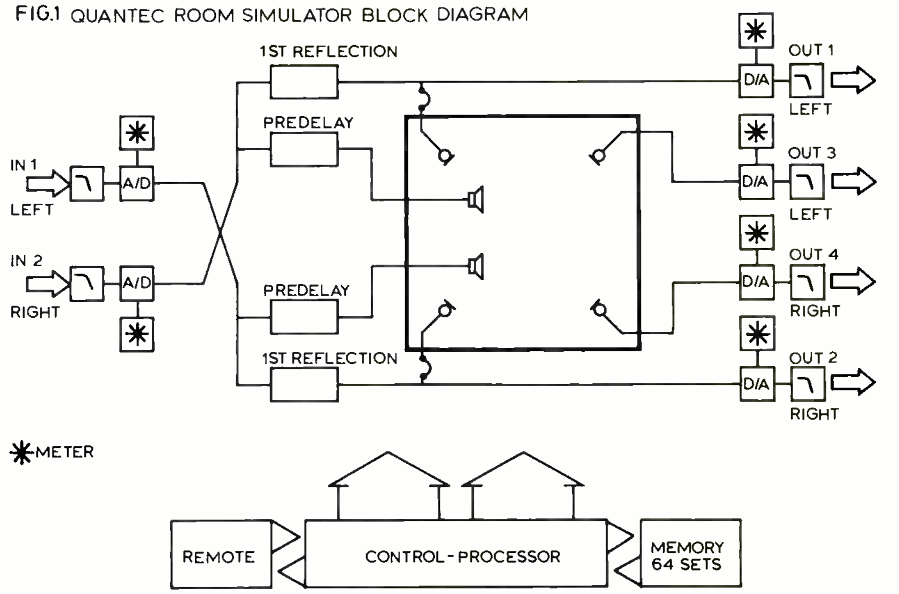
FIG.1 QUANTEC ROOM SIMULATOR BLOCK DIAGRAM



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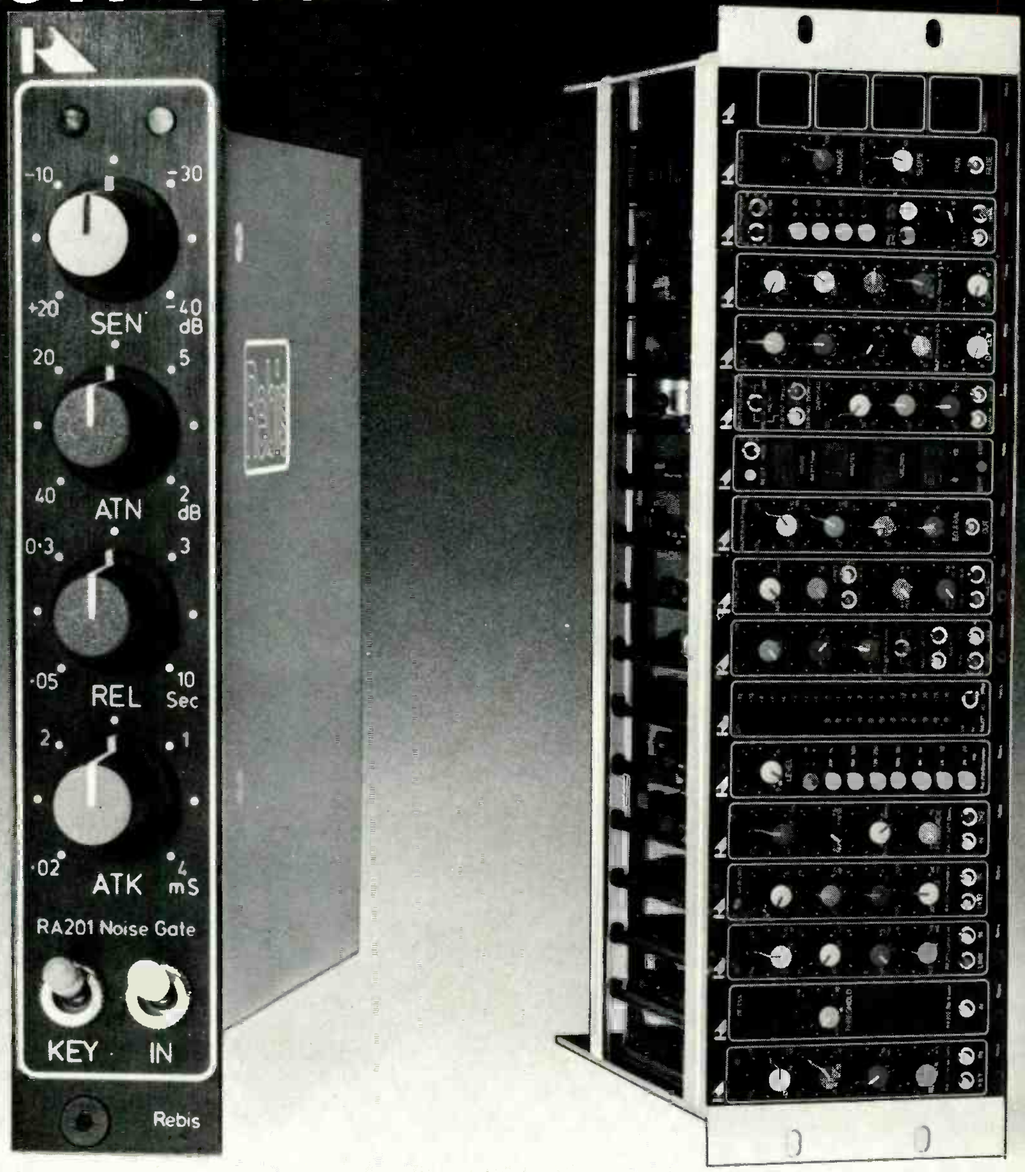
The No1 Gate



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Solid State Logic, Stonesfield, Oxford, UK (099 389) 8282 two devices, both of which are presumably aimed at the same market. A review of the *224X* appears separately.

The Quantec was in use throughout the week, safely mounted in the ancillary rack, and this, along with the fact that we had one of only two units in the country at the time, ensured that I did not attempt to get inside the box, so I have no comments on the internal construction or circuitry. This is therefore an operational appraisal of the unit.

Confronted with the compact unit for the first time, it is noticeable that the rear panel is clearly labelled in German and sometimes English – all of which is clearly understood. Power is via a standard fused IEC socket with voltages and fuse value clearly labelled. Neutrik *XLR-3* connectors are used for the two inputs and four outputs, and these are wired correctly and are clearly labelled in English, including a note that outputs 1 and 2 include the 'first reflection' capability. Interfacing with the Trident *Series-80* console patchbay was simple, with no level or matching problems experienced.

Turning to the front panel there are, from left to right, LED bar-meters for each input and output channel, notable for their calibration from clipping (+12dB) to −74dB, thus allowing visual indication of 'something happening' even if you can't hear it – useful; four 'effects' push-buttons, labelled 'enter', 'freeze', 'mute', and 'enhance', to which we will return later; a set of variable parameters; a memory storage panel; and, lastly, the illuminated power switch. The parameter-control section is the heart of the unit, and needs detailed explanation.

The parameters are in four groups, each parameter being adjustable in two ways as described, the incremental control varying a given parameter after one of the relevant 'nudge' buttons has been depressed. The LED display representing the currently-selected parameter flickers to indicate that it alone will change if the incremental control – a continuously-rotating knob – is twiddled. The four parameter groups are: 1st reflection, which has variable delay (1 to 200ms in 1ms steps) and level (zero to −30dB plus 'off' in 1dB steps); predelay, variable as above; room size – the central feature of this unit – variable between 1 and $10^6 m^3$ in seven steps; and decay time, which is variable from 0.1 to 100s basic. The decay time is modified by two controls, high end and low end. These apply multipliers to the basic decay time between 0.1 and 10 in 11 steps (LF) and 0.1 and 2.5 in 8 steps (HF). The majority of 'straight' reverb sounds are created with the reverb time and room size parameters.

The *QRS* is designed to be thought of as a real room with variable parameters. These are size (volume) and, if you like, acoustic treatment. You can select an almost infinite variety of room simulations from a cupboard (with or without contents) to a cathedral. The room size is set with the control, and then the effective reverb time is set up. The HF and LF 'absorbency' can then be adjusted as appropriate. Here lies the great innovation in the control aspects of the unit: it is very simple to adjust. Having got a sound in this manner it can be stored simply with the memory file system (which is non-volatile) and recalled when you next need it. The ease of setting up, however, meant that I seldom used the memory function, preferring to dial up a sound from scratch for something different every time, much as I use a variable synthesiser with or without storage facilities. There are so many possibilities, that duplication by accident is unlikely, only by design. And the easily-grasped setting-up procedure makes it easy to 'visualise' the type of 'room' you want.

Unlike a number of digital reverb units, this machine offers you room simulation rather than reverb effects (although there are also some specialised effects too). It will not readily simulate a plate, which some might say is just as well. What it *does* allow is a wide variety of real-sounding rooms and acoustic environments. These may be modified by adding an effective reflecting wall at almost any 'distance' (the 1st reflection controls) and the onset of reverb may be delayed without the need for an external tape loop or delay function. This gives you all the simulation you'll ever need of 'real' environments.

The Quantec in use

The Quantec was patched into the system in a conventional manner, as a reverb unit fed by a stereo echo send and brought up either via the monitor echo return or more usually via the line inputs of a pair of channels, allowing the signal to be routed to tape with its source. For musical reasons, echoes were largely applied to tape, as a composite balance between direct and reverb signals. This, incidentally, led to interesting developments that were entirely appropriate for the type of work we were doing: the monitor mix at any given time was virtually as the master turned out after mixing. The advantages of this method were that there was more flexibility with the two reverb units (*QRS* and *224*) and other effects units; the track developed in the studio with all the echoes and effects present, so it was always known how it would sound, and how it was *meant* to sound; and the mixing was very straightforward, leaving all the effects and reverb units in the mixing room free for added, rather than basic, effects. The disadvantages were that the form of the track, and its direction, had to be determined during the recording process only: mixing, whilst very rapid, depended on first finding the balance and direction already established during the recording phase, with only subtle variations of balance being possible. As we had already determined what the balance should be, however, this was not a problem.

The Quantec was used for two main functions: straight reverb and special effects. The reverb was usually applied to solo instruments, such as the *Variophon*, which was used for brass and woodwind lines. Generally, these were recorded with reverb on two tracks, the source instrument being centred in the stereo picture. The first noticeable effect of the Quantec was that, unlike some other reverb units, the device did not 'spit' or 'ring' when presented even with hefty transients, except *very* occasionally on very low room-volume settings, and this was often due to overdriving of the inputs. Only channels 1 and 2 were used as returns, these being the ones with '1st reflection' capability. The room-setting and 'absorbency' controls were found to be very smooth in action, contouring the sound most effectively although, of course, discontinuities were experienced whilst changing the sound with a signal passing through, which is neither surprising nor objectionable. Changes to room volume with an input signal caused muting of the output during the alteration: a good idea. The

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pre-reverb delay was most useful, and offered delays which covered a very satisfactory range, with plenty of adjustment but not too much coarseness. Similarly, the first reflection function performed well and was very useful. It should be noted that the first reflection repeat appears on the opposite side to the input: thus, a transient applied to the left channel will appear 'reflected' in the right output. This is quite logical and a useful effect.

Comparing the *QRS* with the Lexicon *224* it was immediately noticeable that the sound differs. This is hardly surprising: the Quantec was not designed to particularly conventional acoustic theories. It could be said that the *224* is 'warmer' than the Quantec, but this would be misleading. The *QRS* definitely sounds like a selection of variable rooms, and the Lexicon sounds...different. It is difficult to be more precise. Both have their applications: taste and try, before you buy.

Effects

The effects functions of the Quantec put it into a class of its own. Firstly, there is an 'enhance' function which leaves only the room volume control operative. This is designed to simulate rooms 'without perceptible reverberation', and has a very curious effect which is very difficult to describe. At very low room-volumes, it is reminiscent of being in a heavily padded cupboard; at high room-volumes....well, have *you* ever heard a cathedral with no perceptible reverberation? No, neither have I. Hence the description difficulty. I must admit that I did not use this effect very often. It would appear to be excellent on drum tracks, or even on rhythm machine. Only the latter was used, and only occasionally. The button has a push-on, push-off action with LED indicator.

The 'enter', 'freeze', and 'mute' buttons relate to each other. 'Enter' and 'mute' have a push-on, release-off action, while 'freeze' has an LED indicator and is software-latched like the 'enhance' facility. 'Mute' simply dumps the output signal and 'clears down' the reverb system. No sound emerges while the button is pressed, new reverb starting as soon as it is released. The 'freeze' function enables a signal to be input and looped indefinitely – but you don't hear the original signal. What you hear is a constantly-developing reverb signal which shifts and changes with time. Indeed, a simple 3-note chord can be entered and the resulting reverb can be heard for half an hour without boredom. The entering process is simple; pressing 'freeze' holds any existing input signal. Hitting mute then kills any existing reverb. Then a signal is entered by pressing the 'enter' button, inputting a sound, ending the sound, and releasing the 'enter' button, in that order. Pressing or releasing the 'enter' button while an input signal is present can cause glitches, of course: hence the above procedure. The 'enter' button allows a number of signals to be overlaid. For instance, one could sing a note, then another, then another...building up a reverberant chord. This can be allowed to decay by pressing out the 'freeze' function. We used the 'freeze' facility frequently, often playing a string-like chord into the unit and recording the output on a pair of tracks. On one number we did this four times with different chords, and laid out the basis of the track simply by crossfading the chords at interesting times. The effect was simply stunning, and while it could have been achieved in other ways, none would have been so convenient, or easy to accomplish.

Conclusions

There is a danger, with the possibilities of modern digital electronics in the studio, that a manufacturer will discover a unique box that can produce a wide number of effects, all controlled by one processor/memory system. The result is a unit which is supremely cost-effective, in terms of effects-per-pound, but also limiting, because the box will only do a couple of effects at once. In the case of a signal-processing effect unit, this is less trouble, because often effects are put down at the original recording stage, sequentially. You don't need too may effects at once. On the mix, you have hopefully put down enough effects on tape not to run out of them.

Reverb units are a different matter, however. A reverb unit is used virtually all the time on a mix, and is seldom used very much prior to mixing. A manufacturer deciding to release a reverb unit that also does effects, then, must be careful to offer the right sort of effects. They must be those that will have application *before* mixing, and not so much during it, otherwise they will never be used. Alternatively, the unit must be of such a cost that it would be worth the money for reverb alone (usually a more expensive occupation than mere effects), in which case the effects come as a 'free' bonus. And if its primary job is reverb, it must do it well. The Quantec succeeds admirably on all three counts. The 'freeze' effect is very much an overdub tool (or as part of basic tracks, as it was used for this review), and the 'enhance' mode, while related to reverb, could also be used in recording rather than mixing. While in the mix, the *QRS* produces a flexible range of reverbs which will offer most, if not all, of the functions that would be produced by anything other than a plate. The one mod I would suggest would be to add a pair of auxiliary inputs that feed in *after* the predelay, thereby allowing the unit to be used simultaneously for delayed and non-delayed reverb, much as one uses a plate with a tape-delay in the send. This is, however, only a minor observation.

Overall, the unit behaved impeccably, with few untoward effects that were due to other than misuse. Noise and distortion, as far as can be said, were insignificant. The limited frequency response was never of the slightest concern. The comprehensive file-storage facility, offering eight locations in each of eight stores, uses a simple but effective fail-safe logic which makes it easy to use, a typical application being to allocate each 8-position file to one engineer. Only location 1 in each file may be modified, and this is also a useful feature: stored settings may be moved into position 1 for modification without losing the existing stored setting. However, the supreme ease of use of this unit makes extensive memories less necessary than on other, more complex units which offer more facilities (and cost a good deal more too!). It must be said that, apart from the effects, the *QRS* limits itself to simulating rooms, but this is no bad thing at all.

It is worth noting that there is an interesting relationship between the four outputs of the unit. This seems to indicate that the unit would function admirably in ambisonic applications, where artificial reverb has been a problem for some time. Feeding the Quantec outputs into a ambisonic transcoder would yield a useful reverb field, difficult to obtain in other ways.

The Quantec *Room Simulator* is an excellent digital reverb unit at a good price, and will no doubt find plenty of good homes in studios where high quality, flexible reverb is needed within a reasonable budget. **Richard Elen**

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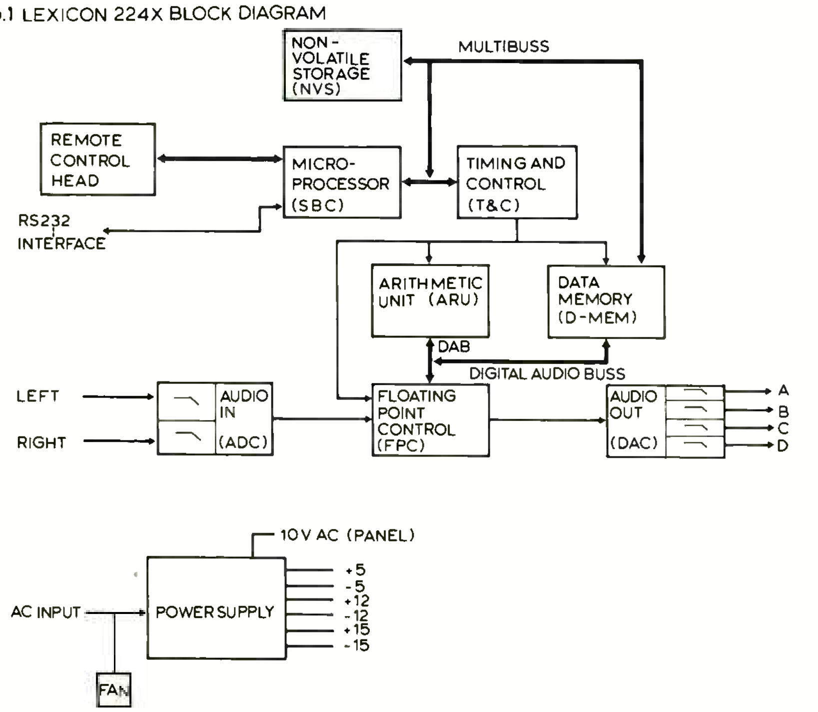
Lexicon 224X

MANUFACTURER'S SPECIFICATION
Program capacity: up to 32 different basic programs.
Register storage: 36 registers (non-volatile), four main and 32 extended; each register stores a basic program and the user's complete parameter settings.
Reverberation time: adjustable in two bands from approx 0.6 to 70s (program-dependent).
Frequency contouring: 6dB/octave filters in digital domain: crossover between reverb bands; treble decay; system bandwidth; all adjustable between 170Hz and 15kHz.
Depth control: controls relative strength of early and late reverb; in acoustic programs, adjusts apparent location of pickup in simulated space.
Pre-delay: program-dependent; minimum values 0 to 24ms, maximum values 100 to 400ms.
Additional controls: eight program-select buttons; page/immediate select; two mode select; shift; four register select; six display select.
Display: 3-digit numeric; dual 5-position LED headroom indicator; overload warning; units indicator.
Frequency response: 20Hz to 15kHz ±1.5dB; 20Hz to 12kHz ±0.5dB.
***Dynamic range:** reverb mode: 84dB typical, 81dB min. rel to reference level at 20Hz to 20kHz noise bandwidth for all reverb times from 0 to 10s.
Non-reverb mode: 90dB typical, 86dB min at 20Hz to 20kHz.
THD plus noise: 0.04% typical, 0.07% max at reference level for all reverberation times between 0 and 35s.
Interchannel crosstalk: −55dB at 1kHz.
Inputs: two, balanced and transformer isolated; impedance 20kΩ; adjustable from +8 to +18dBm.
Outputs: four, balanced and transformer isolated; output impedance 90 Ω; output level +8 to +18 dBm.
Remote console cable: 25ft standard; 50ft optional.
Power: 100, 115, 200, 230V switch-selectable; 50/60Hz; 180W.
RFI shielding: AC power connector, audio connectors, and console cable.
Protection: mains fused; secondaries fused; voltage crowbar and/or current limiting; thermal protection.
Connectors: audio *XLR-3*; power standard IEC 3-wire; remote and option *DB-25*.
Power-on muting: 3s.
Serviceability: field-serviceable; logic modules and each major assembly removable.
Diagnostic programs: automatic at power-on or reset; control and display via remote control panel.
Environment: operating: 0 to 35°C; storage: −30 to 75°C; relative humidity 95% max (without condensation).
Size: mainframe: (whd) 19×7×15 in (483×178×381mm). Console: 5.4×8.8×3in (137.2×223.5×76.2mm).
Weight: mainframe: 34lb (15.5kg); 48lb (22kg) shipping. Console: 2.5lb (1.2kg); 6lb (2.7kg) shipping.

[*In Concert Hall Reverberation Program with input sensitivity set so that 1kHz, +12dBm input corresponds to 0 LED just going out (this is Reference Level). Output sensitivity set to produce +12dBm with 600Ω load in self test mode (unity gain).]

Lexicon Inc, 60 Turner Street, Waltham, Massachusetts 02154, USA.
UK: Scenic Sounds Equipment Ltd, 97–99 Dean Street, London W1V 5RA.

FIG.1 LEXICON 224X BLOCK DIAGRAM



THE Lexicon *224X*, developed from the widely-used and highly-respected *224*, definitely comes into the 'all-singing, all dancing' category as far as effects units are concerned. It offers a far wider range of effects (several of which are totally new, while others are updates) than the original unit. The chances of being able to cover everything in a single article are remote, but the primary functions are less difficult to elucidate. As usual with this type of equipment, we feel that an operational assessment will be of more practical use than a purely technical analysis.

First impressions

I had never totally come to terms with the original *224*, and on seeing the *224X* package, the reason became obvious: I'd never seen a manual for the unit previously and had been forced to work it all out for myself. As a result I missed out on some of the features of the original unit. I was lucky enough to have the manual for the *224X* over a weekend before using the device on a session, and as a result I had already acquainted myself with the machine before I started. The manual is one of the best I have ever seen, which is just as well as its detailed perusal is mandatory. This is *not* a unit to be tackled with no background. Although the operation of the unit is quite simple in fact, you really need to know how to use it, and the manual is designed to give exactly this information.

A 3-ring American-style loose-leaf binder contains all the data in seven sections: installation; operation; programs; theory; service; warranty and specs; and a set of blank register allocation charts for the user. Additionally, a comprehensive contents list makes it easy to find what you want to know.

Turning to the unit itself, we find two parts: the processor section – a 4U rack-mounting beast – and a control head remote unit, the two being

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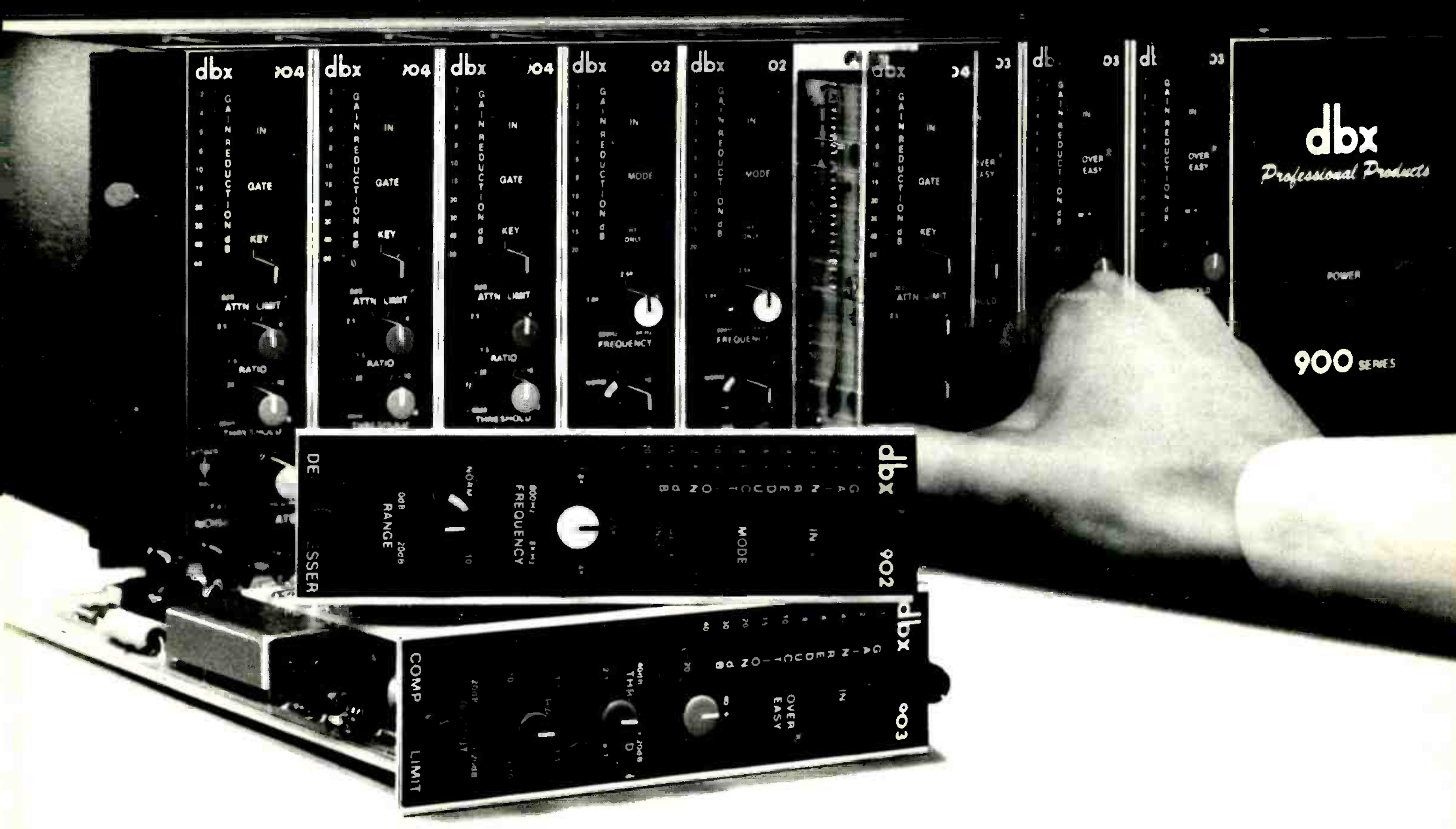

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connected by a multiway screened cable. The processor unit front panel contains very few controls: merely power on/off and indicator, a reset button, and a set of screwdriver presets for level adjustment of ins and outs. Eight captive knurled bolts retain the front panel which is thus simply removed to reveal a neat card frame containing the various boards which make up the unit. These are represented in the block diagram of **Fig 1**. Round the back of the main unit we find a large central heat-sink and to the left a set of correctly-wired *XLR3* connectors, labelled left and right input (female) and outputs A, B, C and D (male). On the right of the heat-sink is an IEC mains socket labelled with the factory-set voltage (voltage adjustments are internal) and an imperial fuse-holder (the manual gives full details). At the top are two D-type 25-pin connectors which are labelled 'option' and 'remote'. The former socket allows extended-distance remote operation and also enables the *224X* to be hooked up to an automation system via *RS232* serial link. The latter connector is designed for the remote unit in normal operation with the supplied cable. This is 25ft in length – quite sufficient for normal use – but a 50ft length can be obtained to order. Being a digital signal-carrying cable it could conceivably radiate digital nasties, or pick up glitches from nearby cables, but in practice this did not occur, although the manual kindly warns about the possibility.

Patching the unit into circuit proved simple, the ins and outs all being transformer-coupled and free from phase inversions. The four outputs are designed for different applications, A and C being the normal stereo returns and B and D providing surround signals on the concert hall programs. Setting up levels was similarly easy, a delay program being used with tone to adjust the level controls, which were precise and easily adjusted.

Usefully, the unit performs a diagnostic routine on power-up. When power is switched on, the unit goes away and checks all the vital functions, returning with an error code (with meanings described in the 'service' section of the manual) if there is a fault, otherwise after a few seconds the unit comes up with the last-used program automatically. The quiet fan in the unit (this must be about the first machine ever devised with a quiet fan) has an air filter which must be cleaned periodically in detergent and warm water under normal studio conditions, but this is virtually the only routine maintenance needed.

The unit is designed to be connected in the normal way for such devices: from an echo send and back into the mix buss via a couple of channels or a specialised echo return. This done, the unit is ready to use.

Operation

All functions are operated from the remote head via a combination of pushbuttons and sliders. There are 13 main programs, each with eight preset 'variations'. In addition, you can create your own variations which may be stored in any of four main registers and 32 shift-key registers, making up 36 locations in all. Many of the controls have multiple functions.

Starting from the top of the remote, we find a multi-function LED display, with headroom indicators labelled 24, 18, 12, 6 and 0dB, the 0dB indicator marking overload conditions. Next to this is a processor overflow light, plus a two-digit alphanumeric display which shows parameters and page selection (more about this later). A set of LEDs to the right of the display panel indicates the units for parameter displays in sec, ms, Hz and kHz.

Beneath the display area are two rows of momentary pushbuttons, eight in each row. The top row is numbered 1 – 8 and these are used to call programs and variations. The four left-hand buttons on the second row are special function keys: Page steps through control pages, with Shift/Page loading slider controls into memory; Set stores programs and parameter settings in registers; Call loads programs or register contents; and the Shift key selects the shift-key registers (with the register and program buttons) while Shift/Program 1, 7 or 8 displays or changes a set of software toggles, namely dynamic decay, mode enhancement and decay optimisation respectively. The four remaining buttons in the row, labelled A, B, C and D, select main registers, or, with the shift key, shift-key registers. At the bottom of the box are six pushbuttons which operate with the short-throw slider controls mounted above them. They are used to control program parameters. The exact functions of these controls depend on the particular control page that has been accessed, but in the 'normal' mode they show, from left to right, reverb time bass and mid; crossover; treble decay; depth; and pre-delay. When a slider is moved, the 2-digit display in the top right of the panel shows the value of the setting and the appropriate units. Additionally, pressing the appropriately-labelled button below a slider displays the current parameter value. In some modes the sliders have a novel and useful 'precision setting' mode, in which a parameter can be changed rapidly by moving the slider, while an exact value can be 'fine-tuned' by overshooting slightly and then coming back down again, when the precision of the slider is increased by a large amount. This makes accurate parameter setting very easy, although the 'fine-tune' facility is only available on parameters that really require it. The operation is not unlike the friction-vernier type of tuning control sometimes found on UHF TV tuners, and is easily grasped.

Accessing a give program and preset variation is simple: the Call button is pressed, followed by the program number. Then the appropriate variation number is selected by simply pressing one of the numbered 1 – 8 buttons. The Call button is illuminated if it is needed to press it to access a program; it goes out after the program has been accessed, so that the operator is aware that pressing a further program button will only access a variation, not a program. The variation setting defaults to 1, and the program selection button remains lit, indicating the selected program number, while the LED in the selected variation-number button flashes. Thus, at a glance, you can see both variation and program number selected. If the program number and variation number are the same (eg, Program 1, Variation 1), the button simply flashes, but as no other program lights are lit, you still know what's been selected. The complete procedure for accessing a program variation is therefore very simple: it's Call, (Program number), (Variation number). The variation can be changed simply by pressing its numbered button. This is simpler than on the *224*, where it was necessary to use the Call button like a shift key (ie, pressed simultaneously with the Program button required). When a program and variation is selected, the major parameter (generally the reverb time) is displayed in the alpha display region.

Calling a setting from a register is equally simple: you press Call (Register). This time the Call button remains lit, indicating the fact that other registers may be accessed simply by pushing the required register button. If there's nothing in a register, the command is ignored. One point to note is that when a sound is called from a register, no variation number flashes. This is no doubt because the sound stored thus is generally a user-defined variation and not a standard one – sensible. You don't need to store things that are already there. The special shift registers are a little more complex to access: but as it's only Call, Shift/(Register number), (Program number), for example, Call, Shift/B – ie, Shift and B pressed together – 2 to access shift register B-2, it's still pretty straightforward. In this mode, any other shift register in the same block (in this case, block B, 1 – 8) may be accessed just by pushing the number. Once again, the Program buttons display only the program numbers from which a given setup is derived, with no variations.

Generating new variations is moderately easy, the straightforward parameters being easily accessed. The sliders are used to changed a parameter, as described, it being important to remember that values do not change until the slider is moved through the current setting value. So, if you press the Bass reverb time button and it reads 3.0 secs, this value will not shift until you move the Bass slider past the point where it would give a value of 3.0 secs (easy to do, more difficult to describe). This means that the sliders won't do any damage if you move them accidentally unless you happen to knock them past their previous settings: useful.

Pages

The thing with this beast is that there is more than one 'page' of parameters available. At the basic default level, the sliders do what they say they do: bass and mid reverb time; crossover frequency; treble decay (the frequency above which the reverb dies away very rapidly); depth (proportion of early to late 'reflections'); and predelay time. But there are 10 pages here. They all do different things to your sliders, and this is one reason the manual needs to be around: it tells you what they do. Pages are accessed, logically enough, with the Page button. Pressing it steps sequentially through the pages, which have their numbers (and a rather weird 'something else') displayed, in the display panel, until a slider is repositioned, when the display reverts to its normal parameter-indication mode. **Table 1** gives an indication of what the sliders do in the various pages. It's only an indication, because I do not intend to transcribe large chunks of the manual: if you are going to spend seven kilopounds on a reverb unit, you will no doubt see the manual first yourself and have someone show you what the thing does.

It will be noticed that the strange characters which appear next to the page number in the display have a certain mnemonic resemblance to what the sliders do on that page, in much the same way as the mnemonic 'LDA' tells my microprocessor to go off and LoaD the Accumulator. There is in fact another page called –Fn which is associated with the Delay page, and provides extremely fine controls for the delays on page – dL. It does not yet exist on any programs. This is why the manual has the capability of adding updates to go with new software ROMs: basically, they can do virtually anything they like in the

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TABLE 1
224X Page/Slider allocations

Page	Slider number & function 1	2	3	4	5	6
1 or 2	LF RT-60 (Running)	MID RT-60 (Running)	LO/MID Transition	HF Rolloff in reverb	Distance of 'pickup' from 'source'	Delay before reverb
– dd	LF RT-60 (Stopped)	HF RT-60 (Stopped)	Chorus (Modulation Depth)	HF B/width	Diffusion	Definition
– LE	Levels of pre-echoes or chorus voices					
– dL	Delays of pre-echoes or chorus voices					
– Fb	Feedback gains					
– LC	LF Cutoff					
– Pd	Pre-delay for effects programs					
– Pn	Pan pots					
– oF	Feedback		High cut		Diffusion	

future, as the *224X* is entirely software-based as far as program capability goes. On past and present performance, it's likely that what *they* like is also something *you'll* find useful.

Some of these pages are both interesting and particularly useful, and often you can determine whether they do anything or not via the software toggles. Thus, Shift/Program 1 toggles the page 2dd (dynamic decay) feature which enables normal functions to be set on Page 1, but different settings of the same parameters (determined on Page 2dd) apply *only when there is no input signal*. Thus you can have, say, a long reverb during the music and a short one during pauses (or the opposite, or anywhere in between). This has its applications; in fact I used a long decay on an otherwise short program to add a reverb 'tail' to an ending: the shorter reverb during the music ensured that the mix didn't get muddy, but I could still have a nice long die-away after the final sting. Nice. Some of the other settings have software toggles, too: for instance, Shift/Program 7 (Mode Enhancement) turns on or off the Chorus control of Page 2dd (slider 3). Pages can be called directly rather than stepped through by pressing Shift/Page, (Program number).

Assuming you've used these gala functions to get the sound you want, you can then stuff it into a register. Normally the *224X* remembers the last program setting you were using before it was switched off. You can, however, command it to power up on a program stored in a particular register (as long as it's C1 – C8 or D1 – D8). This is done with a 4-bit DIL switch inside on the non-volatile storage board (see **Fig 1**). Four bits can store 15 settings, hence the limited choice of registers, one of the settings (all off) being 'normal' ('as you were last'). This is not a limitation, however: a choice of 15 out of 36 is quite sufficient for you to organise storage so that you favourite program can pop up on power-up.

Programs

So what are these programs, then? **Table 2** shows the basic starting points, remembering of course that each program has eight variations, plus any new ones you might devise for yourself. The *224X* offers two basic families of reverb: Halls (with a low density of initial returns) and Plates (which have more early 'reflections'). There are also some which don't come into either category: many of them are effects programs.

The Rooms are rather like Halls, but smaller (hmmm). The Constant-density Plates have a high initial diffusion, but the density remains constant thereafter, unlike ordinary plates where the density increases with time. The split programs turn the unit into two independent reverb units with mono inputs and stereo outputs, where Page 1 controls handle the primary functions related to the Left input (with outputs at A and B) while Page 2 controls right input/C and D outputs parameters. Generally

TABLE 2
224X Basic Programs

1 Halls	1/6 Constant density Plate (A)
2 Plates	2/3 Split Halls
3 Rooms	2/4 Split Plates
4 Chamber	2/5 Plate/Hall
5 Constant-density Plate (B)	3/4 Chorus
	3/5 Resonant Chords
6 Bright Halls	3/6 Multi-band Delays

the programs on either half of the 'split' are ones which occur on their own elsewhere. The Chorus program generates six voices which have variable feedback, delays, panning, doubling, tripling, flanging, echo flanging and other parameters. To facilitate this, the various page/slider functions are changed, drastically. The resonant chord program is also out on its own: it creates, from a transient input, six resonant notes, which are assigned one to each slider. Parameters which may be varied include level, pitch, duration, predelay, panning, feedback and low-pass filtering. The multi-band delay program offers six individually-controlled delays, each with its own levels, delays, LF and HF cutoff, panning, feedback and diffusion.

There is a problem with these effects functions, however. If you use them on the mix, you don't have a reverb unit any more: you have a multi-mega delay line or chorus unit or...You have spent a lot of money on this box. To get the best out of it (and those effects are really neat) you have to know when you want what out of it. You will want to use the *224X* on the mix. Make sure you don't need it for anything else other than reverb, because you will hate every plate in the building, and probably a goodly number of other things too. Put the effects down as you record. This leads us to...

How does it sound?

I know about hating every plate in the building. I had the *224X* on a 3-day mixing session at Marcus Music UK in Kensington. Tim Hunt, my co-engineer on the mixes, kindly set up lots of goodies for effects purposes, plus a plate and the *224X* (thanks, Tim). We put up the *224X* returns to a resounding silence. We put up the plate returns to the familiar hiss we have come to expect from such mechanisms, even the newer ones. We faded them down again. The *224X* was firmly ensconced as *the* reverb unit for the sessions in about five minutes, after we started stuffing signals into it, to be confronted with beautiful, warm, 'roomy' and 'hally' and 'platey' (with no nasties) reverbs. Poor old echo plate didn't stand a chance. I suppose it shouldn't have really either, considering the price differential (the *224X* isn't cheap) but it was still a bit of a surprise. Please may they have bought one by next time.

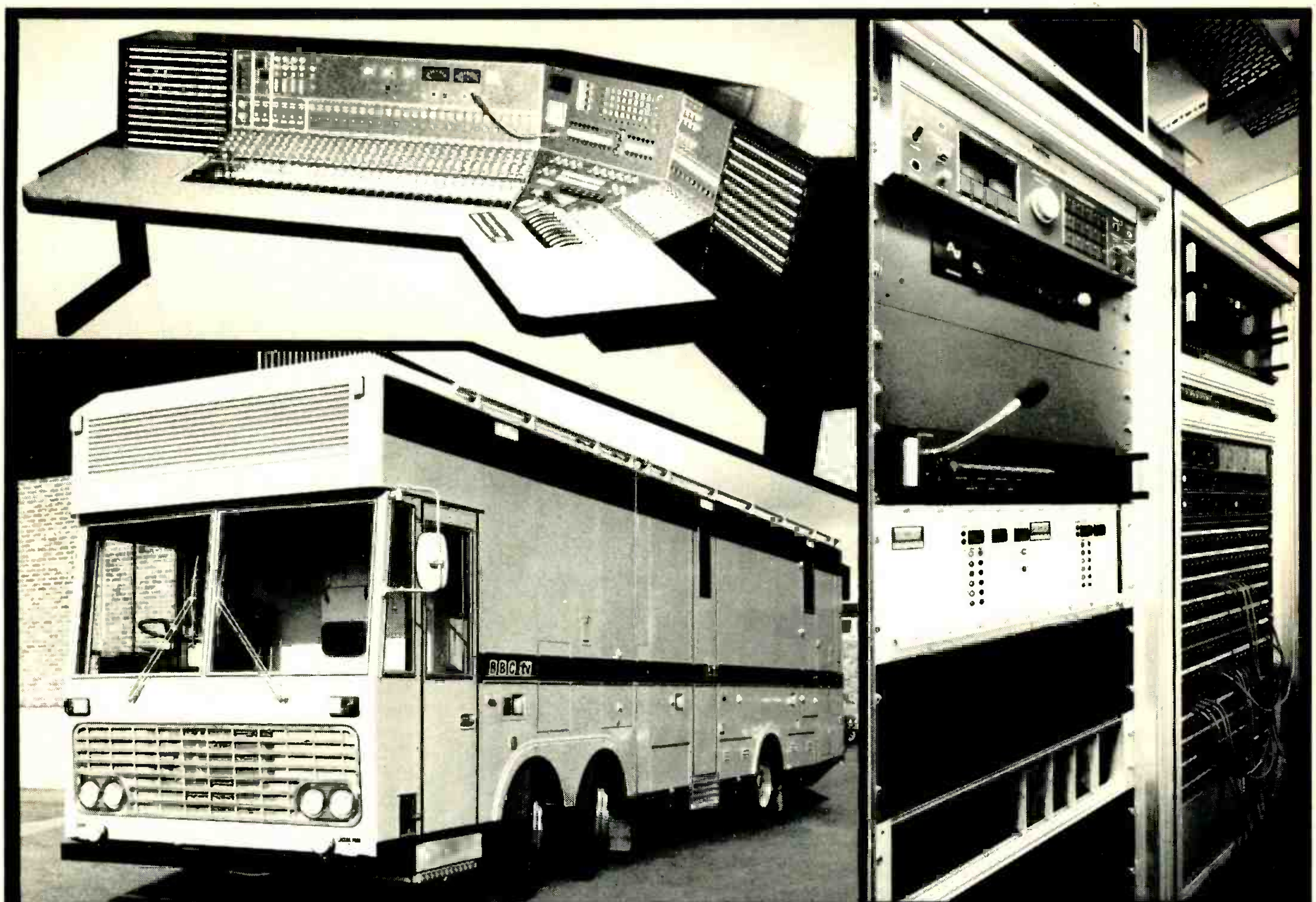
There was this slight problem here, though. There were all these lovely effects lurking in the box, which one would normally have used during the recording, only we didn't have the unit then. We simply couldn't bring ourselves to add six resonant thingummys to a track, or chorus the speakers into their baffles on a lead line, because we'd have lost that lovely hall effect, or whatever it was. It makes you ill, really. Anyway, we'd already got the effects we wanted on tape, although they weren't 6-track chorus sounds, or multiple split delays, or...This is really something to be careful about.

You couldn't fault the sound. Our 'synthesised classical-style' music benefited greatly from large halls and long, tenuous reverb tails on brass-like *Variophon* solo lines, plus plate-like sounds on some of the more up-tempo numbers. The *224X* always behaved, and almost never made any nasty noises, hisses, clicks or bangs. Only on a couple of synth lines did it hiccup slightly, where quiet, low-frequency signals clicked a bit with level shifts. This was expected (in fact it was expected a good deal more than it happened) because page 3.3 of the manual tells us about it: there is a software toggle (Shift/Program 8) called 'Decay Optimization' which is normally on. Toggling it off removed the problem and only affected the sound a little in a rather indescribable way which no-one would notice in a busy track. The *224X* did what it was supposed to do, in a very satisfactory way. I have no complaints at all, but for the fact that I wish I could afford one to take round to studios with me until they have one of their own. As long as you use the effects when you know you want them, rather than on the mix when you want the *224X* as a reverb unit, you can't go wrong. The unit has an excellent sound, *very* low noise, especially for a reverb unit, unnoticeable distortion, very wide dynamic range and frequency response, and it just sounds *nice*. There's no getting away from that essential, and essentially unquantifiable, fact. It clips horribly when you overdrive it, but that is your problem, and anyway, it doesn't sound nasty until you *really* hit it with a lot more level than you'd expect.

Conclusions

It sounds nice. It does its job well. It performs beautifully as a reverberation or effects unit, but not both at once. It can be enhanced with new software with no hassles. It is the answer to every engineer's question 'what is the best reverb unit?' It is expensive. It is so expensive that even though it is the answer to every engineer's question relating to Life, the Universe, and Everything Reverberant, your studio may not be able to afford one, especially since you got the new console (which cost ten times that) and as you already have four plates of varying antiquity plus sixteen reverbs, springs, MexiMegalon *Multi-flangers*, tape loops and other esoteric devices. This is a pity, and your accountants will have to find the Answer to the Ultimate Question (or vice versa). Quality costs money. The machine itself will only reveal its secrets unto you with a few driving lessons, and isn't simple to use in all its complexity *per se*, but if you have the manual handy *and read it*, the only things you have to scratch heads about are the functions of the pages. Of course, you do not spend this kind of money without playing with the box first. I may think it is the answer to all these problems, but you may not agree. Not all engineers are alike, especially with subjective devices like reverbation units. And these days, you often have to cut your coat according to the amount of cloth available. If you can't afford a jacket you will have to start saving.

Richard Elen



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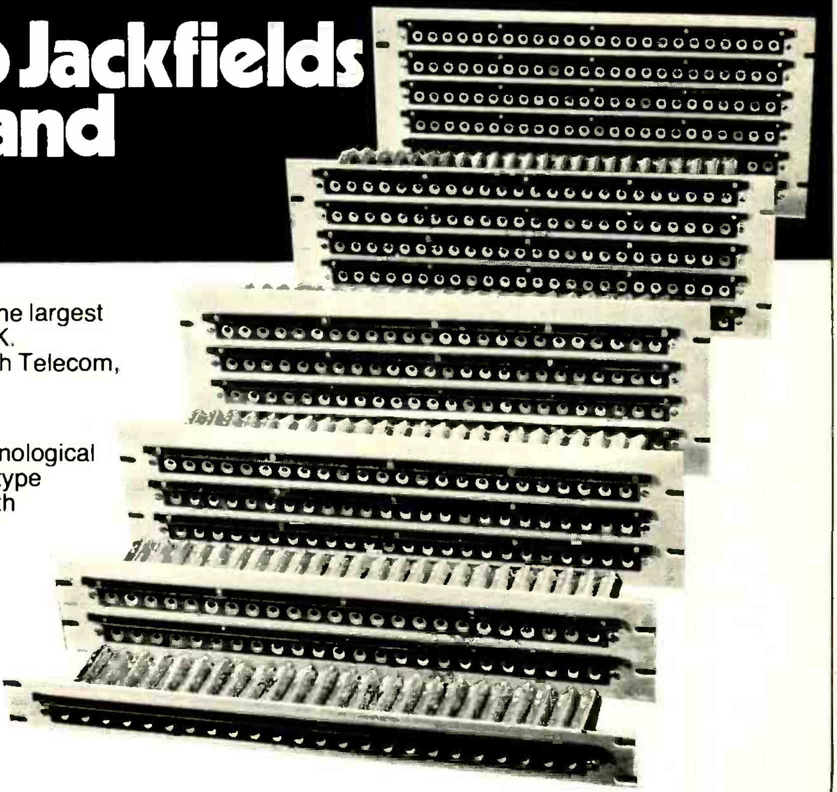

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Loft Model 450

MANUFACTURER'S SPECIFICATION
Delay time at 18kHz bandwidth: 0.5 to 5.0ms in flange mode. 4 to 40ms in delay mode.
Delay time at 9kHz bandwidth: 1 to 10ms in flange mode. 8 to 80ms in delay mode.
Delay time at 6kHz bandwidth: 1.5 to 15ms in flange mode. 12 to 120ms in delay mode.
Delay time at 4.5kHz bandwidth: 2 to 20ms in flange mode. 16 to 160ms in delay mode.
EM-450 Module: doubles delay time in delay mode.
Pre-amp gain: adjustable, up to +20dB.
Input impedance: 20kΩ (balanced) > 90kΩ (unbalanced).
Maximum input level: +18dB (ref. 0.775V).
Output impedance: 10Ω.
Maximum output level: +18dBm.
Total harmonic distortion plus noise (delayed signal only): flange < 0.5% (typically 0.2%). Delayed < 1.0% (typically 0.8%).
Noise: −80dB 'A' weighted.
Input and output connectors: ¼ in phone jacks and *XLR*.
Dimensions: 482.6 × 44.4 × 228.6mm (WHD).
Weight: 3.18kg (shipping 4.09kg).
Manufacturer: Loft Professional Audio Products, Phoenix Audio Laboratory, 91 Elm Street, Manchester, Connecticut 06040, USA.

THE Loft delay line/flanger is based on analogue bucket brigade type delay elements, the delay time of which is controlled by the frequency of the 'clock' applied to the delay element.

Unfortunately, the manufacturer provides very little information in the owner's manual, so **Fig 1** is my attempt at drawing a block diagram of the unit. The front panel of the unit which is designed for rack mounting and one unit high, is divided into four sections, plus the power on/off switch and LED indicator.

To the left of the front panel is the delay control section with three potentiometers called 'delay', 'depth' and 'rate'. These controls derive a fine control voltage for the clock oscillator which is a voltage controlled oscillator.

Reference to the bottom of **Fig 1** shows that the 'rate' control sets the frequency of a sawtooth oscillator the period of which could be set between 190ms and 44s. The DC voltage from the 'delay' control is mixed with the oscillator's output in any desired proportion to derive the control voltage for the clock via the depth control.

This control voltage can be interrupted at a tip, ring and sleeve jack and a single pole jack on the rear panel permitting the use of external control voltage sources.

The central section of the front panel is the 'timebase' section. Within this section a locking pushbutton switch with an associated red LED indicator switches the 'effect' in or out by bypassing the delay section. A second locking button with two LEDs selects the delay or flange modes. This in fact selects one of two delay time ranges which alter the delay times set by four interlocking pushbuttons.

In both modes the delay control has a 10:1 delay time range for each timebase range. In flange the delays for each range are 0.5 to 5ms, 1.5 to 15ms and 2 to 20ms.

In the delay mode with the review sample, which had the optional extended delay module fitted, the delay ranges were 8 to 80ms, 16 to 160ms, 24 to 240ms or 32 to 320ms, these times being halved without the option.

To the right of the timebase section the input/output section contains two level controls and three LED level indicators. both controls are detented potentiometers with calibrations in decibels, the input level control having a 20dB range and the output level control being of the full range type calibrated from 0dB to minus infinity.

The red, yellow and green LED level indicators show headrooms of 5dB (red), 10dB (yellow) and 20dB (green), these indicators being wired sensibly somewhere after the audio mix control.

The right hand section, signal mix and regenerate, had two potentiometers and a locking pushbutton switch. Reference to **Fig 1** shows that the output from the ouptut gain control may be either the source signal alone, the delayed signal alone or any desired mixture, according to the setting of the mix control.

The regenerate control recirculates audio round the delay line in any desired proportion, some form of switchable equaliser being in this path if desired.

To the rear of the unit the audio input may be balanced at a *XLR* connector or unbalanced at a ¼in jack which disconnects the *XLR* input. Similar connectors in parallel are used for the unbalanced main and 'stereo' outputs, the latter

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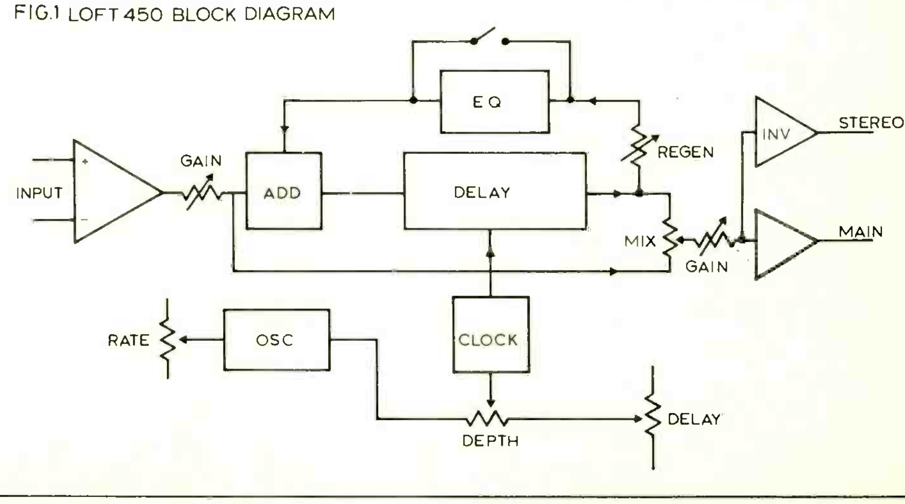
FIG.1 LOFT 450 BLOCK DIAGRAM



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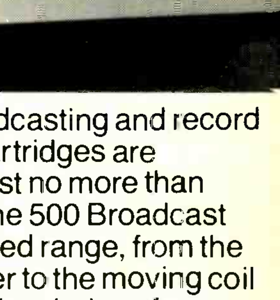

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appearing to be a phase inverted version of the former.

Two further ¼in jacks are used for access to the control voltage. A tip, ring and sleeve jack delivers the control voltage and inverted control voltage out, with a two pole jack disconnecting the internal control voltage and providing a control voltage input. A further jack allowed remote effect in/out switching with a shorting switch.

Finally, there is the fixed, US colour-coded, power line and a properly identified power fuse.

Within the unit all components are supported by a single board which has harmonica connectors for the inputs, outputs and LED indicators.

All integrated circuits were socketed and passive components indentified in value. However, no servicing or alignment instructions were provided. The review sample had five on-board presets, the function of which is unknown.

Inputs and outputs

Both the balanced and unbalanced inputs could accept +21.5dBm at the onset of clipping at lower input gain settings or +2dBm at maximum input gain. The input impedance remained constant with gain setting at 43.3kΩ for the balanced input or 92.3kΩ for the unbalanced input.

Common mode rejection remained fairly constant with frequency, being better than 55dB from 20Hz to 20kHz.

Both the main and the 'stereo' unbalanced outputs had a very low source impedance of about 10Ω with a drive capability of +21dBm loaded into 600Ω.

The maximum gain from the input to the outputs in either the straight through mode or the delay only mode was 19.4dB with the input gain control having quite accurate calibrations over its 20dB range. Similarly, the output level control has reasonable calibration accuracy.

The control voltage output had a range of 1.68V to 12.17V for the positive output or 1.66V to 12.55V for the negative output, the former having an impedance of 250Ω and the latter 1kΩ.

At the control voltage input insertion of a jack disconnects the internal control voltage. External control voltages needed to be in the range +1.8V to +13V with the input settling at +13V in the absence of a load to ground. The input impedance of the input was 12.5kΩ.

Frequency response

The measurement of the frequency response of the delay channel was far from simple as the unit appears to use 2:1 companding for the delay elements in addition to pre-emphasis. This produced some very weird characteristics in the initial sample of the unit. Subsequently the manufacturer provided a second modified sample (a simple modification adding two diodes in the compander) which cured mis-tracking and improved low frequency harmonic distortion.

Fig 2a shows the delay channel frequency response for a 0.5ms time delay setting with the unit at maximum gain. The four curves relate to output levels of +10, −10, −30 and −50dBm in the initial unit with the results for the modified unit being shown in **Fig 2b** where the mis-tracking has been eliminated but the use of pre-emphasis takes its toll at high levels and high frequencies.

In **Fig 3** the more restricted frequency response for 5ms delay is shown for the delay mode.

Fig 4 shows some of the many frequency responses at longer delay settings, including the severe beating at high frequencies due to the lack

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FIG. 2a
LOFT 450
FREQUENCY RESPONSE
AT 0.5ms DELAY

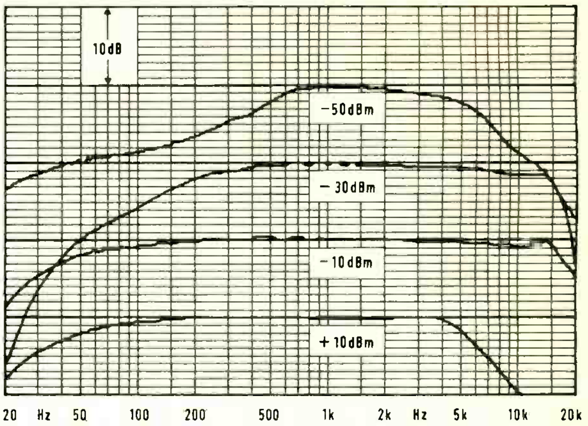


FIG. 2b
LOFT 450
FREQUENCY RESPONSE
AT 0.5ms DELAY
(MODIFIED UNIT)

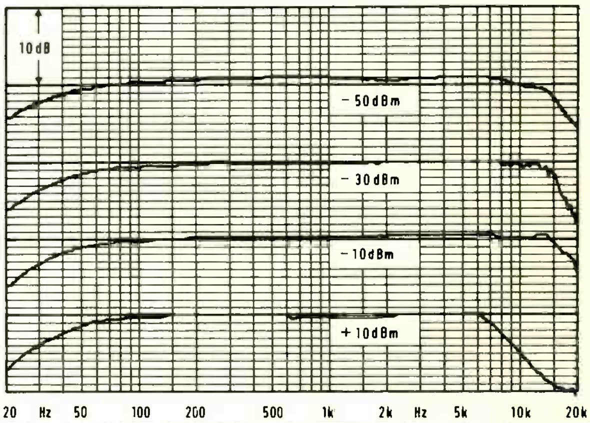


FIG. 3
LOFT 450
FREQUENCY RESPONSE
AT 5ms DELAY

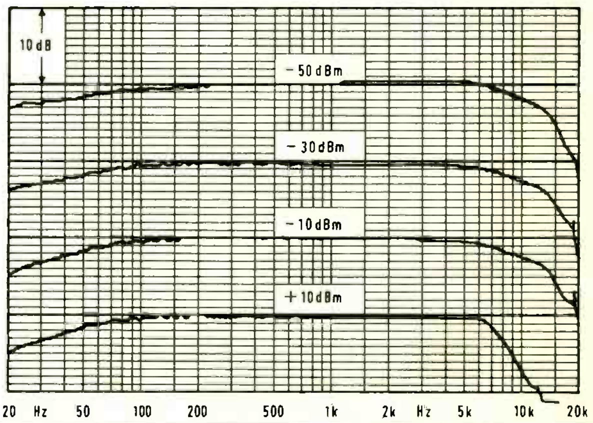
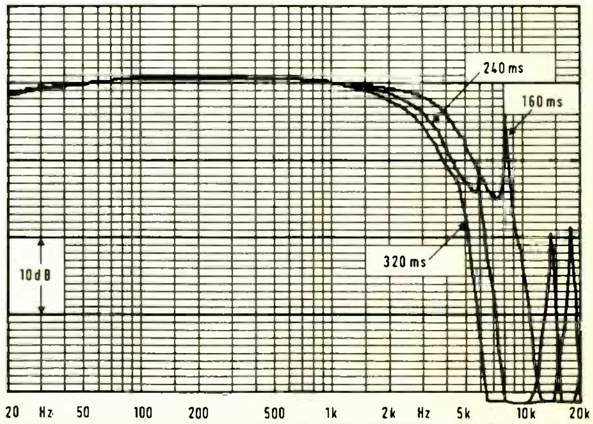


FIG. 4
LOFT 450
FREQUENCY RESPONSE
AT SPECIFIED DELAYS



of bandwidth restricting filters before the delay elements.

Noise

Noise in the output at maximum gain was measured in the straight through mode, the flange mode and the delay mode with no signal input.

As subjective noise breathing was apparent, a spectrum analysis of the noise was undertaken with the spectrum analyser being preceded by a very sharp 1kHz notch filter. A spectrum analysis of the output was then done with no input signal and with a 1kHz input giving −10dBm output.

As can be seen from **Fig 5** there was noise breathing to the extent of greater than 35dB in the subjectively critical region around 7kHz.

It follows that in real terms 35dB or more can be taken off the figures shows in **Table 1**.

Distortion

In the direct mode from the source, harmonic distortion was satisfactory at less than 0.03% at frequencies below 5kHz rising to 0.1% at 10kHz.

In the flange and the delay modes, harmonic distortion was sensitive to frequency, level and delay times. **Fig 6** is typical of what may happen, showing the second and third harmonic products at −20dBm output at maximum gain in the delay mode—a satisfactory situation.

It was found that intermodulation distortion to the CCIF twin tone method was very dependent upon control settings with the second order product (f1 − f2) predominating.

Fig 7 shows the funny things that can happen to a 40ms toneburst superimposed on a continuous low level signal, both at 1 kHz. The lower trace shows the input with the upper trace showing the delayed signal.

FIG 7

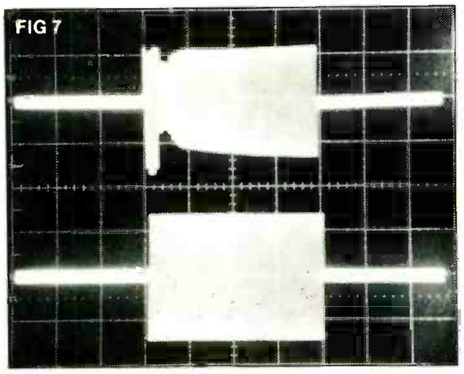


TABLE 1

Measurement method	Flange	Delay	Source
22Hz to 22kHz RMS	−85dBm	−83dBm	−76dBm
A-weighted RMS	−89dBm	−86dBm	−79dBm
CCIR-weighted RMS ref 1kHz	−80dBm	−77dBm	−70dBm
CCIR-weighted quasi-peak	−77dBm	−73dBm	−66dBm
CCIR-weighted ARM ref 2kHz	−86dBm	−82dBm	−76dBm

Other matters

The calibrated delay times were reasonably accurate and the level indicators provided a useful and adequately accurate function.

Summary

The Loft delay line/flanger can produce useful effects and is simple to operate. However, its satisfactory use depends upon the type of programme.

Clearly, the bandwidth is limited for some applications and noise breathing is significant with wide dynamic range material such as isolated drum beats.

It was also found that very fast transients were badly reproduced and lost their attack.

Hugh Ford

FIG.5
LOFT 450
NOISE BREATHING
(80ms DELAY)

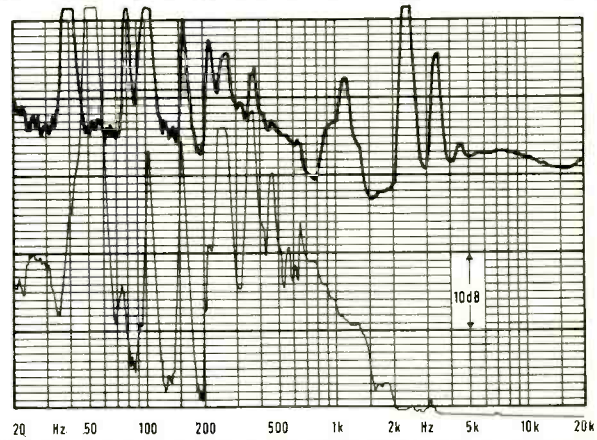
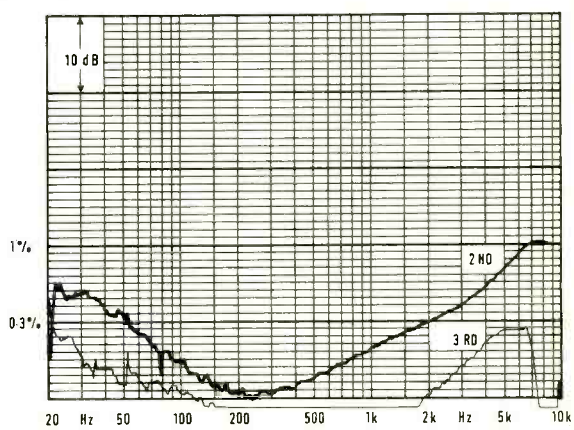


FIG.6
LOFT 450
HARMONIC DISTORTION



Editor's comments

Effects units are notoriously difficult to review, especially when it comes to relating 'objective' measurements to 'subjective' operation. Quite often, devices which measure only satisfactorily in the laboratory, sound excellent in the studio, while occasionally the opposite is unfortunately true! To attempt to get a 'feel' for this unit in a working environment, I took the *450* along to a series of sessions, where it was used for a variety of effects, notably ADT, flanging and stereo synthesis.

The results in a recording environment were impressive. This type of unit is unlikely to be used on its own, and in a mix, any noise problems or 'breathing' effects are obscured by the main signal being processed. Particularly notable was the smooth action of the oscillator: many oscillator-driven time-domain processors have an unfortunate 'hump' at the end of the oscillator's travel, which causes a discontinuity in the effect, but the *450* was fine in this respect. Especially with the add-on memory which is available for the unit, the *450* is capable of a wide variety of effects which, especially in view of the very moderate price of the device, make it an ideal basic effects unit for the studio, and it happily outperforms many more expensive digital effects units. Care should be taken, however, when using both outputs together, as they are 180° out-of-phase, and this should be borne in mind if mono compatibility is required. Other than this, my only reservations on the unit were that it was not too happy ADT'ing exposed drum beats and sounds with heavy transients, where a difference in sound quality may be noted between the original and the effect, as Hugh Ford points out. Overall, however, I would recommend this unit, especially for the smaller studio where digital time-domain processors may be out of the question economically.

As with all effects units, the final decision on purchase should be made as a result of *listening*, in as 'real' a working environment as possible, to see if the unit appeals to the particular people who are going to use it.

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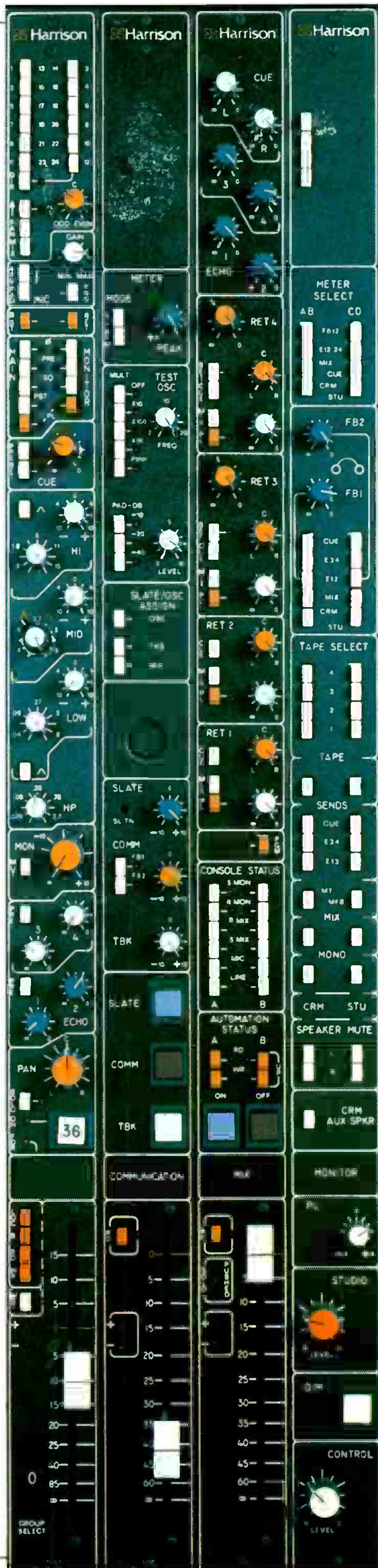

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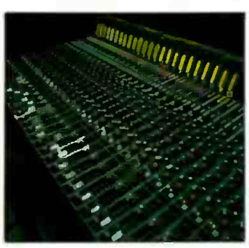

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