AUGUST 1994 $5.00 £2.00

STUDIO SOUND

AND BROADCAST ENGINEERING

INTERNATIONAL PRO AUDIO AND POSTPRODUCTION MAGAZINE

LIVE!

Pink Floyd in Florida
Live Console Roundup

Test and Measurement

DK Audio test set; Prism Sound Dscope;
Sound Check test CD; CMR explained

No Comparison

The other side of loudspeaker testing

08
9 770144 594017

"SCENARIA IS A BRILLIANT CONCEPT! IT COMBINES INSTANT ACCESS AUDIO AND VIDEO IN ONE SYSTEM"

Lloyd Billing, Managing Director, The Tape Gallery, London

The Tape Gallery is one of London's leading audio post-production houses. This multi-room complex was one of the first to recognise the appeal of Scenaria, and decided to add a second system within 12 months.

Specialising in commercials for TV, Radio and Cinema, their award-winning campaigns include those for:
John Smith's
Michelin
Adidas
Texaco
Lloyds Bank

Malcolm Bristow at The Tape Gallery's first Scenaria

Lloyd Billing (standing) and Simon Capes

"We regard ourselves not just as mixers, but as our client's sound design team. We pride ourselves both on the quality of our work and a fast turnaround. Scenaria's ability to recall previous mixes in their entirety makes for consistency and a considerable time saving in the production of multiple versions and re-mixes.

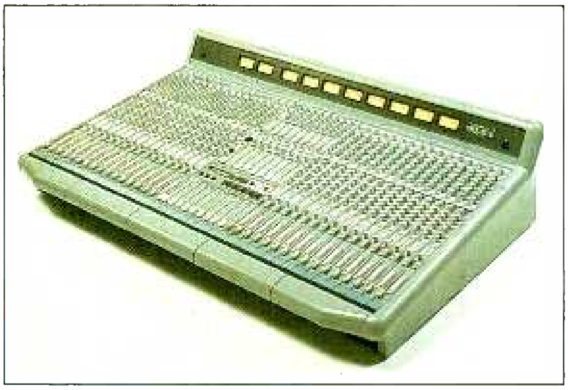
"With the increased level of bookings generated by our first system, we needed a second Scenaria within a year. Both are now upgraded to OmniMix standard to work in surround sound."

Solid State Logic SSL DIGITAL

International Headquarters:- Begbroke, Oxford, OX5 1RU, England. Tel: (0865) 842300
Paris: (1) 34 60 46 66 • Milan: (2) 262 24956 • Darmstadt: (6151) 93 86 40 • Tokyo (3) 54 74 11 44
New York (212) 315 1111 • Los Angeles (213) 463 4444

STUDIO SOUND
AND BROADCAST ENGINEERING

CONTENTS



Unconsoled? See page 33

Beyond compare? See page 9

www.americanradiohistory.com

STUDIO SOUND

AND BROADCAST ENGINEERING

August 1994
Volume 36 Number 8
ISSN 0144 5944

EDITORIAL
Editor: Tim Goodyer
Production Editor: Peter Stanbury
Editorial Secretary: Mary Walsh
Consultants: John Watkinson; Sam Wise
Columnists: Barry Fox; Kevin Hilton; Martin Polon
Regular Contributors: James Betteridge; Simon Croft; James Douglas; Ben Duncan; Tim Frost; Philip Newell; Terry Nelson; Dave Foister; Francis Rumsey; Yasmin Hashmi; Zenon Schoepe; Patrick Stapley

ADVERTISEMENTS
Executive Ad Manager: Steve Grice
Deputy Ad Manager: Phil Bourne
Business Development Manager: Georgie Lee
Advertisement Production: Carmen Herbert
PA to the Publisher: Lianne Davey

CIRCULATION
Assistant Circulation Manager: Diana Rabôt

Managing Director: Doug Shuard
Publisher: Steve Haysom

EDITORIAL AND ADVERTISEMENT OFFICES
Spotlight Publications Ltd, 8th Floor, Ludgate House, 245 Blackfriars Road, London SE1 9UR, UK. Tel: 071 620 3636. Fax: 071 401 8036.

NEWSTRADE DISTRIBUTION (UK)
UMD, 1 Benwell Road, London N7 7AX, UK. Tel: 071 700 4600. Fax: 071 607 3352.

Origination by Craftsmen Colour Reproductions Ltd, Unit 1, James Street, Maidstone, Kent ME14 2UR.

Printed in England by St Ives (Gillingham) Ltd, 2 Grant Close, Gillingham Business Park, Gillingham, Kent ME8 0QB, UK.

Studio Sound and Broadcast Engineering incorporates Sound International and Beat Instrumental.

Studio Sound is published monthly. The magazine is available on a rigidly controlled requested basis, only to qualified personnel.

Subscription Rates:
UK annual subscription: £24.00
Overseas surface mail: £30.50/US:$89
USA airspeeded delivery: $70

Subscription Enquiries
UK: Subscription Dept, Studio Sound Magazine, Spotlight Publications Ltd, Royal Sovereign House, 40 Beresford Street, London SE18 6BQ.
USA: Studio Sound Magazine, 2 Park Avenue, 18th Floor, New York, NY 10016.
US Postmaster
Please send address corrections to: Studio Sound Magazine, c/o Mercury Airfreight International Ltd Inc, 2323 Randolph Avenue, Avenel, New Jersey NJ 07001.
US second class postage paid at Rahway, NJ.

ABC
MEMBER OF THE AUDIT BUREAU OF CIRCULATIONS

PPA PERIODICAL PUBLISHERS ASSOCIATION

Total average net circulation of 18,834 issues during 1993. UK: 7,357. Overseas: 11,477. (ABC audited)

un A United Newspapers publication

Looking forward to looking back

'The video edit suite is a complete artists palette,' Lol Creme said to me back in 1985. 'We took to it like ducks to water. As soon as we saw what they were doing in there and what they could do, we thought, "Gimme. Want one—want one now!"'

At the time, Creme and ex-10CC partner Kevin Godley had just begun to make their collective mark as video directors. Their career was to have a considerable influence over the pop videos with which they started, but was to extend to popular television advertising and beyond. What I missed at the time, however, was the blueprint the duo had made for other musicians with aspirations towards video, and which many have attempted to follow.

In a curious contrast, a recent television documentary series on renowned cinematographers cast certain of its subjects in a decidedly regressive light—much was made of the revival of interest in the old anamorphic film format (British television now regularly 'letterboxes' feature films) and monochrome film-making. Allan Davian, for example, made much of the fact that he has never made a black-and-white film and believes that it will prove a particularly difficult exercise for him. He is, however, desperate for the opportunity.

While we are used to thinking and talking in terms of new technology offering new opportunities to its users, that technology does not necessarily offer us anything the old technology did not. It is also easy to forget that as we move away from older equipment and practices, they can assume the same qualities as those which are brand new equipment and practices. Witness the above examples above.

So it was that we turned our backs on glowing banks of valves in favour of the convenience of transistor technology before demanding valves in our new preamps, equalisers and microphones with an enthusiasm normally reserved for teenagers. So it was that musicians preferred drum machines over their human counterparts before learning that a sense of rhythm is a precious talent. And so it is that we have embraced digital technology—found room in our hearts, studios and budgets for storage and processing equipment which bears no relationship to the analogue equipment that preceded it and then begun to treat analogue with renewed respect.

It is not such strange behaviour, I agree. After all, how are we to fully appreciate the strengths and weaknesses of such developments unless we test it in as many potential applications as possible? It would be more than strange should we be unable to recognise when the new technology was failing to outperform the old—it would be foolishness. To see the launch of such items as the AKG *C12VR* in 1994 is, therefore, a quiet testament to our good judgment and pragmatism.

But with Allan Davian's endearing passion for black-and-white film in mind, I have to raise the question: How long after we have seen the satisfactory marriage of audio and video before we 'rediscover' the elegance of working with sound alone? ■

Tim Goodyer

Cover: Genelec monitor drivers

International News

In-brief

● French AMS Neve distribution
AMS Neve have appointed Systems Audiofrequence Videonique (SAV) as their sole distributor in France. From the 1st July SAV have taken responsibility for sales and service for the complete range of AMS Neve analogue and digital products including the *AudioFile*, *Logic*, *Capricorn* digital consoles; the *VR Legend* and *55-series* of analogue desks. Paris-based SAV's existing equipment portfolio includes products from Lightworks, Pesa, Orban and Eventide.
AMS Neve. Tel: +44 282 457011.
SAV. Tel: +33 1 42 40 5522.
Fax: +33 1 42 40 4780.

● Level 42 go for Tascam
Two members of Level 42 have chosen Tascam *DA88* systems for their own studios. Mark King has 16 tracks' worth, complete with *SY88* sync board, for his Isle of Wight Summerhouse Studio, while Mike Lindup has three *DA88*s also with remote control and *SY88*.
Tascam US.Tel: +1 213 726 0303.
Tascam UK. Tel: +44 923 819630.

● New MD for Turbosound
Steve Revill has become Managing Director of Turbosound, in addition to his role as Managing Director of the Edge Technology Group within Harman. Since becoming Financial Director of EdgeTech he has had special responsibility for Turbosound, overseeing their changes of ownership first to EdgeTech and then to become part of the Harman Pro Group in October of last year.
Turbosound Ltd. Tel: +44 403 711447.

● Sony HD TV sound
Sony have licensed BBE's high-definition sound enhancement for use in all their consumer electronic products, and have released two new stereo Trinitron televisions incorporating the system. Sony call it Orchestra Seat Sound Effect and describe it as 'soundfield processing that duplicates the drama of sitting in front-orchestra seats. The signal is clarified and audio intensity is centred to the front, enhancing the overall realism of your listening experience'. BBE Chairman John McLaren identifies the quality of sound in television sets as a fierce area of competition in the worldwide consumer electronic industry. He points out the system's improvement of speech intelligibility, and cites it as a selling point to, 'older people whose hearing is not what it once was, as well as apartment dwellers who need to be concerned with too loud volume'.
BBE Sound Inc. Tel: +1 714 897 6766.

Deutsche Grammophon go 24-bit

Deutsche Grammophon's Recording Centre have announced the development of a 24-bit multitrack recording capability. The system is already in operation on selected recordings in conjunction with DG's 4D Audio Recording technology and was first deployed to record a live performance of Mahler's 8th Symphony, by the Berlin Philharmonic under Claudio Abbado, on 12th–13th February this year.

The Authentic Multitrack Transcryptor (AMT) provides an extended audio word-length capability on a standard DASH format multitrack recorder. The system consists of encoder and decoder DSP modules and enables 16 tracks of 24-bit recording on a standard 24-track, 16-bit recorder. The encoder and decoder modules each employ 100 MIPS DSP power and a large degree of operational flexibility has been incorporated within the system to allow compatibility with other high-bit multitrack configurations. Twelve-track, 24-bit; 19-track, 20-bit; or mixed 16-track, 20-bit; and 2-track, 24-bit arrangements are all possible. A single AMT system will feed and replay 24-bit digital audio to and from two DASH recorders.

The use of the Yamaha *Y1* digital I-O format for the 24-bit word-length signal allows for ease of interface with the 4D system consoles and the use of passive digital patch networks. Outlining the background to the development, DG's Head of Audio Engineering, Stefan Shibata, explains that having identified a requirement for high-bit storage, 'we recognised that Deutsche Grammophon alone was never going to develop a nonstandard format and neither were the major multitrack manufacturers likely to introduce a 24-bit standard in the near future.

'The solution was obviously to develop some kind of system that did not require any change to the existing format and about five years ago we decided that it was actually possible to store 24-bit digital audio within the existing 16-bit DASH format. Implementing the theory was undertaken during a two-year development programme at the Recording Centre and the resultant system actually provides us with an extremely high degree of operational flexibility without having to replace or in any way alter our recording hardware.'
Deutsche Grammophon GmbH. Tel: +49 40 44 18 11 15.

New Surround Room at GDO. West London audio postproduction house Glynn Davies Oliver Studios have rationalised their name to GDO and expanded their facilities with the opening of a third studio dedicated to audio post for surround sound programmes. The new studio is all digital, based around an AMS Neve *Logic 1* desk. GDO's Nigel Glynn-Davies explains the reasoning behind the new room: 'Last year we began getting a lot of clients coming in asking for surround sound dubs, and we were having to get Dolby to come in and set up the system. We therefore decided the time was right to install our own system for LCRS editing and mixing. Also the Dolby Surround environment is not just a question of putting extra speakers into the room, you actually have to think about it in acoustic design terms.' GDO's primary business area is television audio postproduction including work for the BBC, Channel 4 and Carlton TV. GDO. Tel: +44 81 969 4115

Matsushita license Spatializer

Spatializer Audio Laboratories have announced that their US subsidiary, Desper Products, have signed a licence agreement with Matsushita for use of Desper's *Spatializer* 2-speaker 3-D surround processor technology. Matsushita will incorporate the system into some of their consumer audio products using a *Spatializer* IC manufactured by their own semiconductor manufacturing subsidiary.

Matsushita, whose product lines include Panasonic, Technics, Quasar and National, have already designed several new audio products incorporating the *Spatializer* Audio Processor which will be introduced into the Japanese market within weeks, citing the fact that the processed sound can be recorded by the consumer and reproduced on any stereo system as a significant feature.

The system has also been licensed to Multiwave Innovation Inc for use in their multimedia products and accessories, including an add-on daughter board for existing sound cards and a complete PC peripheral specifically for game applications.

The professional version of the system, Desper's recently introduced *PRO Spatializer*, has already been used on several projects including Disney's *The Lion King*, Bonnie Raitt's *Longing in their Hearts* album, and hundreds of episodes of several Warner Brothers animation series.
Desper Products Inc. Tel: +1 310 268 2700.

ISDN links for Radio Reykjavik

Radio Reykjavik, London's newest radio station, began broadcasting on 10th June as part of the London-wide events for *50 Northern Light Years*—the 50th anniversary of Icelandic independence. For the first two weeks the station was broadcasting from the ship Leifur Eiriksson berthed at St Katherine's Dock near Tower Bridge, while the transmitter was located on Highgate Hill in North London.

Hilton Sound supplied Radio Reykjavik with two 20K bandwidth APT audio codec systems to link the Leifur Eiriksson to the transmitter, enabling the station to give Londoners a unique insight into contemporary and traditional Icelandic music with featured presenters who included Björk and Magnus Magnusson.

Hilton Sound Chairman Andy Hilton comments: 'This project is a perfect illustration of how the boundaries of new technology can be explored by using the facilities of a rental organisation. All of our engineers are greatly enthused by the possibilities of ISDN, and every new situation we are involved in stretches our inventiveness, which we find immensely satisfying.'

Hilton Sound plc.
Tel: +44 71 708 0483.

THE SEASON

September 1994

● September 8th–9th, **Leipzig Radio Show 1994**, Leipziger Messe, Leipzig, Germany. ● September 11th–14th, **PLASA**, Earls Court, London, UK. ● September 16th–20th, **IBC 1994**, RAI Centre, Amsterdam, The Netherlands. ● September 19th–23rd, **Image World Video Expo**, New York, USA. ● September 20th–25th, **Live '94**, Earls Court, London, UK ● September 22nd–27th, **Photokina**, KölnMesse, Cologne, Germany.

October 1994

● October 11th–15th, **Audiovideo-94**, Lenexpo Center, St Petersburg, Russia. ● October 12th–15th, **NAB**, Los Angeles, USA ● October 21st–23rd, **Broadcast India '94**, World Trade Centre, Bombay, India. ● October 29th–31st, **Broadcast Sri Lanka '94**, Marriott Hotel, Colombo, Sri Lanka.

November 1994

● November 10th–11th, **SBES**, Birmingham, UK ● November 10th–13th, **97th AES Convention**, Moscone Center, San Francisco, USA. ● November 16th–18th, **Tonmeister**, Karlsruhe, Germany. ● November 16th–18th, **Interbee**, Makuhari, Japan.

December 1994

● December 1st–4th, **13th Int AES Conference**, Dallas, Texas, USA.

February 1995

● January 30th–February 3rd, **Midem**, Palais des Festivals, Cannes, France ● February 7th–9th, **ISDN User Show**, Olympia 2, London, UK. ● February 28th–March 3rd, **98th AES Convention**, Palais de Congrés, Paris, France.

March 1995

● March 8th–12th, **Frankfurt Musikmesse**, Frankfurt, Germany.

April 1995

● April 26th–29th, **Broadcast Technology Indonesia**, Jakarta, Indonesia.

May 1995

● May 5th–7th, **Theatre World**, Business Design Centre, London, UK

October 1995

● October 5th–8th, **99th AES Convention**, Javits Center, New York, USA.

RTV Slovenia locks up with AK

RTV Slovenia have just completed a major machine control installation at their television broadcasting centre in Ljubljana, using the *ES.Lock 1.11* system from Audio Kinetics. The system comprises a total of 11 *ES.Lock 1.11* modules, supplied with an *ES.Eclipse* controller, to provide dedicated synchronisation across RTV Slovenia's multiroom audio recording and postproduction facilities.

Alongside an Amek *Mozart* console with Supertrue automation, the control room uses Sony *PCM7030* DAT recorders and *APR5003* analogue machines, all controlled by the *ES.Lock* system; the *Eclipse* also handles a *PCM3324S*, Sondor film recorders and a Sony Betacam VTR. Also installed is an *ES.SSU* System Services Unit, providing time-code-accurate triggers for CD players and other non-time-code equipment. The SSU can handle talent cues and provides 'all machines locked' and 'record on' messages to assist operators and artists.

RTV Slovenia's Head of Audio, Drago Perdih, professes himself pleased with the system: 'We believe that the *ES.Lock* system is the best machine control solution currently available, giving us the right match to our requirements. We handled the installation ourselves, with Audio Kinetics finally commissioning the equipment and providing on-site training. The result is exactly as we hoped—our engineers are finding it easy to use and highly flexible in operation, locking equipment up very quickly.'

Audio Kinetics UK Ltd.
Tel: +44 81 953 8118.

German record producers and members of techno band SNAP! Michael Munzing (standing) and Luna Anzilotti (seated) recently scored a hat trick with Euphonix: installing their third, and largest console from the American manufacturer. A Euphonix *CS2000-4-96* board now sits in the duo's Frankfurt base. Munzing and Anzilotti, who also have studios in London and the Mediterranean island of Ibiza, have made their three studios almost identical (each has a 48-fader *CS2000* system), allowing them to transfer complete studio setups from one site to another. The third SNAP! album will be released in October. Euphonix. Tel: +1 818 766 1666.

Contracts

● Studioframe order at APRS

The first European sale of the new TimeLine *Studioframe DAW-80* workstation was announced at the UK APRS Show by Stirling Audio. The system has gone to Videolondon Soundstudios to complement their existing six Synclavier systems.
Stirling Audio Systems.
Tel: +44 71 624 6000.

● Avid for Pinewood

Pinewood Postproduction have installed an Avid *Audio Vision* system, which is already handling ADR and FX recording for Gerry Anderson's *Space Precinct*. The system incorporates removable hard drives which can be taken to the nearby Avid sound editing suite for preparation. Future plans include the installation of an Avid *Film Composer* suite for dry hire, as well as a dubbing theatre based on a *Logic 2-AudioFile-Spectra 16* combination.
Pinewood Postproduction.
Tel: +44 753 656301.
Avid Technology (US).
Tel: +1 508 640 3158.
Avid Technology (UK).
Tel: +44 71 434 0122.
Avid Technology Europe Ltd.
Tel: +44 753 655999.

● Otari *Concept 1* US sales

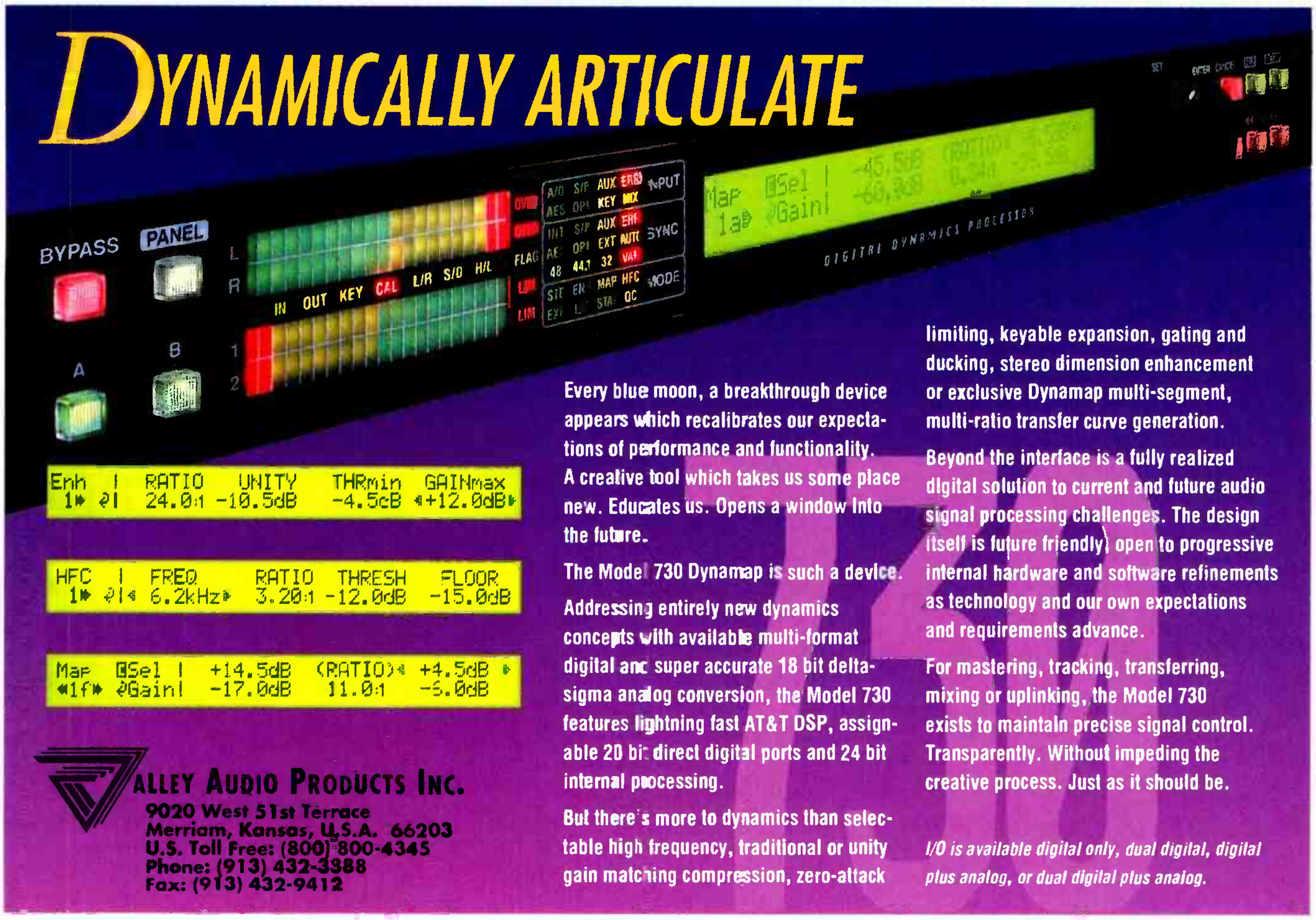
Two US studios have recently upgraded with Otari *Concept 1* consoles: Washington's Studio 5, primarily specialising in radio commercials, audio-for-video and music overdub projects, have bought a 48-input *Concept 1* with DiskMix automation including Snapshot recall functions, while Christian music studio Anthem Recording in Phoenix have installed a 28-input board.
Otari Corporation.
Tel: +1 415 341 5900.

● *Motionworker* upgrades

Motionworks, the studio systems integration specialists, have won an order to re-equip all four studios of Videosonics, the London-based postproduction facility specialising in broadcast television. Studios one, two and four will be fully upgraded with *Motionworker* Super Status Panels, including serial synchroniser options, and a completely new *Motionworker* system will be installed in studio three at the Delancy Street facility.
Motionworks. Tel: +44 865 883001.

● DDA sales

DDA have announced *DCM* console sales in New York and Brazil. A second DCM, a fully-specified 56-channel frame, goes to Mosh Studios for their multiroom facility in Sao Paulo and will be used primarily for mixing, while Matlin Recording in New York have installed a *DCM224V* in their recently opened Suite A. Matlin offer two audio layback suites, and Suite A is also equipped for film-scoring work.
DDA. Tel: +44 81 570 7161.


DYNAMICALLY ARTICULATE
BYPASS
PANEL
A
B
IN OUT KEY CAL L/R S/D H/L
DIGITAL DYNAMICS PROCESSOR
Enh | RATIO UNITY THRmin GAINmax
1 2| 24.0:1 -10.5dB -4.5dB +12.0dB
HFC | FREQ RATIO THRESH FLOOR
1 2| 6.2kHz 3.20:1 -12.0dB -15.0dB
Map 8Sel | +14.5dB (RATIO) +4.5dB
1f 2Gain| -17.0dB 11.0:1 -5.0dB
VALLEY AUDIO PRODUCTS INC.
9020 West 51st Terrace
Merriam, Kansas, U.S.A. 66203
U.S. Toll Free: (800) 800-4345
Phone: (913) 432-3388
Fax: (913) 432-9412
Every blue moon, a breakthrough device appears which recalibrates our expectations of performance and functionality. A creative tool which takes us some place new. Educates us. Opens a window into the future.
The Model 730 Dynamap is such a device.
Addressing entirely new dynamics concepts with available multi-format digital and super accurate 18 bit delta-sigma analog conversion, the Model 730 features lightning fast AT&T DSP, assignable 20 bit direct digital ports and 24 bit internal processing.
But there's more to dynamics than selectable high frequency, traditional or unity gain matching compression, zero-attack limiting, keyable expansion, gating and ducking, stereo dimension enhancement or exclusive Dynamap multi-segment, multi-ratio transfer curve generation.
Beyond the interface is a fully realized digital solution to current and future audio signal processing challenges. The design itself is future friendly, open to progressive internal hardware and software refinements as technology and our own expectations and requirements advance.
For mastering, tracking, transferring, mixing or uplinking, the Model 730 exists to maintain precise signal control. Transparently. Without impeding the creative process. Just as it should be.
I/O is available digital only, dual digital, digital plus analog, or dual digital plus analog.


LONDON - SINGAPORE - AUCKLAND - BERLIN - MUNICH - PARIS - MUNICH - KUALA LUMPUR - HAMBURG
Professional Studio Training
Full Time - Part Time Courses
Established 1977
THE ONLY INTERNATIONAL AUDIO TRAINING COLLEGE
18 Accredited Colleges
Studio Certificate Level
Audio Engineer Diploma
Advanced Music Production
Degree (B.A. Music Prod.)
Tonmeister Masters Program
FULLY SUPPORTED BY THE MUSIC INDUSTRY
Please call for our free course brochure and studio tour
sae
School of Audio Engineering
Individual practical training to the highest standard
London (071) 609 2653. Paris (1) 48 11 9696. Amsterdam (020) 689 4189. Sydney (02) 211 3711.
Vienna (01) 330 4133. Frankfurt (069) 543 262. Singapore (65) 334 2523.

Digidesign's TDM bus extends the *Pro Tools* system

Dynacord DSP 224

Dynacord's latest signal processor is the *DSP 224*, a multifunctional digital sound system processor designed to set up and optimise active multiway systems. It uses linear 18-bit A–D/D–A hardware with Sigma-Delta converters and the 24-bit Motorola DSP 56004, and its functions include a crossover, up to six parametric EQs, horn equalisation, high- and low-pass filters, four digital limiters, and two types of delay. The delays comprise a Master Delay for the main system, together with an alignment delay on each channel to position the individual components of a loudspeaker cluster in one acoustic plane.

Two operating environments are offered, both using a 2x16 character LCD. Preset mode allows the user to select loudspeaker types from the display menu and set the limiters to correspond to the power amplifiers in use. The Preset library contains EPROMs of many standard sound reinforcement systems, simplifying setup, but in Full Edit Mode the user has access to all parameters, allowing the setup and storing of any number of complex configurations which can be recalled at the touch of a button and altered to suit new requirements.
Dynacord. Tel: +49 9421 7060.

dB Optimizm

The *dB3000* from dB Technologies is a multifunction digital audio processor for sample-rate/data-format conversion, monitoring, measurement, debugging and correction for professional and consumer digital audio. dB's exclusive Acoustic Bit Correction is said to assure optimum clean-up of distortion and noise modulation common to digital transfer.

The functions provided include a sample-rate convertor with a 133dB noise floor, a data-format convertor giving bit reduction from 17 to 24 bits with selectable DSP-generated dither and a choice of noise shapes, reference metering with hold and clip indication, digital test tones, and signal analysis showing the actual number of bits being produced, the THD+N, frequency and amplitude of the incoming signal, and an analysis of A–D conversions.
Audio Intervisual Design.
Tel: +1 213 845 1155.

AirPlay 3.0 at IBC

Avid Technology will be introducing *AirPlay 3.0*, a multichannel disk-based playback system for news and commercial playback, for the first time in Europe at IBC '94. With *AirPlay 3.0*, broadcasters can use multiple channels to record and play simultaneously, increase regional advertising revenues via multicasting, and implement redundant broadcast strategies.

'Broadcasters are switching to disk-based playback systems because the benefits are too compelling to ignore,' says Tony Mark, Vice President and General Manager of Avid's Broadcast group. 'By introducing multichannel *AirPlay*, Avid is offering a playback system that is unparalleled in both reliability and flexibility'. *AirPlay* currently supports 1, 2 and 3-channel configurations, and can be expanded from a single to a multichannel system.
Avid Technology (US).
Tel: +1 508 640 3158.
Avid Technology (UK).
Tel: +44 71 434 0122.
Avid Technology Europe Ltd.
Tel: +44 753 655999.

Benchmark Audio World Interface

The new *Audio World Interface* from Benchmark Media Systems combines a 2-way recorder interface with switch selectable operation as a line amplifier or mono mix amplifier, which Benchmark claim is a first in versatility. The unit incorporates balanced instrumentation amplifier inputs and electronic transformer-balanced outputs, and front panel recessed gain controls for both input and output levels.

The half-width rackmount chassis includes LEDs to indicate mode switch position, signal presence, peak overload and power presence. Power comes from the external PS-1 supply, supplied as standard.
Benchmark Media Systems Inc.
Tel: +1 315 437 6300.

Digidesign TDM Bus

Digidesign have announced the availability of the TDM Bus. This is an open architecture 256-channel 24-bit digital audio bus that extends Digidesign's *Pro Tools* system to include advanced digital routing, mixing and patching capabilities, as well as an increasing number of hardware and software DSP 'plug-ins' from Digidesign and their many development partners. ▶

In brief

● **Sennheiser Drum and Instrument Microphone**
Sennheiser's *MD504* miniature dynamic microphone (launched at the UK BMF Show), is specifically intended for close drum-kit miking, with its small size (Sennheiser refer to it as thumb size), tailored frequency response and a claimed frequency-independent cardioid polar pattern. Sennheiser also say it is extremely robust, and capable of handling SPLs in excess of 160dB.
Sennheiser UK Ltd.
Tel: +44 628 850811.

Sennheiser's *MD504* mini mic

● **HHB in Bits**
HHB have followed their successful Indexer CD-R transfer unit with the *Bit Box*, which now features automatic sample-rate conversion, translation of DAT IDs to CD tracks (complete with variable audio delay to compensate for 'late' IDs) and conversion of status bits.
HHB Communications.
Tel: +44 81 960 2144.

● **Sabine Extermination Plan**
Sabine have responded to demand by producing the *FBX1802* 2-channel Feedback Exterminator, effectively two *FBX901*s in a 1U-high housing. Like the *901*, the *1802* features enhancements over the original *FBX900* including improved feedback distinguishing algorithms, better RF shielding, 18-bit resolution and a headroom of 25dBV. The two channels have completely separate controls, metering and filter-activity displays and are controlled independently from each other.
Shuttlesound. Tel: +44 81 640 1900.

● **The Digital Reporter Terminal**
Audio Processing Technology have introduced the *DRT128* Digital Reporter Terminal, a compact portable unit designed to deliver high quality audio commentary during outside broadcasts and in limited bandwidth applications.
The *DRT 128* incorporates a 2-channel ISDN terminal adapter with internal IMUX and uses APT's proprietary audio data-compression technology to provide high quality

Audix Broadcast *DTX*

Among the enhancements to *Pro Tools* are an integrated digital mixing environment with multiple sends, inserts, buses, subs and masters, an open system that allows third parties to develop hardware and software plug-ins that run on the Digidesign DSP Farm card, and flexible routeing, enabling control of sources and destinations from pop-up menus.

The TDM system consists of a piggyback card that attaches to the *Pro Tools* audio card, a DSP Farm card that provides a powerful effects engine for processing and mixing, a ribbon cable that provides high speed data bussing, and a software update which is the interface for the current *Pro Tools* software.

Digidesign Inc (US). Tel: +1 415 688 0600. Digidesign Inc (UK). Tel: +4481 875 9977.

Audix phone-in

The *DTX* from Audix Broadcast is a new phone-in system for radio and television broadcasters which incorporates digital sound processing techniques to improve the sound quality, and includes features designed not only to enhance flexibility and versatility but to make it possible for less experienced announcers to present phone-in programmes with a high degree of skill.

In its simplest form, the *DTX* consists of one digital hybrid with four external lines and an announcer's desk interface, while its maximum size is four expanded digital hybrids, 24 external lines, four studio interfaces and one pivotal control station. The announcer can select calls individually, screening them before allowing them to go out on air, or the system can be programmed for automatic call rotation on a first-in/first-out basis. The status of individual lines is shown on multicoloured solid-state indicators

In larger configurations, the control of call screening and routeing can be handled by an assistant announcer through a separate panel, with ultimate control always in the hands of the announcer.

Audix Broadcast Ltd. Tel: +44 799 542220.

DDA *Network 7*

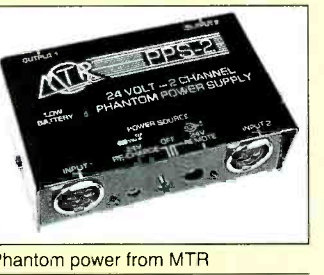
DDA have announced the release of their first dedicated broadcast mixing console with *Network 7*. Aimed specifically at the on-air, one man operation, radio studio, *Network 7* can be customised with any combination of mono, stereo or telephone input modules, each of which can be supplied with or without EQ, routeing and ducker front panel controls. Three frame sizes of 16, 24 and 32 modules are possible, all with ▶

audio over low-capacity circuits.
Audio Processing Technology Ltd. Tel: +44 232 371110.

● MTR's Portable Phantom

MTR have introduced a new portable phantom power supply. Housed in the same steel box as the company's DI boxes, the *PPS-2* has two channels which can be used simultaneously, and delivers 24 volts per channel from three batteries, three rechargeable batteries, or an external DC supply. Rechargeables can be charged *in situ* through a dedicated socket, and the unit also features a low-battery/low-DC supply LED and a recharge indicator LED.

MTR Ltd. Tel: +44 923 234050.



Phantom power from MTR

INSTANT PICTURE FOR INSTANT AUDIO

The **Virtual Recorder®** **(VR™)** is a random-access, digital video recorder that can record and playback timecoded video, plus 2 channels of audio. It also costs significantly less than you may have thought.

VR works with any device (via Sony 9-pin), and for any application where you would normally have used a video tape recorder. And, with VR you get the added advantage of instant access to any point on your video, which can be seamlessly recorded across up to seven 9GB drives. That's around eight hours of video!

Now, with a video recorder that responds as quickly as a digital audio workstation, you'll be amazed at the increase in your production and operational speed.

The time-saving benefits of VR are obvious for a variety of applications:

Post-Production – Now you can do ADR and Foley to video. While laybacks, sound effects, mixing and scoring are all significantly faster.

Broadcast and Cable – Instant search and retrieval makes automated spot insertion extremely economic, with one VR instead of a whole bank of VTRs.

Interactive Video – VR's re-usable data storage, instant retrieval and digital reliability makes it ideal for use in video kiosks etc.

Get up to speed with video today – call Stirling Audio Systems and arrange a demonstration on the Virtual Recorder.

Stirling

071 624 6000

Stirling Audio Systems Ltd • Kimberley Road, London NW6 7SF
Tel: 071 624 6000 Fax: 071 372 7660

desktop, semiflush and flush mounting possibilities.

The console architecture consists of two subgroup buses and two auxiliary buses. A separate record bus with automatic routing override allows an independent safe recording to take place, from any input module, while the console is on-air. All inputs and outputs are fully balanced, and four clean feed buses are available, three automatically routed to the telephone-caller output and one which can be locally assigned. All input modules incorporate a built-in ducker system allowing automatic attenuation of channels when a duck-master channel is in use, and the mono and telephone inputs also include a limiter.

DDA. Tel: +44 81 570 7161.
Mark IV Pro Audio Group. Tel: +1 616 695 4750.

KRK monitors

KRK have introduced optional video shielding on two of their most popular nearfield monitors, the *Model 6000* and *7000B*. The *6000BS* and *7000BS* incorporate mu-metal cans around the drivers to join the growing band of monitors which can safely be sited next to video and computer screens without affecting the picture.

Also new from KRK is the *Model 15P-3* passive 3-way main monitor system. The monitor incorporates a 15-inch polyglass woofer, 7-inch Kevlar midrange and 1-inch Kevlar tweeter, giving a claimed frequency response of 29Hz to 19kHz ±3dB with a maximum power handling of 250W. KRK quote a metric sensitivity of 92dB and a maximum SPL of 115dB, and regard the sub-$9000 price as a breakthrough.

KRK Monitoring Systems. Tel: +1 714 841 1600 ■

KRK Model 7000B with optional video shield

● **Power from Crown**

APRS saw the UK launch of Crown's *MA-3600VZ*, the first in a new series of Macro-Tech power amplifiers. It uses Crown's freshly-patented variable impedance power supply circuitry to deliver 1800W per channel into 2Ω from a 2U-high chassis. Crown claim that it achieves maximum power-matching over a very wide load range thanks to its ability to, 'identify and consequently adapt dynamically to both signal and load requirements.'

All Macro-Tech amplifiers will be fully ODEP protected, and incorporate P I P compatibility allowing the user to tailor the amplifier for specific requirements using a range of P I P accessories.

Crown are now offering for all their amplifiers a 3-year no-fault warranty, promising to repair or replace any amplifier which, for any reason at all, fails or does not perform to original specifications.

Shuttlesound. Tel: +44 640 1900.
Crown. Tel: +1 219 294 8000.

DK Audio *MSD550-SA*

The Danish company DK Audio introduced their software controlled Master Stereo Display *MSD550* metering system in 1992. Since then the product has undergone two revisions—v2 offering a number of additions and enhancements, while the recent v3 further increases functionality and equips the unit with a real-time FFT (Fast Fourier Transformation) spectrum analyser —*MSD550-SA*.

The *MSD550* has been designed to interface to any mixing console as a versatile master metering system, and a number of console manufacturers have expressed considerable interest in incorporating it into their designs—SSL, in fact, are the first company to build-in the display as a standard feature for their *OmniMix* system.

The compact unit (6 x 5 x 2¾ inches) is supplied with a pivoting mount for fixing to a console top, worksurface etc, and all connections (analogue I-Os, the 9V AC or 12V DC power supply, and RS-232 for external computer interface) are made internally. The display itself is a 320 x 240 dot LCD, with adjustable contrast and background lighting intensity, and benefits from a wide viewing angle. Control is from six softkeys placed directly below the display which relate to on-screen menu boxes.

The main screen display is divided into three areas: stereo level bargraph meters to the right, a phase bargraph meter to the left and a vector oscilloscope in the middle. Thus allowing stereo information and programme level to be viewed at the same time without having to switch between screens.

The unit incorporates the five most popular international level scales, each being accessible via single softkey selection. These are: IEC 268-10: Type I (-42dB to +9dB); and IEC 268-17 VU (-20dB to +3dB). There is also the CCIR 468-3 noise meter (-50dB to +10dB). At the moment no provision has been made for full scale digital metering, mainly due to its nonstandardisation, but according to the system's designer, Karsten Hansen, the next software release will offer an FS meter with a -50 to 0 scale which will include a user-definable 0 reference level.

Apart from accommodating different types of metering, the stereo bargraphs can also be set to peak hold, temporary peak hold, fast integration (measuring the true peak of the input signal, which, of course, is very important for correctly measuring signals being output to digital equipment), and can be sensitised by 20dB. The phase meter can be set to operate either at fast or slow speeds depending on user preference.

Level meters may be calibrated either by using an external reference, or the built-in oscillator (33Hz to 21kHz). As calibration data is stored in EPROM, a write-protect function is defaulted to in order to guard against inadvertent adjustment of this and other parameters.

The vector oscilloscope provides a traditional sum and difference type display providing information on phase, amplitude and stereo spread. It uses a crossed axis display, with the horizontal axis (marked +S to -S) showing the difference or 'side' component of the signal, and a vertical axis (marked Mono) showing the sum or 'middle' component of the signal.

Spectral analyser

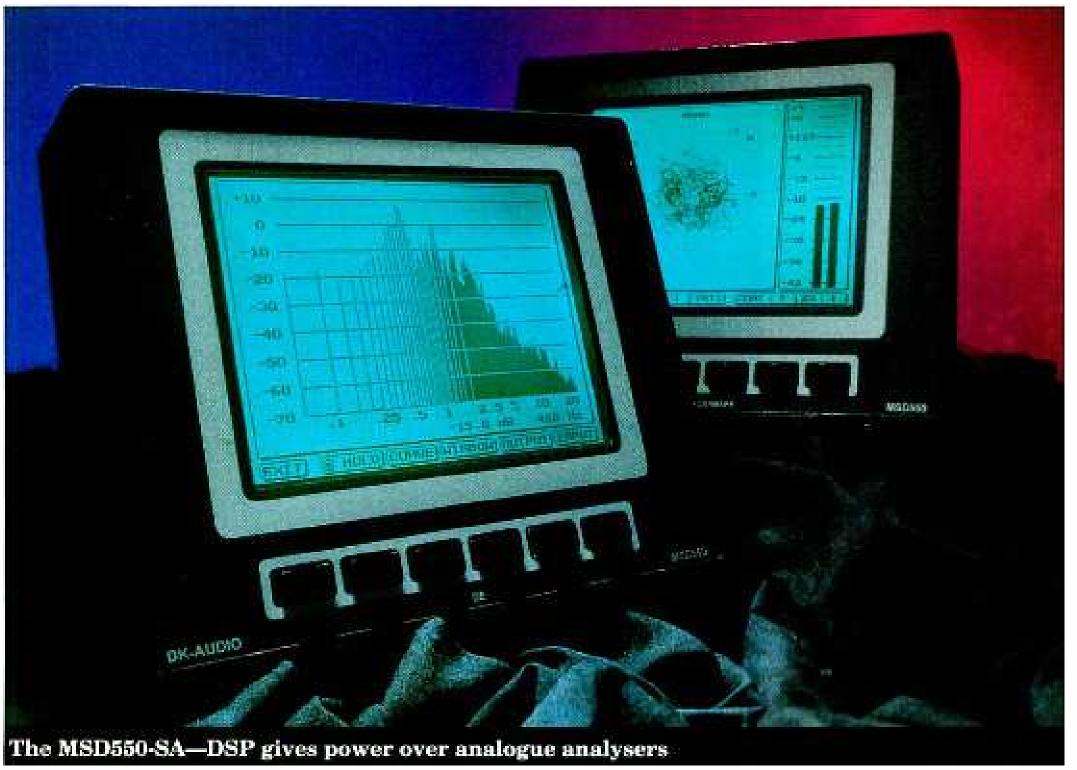
Unlike spectrum analysers based on analogue filtering techniques, the DSP controlled *MSD550* is able to implement FFT thus increasing the number of frequency bands from the traditional 27 to 1,024. Consequently, the unit manages to pack a powerful real-time analyser into a surprisingly small area, and used in conjunction with the internal white-noise generator can be used for testing equipment response, room acoustics and so forth.

Input to the analyser can be Left only, Right only, Left plus Right, or Left minus Right. Dynamic range is 80dB, and the display is scaled vertically from -70dB to +10dB; and horizontally from 46Hz to 24kHz. The graphic display includes a Left-Right cursor that allows any band in the spectrum to be pinpointed giving an exact reading of both frequency and level. The display can be relatively rescaled so that the loudest component peaks to 0dB, or by using the cursor, a particular band can be referenced to 0dB thus providing a clearer indication of its relationship to the rest of the frequency spectrum. A Hold facility is also provided that disables fallback, allowing a peak curve to be built-up over time.

To compensate for background noise generated by the narrow band FFT process, two filtering windows are selectable—Hanning, Hamming —which offer differing levels of filtering to attenuate out-of-band interference and thus improve dynamic response. A third selection—Rectangular Window —allows the analyser to be configured without filtering.

Also included are two memories for permanent and temporary storage of curves. These 'snapshots' can be added to the current spectrum to show-up curve differences.

Existing *MSD-550*s equipped with v2 software may be upgraded to v3 (FFT) at a cost of approximately £732 (UK price). The fully equipped *MSD550* is priced at £2,075, but can also be supplied in 'Demo' mode, whereby the user may try out the spectrum analyser up to 50 times before it becomes software disabled. If the customer decides to take up the option, a 5-figure licence number will need to be purchased from the dealer and entered into the system to permanently activate the facility. ■



The MSD550-SA—DSP gives power over analogue analysers

Danish Pro Audio, Hejrevang 11, DK 3450 Allerod, Denmark.
Tel: +45 48142828.
Fax: +45 48142700.
UK: Plasmec Systems Ltd, Farnham Business Park, Weydon Lane, Farnham, Surrey GU9 8QL.
Tel: 0252 721236.
Fax: 0252 712718.

Prism Sound *Dscope*

Not long ago, competent audio engineers could be expect to be able to keep pretty well all the equipment they used in good order—checking it routinely and realigning where necessary. All you needed to calibrate most gear was something like a Ferrograph *Recorder Test Set*, a test tape an *AVO* meter and, perhaps, an oscilloscope. Armed with these simple weapons, you could measure anything of importance and know what to do in response to any unwelcome results.

Today, all this has changed. There is little that happens inside digital equipment that the average audio person can understand in detail, never mind measure—DAT machines in particular have lost the cosy familiarity of analogue faults and misalignments, giving little indication of deteriorating performance until they become unuseable. At this point, expert help is required.

A digital recorder is an extreme case. There are many aspects of digital signals which could usefully be measured but part of the problem with digital equipment is that its performance parameters are an order of magnitude better than those of the kind of test gear we used to use. To measure digital equipment you need digital measuring equipment, and most of this is too complex and expensive to be a realistic addition to the average studio. Most facilities will, however, own a PC of some sort —which, paired with Prism Sound's *Dscope* system, can become a sophisticated, versatile test set capable of dealing with digital signals of up to 24 bits, at considerably lower cost that comparable dedicated hardware.

There is, of course, hardware involved. This takes the form of a half-length AT-compatible expansion card; one of these can operate as either a signal generator or an analyser interface and two can be installed simultaneously. The card carries a D-connector, and a labelled flail is supplied to break this out into the various inputs and outputs. These connections comprise SPDIF and AES-EBU in and out, and analogue monitor outs on male XLR connectors. The system is designed solely to interface digitally with the equipment under test—there is no analogue input and no analogue generator output. Naturally, Prism Sound suggest their own *AD-1* convertor as an analogue interface.

The principal software associated with the cards is *Dscope* itself, which provides a versatile digital signal generator, a digital oscilloscope, Fast Fourier Transform spectrum analysis and various sweep functions. The user interface is a graphic screen which can be operated from the keyboard or, in most cases, with a mouse (although *Dscope* does not require *Windows*). Most of the screen is taken up with the display which can show any or all of the analyser functions simultaneously in different colours; arranged around this are a number of control areas or 'tiles' which can toggle various functions or open up submenus for detailed control.

The signal generator provides a wide range of waveforms—ramp and square both with variable duty cycles, sine, twin-tone and noise—along with the facility to generate user waveforms for special purposes. An example of this is Prism Sound's own J-test waveform, which presents particularly troublesome data to D–A convertors and highlights problems such as jitter from various sources and the effects of long cable runs in an AES-EBU interface. This is an area of particular interest as Prism Sound have identified THD figures of -65dB to -70dB caused by factors such as the loss of squareness of this signal in 100m of standard twin-screened cable. The output word length can be set from 16 to 24 bits and the output level is fully variable.

The simplest section of the analyser is the oscilloscope, which provides all the expected facilities: variable X-axis, auto or manually variable Y-axis, and a graticule for measurements which is augmented by a cursor which can be placed anywhere on the waveform to extract the amplitude and time measurements at that point. The serious stuff, however, comes with the spectrum analyser.

Dscope employs a user-selectable number of FFT points from 1,024 to 4,096. Obviously, the more points used, the more spectral detail will be visible, the trade-off being the time necessary to perform the calculations. A bar on the screen shows the progress of the calculations and also whether single-shot or averaging is selected. Averaging the calculation of a (user-definable) number of samples of the input reduces the effects of random factors and makes the displays generally easier to read. A panel beside the main display shows, for each reading, peak level and THD+n either in dB below peak or as a percentage. The cursor may be used for specific readings from this.

A sweep function allows various parameters to be plotted against one another. As with the other modes, scale options are available on linear and logarithmic axes.

Comprehensive storage is available for saving traces for comparison with other results, and for saving settings of both generator and analyser for future use. Printouts at various resolutions are available with on-screen previews and a progress bar—printing can take some time.

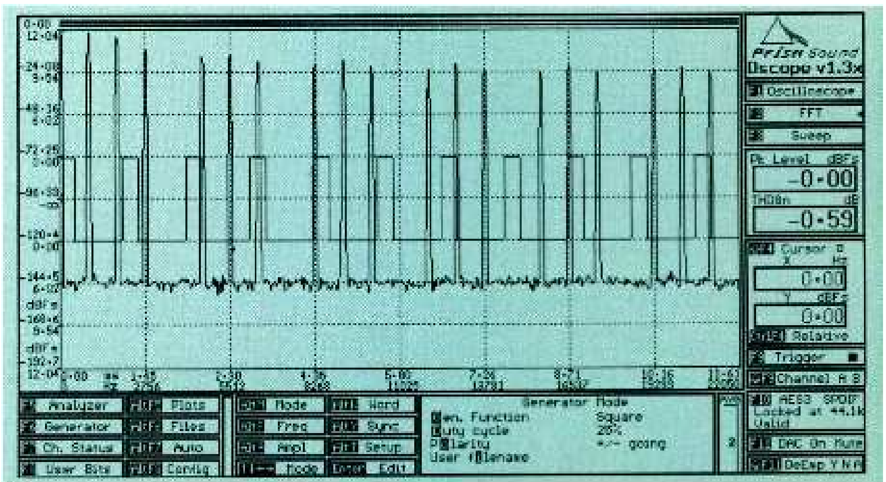
In fact, several *Dscope* operations can take some time. There is a huge amount of number crunching going on, increasing with the number of FFT points chosen. Clearly the faster the PC, the less time the computation will take—never was a maths coprocessor more desirable. That said, I ran the system on my own *486SX25* and while I would not like to use such a system all day long, it is perfectly adequate for occasional measurement and maintenance. Everything bar 64-point FFT sweeps—even ten averaging FFT readings—is acceptable, and the resolution and clarity of the results are worth the wait.

The software reviewed here is a ß evaluation version of the latest package which features the introduction of the sweep function. I encountered bugs which Prism Sound are aware of and curing, but overall the system is friendly and intuitive even if this type of equipment is less than familiar. There are oddities: mouse response can be slow and although almost everything can be controlled by mouse or clearly indicated keystrokes, there are operations (such as switching on the FFT averaging mode) which are 'hidden' on a function key to be remembered or forgotten, as the case may be.

For maintaining a variety of recording equipment and evaluating new equipment, *Dscope*, even on a modest computer, represents a powerful test system, perhaps warranting the purchase of decent A–D and D–A convertors to complement it. On a faster, better specified machine, it should meet the needs of most facilities at a fraction of the cost of the equivalent dedicated hardware. ■

Dave Foister

Prism Sound, The Coulson Building, Cowley Road, Cambridge CB4, UK.
Tel: +44 223 424988.
US: Sprocket Digital, 211 North Victory Boulevard, Burbank, California 91502.
Tel +1 818 566 7700.
Fax: +1 818 566 4477.



Dscope **screen showing square wave generation and FFT**

'We've been using Sony DASH for ten years, and look forward to the next ten.'

Martin Benge, Managing Director, EMI Music Studios UK

The Manor, Oxfordshire, UK

'Ever since we bought our first Sony DASH recorder for Abbey Road Studios in 1984, we've been keen supporters.

'Superbly engineered, easy to use, transparent sound – over the years our Producers and international clients alike have all appreciated what DASH has done for them.

'Our studio group now includes The Town House, Town House III, Olympic, The Manor and Manor Mobiles as well as Abbey Road. Each has its own distinct appeal for recording artists, but all offer DASH.

'More and more artists will now record only on DASH. That's why we decided to invest in some extra PCM-3348s. In virtually constant use since the day they were installed, they are giving our clients the sound quality they need and bringing in more business.

'For us, Sony DASH has been a big success story. Long may it continue.'

SONY

SONY BROADCAST & PROFESSIONAL EUROPE, JAYS CLOSE, VIABLES, BASINGSTOKE, HAMPSHIRE, RG22 4SB, UNITED KINGDOM.

SIMPLY CALL US ON: AMSTERDAM 020 6581911; BRUSSELS 02 724 1711; COLOGNE 0221 59660; COPENHAGEN 43 995100; DUBAI 04 313472; HELSINKI 0 50291; ISTANBUL 0212 224 5961; LISBON 01 837 2566; MADRID 091 536 5700; MILAN 02 61838440; MOSCOW 0501253 4854; OSLO 02 2303530; PARIS 01 4945 4000; ROME 06 549 151; STOCKHOLM 08 7950800; VIENNA 0222 61050; WEYBRIDGE, UK 0932 816300; ZURICH (SCHLIEREN) 01 733 3511; EAST CENTRAL EUROPE, CIS, BALTIC STATES – UK 0256 483294; MIDDLE EAST/NORTH AFRICA – GENEVA 022 7336350; AFRICA – UK 0256 483248.

Peter Cornish

After 20-odd years in the business in London, guitar systems guru Pete Cornish is moving to the country to continue his services for players who know what they want.

Cornish's career started in the service department at Sound City when it was importing Fender, Acoustic and MXR and he distinguished himself as someone who could build solutions as well as fix things. A string of 'little boxes' led him to branch out on his own in 1972 with his first pedalboard, for Yes, and establish an association with name acts that has never waned. His client list includes King Crimson, Genesis, Roxy Music, Pink Floyd, Bebop Deluxe, UK, Police, Status Quo, Queen, Judas Priest, Black Sabbath, The Pretenders, Greg Lake, Iron Maiden, Dire Straits, Dave Gilmour, Moody Blues, Eric Clapton, Paul McCartney, Marillion, Chris Rea, Sting, Bryan Adams, Lou Reed, John McLaughlin, and Jimmy Page. For these he has built everything from voltage stabilisers, power distributors, DIs, preamps, routeing systems and complete systems.

However, it was with pedalboards that he initially carved his reputation. In the first instance this involved mains-powering the board and doing away with multiple plug-and-wire connections but led to removing pedals from their boxes, reworking the circuitry to improve reliability, and building them into the pedalboard with the knobs and switches brought out to the surface.

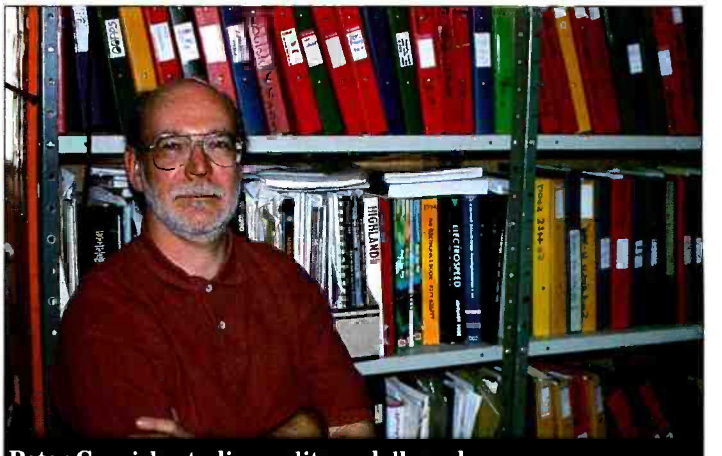
And they were built to last, although Cornish cannot hazard a guess at their life expectancy. 'None of them have worn out yet,' he says. 'David Cross from King Crimson came in recently with his violin pedalboard from 1973 and it was still perfect.'

Cornish plays down his own contribution and insists that he only builds what players ask for. Thus Bill Nelson got a light on his pedalboard and a sloped footrest to counter the forward pull of raked stages while John Whetton's 'goal-post' pedalboard was built to accommodate Moog Taurus bass pedals.

'It's customised to the player, we measure their output and the size of their feet for the pedalboard,' he says waving a cut-out of Mark Knopfler's foot. Boards are built from water and temperature-proof birch ply after numerous full-size mock ups.

'I've never hidden the fact that there were manufacturers' pedals inside my boards,' explains Cornish. 'The fact that I've changed them and added to them is really part of getting the whole thing, the overview, and designing it from start to finish.'

The move into complete systems with amps and speakers started with Pink Floyd on The Wall in the late 1970s. 'I made them better, run quieter, more reliably,' he says. 'My clients have a very good idea of what they want and they've tried everything over years and years, but it still isn't right. There's a noise problem. The biggest thing is getting rid of hum, and I can get rid of all of it and the noise without gates,' he states. 'Connecting things together is the problem. Any one thing on its own is usually fine, but 12 pedals with all that wire hanging on a passive instrument? I can get rid of all the noise on any rig—never been beaten yet,' he laughs, and claims that this is the drawback of modern affordable pedalboard-switching systems that mimic the functionality of his creations. 'They'll still hum, I'm afraid. You can't hook them up without it making noise, it's impossible.'


Peter Cornish: studio-quality pedalboards

A visit to Iron Maiden recording in Amsterdam in the early 1980s made him realise that his pedalboards were good enough for studio use but he believed he could make them better. Every circuit was redesigned to get the noise down still further and all Cornish equipment now carries the *Studio Series* monicker. His work with routeing systems started in the studio with Producer Mutt Lange and Bryan Adams for whom he built a unit that allowed amps to be placed in the recording area and controlled remotely from the control room. This involves an interface between the guitars and pedals to bring them up studio level, another for preamps and effects, and a third for signal to amp routeing with reverse phase, gain adjustment and mutes. Level adjustment has become critical now that digital rack-effects are integral to rigs. 'Every single independent circuit has its own in and out gain control so you can use any combination of pedals and rack effects together,' he says.

While he refuses to redo or modify other people's pedalboards, systems or lash ups he is often called to rework his own designs in line with a player's changing requirements, as was the case with Dave Murray's enormous pedalboard from Iron Maiden.

'A few years later he came back and asked for it to be made smaller because particular bits of kit were never turned off and could be put somewhere else off the stage. We did that again several years later and made this tiny little board. He wasn't using any less equipment it was just that it was being removed from the pedalboard,'. Things like MIDI have also enhanced the degree of remote control in big rigs.

However, he reports a subtle move back towards traditional pedalboards incorporating modern effects boxes. 'It's has to do to with fashion,' he says, 'but I think players like the convenience of being able to plug in and get on with it.' And that after all is what Pete Cornish has always aimed to achieve. ■

Pete Cornish, Ailies Buildings, Whitesmith Lane, East Hoathly, East Sussex BN8 6QP.
Tel: +44 825 873033.
Fax: +44 825 873044.

Music News is compiled by Zenon Schoepe

HHB DAT TAPE

INTERNATIONAL DISTRIBUTORS

AUSTRALIA

BELGIUM

CANADA

CZECH REPUBLIC

DENMARK

FRANCE

GERMANY

ISRAEL

JAPAN

NETHERLANDS

NORWAY

POLAND

SINGAPORE, MALAYSIA AND THAILAND

SOUTH AFRICA

SPAIN

SWEDEN

USA

SIMPLY THE FINEST DAT TAPE IN THE WORLD

HHB COMMUNICATIONS LIMITED

IN INDEPENDENT TESTING OF EIGHT LEADING DAT TAPES CONDUCTED BY STUDIO SOUND MAGAZINE, HHB DAT TAPE EXHIBITED EXTREMELY LOW BLOCK ERROR RATES, UNPARALLELED CONSISTENCY AND SUPERIOR ARCHIVING STABILITY.


ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT15
DAT
ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT30
ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT48
DAT
Digital Audio Tape
ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT62
DAT
Digital Audio Tape
ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT92
DAT
Digital Audio Tape
ADVANCED MEDIA PRODUCTS
For Professional Audio Applications
DAT122
DAT
Digital Audio Tape

MANAGING THE BIT BUDGET

Francis Rumsey reports on the recent AES UK Conference on digital audio techniques concerning bit rates in conversion and transmission, and bit-rate reduction

Digital audio used to be a relatively simple business. But now that the technology has begun to mature, and digital audio is becoming the norm rather than the exception, engineers and managers are increasingly concerned with the economic trade-offs between digital audio resolution and sound quality. Research is developing in two apparently opposing directions: one towards higher sampling rates and number of bits per sample, and the other seeing how few bits can be used to represent audio signals. There are economic benefits to broadcasters if more digital broadcasting channels can be squeezed into a given bandwidth, but the quality must not drop too low, thus fuelling the development of bit-rate reduction systems. Conversely, there are advantages to recording engineers if they can make master recordings with a higher resolution than that of the CD, leading to such things as 20-bit convertors and higher sampling rates.

For each person involved with audio, balancing this bit budget involves trade-offs between financial expenditure, operational convenience and sound quality. A knowledge of psychoacoustics is beneficial in helping engineers to understand what distortions may be audible and what may not. Only when this knowledge is available can the right compromises be made in engineering design.

During the course of a recent 2-day conference entitled 'Managing the Bit Budget', organised by the British Section of the AES, a group of international audio experts tackled a range of topics intended to provide delegates with information that would allow them to make more informed judgments. Held in London, the conference attracted delegates and speakers from the UK, continental Europe, Scandinavia and the US.

Opening session

Opening the proceedings the conference Chairman, Neil Gilchrist (BBC R&D), welcomed the first speaker Louis Fielder, President Elect of the AES, who was to give a paper setting the scene for the remainder of the conference. This most excellent paper attempted to determine the required dynamic range for the modern high-quality digital audio environment, by investigating the extremes of sound pressure available from acoustic sources and comparing these with the perceivable noise floors in real acoustic environments. Using psychoacoustic theory, Fielder was able to show what was likely to be heard at different frequencies in terms of noise and distortion, and where the limiting elements might be in a typical recording chain.

Having defined dynamic range he proceeded to show that the just audible level of a 20kHz bandwidth noise signal was about 4dB SPL, and that a number of musical performances reached levels of between 120dB and 129dB SPL. From this he determined a dynamic range requirement of 122dB for natural reproduction, a factor which he felt might worry some designers. Taking into account microphone performance and consumer loudspeakers, this requirement dropped to 115dB for consumer systems. He pointed out that most loud signals were predominantly low frequency in content, thus making pre-emphasis curves such as J17 acceptable in many cases.

Conversion and processing

Chaired by Francis Rumsey (University of Surrey), the second session dealt with conversion and processing of digital audio, looking at some of the issues involved with specifying and enhancing convertor performance. As introduced by the chairman, the industry has seen considerable controversy in the last year concerning the meanings of terms used to describe convertors, leading the Deutsche Grammophon recording company to challenge the AES to define certain key issues.

Jim MacArthur of Lexicon proceeded to describe a number of techniques used to enhance the performance of A–D convertors. DSP could be used to remove DC offsets and introduce limiting characteristics for example. He encouraged delegates to stop talking about convertors in terms of the number of bits involved, because this really told very little. What were important, he said, were aspects such as linearity and dynamic range.

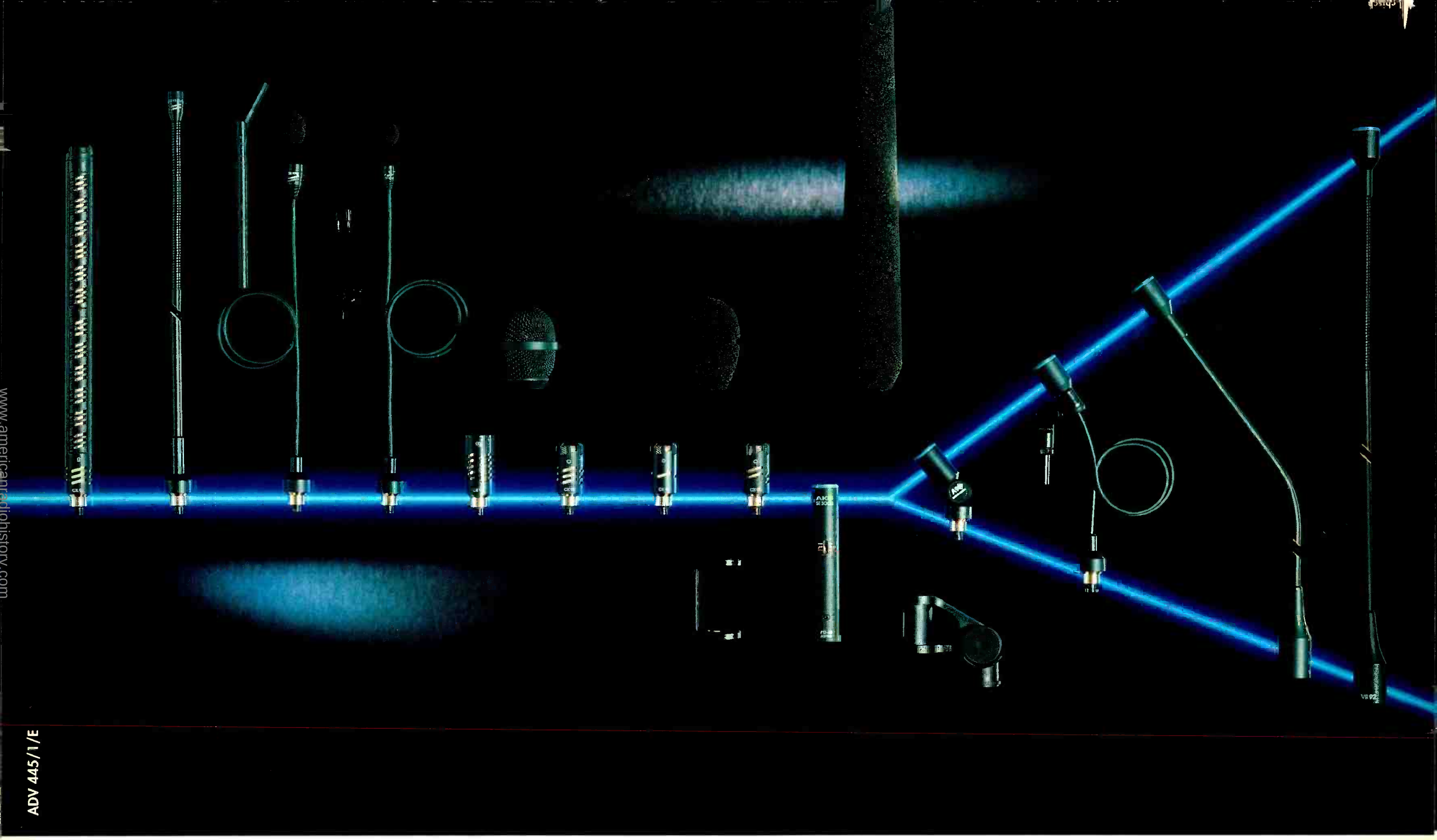
Mike Story of the UK convertor manufacturer Data Conversion Systems (dCS) spoke about issues which pertained to the specification and characterisation of high-resolution A–D convertors. He discussed differential nonlinearity (DNL) as a key factor affecting sound quality, and showed how the addition of dither noise to improve linearity might actually be counter-productive in cases of bad DNL. Story's published paper included a number of examples of curves and responses which could be used to describe convertor performance, and made recommendations for appropriate measurements.

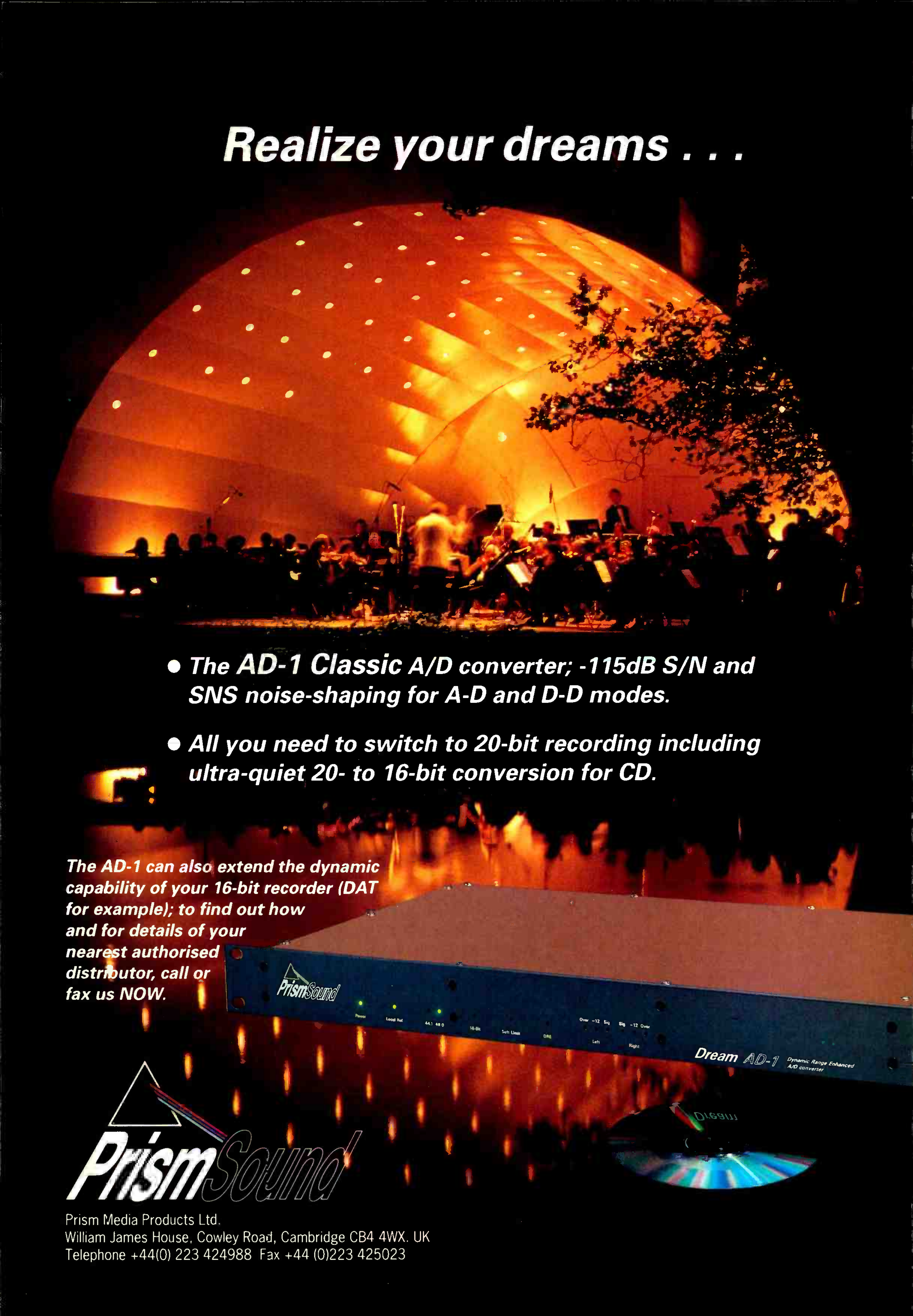
Proposing a somewhat unusual but valid concept, James Angus from the University of York (UK), described what he called the 'one bit alternative' for audio processing and mastering. Why not, he asked, perform storage and processing on single bit signals instead of multibit PCM data? Many convertors now operate as highly oversampled devices whose outputs are normally filtered and decimated to produce multibit PCM at the nominal sampling rate. Why not leave the audio data in the oversampled state? There would be many advantages. The frequency response would not stop at 20kHz, the audibility of errors could be lower than for multibit signals, the phase response would be improved, and processing and interfacing would be simpler. Concluding the session, Malcolm Hawksford (University of Essex, UK) provided an excellent overview of D–A convertor techniques, all the way from simple R/2R ladders to advanced 'bitstream' convertors. He showed how oversampling and digital filtering could be used to advantage in D–A conversion as well as at the A–D stage.

Dynamic range

Chris Travis (Division Ltd, UK) chaired the first day's final session, which dealt with issues relating to dynamic range in digital audio. Rhonda Wilson of Meridian Audio opened by demonstrating and discussing a variety of different dither spectra which were appropriate in digital audio systems, and provided a description of different approaches to noise shaping as a means of improving perceived dynamic range. Following this paper, Paul Levin of Apogee Electronics (US) spoke about his company's search for the best method of getting high dynamic range audio information onto the 16 bits of a CD so that all listeners could benefit. Apogee had chosen a narrow-band dither signal at half the sampling rate, modulated with a random number sequence which had taken many days of computing time to determine, the result of which was to improve the audible dynamic range and allow the reconstitution of signals which would normally have been well below the noise floor. The technique was said to have been very well received by US mastering engineers.

Three papers from the broadcasting community described methods that had been adopted for measuring and controlling the dynamic range of transmissions. Charles Girdwood (ITC) and John Emmett (Thames TV) described the development of a loudness meter to give readings which correlated more closely to people's perception of programme loudness than the standard PPM. Neil Gilchrist, the conference Chairman, then proceeded to describe a system known as DRACULA (the BBC acronym strikes again!) which could be used as an automatic controller of programme level without the unpleasant side effects associated with other forms of compression. To some extent DRACULA worked as a balance engineer would, to ride the gain of faders so as to reduce dynamic ▶


AKG
VR92
ADV 445/1/E


Realize your dreams . . .
• The AD-1 Classic A/D converter; -115dB S/N and SNS noise-shaping for A-D and D-D modes.
• All you need to switch to 20-bit recording including ultra-quiet 20- to 16-bit conversion for CD.
The AD-1 can also extend the dynamic capability of your 16-bit recorder (DAT for example); to find out how and for details of your nearest authorised distributor, call or fax us NOW.
PrismSound
Dream AD-1
Dynamic Range Enhanced A/D converter
PrismSound
Prism Media Products Ltd.
William James House, Cowley Road, Cambridge CB4 4WX. UK
Telephone +44(0) 223 424988 Fax +44 (0)223 425023

range for broadcasts. His demonstration showed that although delay was necessary in the device, so as to allow the looking ahead required to anticipate changes in level, this could be reduced as low as 30ms while still sounding reasonably acceptable. One intention was to use the device to provide gain control information that could be transmitted with future digital audio broadcasts. This information would then be used by receivers to reduce dynamic range if required for listening situations such as those encountered in cars.

Finishing the day, Robin Cherry of BBC Radio gave a relatively light-hearted view of the creeping increase in dominance of automatic modulation controllers such as *Optimod*. Using recorded examples he showed how the sound quality of radio broadcasts was gradually being subverted by managerial pressures to boost the loudness of radio stations for competitive reasons. Delegates noted wryly that the day had started with a requirement for a dynamic range of 122dB and finished with a dynamic range of around 10dB!

Bit-rate reduction

Day Two of the conference was opened by John Nunn (BBC), chairing the session on bit rate reduction. Rather than concentrate on the now well-worn topic of ISO-MPEG standards and psychoacoustic masking, this session attempted to look at some alternative approaches and introduced some issues relating to implementation in practical situations.

Claude Cellier of Merging Technologies (Switzerland), gave a lively talk and demonstration about lossless compression, showing how significant real-time reductions in bit rate could be achieved with absolutely no loss of digital audio information. He noted with some amusement, and with tongue firmly in cheek, that analogue audio in the 1970s had offered a spectrum with modulation noise and distortion components, whereas nice, clean digital audio had come along in the 1980s. Now, in the 1990s, psychoacoustic coding had brought back a spectrum similar to that of analogue. Using examples of real programme material he showed how many types of audio allowed coding gains of over 2:1, which was significant and useful for professional purposes.

Werner Oomen, of Philips in The Netherlands, then described a means by which a variable rate buried data channel could be encoded onto normal CDs by using psychoacoustic effects to 'hide' the data signal under the masking curve of the audio signal. The data signal is randomised so as to appear like shaped noise under the audio signal. This data channel could be used for many purposes, including the carrying of 5-channel MPEG surround sound data on CDs.

Audio standards for multimedia computers were then reviewed by Mark Taylor from Crystal Semiconductors. He showed how varying degrees of sound quality were available from multimedia PCs (he restricted himself to 80 x 86 machines), and how MIDI and various ADPCM audio options could be used to keep the data rate required for sound applications low. The upcoming MPEG standards would be used widely in the future for multimedia applications.

Following a break for coffee, Colin Spicer (BBC R&D) described a 7.6kbit/s system based on speech coding principles, which could be used to insert a 2-channel talkback signal into lines of the TV vertical blanking interval. The CELP coding principles were similar to those used to code audio description information for visually impaired people in the European AUDETEL project. Subsequent to this presentation, Fred Wylie of APT compared a number of different approaches to high-quality data-rate reduction, and looked at ways of linking codecs with different sample rates and resolutions at various points in the signal chain. He finished with the recommendation that a minimum data rate of 192kbit/s should be used at the first stage in a complicated signal chain, with a minimum of 128kbit/s per mono channel if further processing of the signal was anticipated.

Of great interest to many delegates from the operational side of the audio business was The Audio Exchange's Bill Foster's talk on using ISDN to carry high-quality audio information. He gave an overview of ISDN and showed how the digital telephone network could be extended to people's homes and businesses, providing either 2 or 30 lines, each of 64kbit/s data rate, in both directions. Problems arose in some cases when linking with US sites because much of the US network had bridges running at 56kbit/s, owing to the older nature of the US network. He showed how multiple ISDN lines could be synchronised using an inverse multiplexer (IMUX) in order to obtain higher overall rates, and explained that the number of lines required depended on the quality of sound desired. Outlining the most ►

common codecs in use today, Foster proceeded to give examples of uses for this technology, claiming that ISDN allowed costs to be saved on many occasions where it avoided the need for artists or other staff to fly long distances and stay in expensive hotels. Although the cost of the telephone calls was not low, it was still often much cheaper than the alternatives. He also showed how a wireless modem could be used to connect sites without ISDN lines to nearby sites which did have them, thereby avoiding the cost of installing temporary lines in live broadcast venues such as concert halls. Discussing incompatibilities between ISDN terminal equipment, he pointed out that it was vital to use the same equipment at both ends of the line, because the multiplexing methods used to split data between the lines varied, as did the audio coding processes.

Assessment

Chaired by Rhonda Wilson, the last session of the conference dealt with ways of assessing the performance of digital audio systems. The tests which might have been appropriate for older analogue equipment are unlikely to be very revealing of the quality of a digital system, especially in the case of low-bit-rate systems.

Julian Dunn of Prism Sound (UK) discussed and demonstrated the effects of the demon timing jitter, especially on convertors which did not properly reject the effects of jitter passed on from digital inputs. Using the *DScope* analyser he showed how a specially-built jitter signal generator could be used to superimpose a controlled amount of timing instability with a view to measuring its effects on the audio output of a subsequent D–A convertor. Dunn had developed a test signal called the 'J-test signal' designed to stimulate a worst case situation on a digital interface, in order that consistent comparisons could be made between devices. Bob Stuart of Meridian followed, describing ways in which auditory modelling could be used to predict the audibility of distortions in digital audio systems. He indicated that considerable further work was required to model aspects of the cognitive processes which determine how the brain interprets the information receives from the ears.

Presenting the last formal paper in the conference, Richard Cabot, currently the President of the AES, spoke on methods of assessing the performance of low-bit-rate codecs. Conventional audio tests did not give much useful information, he proposed, but beta software had been developed for the Audio Precision system which used an auditory model to give an indication of the seriousness of errors in low-bit-rate systems. By using an error display and a test signal especially designed to exercise the codec maximally, it was possible to see how noise and distortion generated by the low-bit-rate coding process compared with the masked threshold. Cabot described a number of tests that might be appropriate for different operational situations. He explained that such tests might give useful information about the effects of combining codecs in series, and might provide preliminary data on which to base listening tests.

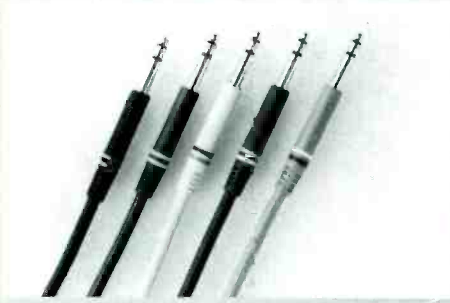
During the panel discussion which followed, a number of the speakers from the two days of the conference discussed questions raised by delegates and by each other. Michael Gerzon pointed out that masking models might not always be applicable in cases when distortions were highly correlated with the signal, because their audibility was not controlled entirely by masking. The use of dither was vital to the decorrelation of quantising errors, and it was not clear whether codec manufacturers always used dither correctly in requantising procedures. This point was generally agreed by the panel, although Louis Fielder was not certain that there was enough evidence to suggest that the benefits would be worth the work. In discussing the effects of jitter, Malcolm Hawksford reinforced a point from the floor that one of the key audible results of jitter on D–A conversion was often an effect upon the clarity and spaciousness of the stereo image. The correlation of timing instability between channels, or lack of it, was likely to have an important effect in this area. ■

Dr FRANCIS RUMSEY is the past Chairman of the British Section of AES, and a lecturer on Surrey University's Tonmeister degree course in Music and Sound Recording. He is the author of numerous conference and convention papers for AES, the Institute of Acoustics and the Royal TV Society, and six books on audio technology including *Digital Audio Operations* and *MIDI Systems and Control*

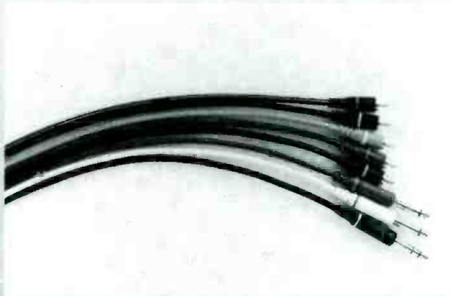
THE LONG-LIFE, ECO FRIENDLY CABLE

MOGAMI

Any cable failure often results in problems where the original cause is extremely difficult to detect. The reliability and quality of Mogami cable ensures those problems do not arise. Long-life, safety, efficiency and economy make Mogami the No. 1 choice for cable.

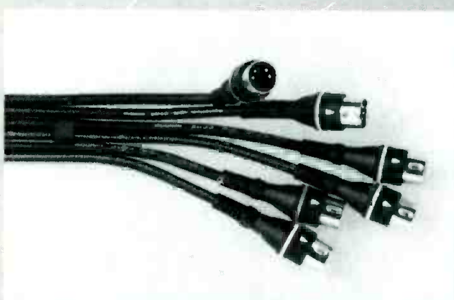


BANTAM PATCH CORDS

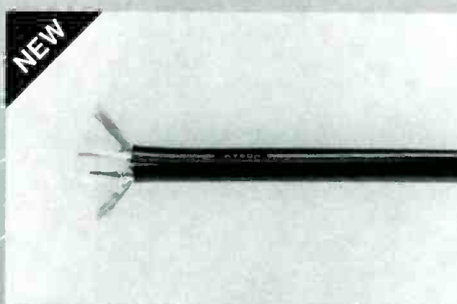


HIGH DEFINITION 75Ω AUDIO/VIDEO CABLES

LONGFRAME PATCH CORDS

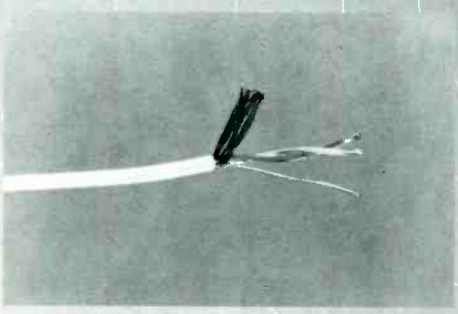


MIDI SYNCHRO CABLES



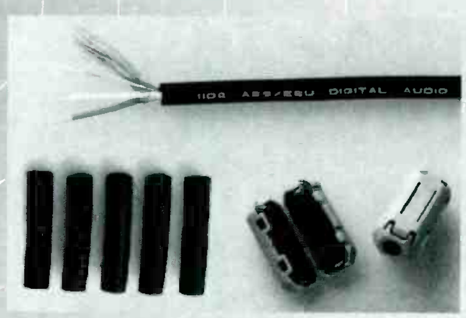
STEREO MIC. CABLE AWG #24 (0.226M²)

CL2 RATED SNAKE CABLES

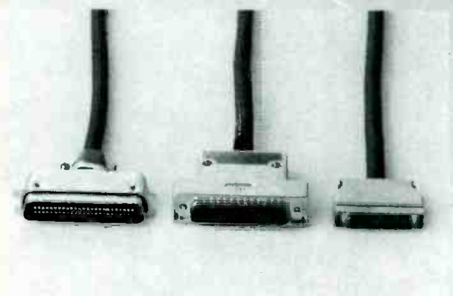


CONSOLE WIRING CABLES

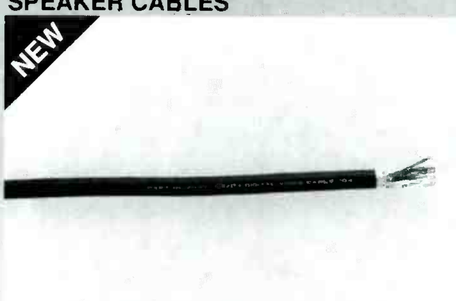
CL2 RATED SUPER-FLEXIBLE STUDIO SPEAKER CABLES



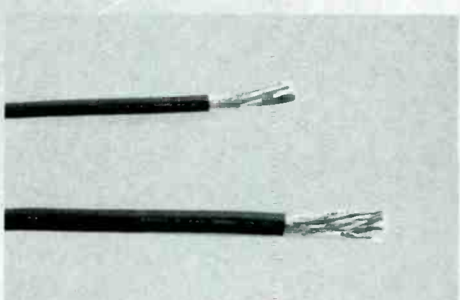
AES/EBU 110Ω DIGITAL AUDIO CABLE



SCSI -II SINGLE ENDED A CABLE



PARALLEL DIGITAL VIDEO CABLE

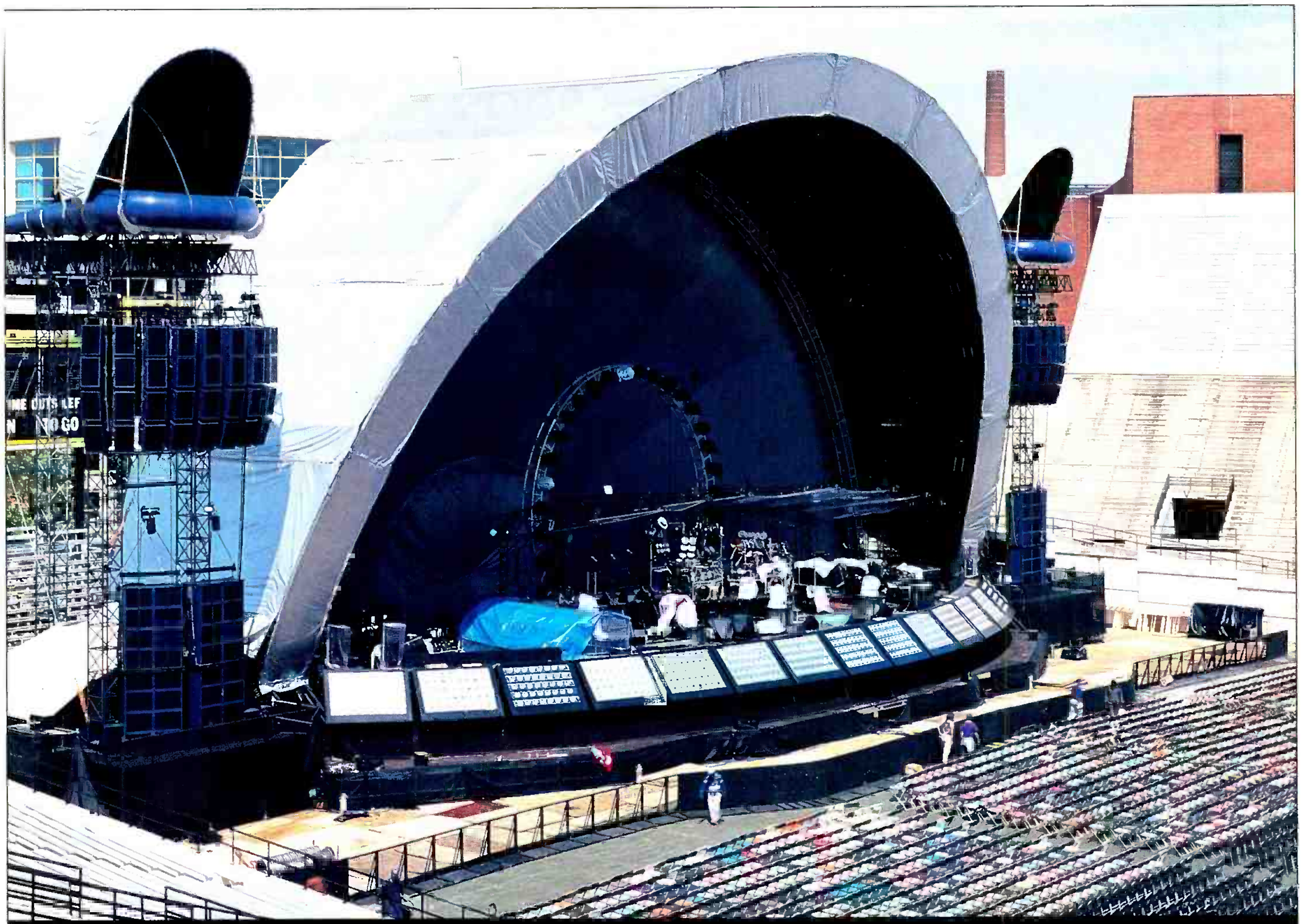


MULTICORE CABLES WITH AND WITHOUT OVERALL SHIELD

CALL OUR DISTRIBUTORS FOR A FREE CATALOGUE AND/OR MORE INFORMATION:

AUSTRALIA: *GREATER UNION VILLAGE TECHNOLOGY (02) 550 5488* • **AUSTRIA:** *DIETMAR KOLLER RECORDING EQUIPMENT (0222) 450 43 43* • **CANADA:** *CABLETEK ELECTRONICS LIMITED (604) 942-1001* • **CROATIA:** *AUDIO VIDEO CONSULTING (38) 41 624 622* • **DENMARK:** *D A DISTRIBUTION APS (31) 61 07 55* • **FINLAND:** *SOUND MEDIA OY (0) 629 669* • **FRANCE:** *D.M.2.J. SARL (1) 39 57 90 44* • **GERMANY:** *AUDIO-VERTRIEB PETER STRUVEN GMBH (04106) 6 99 99* • **HONG KONG:** *ODYSSEY ENGINEERING CO. LTD. 5-8988111* • **JAPAN:** *MIT INC (03) 3439-3755* • **KOREA:** *AVIX TRADING COMPANY (02) 565 3565* • **NEW ZEALAND:** *DENCO AUDIO LTD (3) 3790743* • **NORWAY:** *LYDROMMET A.S. 22 37 02 18* • **SINGAPORE:** *TEAM 108 TECHNICAL SERVICES PTD. LTD. (065) 748 9333* • **SOUTH AFRICA:** *EMINENTLY MORE SUITABLE S.A. (PTY) LTD. (011) 482 4470* • **SPAIN:** *MEDIA-SYS, S.L. (93) 426 65 00* • **SWEDEN:** *TAL & TON ELEKTRONIK AB (031) 80 36 20* • **SWITZERLAND:** *ZAP SONO (22) 340-05-70* • **TAIWAN:** *SEA POWER CO. LTD (02) 298 2688* • **THAILAND:** *VIDHYA SUVANICH (2) 321 4426* • **THE NETHERLANDS:** *MAARTENS SOUNT & VISION (0) 5130 29120* • **U.K.:** *STIRLING AUDIO SYSTEMS LTD. (071) 624-6000* • **USA:** *MARSHALL ELECTRONICS INC. (310) 390-6608* •

FLOYD IN FLORIDA



The Pink Floyd stage rig in Nashville showing the 'skirt'

In keeping with the band's profitable profile Pink Floyd's recent US tour took $83m. Kevin Hilton flew to Florida to catch their sophisticated stage show in Tampa

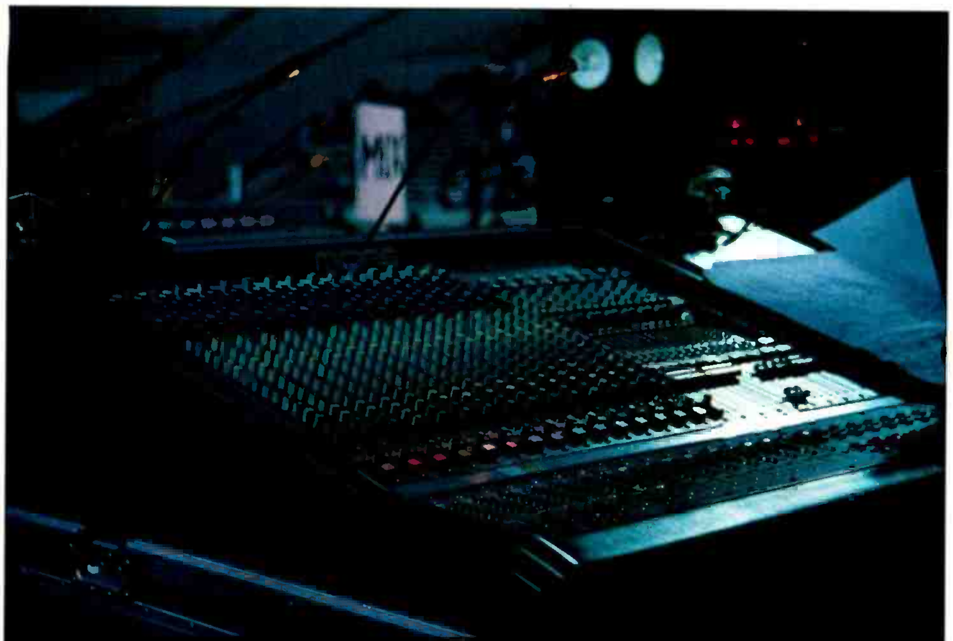
That it was a beautiful clear, warm evening in Tampa came as much to the relief of the Pink Floyd crew. The shows on the way to Florida, in Houston and Atlanta, had been drenched by the rain but for this one the heat of the day was still lingering, putting the 70,000 crowd, covering the age range 10 to 45, in a relaxed and good-natured mood.

It had been seven years since Pink Floyd's last studio album and supporting tour but they are back with *The Division Bell*, which is number one in 16 countries around the world, and a tour that practically sold out when it was first announced. The band—still featuring David Gilmour, Nick Mason and Rick Wright—started the European leg of the tour in Portugal at the end of July and are due to play ten dates at Earl's Court in London during October.

With the new tour, the band wanted to get back to the scale of production that they became famous for with *The Wall*. While it does not have the same theatrics, it certainly delivers a huge production, meshing lights, lasers, pyrotechnics, film projections and a massive sound system.

'They kinda make their own stadium, man,' an awe-struck fan said to me at Tampa Airport. It sounds extravagant but that's the intention, as Monitor Engineer Seth Goldman confirms: 'We're very much in our own environment.' The foundation of this is the massive band stage, with its in-built surrounding 'skirt'; there is only one of these and it is broken down and rigged for each show. Rising above this is an arch which supports the lights and projection screen; three of these alternated during the US leg, while two will leap-frog each other in Europe.

On either side of the stage are two towers, which serve the dual purpose of containing the main front-of-house sound system and being home to the trademark inflatable pigs, which pop up during 'One of these Days' at the end of the first half of the show. Because it's Floyd, there can't be just ▶



The customised Midas *XL3* console performs quadraphonic duties

straightforward left and right stereo; the band has been using 'live' quadraphonic since the mid 1970s, and this tour features three quad stations, plus a delay stack directly behind the mixing tower.

An emphasis on sound is something that Pink Floyd have always shown, down to forming their own sound rental company during the early 1970s. Although the band no longer owns Britannia Row Productions, it still provides equipment for their tours, which offer the perfect outlet for its specialist systems. Bands having been playing huge venues, both outdoor and indoor, since the late 1970s, it is only during the last ten years that dedicated long-throw loudspeaker systems have started to appear.

Brit Row holds the world's largest stock of one of the leading models, the Turbosound *Flashlight*, which dominates the current Floyd tour. The front-of-house system is made up of 32 *Flashlight* highs and 32 low-end units per side, supported by three units left and right of the wider dispersion *Floodlight* for near audience in-fill.

This reinforcement is further reinforced by nine of the new underhang cabinet, which, like *Floodlight*, was designed by research house Funktion One (run by original Turbosound founders Tony Andrews and John Newsham) and licensed to Turbosound for manufacture. These are arranged in three lots of three along the 'skirt' of the stage.

One of the design criteria of *Flashlight* was to deliver clear, well-defined sound without the need for delay towers. This the system does, but a delay stack is still need to fill in the area directly behind the mixing tower, a 4-level construction that houses the sound mixers, a hospitality suite, lights-lasers and film projection. Right at the top of this lurks a mirrored rotating ball, which makes a startling appearance during 'Comfortably Numb'.

Ironically, the audience probably noticed the three quad stations more than the visually commanding left and right stacks. Each of these contains eight *Flashlight* cabinets and four *Floodlight* boxes. Early on in the preproduction of the tour, it was intended to have the fourth quad cluster suspended over the stage from the semicircular truss that 'contains' the show. However, the height at which this is set made it impossible, so the fourth quad signal comes from the middle of the left and right stacks, fed from a subgroup on the main desk.

Up front

The front-of-house mixing position is oversubscribed with consoles, featuring two Yamaha *PM4000*s, which, over 80-channels, handle the instruments and vocals; a *PM3000* for all effects returns and tape playback, still a major part of the Floyd's shows after all this time; and, although the smallest of all the desks, perhaps the most innovative and important, a custom-built quadraphonic Midas *XL3*.

This one-off was designed by Brit Row, with the blueprints being faxed to the US so that the man who operates it knew what was going on. Dave Lohr ran the quad and effects on the last Floyd tour, which makes him, along with the band's Mixer Andy Jackson, a front-of-house veteran. The other two Engineers, Colin Norfield, who shares front-of-house duties with Jackson, and Engineering Manager Paddi Addison, are making their first appearances with Pink Floyd.

The *XL3* is a dedicated 16-channel quad board, which features two manually controlled joysticks. 'Because we need four outputs and joysticks to pan, we needed a board that was designed to do quad, rather than making do with a conventional stereo desk,' explains Dave Lohr. 'The quad is used to heighten what is going on on stage—it adds ambience to the performance.'

The effects and various voices come from two sources: either one of the two Otari 8-track tape machines nestling at the back of the mixing position, or from digital samplers on stage, which are triggered by the musicians. This last move would appear to be a fairly recent one and is used for much of the newer material, including the samples of Professor Stephen Hawking on 'Keep Talking'. These, along with four channels of reverb and four channels of autopan, are fed to the joysticks and then panned around.

Any of the sources coming through either of the two *PM4000*s or the *PM3000* can be routed through the *XL3*, and either be panned or have ambience added to them. Likewise, anything from the *XL3* can be fed back to any of the other boards; in this way the fourth quad feed is sent to the main *PM4000* and then to the front-of-house stack. 'You've got to mix the quad for the distances,' says Lohr. 'However, you don't want to kill the people right at the back but you want to make sure that the people below get it as well. The quad wasn't really designed for stadium work, though, and I told Dave [Gilmour] this. It works great in arenas.'

Surprisingly, this whole high-tech setup was designed around analogue open-reel tape machines, but, as Lohr explains, this is what Pink Floyd know and are used to. 'They wanted to use the two-tracks again and they're a tried and trusted method. With SR it's close to what digital would sound like. Digital systems have transportation problems —they're not really practical for touring work yet, although the band is generating some of the effects on stage using samplers.'

Despite this pivotal role, the *XL3* is not the main console in the front-of-house setup: it was designed so that any board could feed into any other. 'The whole thing is linked up as one,' explains Colin Norfield. 'The chain comes through the instrument and vocal desks [the *PM4000*s], from which you can either send to the effects console [the *PM3000*] or to the quad panner. The Yamahas have their own buzz link for the VCAs and mutes, while the rest goes via XLRs to multiconnects. All three of these desks then link into the quad. Anything can be sent to any of them.'

The instruments and vocals build up across the two *PM4000*s from left to right, starting with the drums and percussion (which Norfield calls his 'baby') and progressing through the other instruments. The right-hand *PM4000* primarily handles the vocals and keyboards, with the output from both being sent to the *PM3000* to be 'treated' and then, finally, routed to the quad board to be distributed.

Cause and effect

Pink Floyd's sound has always been very effects orientated, both in terms of voices and noises suddenly popping up out of nowhere, and weird and wonderful ways of processing the vocals and instruments. Despite the sophistication of much of the system, most of the tape effects are played in manually. Only the introduction of 'Money'—cash registers, the clanking of coins—is hooked up to time code. This section is driven by the tape, the band taking its cue from a SMPTE click track that also runs the projection. 'In the past, the band have used a lot of time code,' says Lohr, 'but they felt locked into it, so they use less now.' ▶

The Power Of Digital Editing

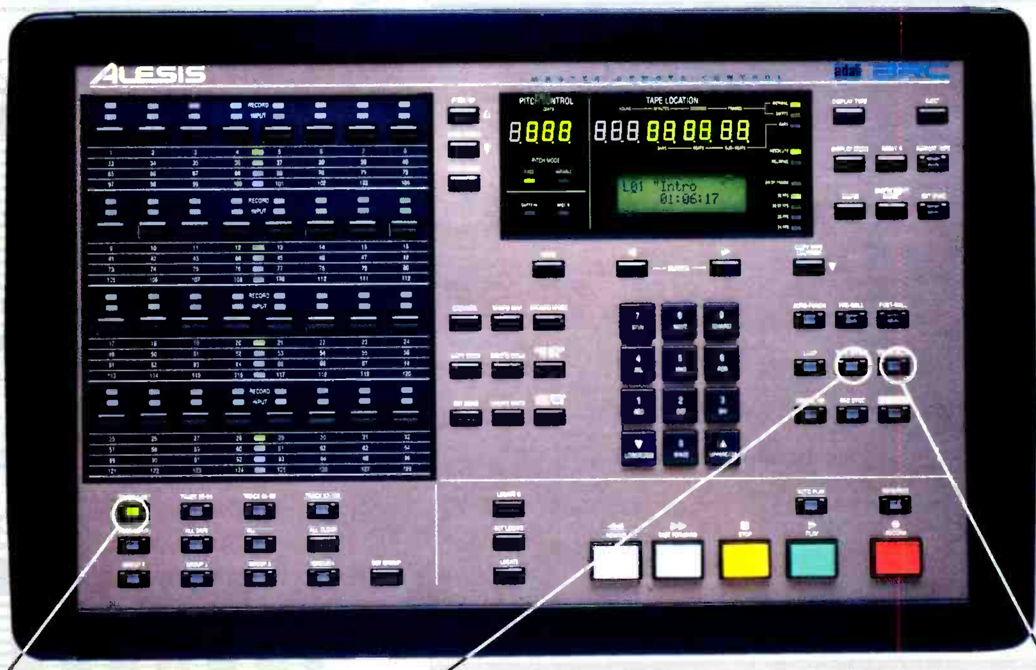
Part One Of A Series

adat® BRC

MASTER REMOTE CONTROL

The Alesis **BRC™ Master Remote Control** puts the power of digital copy-and-paste editing in the hands of all ADAT® users. Take that great backing vocal on the first verse and fly it into verses two and three. Or, create the ultimate montage sax solo by copying small sections of many different takes with single-sample accuracy. The BRC gives you the advantages of a hard disc recording system with the security of a tangible tape-based storage medium.

But advanced digital editing is only one part of the BRC. You can use the BRC to control 128 ADAT tracks from a single location. It also offers SMPTE, video and MIDI synchronization, stores session notes to the data header of ADAT tapes, and remembers 460 locate points. And much more. Can't believe it? Go to your Authorized Alesis ADAT Dealer and try it yourself. The power of digital editing is only the beginning.



Select source
with trk buttons

In the ADAT Digital Multitrack Recorder, digital audio not only sounds better, it becomes an essential creative tool, thanks to the patented Alesis technology that connects all ADATs in a system. The ADAT MultiChannel Optical Digital Interface™ on each ADAT transmits and receives eight channels of digital audio simultaneously using high-quality fiber-optic cables. The BRC can direct the "traffic" on this eight-channel interconnect, offering advantages not available on any other tape-based recording system.

Offset Tape 02
from L04 to L08

L04 "Chorus 1"
+ 16-01/00

L08 "Chorus 2"
+ 48-01/00

Because the BRC can think in musical terms, you can offset ADATs by bars and beats. Set ADAT #2 to be at the second chorus while ADAT #1 is at the first chorus, using Auto Punch, Tape Offset and DIGITAL I/O to "fly in" sections to eliminate tedious overdub time. It's easier and more accurate than using a razor blade or sampler, and there are no computer screens to distract you.

Track delay:
170.0 ms

And even if you don't want to perform copy / paste editing, you can manipulate any track independently. Each track can have up to 170 milliseconds of digital delay to compensate for microphone placements, musicians ahead of the beat, or to create master-quality digital delay effects. With the BRC's Track Delay, you can even create an entirely new rhythmic feel for your recording.

AUSTRIA: ATEC: Audio & Videogerate +43.223474004
DENMARK: Kinovox A.S. +45.42187617
FINLAND: Oy Landola Marketing +358.677230407
FRANCE: Importation Musicale Lyonnaise +33.149381700
GERMANY: Soundware Audio Team GmbH +49.607489150
ITALY: M. Casale Bauer s.p.a. +39.51766648
NETHERLANDS: Electric Sound B.V. +31.365366555
NORWAY: Norsk Musikk Instrument Co. A.S. +47.22235680
PORTUGAL: Americo Nogueira L.D.A. +351.22004616
SPAIN: LETUSA: Distributora Musical +34.16410812
SWEDEN: Muskantor & Co. A.B. +46.31878080
SWITZERLAND: SDS Music Factory A.G. +41.14342270
UNITED KINGDOM: Sound Technology p.l.c. +44.462480000
UNITED STATES: via e-mail: ALECORP@Alesis1.usa.com

To be continued...

ALESIS
STUDIO ELECTRONICS

Processing effects appear on just about everything going through the desks, although the drums and percussion get most of this. There are 24 Drawmer gates across all the drum feeds, AMS reverberation on the snares, Roland *SRV2000* reverb over the toms and percussion, and *DPR 402* compression on the bass drums.

The vocals are not left out, with a mixture of tc electronic *2290*s and the new Roland *SDE330* being used for effects on the lead and backing voices. The delay returns from these units are fed to the *PM3000* and then into the quad board. A quad-panner, the *SPX900*, is used for a specific effect during 'One of these Days', where the signal is panned between front and back and left and right. Two Lexicon *PCM70*s are used for quadraphonic reverb. Lead Guitarist and Vocalist David Gilmour has a BSS *FCS3960* specifically for his vocals.

The delay and quad stations are equalised by a bank of BSS *Varicurve* remote units, controlled through a remote interface. 'We can control these either from the front-of-house position or by a radio unit wandering about the field when we're setting up,' says Paddi Addison.

Power for the front-of-house system comes from 14 racks of amplification, each one containing two BSS *780*s for the low and low-mid cabinets, and two *760*s on the high-mid and high units. The delay stack and three quad stations each have two racks to drive their loudspeakers.

The monitor mix is equally complex and important, but only has one person in charge of it. Seth Goldman has worked with the Floyd for over 20 years, and in the sometimes long gaps between their tours has also monitored for David Bowie and Mariah Carey.

Two Midas *XL3*s by the side of the stage take 80-inputs and give 22 mixes. These are distributed to the band through a mixture of floor wedges (the Turbosound *1x15*) and the Garwood in-ear monitoring system. 'It's a sophisticated system, but it's still very straightforward, although a little on the large side,' explains Goldman. 'The main desk is mostly used for the floor wedges, which everyone has. The in-ear moulds complement these and are used mostly by the two drummers and the sax player, who wear them throughout the show. Others in the band may want to use them if they're getting a lot of slap-back from the arena. I use them all the time, which allows me to walk around the stage and keep tabs on what is going on. They give a lot of freedom.'

Flexibility appears to have been built into this whole system, giving the players communication with Goldman and, in some cases, control over their own feeds. Brit Row have designed a buffering foot-pedal that takes a microphone out of the main mix and turns it into a talkback unit. David Gilmour has the luxury of a VCA crossover which allows him to adjust the level of his vocals in his own wedge by means of an external button on the mic stand.

'This gives him total level control over what he wants to do,' says Goldman. 'I set things up and tweak it, but it's usually only very minor up and down movements, especially when the vocal is a little more laid back.' This philosophy extends to the whole setup, with the BSS *960* graphic equalisation, *Varicurve 9260*, BSS *402* compression, *502* gates and Yamaha *SPX990* effects units being set up every day and only adjusted where necessary.

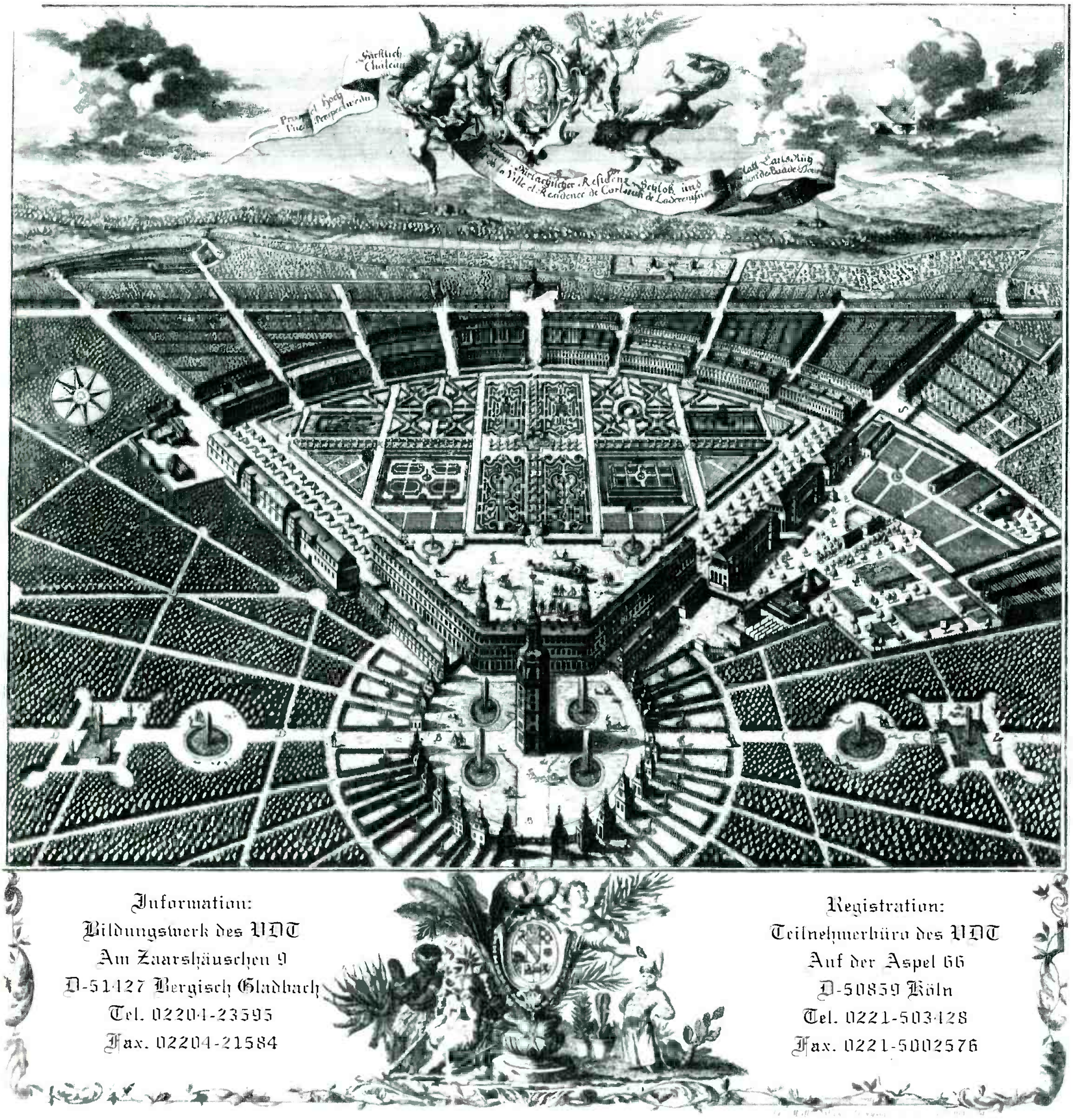
'The *Varicurves* are pretty much set, while the EQ on the floor monitors changes a bit each day, but not much because everything is very much in our own environment,' Goldman says. 'I fine tune all the monitors everyday but it's still a complex mix, with lots of cues coming in.'

Even with all this going on, it all seemed to work well, producing a clear and fully extended sound, the track 'Sorrow' in particular reaching the parts you'd rather it didn't. The Floyd themselves were a little on the loose side—not that this bothered the crowd, who flooded out into the Tampa night proclaiming it the best gig. Ever. ■

18. tonmeistertagung

INTERNATIONAL CONVENTION ON SOUND DESIGN
15.–18. November 1994 – Stadthalle Karlsruhe

VERBAND DEUTSCHER TONMEISTER VDT e.V.



The Castle of Karlsruhe. Engraving by J. Thran 1739

LIVE CONSOLE ROUNDUP

The options facing users of live mixing consoles reflect changing requirements and advancing technology.
Patrick Stapley presents a comprehensive roundup of current consoles

Allen & Heath

A&H currently manufacture two low-cost live consoles, the ***GL2*** and ***GL3***, but will also be launching the ***GL4***, mid-priced, touring console in September.

The ***GL3*** functions as either a FOH or monitor console, or both simultaneously. It comes in 16 and 24-input sizes and can be expanded in groups of 8 channels. The 4-bus console has 4-band EQ, 6 aux sends, and 4 effects returns. A recessed switch changes the configuration of the console between FOH and monitor. In Monitor mode, the auxiliary mix is fed through the main signal path in place of the group mix, providing 6 stage-monitor sends; also the mono output (normally used to drive a centre fill speaker in FOH mode) can be used to feed the engineers wedge monitor. In 'Multimode' operation the console acts as a stereo and mono FOH desk with 4 fully-featured stage monitor sends and 2 effects sends.

The ***GL2*** is a nonmodular, single size, rackmounting budget mixer which like the ***GL3*** can function both as an FOH console (18:4:2:1), or a monitor desk (14:6). The 4-bus console has 4-band EQ and 6 aux sends, and although basically a scaled down version of the ***GL3***, includes unique features such as dual input stereo modules, and a subgroup facility allowing the 4 effects returns to source the group outputs. This provides separate control of the groups to the L-R *mix*, and apart from offering a true submixing facility is useful for 4-track recording. ***GL2*** consoles can be linked without losing inputs via Allen & Heath's ***SYS-LINK*** system.

Amek

Amek have been making sound reinforcement consoles since 1974 when they first supplied hippy band Gong with a quadraphonic FOH mixer. The current range consists of two ***Langley*** desks and four TAC mixers.

Recall by Langley, which heads the range, was introduced earlier this year becoming the first live console to offer a recall system. The console can be used either in FOH or Monitor modes and is supplied in 40 or 56-channel frames. Facilities include 12 aux buses, 4-band EQ with variable high and low-pass filters, 8 groups, 8 VCA groups, 10 x 8 matrix, and 8 mono and 2 stereo returns. The console also features Amek's *Showtime* snapshot automation controlling VCA levels, various switches and triggering external MIDI events. Another standard feature is Amek's *Virtual Dynamics* which provides each channel with one of nine software-controlled dynamic processors. The console is supplied with main and backup power supplies, and desks may be linked so that control is governed by one master computer.


Amek *Recall by Langley*

Released this month is a scaled-down version of ***Recall by Langley***, that reduces the price by nearly half. ***501 by Langley*** comes in 24 or 40-channel frames offers 8 aux buses, 4-band EQ with HPF, 8 groups, 4 VCA groups, 4 mute groups, 10 x 4 matrix (also directly assignable from input channels), and 2 returns. *Showtime* automation and recall comes as standard although *Virtual Dynamics* is an option.

The two largest TAC consoles are the mid-priced ***SR6000*** (FOH) and ***SR6500*** (monitor), both available in 24, 32 and 40-input sizes. The ***SR6000*** is a conventional manual desk equipped with 8 aux sends, swept 4-band EQ with parametric mids and variable HPF, 8 groups, 8 VCA groups (with muting), 10 x 8 matrix, and complete input-output metering. A unique feature is the 'Split Auxiliary' function which allows input module aux sends to be split into two groups each feeding a separate master, effectively doubling the number of sends to 16.

The ***SR6500*** provides 18 outputs plus a wedge monitor output, includes 8 VCA-Mute groups, 4-band swept EQ with variable HPF on all inputs and outputs, and a comprehensive communications and talkback system. Both the ***SR6500*** and ***SR6000*** include bus linking for connecting additional consoles.

The TAC ***Scorpion*** was introduced in 1984 and was updated five years ago as ***Scorpion II***. This 'workhorse' desk is available in FOH or monitor versions offering 40+ inputs. There are mono and stereo inputs, and many configurations available including different subgroup and output formats.

TAC ***Bullet*** and ***B2*** are at the bottom of the Amek range offering configurations from 10:2 through to 28:8:2 over three chassis sizes. The consoles feature mono inputs with 4-band EQ and 6 aux sends; stereo inputs with RIAA, 3-band EQ and 6 aux sends; and dual group modules with a stereo return.

Cadac

Cadac are the oldest manufacturer of consoles listed in this survey, having being in existence since the late-1960s. However, it was in 1983 that Cadac built their first theatre console, since then the company have specialised in live sound gaining an enviable reputation particularly for theatre consoles.

The current range comprises the ***J Type*** and the newly introduced ***Concert***, both are highly specified designs.

The ***J Type*** follows on from the successful ***E Type*** offering increased flexibility as well as a reduction in price. Available in standard frame sizes of 38, 46, 54, and 62 modules, the console can also be expanded to four frames providing a maximum of 120 channels—any module can be used in any position in any frame, and modules can be removed-inserted without having to power down. Input channels have 16 group sends, 12 aux sends (10 mono, 1 stereo), 3-band separately switchable parametric EQ with variable HPF, can be controlled by any one of 15 VCA master faders, and have pre-post selectable direct outputs. There is a 17 x 32 matrix with switchable insertion, ▶

Crest *Century LM* monitor

4 group-routeable stereo returns, and an optional quad panning module. Other features include high resolution integrated LED metering, comprehensive communications, oscillator/pink-noise generator, and dual power supply. The console also offers Cadac's proprietary automation system which provides snapshot control of faders, mutes, routeing, MIDI and Events; it can also run dynamic sequences if optional motorised faders are fitted.

The more expensive *Concert* is manufactured to order, and up to four desk frames may be bussed together. Input modules have two inputs which can be active at the same time, each sharing 16 aux sends (all individually pre-post, on-off selectable), 3-band, separately switchable, parametric EQ with variable HPF, routeing to 12 groups and 12 matrix groups, and connection to a maximum of 15 VCA groups. There are separate line-level inputs and switchable insert points for all groups and matrixes. Console switching can be accessed from a central assignment module, or programmed off-line using *Concert*'s automation system. Console parameters are stored in computer memory allowing instant reset of switches, and manual reset of knobs with the aid of null indicators. The computer also controls MIDI. Events and Cadac motorised faders which are supplied as standard. As with the *J Type*, *Concert* may be powered by two independent power supplies, and can be supplied with a backup computer.

Crest Audio

American-based Crest Audio manufacture two ranges of live console: the *Century Series* and the *Gamble Series*—both of which offer FOH and Monitor versions.

The *Century Series* is made up of three FOH consoles (*GT*, *TC* and *SP*) and three monitor consoles (*LM12*, *LM8+4* and *LM20*). The more expensive and more esoteric *Gamble* Series (acquired by Crest in 1989) consists of a FOH and Monitor console.

All *Century* FOH desks share the same type of frame allowing modules to be interchanged between the three desks and positioned to suit the user. Each console can be supplied with up to 52 inputs, and in 4 or 8-bus formats. The fully-featured *GT* has been primarily designed for road use, but is also suited to fixed installations. *GT* features include a high-quality 4-band swept equaliser with fixed HPF, 8 auxiliary sends, a 'clean' mono bus, an 11 x 2 matrix (expandable with optional matrix modules), and a mute group system for channels and effects returns. The lesser featured *TC*, has been designed for installation work, but again can crossover and be used for touring. The *SP* is a simplified version of the *TC* and is pitched at installations requiring an easy to use budget desk.

The *LM12* and *LM20* Monitor consoles provide up to 12 and 20 discrete mono mixes respectively from up to 52 inputs. The *LM8+4* offers 8 stereo and 4 mono mixes making it suitable for in-ear monitoring. All *LM* consoles include 4-band EQ on each input and 3-band EQ on each mix group both with HPF, have integral mic splitters per channel, and offer a 2-way monitor submix. They also share the same build and sonic quality found in the *Century* FOH desks.

The two *Gamble* consoles are fully-featured, compact, quality conscious designs aimed at the high-end market. They are very often customised to suit the user, but the standard *EX56* is a 56-channel board offering 8 stereo group buses and 8 stereo matrix groups, thus providing a fully controllable 16 into 16 configuration. Facilities include 4-band parametric EQ and variable HPF, 10 aux sends (8 mono, 1 stereo), 8 mute groups, a built-in meter bridge (20-segment bargraphs for all I-Os) and an integral Mogami-wired ADC patchbay. The monitor desk provides 48-input channels with 16 mixes that feed a 16 x 16 matrix, it also includes a stereo bus. All 16 outputs are equipped with 5-band parametric EQ.

A peculiarity of *Gamble* consoles is the back-to-front operation of the pots—gain is attenuated by clockwise movement and boosted by anticlockwise movement. The channel layout is also unusual in that the input controls are all positioned at the base of the strip.

D&R

Dutch manufacturer D&R make two SR consoles: *Axion* and *Vision*.

Axiom is a mid-priced, fully-featured, FOH console that comes in two standard sizes (32 and 40-input) but can be optionally supplied in larger frames. Mono and stereo input modules are available each offering 12 aux sends with global pre-post switching per send sand individual mutes, 4-band EQ with a variable HPF, switchable insert point, and routeing to 8 groups, LR bus, and a mono bus. There are 8 VCA groups with grand master control, and a programmable mute system that allows up to 64 mute scenes to be stored and recalled either manually or against MIDI time code. Also included is a 12 x 8 matrix, an elaborate communications system with call lamps and electronics to power and feed a beltpack comms system, and 25-segment metering. Consoles may be 'bus' linked without loss of channels.

Vision is a general purpose SR desk available in four frame sizes ranging from the 12-channel rackmount to a full 44-input desk. The console offers both mono and stereo input modules, and can be specified either in Standard or Deluxe configurations. The Standard version is a 4-group desk with 4 aux buses, and 3-band EQ, while the Deluxe is an 8-group console with, 8 auxes and 4-band EQ with a fixed HPF, input metering and talkback facilities. Additional features include an 11 x 8 matrix (Deluxe version), optional MIDI-controlled mute system, and effects returns which can double as tape returns for recording.

DDA

DDA currently have three sound reinforcement consoles—the competitively priced *Forum* that replaced the *Arena* console, the *Interface* (see Dynacord) and the latest *Q Series* console, the mid-priced *Q2*.

Forum FOH features include 8 group buses, 6 auxiliary buses, LED metering on every input, 4-band EQ with swept mids and fixed HPF, direct outputs on each channel, and delayed switch-on for main outputs to protect monitors from power-up surge. The console is available in 24, 32, and 40-input frame sizes, and can expanded by adding (one or more) 8-channel extender frames.

There are two *Forum FOH* models, *Forum Matrix* and *Forum PA*, which differ only in the type of output modules fitted. *Matrix*, as the name suggests, includes a 10 x 8 matrix and is suited to theatres and FOH applications, while the *PA* console has 16 auxiliary returns in place of the matrix and is more of a general purpose desk. Both consoles can be fitted with an alternative Mute input module which includes 8 mute groups. Additionally, there is a Stereo module with M-S decoding, a Digital module providing an SPDIF input, and a double width multi input module that submixes 6 mic inputs.

Forum Monitor is available with 22 to 46 inputs, with 12 monitor buses and a stereo mix for side fills. It also includes a matrixing system, ▶

D&R *Axion*

Because 8•Bus consoles have 20dB of gain above Unity (and mix amp architecture that resists overload), **Trim** is a "set-and-forget" control.

Flip works with the MIX-B section to double inputs during mixdown.

6 mono **Aux Sends** from 4 controls. All are Pre/Post Fader-Mute switchable. Aux 3&4/5&6 can tap their signal from the MIX-B/Monitor section.

4-band equalisation. True parametric **Hi-Mid** with 1/12 to ultra-wide 3-octave bandwidth (previously found only on far more expensive desks), comprehensive 500Hz to 18kHz sweep (15dB boost/cut). Sweepable 45Hz-3kHz **Lo Mid**, shelving **Hi** EQ at 12kHz, shelving **Lo EQ** at 80Hz & 18dB/oct. **Lo Cut filter** @75Hz.

–20dB activity LED's "stacatto" effect is so distinct that you can tell vocals from instruments.

+20dB **OL LED.**

Triple bussing! All 8•Bus consoles have **24 Tape Outputs** (3 sets per bus) so you can "send" an output to your choice of tape tracks without re-patching. For example, Sub 1's output appears at Outputs #1, #9 and #17. All outputs are **switchable** between "pro" +4dBu & "semi-pro" –10dBV in banks of 8.

MACKIE RE-DEFINES THE 8•BUS CONSOLE. MORE FEATURES. MORE HEADROOM. LESS NOISE.

If you've been searching for affordable recording consoles that truly do justice to the new generation of digital recording media, your search is over. A complete mixing system.

Mackie's 24•8 and 32•8 are expandable in groups of 24 channels. And all 8•Bus models are level and mute MIDI automation ready.

Sealed rotary controls resist contamination.

Optional **MB•32 meter bridge.**

Mackie's famous **discrete mic preamps** with –129.5dBm E.I.N. & awesome headroom.

Elaborate monitoring including **two separate headphone sections** with multiple sourcing & solo, individual **Studio & Ctl Rm** level controls, **Talk back** with built-in **mic, MIX-B & Solo** sections.

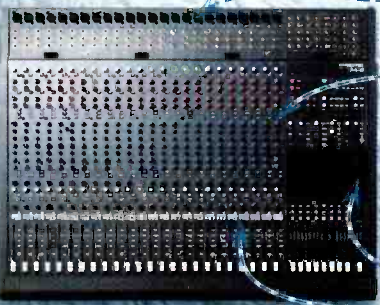
Optional **stands** available for all models & 24•E Expander.

Constant loudness pan controls' active buffered circuitry & special pot design provide up to 74% more attenuation than comparably-priced consoles.

MIX-B bus is a "channel strip within a channel strip," with its own **Level, Pan, Source** and **EQ** options. During mixdown, Mix-B **doubles your inputs** (for example, 64 on our 32x8x2 **32•8** shown above).

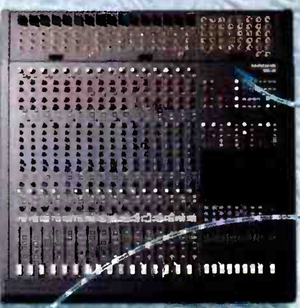
All channel inputs include **mic preamp** with +48VDC phantom power, bal./unbal. **line input, mic/line** switch, **direct out** & **channel insert.**

Mackie exclusive, **accurate-taper 100mm faders** with extra resistive elements closely approximate the logarithmic taper of mega-expensive consoles. Our faders let you do precise fade-outs all the way down to silence (versus conventional D-taper faders that give up at about –20dB). Compare with your own ears to appreciate our custom design.



Mackie **24•8** (24x8x2) is expandable in groups of 24 channels (complete with tape returns) via our **24•E Expander.**

In-place stereo solo on all three console models lets you solo any channel(s) to the main L/R LED meters at the touch of a button. All solo assignments in **true stereo perspective** even if you have a source panned hard right & an effect panned hard left.



Mackie **16•8** (16x8x2) is perfect for video post and scoring applications.

Tape returns are **switchable** between "pro" +4dBu & "semi-pro" –10dBV operating levels.

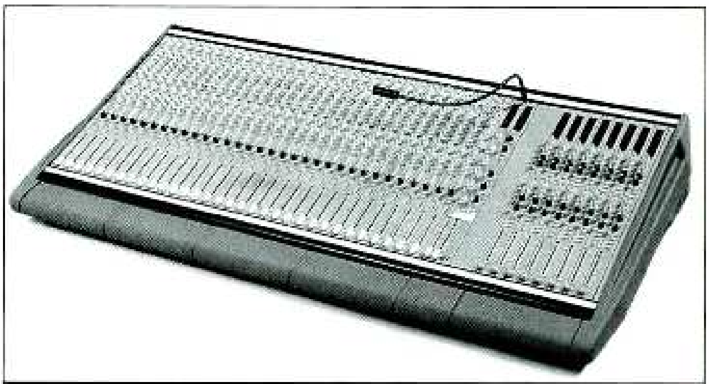
Exceptional specifications including 0.0014% THD (1kHz @+4dBu, main outputs), –95dBu output noise (master fader up), 10Hz-120kHz (+0dB/–3dB) frequency response, –129.5dBm mic E.I.N. (20Hz-20kHz, 150Ω), –83dBu mic C.M.R.R. and 91dBu crosstalk (line in to adjacent ch.).

MACKIE.

Mackie Designs Inc. • Woodinville • WA • USA • Represented internationally by MMS,INC. P.O. Box 1071 • Neptune • NJ • 07754 • USA • (908) 988-7800 • FAX (908) 988-9357

8 auxiliary returns, 4 submix inputs, and a dedicated fader for the engineer's monitor wedge.

The *Q2 FOH* is equally well suited to fixed installations or touring, and comes in four sizes ranging from 24 to 48 inputs. In standard configuration, the desk is fitted with 8 mono groups, however it can alternatively be specified with 8 stereo groups, 4 left-centre-right groups, or 16 mono groups (paired routeing). The console is equipped with a 15 x 8 matrix—if stereo group modules are fitted, each matrix output will be in stereo.



DDA *Forum PA*

Other features include 4-band swept EQ and swept HPF, 8 aux sends, comprehensive solo-cue system, 8 mute groups with safety lock and built-in meter bridge (12-segment bargraphs).

The *Q2 Monitor* console has a number of features found in the *FOH* desk, and provides six stereo sends and eight mono sends from each input. Two different output modules are included giving a reassign facility for two of the mono groups—this feature is also used on the *Forum* monitor console and offers a quick way of generating mixes.

Dynacord

Dynacord offer a number of budget consoles for live use. The top-of-the-range *Interface* (designed by DDA as part of the Mark IV cooperative) is a general purpose FOH desk supplied in five frame sizes ranging from 8 to 40 channels. The 4-group desk has five frame sizes ranging from 8 to 40 channels. The 4-group desk has five types of input module which may be specified to suit the application. These consist of mono and stereo mic-line inputs, a dual line input, a digital input (providing 2 digital and 3 analogue stereo inputs), and a 6:1 mic input. All inputs have 4 aux sends and 4-band EQ apart from the dual input which has two 3-band sections. Either standard group modules can be fitted which each include two stereo returns, or alternatively matrix modules can be fitted providing a 4 x 4 matrix plus four effects returns.

Other Dynacord desks include the *MCX 1900* 20-input rackmount mixer designed for stage applications and as a sub-premixer; the *MXC 2100* available with 20 or 28 inputs, and the *PSX 50* processor controlled range which incorporate built-in 16-bit reverb and effects, plus power amplifiers making them suitable for small PA applications.

Midas *XL3*

Midas

Midas launched their current *XL3* console in 1990. The desk is a high-quality, highly specified, upper-to-mid-priced console suitable for both FOH and monitor applications. It is supplied in 24, 32, and 40-input frames and can be expanded with a 16-input extender console.

There are seven different modules available for the console. Mono input and stereo inputs each have 16 rotary-controlled mix output-sends (each pre-post selectable), which can be used as either output mixes, subgroups or aux sends. They also have 8 mute groups, 8 VCA groups (featuring two grand masters), 4-band parametric EQ with swept HPF (mono module only) and switchable inset points. The stereo module includes MS circuitry and arranges the 16 mix outputs in pairs to facilitate proper stereo sends.

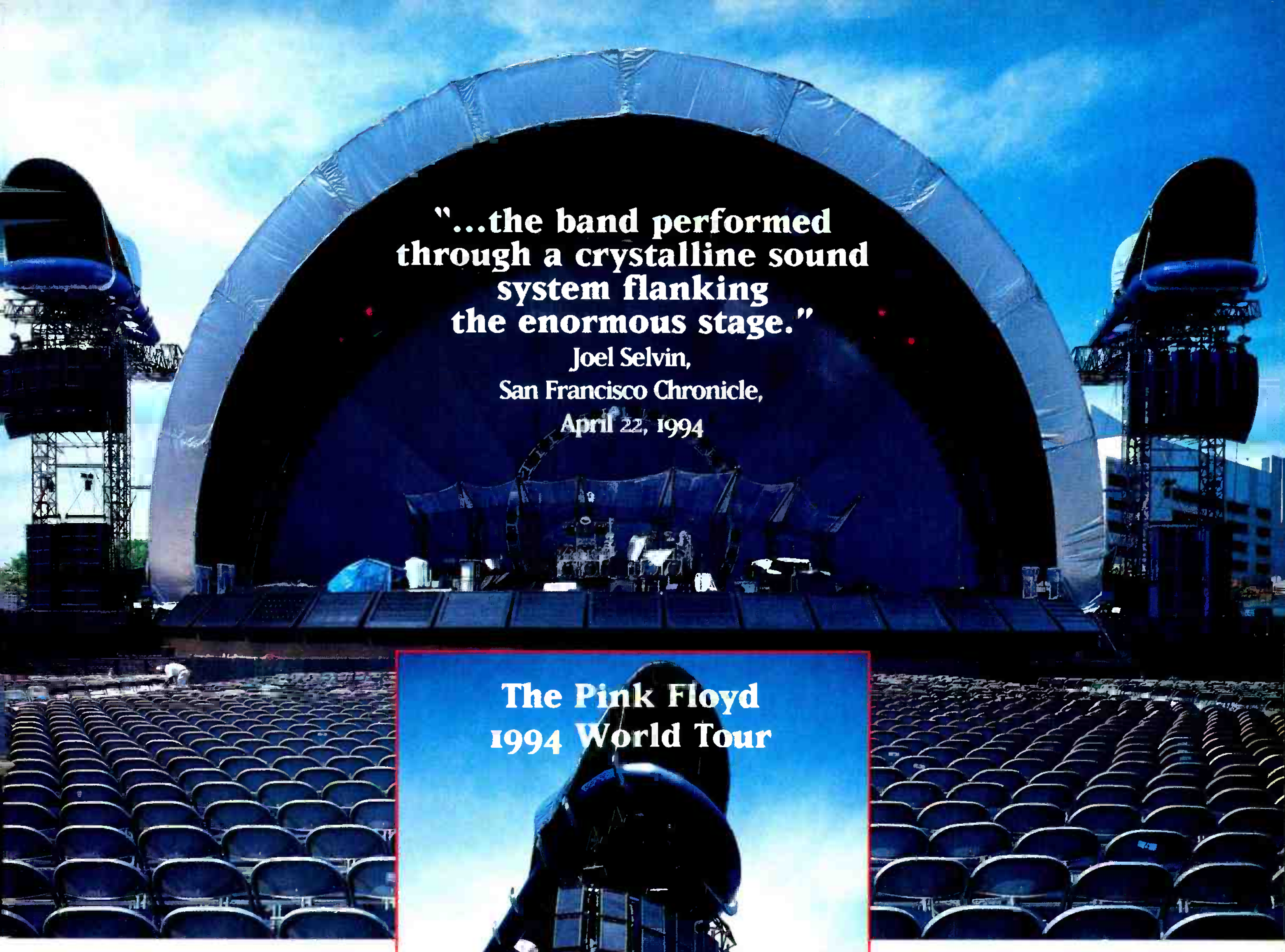
The 16 mix masters are fader controlled and organised across eight dual group modules. These modules also have an external return input for each mix group, and additional submixing allowing each group to feed a stereo matrix. The group modules may also be assigned to the 8 mute groups, but a safety switch is provided for 'panic' operation.

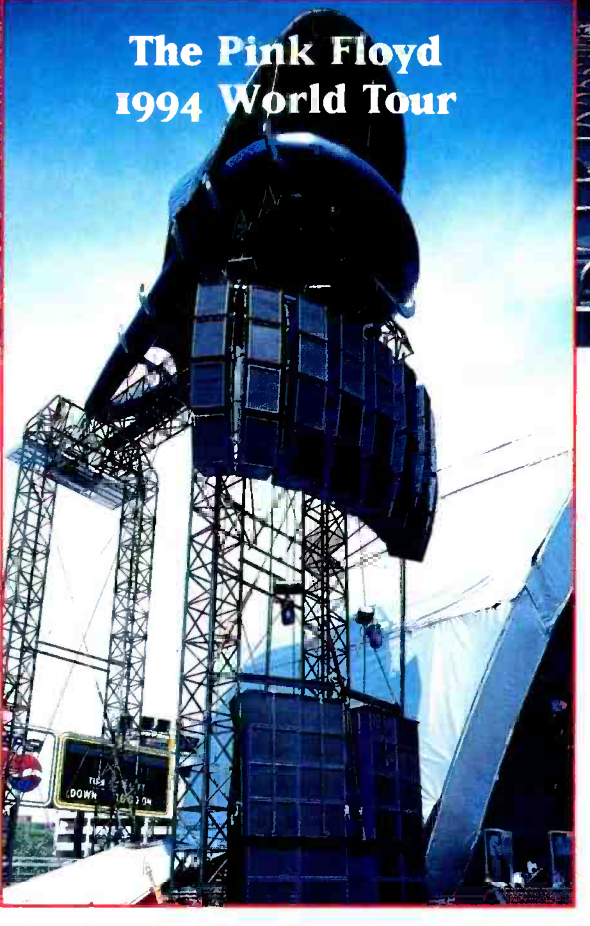
The console also features an advanced communications system, input and output metering, and an optional moving fader system (from Out Board Electronics) is available providing both snapshot and dynamic automation, plus MIDI control.

Peavey

Peavey live mixers fall into four groups. Rackable mixers: the *Unity* series, the *MD* and *SRC* desks, and the top-of-the-range *Mark VIII*. All consoles are low in cost and, apart from *Mark VIII*, are nonmodular designs.

The most comprehensive of the rackmountable range is the 16:2 *Versamix*. Facilities include 3-band fixed EQ, 6 aux buses fed from 4 sends (aux send 1 is pre-post selectable, others remain post), 4 stereo returns, and a combined tape send-return. The desk features a hinged I-O panel that can ▶





"...and the monstrous speaker system produced the delicate sound of Pink Floyd's thunderous music nearly to perfection."
Gerald Defiltch,
Pittsburgh Tribune Review,
June 1, 1994

"...a quadraphonic sound system that was near perfection."
Craig Marine,
San Francisco Examiner,
April 22, 1994

"...a quadraphonic sound system that rendered the 27-year-old band's music with fidelity previously unheard in stadiums.."
Sam Wood,
Philadelphia Enquirer,
June 3, 1994

Exclusively Turbosound.

"...pristine high-fidelity sound..."
Jim DeRogatis, Chicago Sun-Times,
July 14, 1994

"Production ruled the performance, and the sound quality was nothing short of amazing. When Tim Renwick strummed his acoustic guitar to start 'Wish You Were Here', the strings snapped crisply and clearly, as if he was sitting in his living room in front of the fire."
Joel Selvin,
San Francisco Chronicle,
April 22, 1994

"...The sound may have been the true star of the show, however... No rock band can match Pink Floyd when it comes to making a stadium show come off sounding as if it's being held in your living room."
Michael Norman,
Cleveland Plain Dealer,
May 27, 1994

Photos: On May 8, 1994, Pink Floyd played to a sold-out audience of nearly 50,000 at Vanderbilt Stadium, Nashville, TN. With Britannia Row & Turbosound, there wasn't a bad seat in the house.

Turbosound
Once Heard, Never Forgotten.

*The quadraphonic reinforcement system for **Pink Floyd's** worldwide 'The Division Bell' tour consists exclusively of Turbosound **Flashlight** and **Floodlight** enclosures, loudspeaker management systems, **BSS/Turbosound** amplifier racks, rigging and cabling.*

Turbosound Ltd., Star Road, Partridge Green, West Sussex, RH13 8RY England Tel +44 (0) 403 711447 Fax: +44 (0) 403 7101155
Turbosound wishes to express its sincere thanks to Pink Floyd, Robbie Williams & Steve O'Rourke. *Floodlight is manufactured under licence from Funktion One*


3M Professional Audio
Capturing Creativity...
Producers, Engineers and Musicians provide the creativity - 3M captures the results. Accurately, Consistently, Reliably.
Very high operating level enables massive headroom and low saturation effects.
A low noise floor that is subjectively as good as digital - even without noise reduction.
A smooth top end, complemented by clean, rock-solid bottom end punch.
Combining the best of analogue warmth with high clarity.
3M 996 - a tape with a sound so popular that even digital projects use it for mixdown.
Supported by the range of 3M Professional Audio products, covering all recording formats.
3M 996 - Capturing every level of creativity.
3M
996
3M Professional Audio/Video group
3M United Kingdom PLC,
3M House, P.O. Box 1,Bracknell,
Berkshire, RG12 1JU.
Tel: (0344) 858614 - Fax: (0344) 858082
3M Innovation

be orientated in different positions to optimise rack space and access to connectors, and has an external power supply. The other rackmounting mixers are the 7-input *7701R*, the 2U-high ***Line Mix 8*** and the 1U-high ***CMX 602***.

The ***Unity*** series consists of the ***2000*** (12 or 16 inputs) and the lesser-featured ***1000*** (8 or 12 inputs). Both consoles have single circuit-board construction, 60mm faders and offer 3-band fixed EQ. The ***1000*** has 2 aux sends (Mon and FX) and a single effects returns that can be sent separately to either the monitor or LR mix. The ***2000*** has 4 aux sends and 2 effects returns.

The 4-bus ***SRC*** console is available in 16 or 24 channels. On offer are 3-band EQ with swept mid, 6 aux sends, 6 effects returns, a stereo record send (pre main LR faders), and four 10-segment LED meters switchable between groups, LR outputs and PFL bus. The ***MD-III*** is a scaled down version of the ***SRC*** supplied in 12:2 or 16:2 sizes. Also available is the ***MD*** 16 x 6 dedicated monitor mixer which features splitters for each channel, 3-band EQ on inputs, variable HPF for outputs, and a separate circuit for the engineer's monitor.

Mark VIII is an 8-bus FOH console supplied in 24 and 36-channel frames. Channel features include 4-band swept EQ with fixed HPF, and 8 aux sends (pre-post fader or EQ in groups or 2 or 4). There is a 10 x 2 matrix with an external input to each matrix, 8 fully-featured returns, and a dedicated CD-Tape input with EQ and mono switching. The desk contains comprehensive talkback and intercom facilities, and there is separate control for a remote monitor.

Ramsa

Panasonic's Ramsa consoles are available in two series—the mid-priced ***S840*** (FOH and Monitor versions), and the budget priced ***S4400*** FOH console which was introduced last year.

The ***S840*** FOH desk in its standard configuration fits seven types of module into a 52-module frame including 36 mono inputs with 4-band swept EQ, variable HPF, 8 individually switchable aux sends, and switchable insert. Two double-width Submix Modules each source 8 mic-line inputs, and two effects-return modules provide four fully-featured returns. The eight group modules also include a 10 x 8 matrix with two external inputs and switchable insert. The console has a Talkback-Oscillator module, and offers 18 mechanical VU meters (8 switchable between matrix and group outputs).

By replacing standard input modules with Stage Monitor modules a 40 x 18 monitor console is created. Desks can be bus-linked for expansion.


Soundcraft *Vienna II*

The 4-bus ***S4400*** is available in 12, 16 and 24-channel frames. Facilities include 3-band EQ with a swept mid range, 4 aux sends (plus direct out option), 4 stereo effects returns and 6 mono returns. Six VU bargraph meters are fitted for monitoring the group and LR outputs, or the aux outputs.

Soundcraft

Soundcraft began as a PA console manufacturer 21 years ago, and although they have expanded into other fields, the live console market still accounts for a large proportion of the company's sales.

Europa was launched in 1991 as the company's flagship FOH console, and although top of the range, remains mid priced. ***Europa*** is available in 24, 32 and 40 channel-sizes, and has 8 groups, plus stereo and mono buses, which along with external inputs can feed an optional matrix. There are 12 individual aux sends, 4-band parametric EQ with variable high and low-pass filters, and a noise gate on each input channel. The desk also includes 8 VCA groups, 8 mute groups, 8 fully-featured stereo effects returns, and an optional stereo input module. Another option is Out Board Electronic's *SS2* automation system, which provides level control for VCA masters and various switches.

Soundcraft's top-of-the-range monitor console is the recently introduced ***SM24***. Available with 32, 40, and 48 inputs, the desk is based on the successful ***SM16***, and offers 26 buses which can be configured to provide both mono and stereo mixes. Other features include dedicated stereo send for sidefills, 4-band parametric EQ and HPF on all inputs, logic-controlled solo system, and full audio and logic console linking. Although primarily designed for monitor use the ***SM24*** may also be used for FOH applications.

Vienna II superseded the original ***Vienna*** a year ago, offering enhanced 4-band EQ with variable HPF, more flexible input compatibility, pre-post switching on each of the 8 auxiliary sends, and Grand Master VCA control for the 8 VCA groups.Other key features include 8 output groups, 8 mute groups, a 12 x 8 matrix, optional stereo input modules and a sophisticated linking system allowing two ***Vienna II***s or a ***Vienna II*** and ***Europa*** to be connected. ▶

Soundtracs *Solo Live*

Vienna II is an affordable, general purpose FOH mixer has been designed for medium-sized events. The 8-bus console comes in four frame sizes—16, 24, 32 and 40—and can be specified with a number of different modules to suit application. There are mono and stereo input modules each with 6 aux sends and four mute groups, a choice of dual group modules (either with EQ, or a stereo FX return), and a dual matrix module which can submix 10 sources. An additional module, the Theatre Input Module, provides discrete rather than paired output routeing, and adds pre-post switching to pairs of aux sends. A meter bridge is optional for all but the smallest frame size.

The baby of the range, the budget ***Delta SR***, took over from the ***200SR*** in 1992. Suited to small PA applications the console comes in 8, 16, 24 and 32-channel sizes plus an 8-channel rackmount version. It offers 4 groups, 4 aux sends, 3-band EQ with fixed HPF on both mono and stereo input modules, a 4 x 4 matrix, and 4 effects returns. Also available is the ***Delta Theatre*** which, like the theatre version of ***Venue II***, offers individual group routeing and pre-post switching for the auxes; it also expands the matrix to 6 x 4.

In addition to the ***SM24*** and ***SM16*** monitor consoles, Soundcraft produce the ***Delta Monitor*** and will shortly be releasing the ***SM12***.

Soundtracs

Soundtracs offer three ranges of SR consoles: ***Sequel II***, ***Megas*** and ***Solo***.

The top-of-the-range ***Sequel II*** is available in four sizes from 24 to 48 inputs. The 8-bus console offers 8 aux sends, 8 VCA-mute groups, an optional 11 x 2 matrix, and 8 fully-featured stereo effects returns. Each input channel is equipped with Soundtracs FdB 4-band swept EQ and a fixed HPF, and can access an optional Assignable Dynamics Processor providing up to two recallable dynamic effects per channel (gate-exp, comp-lim, automod/autopan). Another option is a stereo input module. The console also features a two-way communications system with individual group and aux routeing, and a VU meter bridge.

Megas comes in three variants: the recently released ***Megas II*** (stage and monitor desks) and the general purpose ***Megas Mix***.

Megas II Stage is a dedicated FOH console in four frame sizes (24 to 48). It may be configured with mono and stereo input modules, and up to four dual group modules and four dual matrix modules providing an 8-bus, 11 x 8 configuration. All inputs feature 4-band EQ with swept mids groups with SAFE button. Other features include separate group to LR bus submixing, up to four fully-featured, fader-controlled, stereo effects returns, individual talkback to groups and aux buses, and VU metering as standard.

The 12-way ***Megas II Monitor*** comes in the same frame sizes as the FOH version. Inputs have 4-band EQ with swept mids and fixed HPF, 10 monitor sends (pre-post selectable in groups of 5) plus access to the LR bus enabling console to double as a FOH mixer. The outputs have 2-band parametric EQ, and individual talkback is provided for each. An integral meter bridge provides 13 10-segment bargraphs for the monitor outputs, LR output and solo.

The 4-bus ***Megas Mix*** was introduced four years ago and is available with a range of modules options including mono and stereo inputs and a dual returns-group module. Inputs feature 4-band EQ (swept mids) with fixed HPF, 6 aux sends, and stereo modules include MS decoding.

The budget ***Solo*** range offers four live mixers: ***Rack***, ***Live***, ***8 Live*** and ***Monitor***.

The entry level ***Solo Rack*** is a 12:2+1 rackmountable design with 4-band EQ (identical EQ is used throughout the range), and 6 aux buses (pre-post selectable in pairs). It also features two stereo effects returns, separate faders for Left, Right and Mono outputs, and talkback to auxiliaries. The ***Live*** console is similar in design although it offers a 4-bus format and can be supplied in 16, 24 or 32-input frames. It has four stereo effects returns and a slightly more elaborate talkback system. The ***Solo 8 Live*** is an 8-bus version (24:8:2 or 32:8:2) with dedicated (rather than switchable) sends to the aux buses,4 mute groups and a SIP solo system as an alternative to the standard PFL. The four stereo effects returns are equipped with 65mm faders and can access the mute group. Talkback is individually assignable to each group and aux bus, and unlike the other ***Solo*** consoles, ***8 Live*** features a full length meter bridge with mechanical VUs.

The ***Solo Monitor*** has 10 monitor sends (individually pre-post selectable via internal jumpers), and 65mm faders controlled sends to the main monitor bus. Again there are 4 mute groups and every input has a built-in splitter output. Monitor outputs include a single band (50Hz–10kHZ) parametric equaliser, and phase reverse. Monitor output select buttons allow the operator to listen to any combination of outputs, and like the ***8 Live***, talkback can be individually assigned to buses. Ten-segment LED meters are built into the monitor and main outputs sections.

Studiomaster

Studiomaster offer a range of compact, nonmodular, budget consoles designed for smaller SR applications.

The top of the range is the ***Showmix*** FOH console which comes in three versions: 16-2, 16:4:2 and 16:8:2. It has 6 aux buses, 4-band EQ with swept mids (plus 2-band EQ for the LR outputs), 4 stereo returns, and can be extended with an 8-input expander. Complementing ***Showmix*** is the 16-8 ***Stagemaster***, which offers 3-band EQ for all inputs and outputs and can also be extended in blocks of eight channels.

The ***Diamond Pro*** series consists of five models: 8:13, 12:3, 16:3, 16:4:3 and 24:4:3; the two smallest mixers can be rackmounted. Input channels feature 3-band fixed EQ, 4 aux sends, and direct routeing to both the stereo and mono buses. The desks are also equipped with two fully-routeable stereo aux returns. ***Powerhouse Vision*** is a powered mixer (2 x 350W) with a built-in MIDI controllable reverb-effects unit, and two assignable 7-band graphic equalisers. It is available in 8 (rackmountable), 12 or 16-channel versions which mix down to LR outputs via a stereo subgroup. Input channels have 3-band EQ with swept mid, 3 aux sends (one dedicated to the internal effects processor), and one of the channels can be configured as a stereo module—there is also a dedicated stereo aux return. The console offers 20 user-memory stores which allow instant reset of effects programmes and various console switches, either via manual or MIDI control.

3G

German manufacturer 3G offer a series of low-cost, quality consoles: ***Silk*** which includes ***Live*** and ***Studio*** variants; and ***Signet***, which is dedicated to PA applications.

The ***Silk Live*** console comes in 16-input, 8-bus and 24-input 8-bus formats and offers 3-band EQ with swept mid on each input channel, 4 aux buses (each assignable pre-post) and 2 dedicated stereo aux returns. Power supply can be fitted as external units to reduce noise on request and the company specify different faders for studio and live operations in keeping with the demands of the application.

Signet is a 4-bus console which comes in 16, 24 and 32-input sizes, but is expandable up to the 32-channel maximum. EQ is 4-band with parametrics sweeps on high and lo-mid bands; the 8 aux sends are configurable (pre-post) and 4 stereo returns may be optionally configured as 8 mono returns assignable to any combination of groups and masters. Bargraph metering may be internally optimised for +4dBm or -10dBv ref.

TOA

The Japanese company TOA currently manufacture the only all-digital console specifically designed for SR applications. The ***ix-9000*** was released in 1990 exclusively for fixed installation applications particularly in opera houses—the first ▶

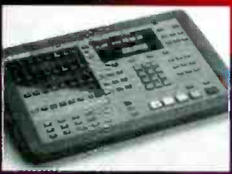
Pro-Audio
041-204-2774

For ALL Your Professional Audio Requirements from the 'Pro-Audio' Experts

ALESIS ADAT

- 16 Bit linear SVHS
- >92dB dynamic range
- Complete with LRC
- 1/4"jack ins/outs

ALESIS BRC



- Controls 16 ADATS
- Reads/Writes SMPTE
- Generates MTC/Clock
- Rack-mountable

TASCAM DA88

- 16 Bit linear Hi-8 video
- >92dB dynamic range
- Optional Remote
- Jog/Shuttle control

FOSTEX RD8

- 16 Bit linear SVHS
- >92dB dynamic range
- Complete with LRC
- 99 Locator Points

AIWA XDS1100

- 3 Motor Transport
- 93dB dynamic range
- Infrared remote control
- Optical/Coax In/Out

DIGIDESIGN Session8

- For Macintosh or PC
- >93dB A/D. >98dB D/A
- Random access edit
- Digital mix/trk. bounce
- DSP/parametric Eq.

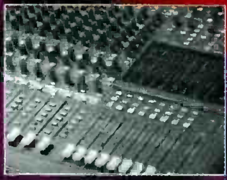
OKTAVA Microphones

FULL RANGE AVAILABLE OF BOTH THE RUSSIAN BUILT OCTAVA AND THE PRO SERIES NEVATON MICS INCLUDING:-

- MK219 • MK011
- MK012 • MD186
- MD87a • CMC401
- CMC51...

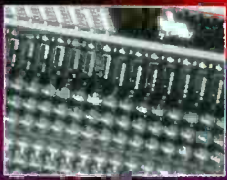
CALL US NOW FOR FULL DETAILS.

MACKIE 8-BUSS



- 16, 24 or 32 Channel
- Optional 24ch Expander
- Fully Parametric Eq.
- Superb Headroom & SNR
- Optional Meterbridge

AMEK - BIG by Langley

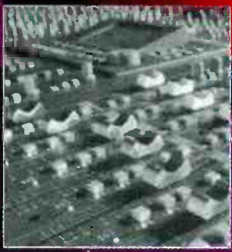


- Control MIDI from desk
- Total Mix Recall
- Automate Faders and Mutes against SMPTE
- C/w Meterbridge

SOUNDTRACS Solo Logic

- 24 or 32 Channel
- Control Tape Transports
- VCA Facer Automation
- Excellent Monitor Path
- 6 Seperate Aux sends
- C/w Meterbridge

SOUNDCRAFT DC2000



- 24 or 32 Channel
- Dedicated Touchscreen
- Auto Moving Faders
- Control M-track Transport
- Meterbridge with SMPTE

PEAVEY QFX4 x 4 Effex

- 4 independent digital multi effex in 1 unit rack
- 5 Simultaneous Effex per Processor
- Full Stereo In/Out per Processor
- Custom 40 Columns x 2 Lines Display
- 128 Presets per Processor
- Full MIDI In, Out and Thru
- Better than 96 dB Dynamic Range

Effects Include - Standard Reverb, Gated reverb, reverse reverb, chorus, flange, delay, echo, parametric EQ..

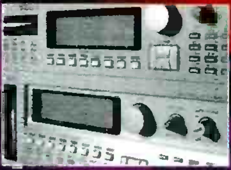
Original RRP - $918.00

Our Price - $349.00

AKAI MPC3000

- 16 Bit Stereo Sampling
- 2 Meg sound memory
- Expandable to 16 Meg
- SCSI, SMPTE, Sep. Outs
- Read/write Std. MIDI files
- Reads S1000/3000 disks

AKAI S-Series



- S2800 • S2800i
- S3000 • S3000i
- S3200 • CD3000

Full Range Available...

SOFTWARE Packages

WE ARE STOCKISTS FOR ALL THE LEADING BRANDS OF SOFTWARE - INCLUDING:-

- STEINBERG
- E-MAGIC
- DIGIDESIGN etc...

CUBASE AUDIO & AudioMedia-2

PACKAGE DEAL FOR MACINTOSH COMPUTERS:-

- Integrated MIDI and direct-to-disk recording
- 64 Sequenced MIDI tracks, 4 Digital Audio tracks
- Stereo Compression & time expansion
- Analog/Digital I/O's
- Digital Equalization
- Non-Destructive Editing

now - $1399.00

SOUND CONTROL Pro Audio

- Jamaica Street • Glasgow
- Strathclyde • Scotland
- G1 4NN • 041-204-2774

Freephone Sound Control 0800-52-52-60


Yamaha *M2000*

installation being at the prestigious Vienna State Opera.

The console offers 256 inputs and 256 outputs with 4-band parametric EQ, dynamic processing and delay line on all I-Os. there are 16 aux sends, 48 groups, and a matrix offering up to an 80 x 68 capability. All console parameters can be reset in $^{1}/_{120}$th of a second, and unlimited numbers of scenes may be stored-recalled. 512 built-in control relays (momentary or fixed), along with MIDI control messages can be included in scenes, and scenes themselves can be edited and sequenced in any order.

Other features include software-controlled metering for all I-Os, moving fader groups, and channel linking for ganging together channel parameters. The assignable nature of the console allows it to be very compact, and each fader channel only incorporates four physical controls: much of the control is implemented from two touchscreens placed in front of the operator. Two separate racks for interface and DSP are sited remotely to the console.

The ***ix-9000*** is a specialist, high end product, commanding a high price dependent on requirements. Although not officially confirmed, the next offering from TOA is likely to be a console aimed at the touring market.

Yamaha

The Yamaha range consists of three types of console: at the top is the ***PM4000***, in the middle is the brand new ***M2000*** and at the bottom is the ***MC Series***.

The ***PM4000*** took over from the popular ***PM3000*** in late-1992 making improvements to audio quality as well as reducing the overall size of the desk. There are three versions—***FOH***, ***Monitor*** and a special ***Hall*** model (***PM4000H***) which is made to order.

The ***PM4000 FOH*** mixer is supplied in 24, 32, 40 and 48-channel frames with 4 stereo inputs as standard. There are 8 groups, 8 VCA groups, 8 mute groups, 12 aux buses (8 mono, 2 stereo), and an 11 x 8 matrix. Other features include an input priority 'in place' cue system, plus a solo mode that mutes other channels, parametric 4-band EQ with variable HPF on all input channels, switchable channel insert, comprehensive metering, generous use of LEDs, and extensive internal switching to alter standard defaults. Consoles can be linked, and I-O ports are provided for interfacing mute and VCA groups, or to facilitate limited user-supplied automation.

The ***PM4000M*** monitor mixer shares most of the features of the FOH console and comes in 36, 44, and 52-channel sizes. It offers 18 mixes groups plus two stereo buses that can be directly assigned from inputs or groups. Four-band parametric EQ and variable HPF are available to all input and output modules.

The economical but well featured ***M2000*** (released last month) is produced in four frame sizes varying from 16 to 40 channels, and is equipped with two stereo modules as standard (more are optional). The 8-group console has stereo and mono output buses, a 13 x 4 matrix, 6 auxiliary buses and 4 aux returns, 4-band EQ with HPF for mono input modules and 3-band for stereo inputs, switchable inputs for both mono and stereo modules, and a comprehensive cue system. The desk features a 128-scene channel on-off memory which can be recalled either directly from panel controls or by an external MIDI controller. The ***M2000*** also outputs MIDI program change messages.

There are currently three ***MC*** consoles—the ***MC04 II***, ***MC03*** and ***MC10M*** dedicated monitor desk—all of which are budget products. The ***MC04 II*** is a 4-group FOH console (although it can be used for stage monitoring) available in 12, 16, 24 or 32-input sizes. It has 4 aux buses, 2 auxiliary returns, a 6 x 2 matrix, and 4-band EQ (fixed high and low, swept mids) with HPF. The more basic ***MC03*** is supplied with 8, 12, 16 or 24 inputs which mix down to stereo and mono outputs. There are 3 aux sends which can be each internally jumpered to operate pre or post, and 3-band EQ with swept mid frequency. The ***MC10M*** monitor desk, which supersedes the ***MC08M***, provides 10 monitor mixes and two aux mixes and is available in 24 and 32-input configurations, although consoles may be cascaded together to provide more inputs. The desk also features 4-band (swept mids) with HPF, input phase reverse, 4 aux returns, and versatile talkback facilities. ■

CONTACTS

● **Allen & Heath**
Tel: +44 326 372 070.
US. Tel: +1 801 566 8800.

● **Amek**
Tel: +44 61 834 6747.
US. Tel: +1 818 508 9788.

● **Cedar**
Tel: +44 582 404202.
ProMix.
Tel: +1 914 668 8886.

● **Crest Audio**
Tel: +1 201 909 8700.
UK Tel: +44 273 325840.

● **D&R Electronica**
Tel: +31 2940 18014.
US Tel: +1 409 588 3411.

● **DDA**
Tel: +44 562 741515.
US. Tel: +1 616 695 4750.

● **Dynacord**
Tel: +49 9421 7060.
Fax: +49 9421 706265.

● **Midas**
Tel: +44 562 741515.
US. Tel: +1 616 695 4750.

● **Peavey Electronics**
Tel: +44 536 205520.
US. Tel: +1 601 483 5365.

● **Ramsa**
Tel: +1 714 373 7277.
UK. Tel: +44 344 853920.

● **Soundcraft**
Tel: +44 707 665000.
US. Tel: +1 818 893 4351.

● **Soundtracs**
Tel: +44 81 399 3392.
US. Tel: +1 516 932 3810.

● **Studiomaster**
Tel: +44 582 570370.
US. Tel: +1 714 524 2227.

● **3G**
Tel: +49 6081 94190.
Fax: +44 702 420647.

● **TOA**
Tel: +44 277 233882.
US. Tel: +1 415 588 2538.

● **Yamaha (UK)**
Tel: +44 908 366700.
US. Tel: +1 714 522 9011.

AURA STUDIO

From modest origins, Portugal's Aura Studio has become a sophisticated recording facility. Sergio Castro investigates the all-digital connection

Portugal's Aura Studio first opened as a semiprofessional facility in the mid-1980s. Back then, the single-room studio was based on two synchronised Fostex *E16* multitrack recorders and was intended primarily to serve as a demo studio for the owner, Fernando Rocha, and local bands. Rocha—then a young pianist with one such promising band—had been to the United States to study sound engineering. On returning to Portugal, he built Aura (his first studio) in the small house at the back of the family farm. I visited the place in 1986, when I used it to record music for television, and although limited in its facilities, I could sense Rocha's interest in the latest technology and innovation.

In 1987, Rocha bought one of the very first Yamaha *DMP7*s into Portugal and a year later he exchanged it for a number of '*D*-version' *DMP7*s—he had defined, and was beginning to build on, his philosophy of recording. Today, the policy of avoiding processing between the source and the tape is relatively commonplace; in 1987 in Portugal, it was visionary and was to lead directly to the concept of the all-digital studio.

To this end, Rocha had Yamaha *MLA7* preamps racked on the recording area walls, so that the mic outputs could immediately be routed to any of the Fostex' available 30 tracks and then through *AD8X* 19-bit A–D convertors to the mixers. The *DMP7D*s were replaced by *DMC1000*s in 1991, and this hybrid system then was further modified by the exchange of both *E16*s for a 1-inch, 24-track Tascam machine.

In 1992 the digital chain was completed with the acquisition of a Sony *3324A* DASH machine. This time the A–D convertors were placed in the recording areas, and employed immediately after the preamps, thereby ensuring a clean signal chain from the mic output to the digital multitrack, monitored or digitally mixed to DAT through the cascaded *DMC1000*s.

After being converted by the Yamaha A–Ds, the signal was converted to the 20-bit DASH format, using a Yamaha *FMC8*, bypassing the *3324A*'s convertors and going straight into the digital SDIF2 inputs, via a custom-built digital patchbay, which allows assignation of any of the eight channels of the three *AD8X*s to any of the 24 tracks of the multitrack.

'The next stage is to cascade the two *DMC1000* desks,' Rocha explains, 'and digitally feed them from the Sony multitrack as the Yamaha desks will accept any resolution from 16 to 24 bit.'

Aura's mastering format is the Tascam *DA30* DAT machine. 'We chose the *DA30* because we needed a machine that could bring out monitor without tape-head contact being necessary,' explains Rocha. 'We knew this was not the best-sounding machine, but we were determined not to use the A–D or the D–A convertors, which is normally where these machines fail.

'The DAT machine feeds the last digital device in the chain —a Nakamichi preamp—and from there it is converted back to analogue and fed through the amps to the B&W main monitors.

'Following mixdown we can do in-house editing by using our Roland *DM80* hard-disk unit. The production master may as well come out in DAT format or as a CD-R, which some CD plants are capable of working with. It gives less alignment and error correction problems than DAT.'

Processing

One striking feature of Aura is the absence of the extensive racks of outboard gear which frequently characterise a commercial recording facility.

'The *DMC1000* has built-in 32-bit effects,' says Rocha by way of explanation, 'but the reverb sounds have a Yamaha-ish character which is not particularly pleasant. For this reason we will be adding a Lexicon *300* very soon.

'We still try to simplify the signal path as much as we can,

The twin Yamaha *DMC-1000* consoles dominate the control room

starting from the microphone,' continues Rocha, 'through the multitrack, and then to the final master. The option remains to mount both the preamps and the A–D convertors in the recording areas to achieve the minimum length of analogue signal displacement.

'Sometimes you may need to process the analogue sound coming out of the microphone preamp with compressors or EQ, but I try to keep the signal as clean as possible. When it comes to recording acoustic material, we would rather have it go straight to the multitrack with no processing at all because we can do changes on the mix with the built-in dynamics—I am not very happy with digital compressors.'

Rocha's approach to recording realistic sound relies on the choice of microphone and its positioning.

'In extreme cases we may need to use equalisers and-or dynamics between the preamps and the A–D convertors,' he concedes. 'The problem is to adjust them remotely. At present it takes two people in constant communication to make an adjustment. A more technical solution involves a very high investment, but it will soon be made—the use of MIDI remote-controlled analogue equalisers, to make possible their insertion in the preamp stage, or possibly new analogue preamps with EQ and setup memories, all remote-controlled from the control room.

'I believe that there are tremendous advantages in sorting out the sound before A–D conversion, at least in terms of recording resolution. If you start with the wrong sound it is very difficult at a later stage to modify it without compromise. I do not really believe in the idea of the free manipulation of the digital recording. I think that the "digital print", creates a pattern with its own characteristics, which is hard to modify in musical terms. It is definitely acceptable to add effects or gently correct the program EQ at the mixing stage, but dramatic changes must be avoided.

'The dynamic range of our recordings is well over 96dB, bearing in mind that we are recording at 20-bit rate, processing it at 28-bit, internally noise-shaping it down to 24-bit, and then, with the Super Bit Mapping, optimising it to the consumer CD 16-bit standard. The Sony *K1203* SBM was our only option at the time. We now know of some other directions we could have taken.

'From the AES-EBU bus output we can feed the Roland *RSS* interface, and come back through two of the remaining digital AES-EBU inputs of the second *DMC1000*. This way we can add some spatial effects to the mix.

'It must be done extremely carefully. We normally use it on string pads or on instruments which previously had been added to the arrangement bearing it in mind. However, we constantly check for the mono compatibility, especially for broadcast purposes. We absolutely avoid using it on basic instruments such as bass or drums. I think that by processing some less relevant sounds of the mix with the *RSS* we end up highlighting them without interfering with the front mix, thus giving a psychoacoustically wider stereo mix. In fact some people even cannot perceive the effect on some records.'

Now and next

The acoustics of the studio have also undergone some transformations since the mid 1980s.

'My main concern when building the studio, was the control room acoustics', recalls Rocha. 'I had a few limitations on the available space. Luckily, however, this room came out with a very low frequency resonance, and ended up with good overall dimensions. So, considering the working area, I tried as much as possible to deaden the surfaces, keeping the sound path clear from the side reflections, as well as controlling the bass response of the room with some bass traps.'

Aura's grand piano is a reminder of the studios origins

The resultant acoustic relies on a simple treatment; by careful combination of soft and hard materials, effective control on the entire frequency spectrum has been achieved inside the room.

'There are no surprises when it comes to the mixdown,' Rocha asserts. 'Anyhow, I intend to swap the passive B&W crossovers for actives. I will try the new Yamaha *D2040* digital 4-way processor, wired three ways. It has its own 20-bit D–A convertors, accepts a digital input and is a variable frequency and slope unit with built-in delay adjustment. This way, I believe I will be able to optimise the response of each individual driver, without using equalisers.'

Another development of Aura's activities is the introduction of a location recording facility. Primarily used for classical projects, this facility employs a Tascam *DA-88* tape-based digital recorder.

'One of the reasons for choosing this is that it records at 44.1kHz. I had the option of the *ADAT*, but again the Tascam gave us more usable time per tape, and it also allows immediate transfer of the eight tracks over AES-EBU or SDIF2, so it can be used to drop directly onto the multitrack. It can also be considered an 8-track extension to the Sony, allowing us to use 32 synchronised digital tracks. When recording classics we use eight microphones—most of the time these are routed straight through the preamps to the eight tracks of the *DA-88*. We then bring the recordings into the studio, choose the right takes and drop them into the Sony. If necessary I can even edit them, prior to mixdown, on the *DM-80*, which I think makes it a lot easier to match the sound of the different cuts.'

In 1994, you can still lie on the lush grass of the farmyard between recording sessions at Aura Studios. The studio continues to grow, however—the building has developed to house three recording rooms, a control room and some other functional areas. Full kitchen facilities and an editing suite are the latest additions. In the same premises—and under Rocha's direction—a record label called Numerica is doing good work with a wide range of musical projects, from new age and rock music to fado, a traditional Portuguese music. Numerica intend to release no less than 12 CD albums during 1994, using its own marketing and distribution. Madrid-based Portuguese jazz pianist, Emilio Robalo, is releasing his new album through Numerica. It is one of the first SBM-processed releases from Aura.

Professional life is rather busy for Fernando Rocha—perhaps it is a good thing that he does not have far to travel for a holiday in the sun. ■

Aura Studio, Rua do Barroso, 673, PO Box 21, Paços de Brandão, Portugal. Tel: +351 2 7446880. Fax: +351 2 745 0450.

TESTING TIMES

Patrick Stapley auditions a new audio test and setup CD from Producer Alan Parsons and Designer Stephen Court

There have probably been occasions in every engineer's professional life—whether dealing with music recording, broadcast, postproduction, sound reinforcement, theatre or R&D—where a reliable and comprehensive collection of easily accessible test and reference material at hand would have been a great advantage. This had certainly crossed the minds of Producer Alan Parsons and Sound Designer Stephen Court when they conceived Sound Check.

'Test discs have either been all technical or all musical, and it was about time to fill the gap by combining both on one CD,' says Parsons. 'In general I don't think previous discs have targeted the right audience, and many have actually been far too technical for the average engineer. What we've aimed to do with *Sound Check* is to provide a 'do everything' disc, at a sensible price that people in the real world—studio engineers, live sound engineers, home studio owners and so on—can all make good practical use of.'

The disc is split into two-thirds technical material and one-third music-SFX, and contains a broad selection of tracks ranging from test tones to tanks. Parsons makes no secret of the fact that he has chosen many of the items purely on the basis that he would use them himself, but he is confident that others will find them equally useful.

'I find that I use the disc almost every day for various purposes, but the thing I probably use the most are the solo instruments. They've all been recorded totally dry, and are great for testing and setting up outboard equipment. Having everything on CD is wonderfully convenient and it means I no longer have to search through tapes for suitable test material and spool it backwards and forwards.'

As might be expected with Stephen Court's background in loudspeaker system design and live work, the disc also contains a comprehensive selection of reference tracks designed not only for equipment alignment and test applications, but also more specifically to set up speaker systems—checking for resonances and room effects. In fact, the CD used in conjunction with a good quality microphone and standard metering (mixer, tape machine), can provide a spectrum-analysing facility capable of plotting system response curves. Additionally, *Sound Check* is to be made available in a double CD pack with the rear half containing a built-in, miniature spectrum analysis unit.

All aspects of *Sound Check* from production to sales have been handled solely by Parsons and Court who formed Soundcheck Productions specifically to deal with it. However, this was not their original intention as Parsons explains.

'Originally we had a deal with Polydor to put it out, as we felt a major label with a good distribution network would be beneficial. However, due to politics, this didn't happen and instead we decided to set up our own label and do the whole thing ourselves which has turned out to be far more profitable.'

Contents

The *Sound Check* CD has been arranged into nine sections offering a total of 92 tracks with an overall playing time of 66½ minutes. The first five categories contain test signals generated from a Bruel & Kjaer *1049 Sine-Noise* Generator which were digitally recorded via 16-bit A–D delta sigma convertors. All levels indicated on the disc relate to OdBFS (Full Scale Digital), and directly before each group of test signals is a 1kHz tone recorded at -14dBFS. This references the recorded level of the majority of the technical bands; the exceptions being square waves (recorded at -20dBFS) and a maximum level (OdBFS) 1kHz sine wave. The 14dBFS level was considered the most appropriate of the industry standards, offering good headroom.

The first group contains seven tracks of pink noise which are intended for equipment alignment and in particular for setting up active speakers systems. They provide left channel only, right channel only, both channels plus phase check, and 4-band limited tracks: Bass (0–200Hz), Low Mid (200Hz–1kHz), Mid (1kHz–20kHz), and High (7kHz–20kHz)—these also include a 180° phase check on CD Index 2.

The next section contains 31 bands of third-octave pink noise across the 20Hz–20kHz range, and these are used for spectral analysis. Normally speaking, one would expect to use costly, precision-built equipment, but by splitting the spectrum into 31 individual bands, *Sound Check* provides the means of measuring response using 'conventional' equipment—CD player, microphone, and meter.

The procedure is as follows: first of all the -14dBFS 1kHz reference tone is played to establish a good working level. Next the measuring equipment is set to read zero: this can be an SPL meter, tape or cassette recorder level meter, a mixer output meter, a volt meter across the power amplifier input, or the *Sound Check* mini spectrum analysis unit (more of which later). Once set up the 341 10-second bands can be played through while making notes of peaks and toughs, thus enabling a system response graph to be plotted.

However, although the system appears extremely straightforward, there are some fundamental points that should be taken into consideration. Firstly, it is important that the microphone being used has as flat a frequency response as possible; failing this, allowances should be made for any irregularities in response—for example, it is likely (unless using a specialist microphone from a manufacturer such as B&K) that the extreme high and low frequencies will be rolled off, and that there may be a boost somewhere in the mid-frequency range. If the frequency response for the mic is unknown, it will be prudent to test it or alternatively ask the manufacturer for details. It is also advisable to make a number of plots using different mic ▶

placements to ensure a more complete picture of the room's effect on the system's frequency response.

The *Sound Check* notes offer some basic pointers, explaining that peaks at around 100Hz where there is a concentration of energy in music, and 10kHz where the fidelity of music is accentuated, are quite acceptable as is a slight dip between 1kHz and 4kHz where speakers and ears tend to be more sensitive. If Sound Check is being used to configure equalisers, it is generally best to avoid taking the readings too 'literally', and in many cases a compromise between settings will produce the best results. If allowances are made for these points and a common sense approach adopted, the system is fully capable of producing good results, but it should be stressed that it has not been designed as a replacement for top-end analysing equipment, as Parsons is quick to point out.

'I should emphasise that the system is not in anyway meant to replace laboratory analysers, reference microphones and so on. If fills a niche by allowing a series of simple tests to be run that will clearly show up problems without being over analytical.'

The next section has two swept frequency tracks. The first sweeps down in octave steps: 1kHz–500Hz, 500Hz–250Hz, 250Hz–125Hz, and 125Hz–20Hz; while the other sweeps up from 1kHz–20kHz in four steps (although according to the sleeve notes there are only three). Swept tones are useful for showing up system resonances, but pinpointing the actual offending frequencies will involve a certain degree of guesswork as the sweep is obviously not manually controllable or tunable as with a standard sweep generator.

A collection of 11 sine waves from 60Hz–15kHz comes next. Each lasts ten seconds and has obvious uses for alignment and system testing. It is worth noting that all the test tone-noise tracks have been indexed to allow continuous play (using the Repeat function found on most CD players) without playing the trac announcement each time. Following the fixed sine waves is a full 20Hz–20kHz swept sine wave, two square waves (1kHz and 5kHz) recorded at -20dBFS (to compensate for their perceived increase in loudness compared to the sine waves), and a 1kHz sine wave recorded at 0dBFS which depending on how system levels have been set, should be treated with great caution.

This brings us to the musical items which are preceded by a track of speech (Parsons talking for a minute about the disc). The importance of the spoken word as a reference is generally underestimated, and it can give a great deal of useful information—I know of two very successful recording engineers who always check monitors using speech as their reference. The voice track along with the instrumental excerpts have all been recorded without any dynamic or reverb processing (although EQ has been added), thus retaining natural transients and acoustic. They originate from two sources, some have been specially recorded for the CD by Parsons at his home studio (drums, guitars, flute...), and some have been taken from a KPM Library Music CD called *Tradewinds* by Graham de Wilde and Mitch Dalton (piano, bass, strings...). The library music was recorded at CTS Studios in London and was chosen for being 'a good clean, all-digital recording, that lent itself very well to test purposes.'

The solo instrument tracks comprise—Piano (Steinway Model D Concert Grand), three separate acoustic guitars (steel-strung picked, steel-strung strummed, and nylon-strung Spanish guitar), two electric guitars with clean and distorted sounds, bass guitar, flute, sax, vocal, bongos, tambourine, drum kit broken down into its components parts plus as a whole, violin section, and cellos and violas (starts with cellos panned right, later joined by violas panned left). All are in mono apart from the piano, drums and strings. These 19 tracks provide a good cross section of real instruments, perhaps with the exception of a brass section, and are useful as an instant source for setting-up or assessing outboard equipment such as effects processors, reverb units, EQ units, dynamics processors and so on—they can also be used for reference purposes once the user has become more familiar with them.

The next four tracks are extracts from finished mixes transferred from CD. They cover a range of musical styles from Yello's 'The Race' to Bach's 'Toccata and Fugue in D minor' played by Daniel Chorzempa.

'The Yello track has been very popular with a lot of people for showing-off systems—including Stephen Court who uses it to assess his speaker systems,' says Parsons. 'It's got a good overall sound and dynamic, and generally sounds impressive. My track 'Limelight' (taken from the Alan Parsons Project album *Stereotomy*) has a drum sound and vocal sound (Gary Brooker) that I'm pleased with, and the track from Graham de Wilde and Mitch Dalton provides a good contrast with well recorded piano and orchestra. The Chorzempa organ recording is simply magnificent; we specifically wanted a cathedral organ and listened to a lot of recordings before we chose this one; the bottom end is wonderful and it really makes the room shake—Stephen in particular has always been very keen on shaking rooms!'

The five sound effects tracks which appear next are certainly the most entertaining items on the CD and have already been heard resonating around various trade shows throughout the world. In fact the *Chieftain* tank recorded by Stephen Court in the late 1960s has a certain notoriety: during the 1980 AES held at the Park Lane Hotel in London, the *Chieftain* recording along with sub-machine-gun (also recorded by Court and included on *Sound Check*) was responsible for literally bringing the ceiling down in the Court Acoustics demo rooms. It was also responsible for bringing down a very nervous group of hotel security men who were convinced the building was under siege by Iranian terrorists.

The other sound effects are a steam train montage; a thunderstorm which again has been created from different sources including German thunder and English rain; and fly-pasts from *F-16* and *Tornado* jets complete with afterburners recorded at Biggin Hill and RAF Cotteshall. With the exception of the army effects and a couple of steam train 'chuffs', all the SFX items have been taken from digital recordings (Sony *F1* and DAT).

The final category on the CD is called Utility Tracks; these include an A-440 tuning reference and three time-code stripes for EBU, SMPTE, and SMPTE drop-frame; each running from 10:00:00 to 10:05:00. These have uses where a time-code generator is unavailable, or can be used instead of a generator or striped tape as a convenient source for testing automation systems and the like.

The disc was compiled and mastering using a Sonic Solutions system at Abbey Road Studios.

Spectrum analysis

Later this year, *Sound Check* will be available in a double CD case format—the front half containing the disc while the back houses a miniature spectrum analysis unit consisting of a battery operated LED sound level meter with a built-in microphone. The system offers a neat, portable alternative to setting up microphones and meters and has obvious novelty value appeal.

Strictly speaking, the unit should not be viewed as a spectrum analyser as this suggests the system is capable of measuring and displaying frequency response across the whole spectrum at one time. The *Sound Check* unit will simply enable the user to plot response by individually measuring bands one at a time.

The display is made up from a vertical array of ten LEDs calibrated in 3dB steps (-15dB to +12dB) which shine through the perspex back cover—all are red apart from 0 level which is green. The built-in electret microphone is specifically manufactured for calibration purposes being substantially flat between 20Hz–20kHz, and is placed at the side of the case along with the on-off switch. The sensitivity of the unit has been factory set to read 0 level at 85dBA, but this can be trimmed internally. To guard against flat batteries (four AAA cells) an autoshutdown facility has sensibly been included that switches the unit off after 15 minutes of use.

Conclusion

The *Sound Check* test disc has many uses whether referencing monitors, aligning equipment, fault-finding, assessing new equipment, measuring system response, or simply for impressing clients by firing off a few round from your *Chieftain* tank or showing off the realism of your new train set!

It has been aimed at the working professional who requires a practical cross section of easy to use and quickly accessible test material. For the first time it allows certain functions, such as spectrum analysis, to be basically performed without the need for expensive, specialist equipment—although it is not intended as a replacement.

Sound Check will be useful in all areas of professional audio and offers the busy engineer a reliable and reasonably priced test and reference package which could prove invaluable. ■

Soundcheck Productions.
Tel: +44 753 833 524.
Fax: +44 753 832 812.
Available through *Studio Sound* see the advertisement in this issue.


NEW PRODUCTS. STUDIOSPARES. 61-63 ROCHESTER PLACE, LONDON, NW1 9JU. tel 071 482 1692. fax 071 485 4168.
AKG C3000
SECURITY SYSTEMS
CARVER AMPLIFIERS
1961 DRAWMER TUBE EQ
STUDIOSPARES MULTIWAY
dbx EFFECTS UNITS
NEUMANN TLM 193
ULTIMATE A FRAMES
RADIO HEADPHONES
BANTAM PATCHBAYS
ULTIMATE STANDS
BEYER INFRA RED SYSTEM
WORK STATION
SONY MINIDISCS
The new 100 page Studiospares Catalogue was released on 23rd July.
If you are without your copy phone 071 482 1692 now.
19' EQUIPMENT CASES
PLASTIC CABLE DRUMS
STUDIO BUG MIC
PZM MICROPHONE
AKG TRI POWER
METAL HOLE PUNCH
MUSIC STANDS
MINIDISC RECORDER
Please mail my catalogue to
Post Code

BEYOND COMPARE

Philip Newell discusses problems in the comparison of loudspeakers and argues the case against reviews

Loudspeaker manufacturers frequently comment that their small monitor systems do not receive the quantity of editorial reviews in the professional recording press as do many other products. Those magazine editors who understand the problems (along with many users) would doubtless love to offer truly meaningful comparative tests, but such tests as are often envisaged are usually either-both flawed and controversial or not taken too seriously. In general, different loudspeakers sound more different than different amplifiers, and while much can be gleaned from the specification and general review of an amplifier by an experienced individual, such is not the case for loudspeakers, where conventional written specifications relate only very poorly with perceived sonic performance. The only seemingly viable means of comparison would be listening tests, but if results are inconclusive, then what real purpose would they serve?

No one loudspeaker can be all things to all people. As with musical instruments, different manufacturers largely judge the success of a design by the degree of acceptance by professional users. However, the 'rightness' of a musical instrument is generally accepted as the entirely personal, subjective choice of the individuals concerned, whereas the 'rightness' of monitor loudspeakers is usually related to a more objective, definitive 'rightness': 'the closest approach to the original sound', as one manufacturer so aptly put it. Nonetheless, even this is subject to user priorities, as a person recording computer music at home and needing a 'vibe' with which to work may need a very different monitor to a person recording a chamber orchestra. Initially, it would seem that the 'best' loudspeaker would suit all purposes but this is not necessarily so. To use an analogy, one could decide that one wanted the best aeroplane wing. It would be reasonable to assume that British Aerospace were likely to have the know-how for such a requirement, but it would be futile asking them simply to produce the 'ultimate' wing. Which ultimate wing? For what flying speed? To fly at what altitude? To carry what weight? For what minimum landing speed? To fit in which hangar?

Location

Before attempting any comparative assessment of small monitors in particular, one must ascertain with great accuracy the conditions of location, mounting and driving for which the manufacturers have designed their systems.

Loudspeakers designed for mounting in a relatively free space will have substantially different low-frequency characteristics if mounted next to a wall. Indeed those characteristics will be different again dependent upon whether the aforementioned wall is either behind or to one side of the loudspeaker. If the low-frequency directivity is 360° or thereabouts, then in free space, the low-frequency power output is balanced such that a more or less uniform overall response exists on axis. With a wall behind the loudspeaker, the mid and high-frequency output will still continue to project in a largely forward direction as before, but the low-frequency energy cannot propagate through the wall—at least not to any significant degree, so it is reflected back into the room and reinforces the forward radiating part of the low frequency output. The on-axis response will thus be raised by something in the order of 3dB.

If the wall in close proximity to the loudspeaker is to one side, then the same sort of low-frequency augmentation will take place. This time, the energy which will have travelled to one side of the loudspeaker will be reflected back to reinforce the axial response. In this case, however, some mid and high-frequency energy may also be reflected off the side wall, dependent upon its nature, and while the overall on-axis energy may remain reasonably uniform, the reflections at the shorter wavelengths of the mid and high frequencies will be time delayed. Due to the additional length of their reflected paths being a greater proportion of their wavelength than the low-frequency reflections, the resultant phase discrepancies will cause comb filtering, and hence both coloration and time smearing of the signal (**Fig.1**).

An extension of this location problem occurs when the choice is made arbitrarily as to whether to mount the small monitors on top of the mixing ▶

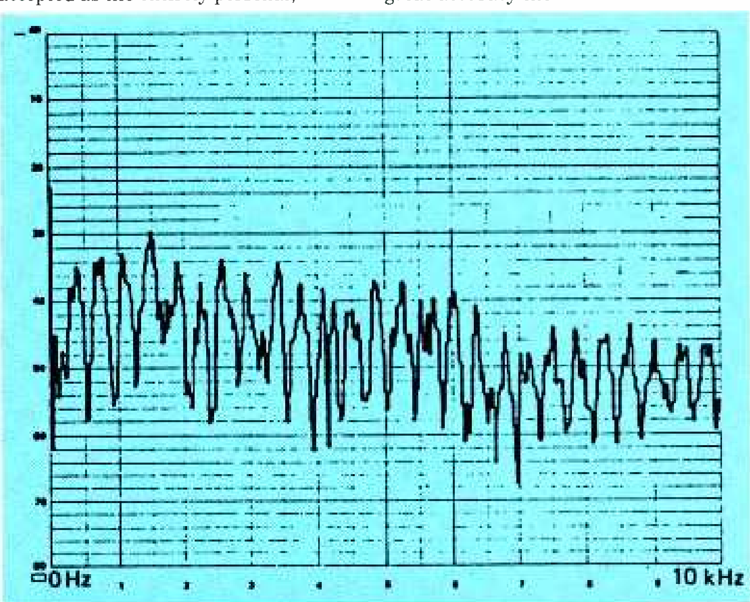


Fig.1: Above is the averaged power spectrum of a signal with one discrete reflection comb filtering is revealed clearly. The additional path length of the reflected signal over the direct signal is just under one metre, producing comb filtering with dips at a constant frequency spacing of just under 400Hz

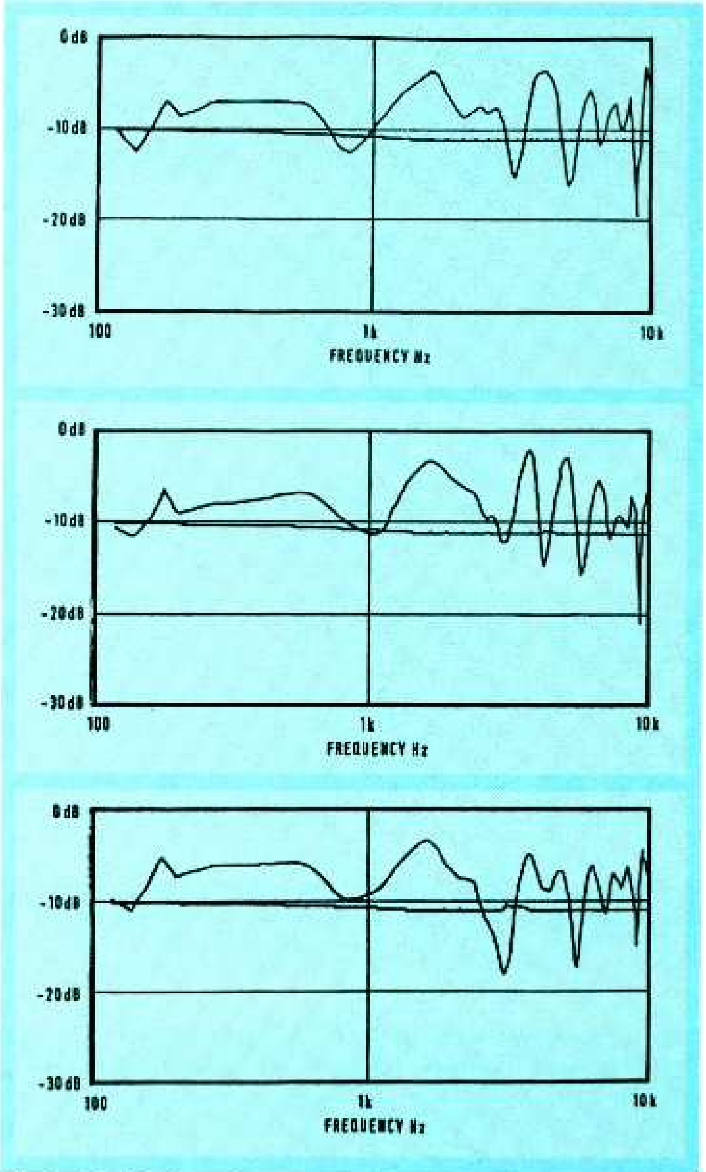


Fig.2a: One loudspeaker measured in three rooms, on top of three different mixing consoles. Note different high frequency responses due to varying reflection patterns

console, or on the stands behind the mixing console. When mounted on top of the console (typically on the meter housing), the flat surface of the console will act something like the side wall described above, largely reflecting at all frequencies. If the loudspeakers are mounted on stands behind the console, there will be a space between the loudspeaker and the console which will allow the low frequencies to 'breathe' and escape below the console, approximating more to free-space mounting, but here, the middle and high frequencies still encounter a reflective surface in the top of the mixing console. Different consoles, depending on design and construction, will reflect to different degrees at different frequencies. Panel resonances will be different in different consoles, and absorption or resonant overhang will thus also be different from console to console.

While the side wall or the console reflections may measure quite similarly, the reflections from the side wall will most definitely be perceived differently to the vertical reflections coming back up from the console, both in terms of perceived coloration, and the effect in stereo imaging. Given all of these variables—and even scratching the surface to add a few more, such as the proximity of corners or other low frequency reinforcing structures and mid-high frequency reflections off equipment racks—it is evident that a small monitor system is unlikely to represent similar listening conditions for any two pairs of units sold. This is one reason why I have avoided manufacturing or selling loudspeaker systems unless I have had some control over the acoustic design of the room in which they are to be used. I sympathise with loudspeaker manufacturers who face totally uninformed criticism of their products which may have been auditioned in entirely inappropriate circumstances.

Practicalities

The first problem in any meaningful test setup for auditioning small loudspeakers is therefore to locate them in a position preferred by their manufacturers; the location in which they were designed to perform best. If one attempted to set up, say, six pairs for A, B, C, D, E, F comparative testing, then what happens if say two or more loudspeakers demand the same location for optimum performance, or the preferred location for some of them obscures the direct signal path for the others? No loudspeaker manufacturers can be criticised for their units sounding less than optimum in locations for which they were not intended to be used.

The second problem which arises with any sets of loudspeakers in close proximity to one another is that the drive units, or their tuned boxes, or both, of the systems not being driven, may resonate in sympathy with the driven units, hence they can cause either absorption or resonant overhang. If the amplifiers are left switched on and connected to all of the loudspeakers, then some electrical damping will be effective in vastly reducing the vibration of the nondriven loudspeaker cones, but there is little that can be done to completely prevent a tuned box from resonating.

Setting up more than a couple of pairs in optimised locations begins to appear as something of an intractable problem.

A few years ago, Tannoy conducted a series of listening tests. The intention was to audition four different models of a new range of loudspeakers, and also to help to decide on which one of four short-listed amplifiers to recommend for use with the new range. The outcome was quite unequivocal: no one amplifier performed best on all of the four loudspeakers. Indeed only one amplifier was not chosen as being the best on at least one of the four loudspeaker pairs. This being the case, whenever a loudspeaker is auditioned, its manufacturer should be referred to in order to find out which amplifier(s) is (are) recommended for optimum loudspeaker performance. Some loudspeakers have benign impedance characteristics while others are nightmares to drive. Two different amplifiers which may be deemed to sound the same when driving the benign loads, may perform radically differently on the difficult load. So, if we must consider the room position and the amplifiers to be used when auditioning loudspeakers, are we ever actually auditioning the loudspeakers *per se*, or are we auditioning an electroacoustic system comprising amplifier, loudspeaker and room? In reality, can we ever just audition comparable loudspeakers? I think not.

Design priorities

In previous articles I have discussed the different order of priorities which face loudspeaker designers, dependent upon the type of music, be it rock, classical, electronic or whatever; and the type of recording technique such as close mics, stereo pairs, or direct injection. Furthermore, would the loudspeakers be the original source of sound as would be the case for computer-generated music, in which case the monitor loudspeaker during recording would be an extension of the music production system; or would the monitors always be in the reproduction chain, as when recording an actual acoustic event? In the first instance a 'buzz factor' for the musicians is a possible requirement when the loudspeakers are part of a music production chain, whereas when recording an actual acoustic event, the loudspeakers will not be in the same room as the musicians so no 'buzz factor' will be relevant. Given these different priorities, the amplitude-phase characteristics are likely to be different for different designers and users, dependent upon their own compromise points in terms of cost, size, weight, sound-pressure level, low-frequency linearity or extension, music, or personal concept of what is achievably 'right'.

What is right is a remarkably contentious issue. In a very extensive set of listening tests in 1989, my colleagues and I were astounded by the variability in the opinions of audio industry professionals who could not agree on which two of a sequence of sounds were most similar. Although on a large sample of listeners, agreement was quite strong, within those samples it was possible to have two eminent listeners entirely disagree on whether or not any two given sounds were similar. The fact is, different individuals simply do not necessarily perceive anything, by whatever senses, in the same way; and from this it follows that even from the same set of unanimously agreed outline specifications, there will be differences in the end product. ►

The VMP™ 2 Two Channel Vacuum Tube Preamp

The ~~Sole~~ Soul Requirement

Digital technology has come a long way. The clean, crisp sound survives from creation to studio reproduction without degradation or signal loss. But in the golden analog days, some of the signal changes picked up through the signal flow actually enhanced the sound, especially on vocals. A sort of warm, fuzzy presence with soul.

Not to worry. . .by request from studio technicians, Peavey Electronics has developed the VMP 2 to add that vintage analog sound. The VMP 2 is dual-channel 12AX7 vacuum-tube microphone preamp with a unique input-transform design and exceptional frequency response. The front panel features a -20 dB input pad, switchable +48 volt phantom power, a 1/4-inch high-Z line input, high and low shelving-type EQ, and an EQ-bypass switch.

The output stage of the VMP 2 is all-tube, as well, and employs a power-amp design to create the +19 dBm output capability which is available on either a fully transformer-balanced XLR connector or a 1/4-inch phone jack. And a custom-designed power supply consisting of a special low-hum field power transformer with magnetic shielding and high-voltage regulator circuitry delivers the juice without power-line transients and voltage variations.

The VMP 2... puts the soul in the signal flow.

PEAVEY®

AUDIO MEDIA RESEARCH™

Peavey Electronics (U.K.) Ltd. • Hatton House, Hunters Rd. • Weldon Ind. Est.
Corby, Northants NN17 5JE United Kingdom • Tel: 0536 205520 • Fax: 0536 269029

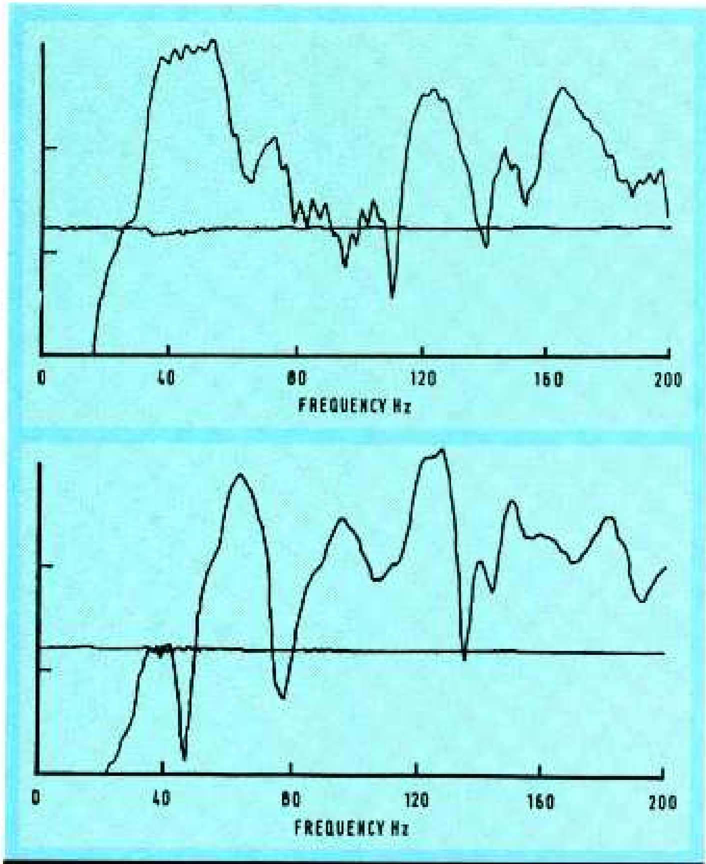


Fig.2b: Low frequency response plots of two identical loudspeakers in two different control rooms. Measurements are taken at engineer's position

manufacturers will tend to go in their own directions. It is, of course, this fact which creates such a diversity of choice of pianos, guitars, drum kits, hi-fi systems and many other things.

Two additional points relating to individual choice may help to further develop this point. I remember one well known and successful studio in London where two of the equally respected and successful engineers swore by their own desired settings of the high frequency drivers above 6kHz, but their individual settings were 3dB apart. As they both considered their individual settings to sound most accurate when compared to an original sound, given that the same signal path was used, then each of their perceptions of the original via loudspeaker sound fields must have been very different, with their own heads and pinnae masking in different ways the different fields of the live and reproduced sounds. Remember, no loudspeaker even vaguely represents an original sound field unless that original sound was itself produced 'synthetically', electronically generated, and first heard via a loudspeaker.

Human senses

Most designers tilt their compromises to their own beliefs and preferences. Their customers are frequently people with similar sets of preferences and priorities to themselves, who make many of their choices based on this empathy. The refinement which a designer seeks for future models follows this path, and the users and designers reinforce each others' opinions so the refinement follows an almost deterministic route. From the point of view of some people, the refinements of different designers should eventually converge, but for a number of factors, this is not necessarily the case.

As stated elsewhere at other times, loudspeakers as we know them are so far from a true reality that we are far too far away from any point of convergence for it to be realisable in current practice. Michael Gerzon, a mathematician and acoustician, one of the original brains behind the concept of Ambisonics and the *Soundfield* microphone, is of the opinion that to truly begin to represent an accurate reproduction of an original sound field, it would take around one million microphones placed around a one metre sphere, each coupled to one track of a one million track recording device, probably requiring a frequency response from many hundreds of kilohertz down to 'weather frequencies' of around four cycles per hour.

The whole recording would then need to be reproduced via one million amplifiers and through one million loudspeakers placed on the surface of a sphere of one metre in diameter, centred on the head of the listener. Even then, if the listener was not present in the original sound field, then his or her introduction into the reproduced sound field would...no, let's just forget it!

Until we approach anything even close to the above, individual designers, clients, and

Comparisons

The hi-fi fraternity are often also well off the beaten track when trying to assess monitors. One technical engineer to whom I recently spoke in a studio, was trying to compare his hi-fi with a type of studio monitoring system which he had not previously heard. His first comment was that his hi-fi 'specials' at home went deeper. Whether they did or not I am not sure, most likely he perceived them to go deeper in his room. When I related this story to a musician friend of his, the friend said to me that the technician really believed in his home system, and on one occasion had brought his loudspeakers to a studio to demonstrate. The musician, intrigued by the expectation of some wonderful sound, asked a recording engineer to start a DAT machine, then went into the studio to begin playing the drums so that he could return to the control room to listen to the recorded drums, via the wonderful loudspeaker system. Once the technician realised what was about to happen, he shouted, 'No! You will destroy them!'. Clearly he was trying to make an unqualified and misleading comparison by criticising a monitor system not by others of its kind, but by judgement alongside something which was entirely unsuitable for monitoring purposes. That which can be achieved on a small scale at low levels cannot always be achieved on a large scale, and what is achievable from a small hi-fi loudspeaker, is not always realisable from a large monitor system. For example, a 24-foot high human being, would be four times normal height, but 4 x 4 x 4 times, or 64 times normal weight. Human bodily systems would not function with such disproportions of surface skin to weight ratios and other mismatches. Such a human could never exist; the ratios are too wrong to be human.

It is little wonder that monitoring is one of the most contentious issues in the recording world, because despite being expected to be the major point of reference for any recording process, they are inherently 'inaccurate' in absolute terms, and disagreement exists widely on precise specifications. If such conditions exist for the 'experts' then little wonder that many studio personnel who have not spent a lifetime dealing with the problems, feel a little insecure at times with their 'elastic tape measures'.

Choice

A couple of years ago, I was telephoned by a man in Yeovil, Somerset, who asked if I could visit his studio because something was wrong with the monitoring. He had bought a pair of medium-sized Genelec monitors which he had liked very much when using them in a studio in London. I went to see his studio and he explained that he had paid a great deal of money for these loudspeakers and was most disappointed in their performance. How could they be so different from the pair in London? The room had little acoustic control and was greatly colouring the sound. The sort of acoustic control which the room required was out of the question because the owner was in the middle of a long-running project, the room had little space to spare, and after the purchase of the Genelecs, money was tight. I moved the loudspeakers about four feet and he suddenly exclaimed, 'That is it! That is the sound I heard in London!' Had it been a room of different shape, size, or construction, the same solution may not have been possible. Obviously, the position of the loudspeakers was such that they were driving some problem room modes; but in a different room, it may merely have been a case of the move driving other, different problem modes, so out of the frying pan and into the fire.

Somewhat similarly, I recently refurbished a small control room in Miraflores where the room had been too live to produce any real clarity, particularly in the mid and high frequency regions, and was of such a shape and size that the best solution seemed to be to reduce the overall reverberation time to very low figures. There was a large window in the room, out of which went much of the low frequencies, so the low frequency build up was not too great. When the work was completed, the owners were delighted by the new clarity and imaging, but now they considered their monitors to be bass heavy. What had happened was that the mid and high-frequency reverberation of the room was now much reduced; it no longer reinforced the overall energy in those frequency bands. The large window still 'lost' the same amount of low-frequency energy as before, so the subjective result was an ▶

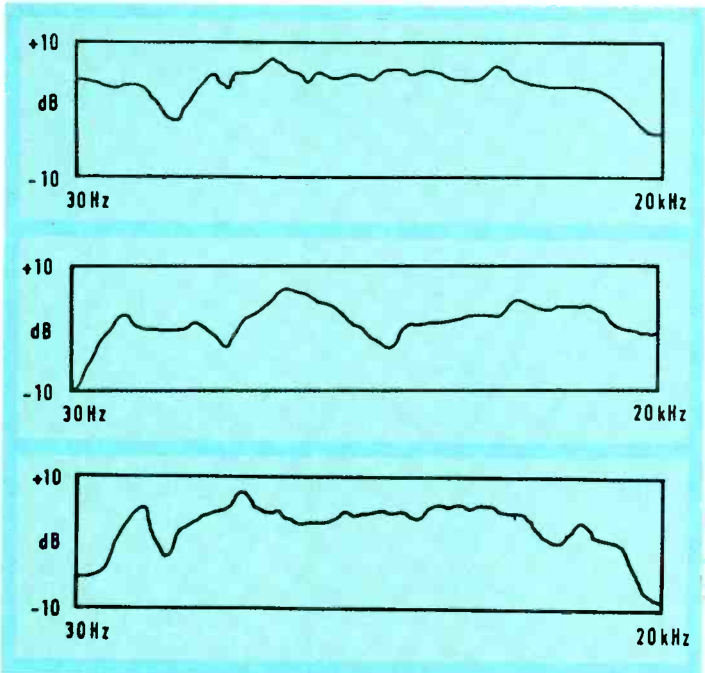


Fig.3: Response of one loudspeaker at three different locations in the same room. Differences are entirely due to room position

overall reduction in the perceived mid and high frequency levels, though what was perceived was heard with much greater definition.

The loudspeakers which they were using were Audix with a bass port on the rear of the cabinet. Such a loudspeaker was no doubt designed for mounting in a relatively free space, but given the very narrow front to back nature of the room described here, less than 3m, inevitably the loudspeakers were forced back, very close to the front wall. As mentioned at the beginning of this discussion, when a loudspeaker is balanced by the manufacturer for a uniform on-axis response when mounted in a relatively free space, it is assumed that some bass energy will be 'lost' behind the loudspeaker. When a wall is placed close behind, this rear radiated energy will be forced forward to produce an excess of low frequencies on-axis. Once again, these people were also lucky, but in this instance, moving the loudspeakers was out of the question.

The purpose of the port on the cabinet was to augment the low frequency response of the loudspeaker as compared to a sealed box of the same size. It turned out that this augmentation was not dissimilar to that produced by the location of the loudspeakers too close to a wall. Given the 'live' state of the room prior to refurbishment, especially in the mid and high-frequency ranges, the 'double boost' provided by the port and the positioning, had helped to provide a reasonably uniform though muddy frequency balance, but when the mid and top were controlled in the refurbishment, the bass became predominant. By sealing the port, the overall balance of the axial energy was restored to something more akin to a lifelike response. Certainly the owners of the studio and the engineers using it considered that they now had an overall sound vastly superior to that before the refurbishing, and found the system very workable. Had the loudspeakers not been of the tuned-port type, but of similar overall frequency balance to the open-port response, such a rectification would not have been possible, and a change in loudspeakers would have been required.

In the first of the two cases mentioned above, Genelec could easily have been accused by the uninformed of the lack of consistency in their production batches, implying that two nominally similar pairs could sound very different. This would patently not be true, and I have personal experience of the excellent consistency of Genelec products. In the second case, the uninformed could equally have made statements to their colleagues to the effect that: 'This room is now very smooth, but the Audix loudspeakers are bass heavy', which would also not necessarily be true. So many loudspeaker manufacturers must receive enormous amounts of entirely unwarranted, uninformed, negative criticism, often as a result of the misapplication of their products, when in reality no blame could be attached to the loudspeakers themselves. To the credit of both the above mentioned studio owners, they had the wit to consult an appropriate person before making rash judgements.

The audio industry is moving apace into the realms of more 'home' recording in untreated and frequently unsuitable rooms. This situation cannot be controlled, so each and every room will be different. Half a dozen such rooms may perform as best they can by choosing different loudspeakers for each room. The smoother rooms may suit smooth, wide directivity loudspeakers; rooms with problematical side-wall reflections may suit loudspeakers with a response more concentrated on axis. A room where the loudspeakers must be placed in corners may suit a loudspeaker which is nominally bass light, to offset the 'room gain' caused by the augmentation of the response by the corners at low frequencies. A larger room, on the other hand, may suit a loudspeaker with a nominally more 'bass heavy' responses, as much of the low-frequency energy will spread far and wide. No one loudspeaker design can suit all of these rooms, and while acoustic control of the rooms may be the 'correct' answer, the real world tends to ask for a different solution.

Around 50 years ago, Gilbert Briggs—founder of Wharfedale Loudspeakers—stated his opinion that the overall response of a loudspeaker should be balanced around the median of the audio frequency range. If the response goes down to 30Hz, then it is probably alright to take the high frequency response to 20kHz, but if the response only goes down to 40Hz or 50Hz, it is possibly better to restrict the high frequency response to 15kHz or so in order to create a perception of a frequency-balanced overall sound. To go all the way out to 20kHz with a low-frequency response restricted to 50Hz would tend to produce a loudspeaker which would be perceived as 'bright' or top heavy, and if used as a monitor, this would probably lead to dull mixes. There is a great amount of wisdom in Briggs' statement, and consequently if somebody possesses a bass-light room, it may also be sensible to chose a loudspeaker which may in other rooms be deemed lacking in top if a balanced overall frequency characteristic was desired as an aid to producing balanced mixes.

Realistic tests

Loudspeaker manufacturers seek gaps in the market for which they produce various systems to fill. There is a vast array of different loudspeakers which are used for monitoring, and where room ▶

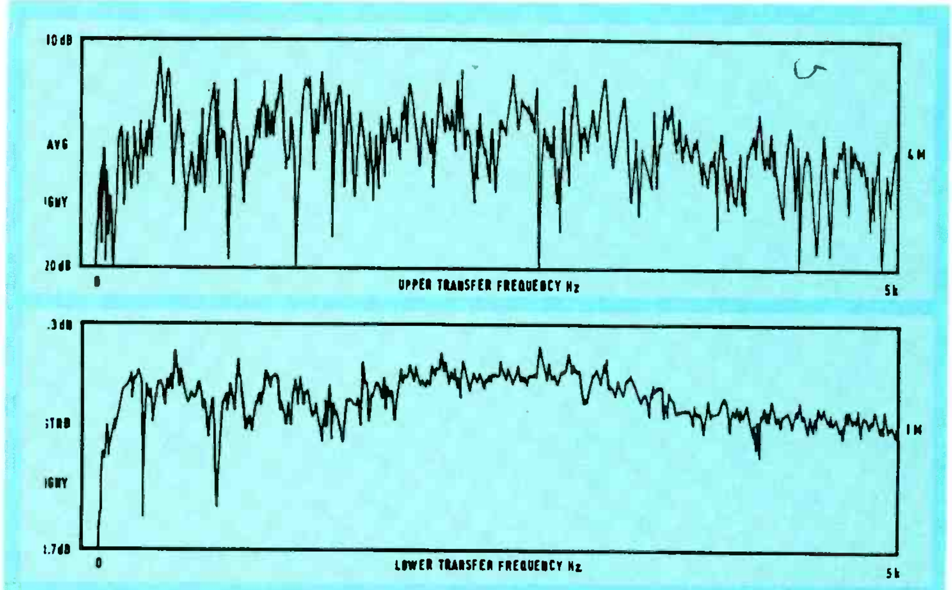


Fig.4a: One loudspeaker at one location in one room. Lower trace measured at a distance of 5ft. Upper trace measured at a distance of 15ft. Note the predominance of the room at low frequencies and the comb filtering in the upper trace caused by room reflection


ONLY £499*
Write On Target
AUTOGRAPH II
PEAVEY
"The Autograph II, a MIDI, 28-band graphic EQ, utilizes built-in RTA, pink noise generator, room sampling, storage for 6 target and 128 venue curves, feedback finder, & more to write perfect EQ curves and hit the bull's-eye every time!"
AUTOGRAPH
COMPUTER CONTROLLED MIDI GRAPHIC EQUALIZER
EQ RTA BYPASS
GLOBAL STORE RECALL
PEAVEY
MADE IN U.S.A.
The Peavey Autograph™II contains all the easy-to-use features necessary for accurately EQing PA systems. The MIDI controllable, 28-band, 1/3 octave graphic EQ features constant Q filters on standard ISO centers. The built-in real-time analyzer and pink noise generator perform RTA using multiple discrete and continuous room sampling. This amazing feature samples a room's frequency response at different locations, and averages them together for a final EQ curve. This final curve matches the system frequency response to one of six different factory or user target response curves stored in the Autograph II. You can then name and store this new curve in any of 128 available memory slots. It's that simple. For best results use the Peavey PVR-1 reference mic (sold separately), or store the curve from another mic and use it for the RTA. Another exciting feature is the Feedback Finder which automatically locates a "hot" frequency and moves the cursor under it...press the down parameter button and the feedback is eliminated. The Curve Compare feature allows you to take an EQ curve, modify it, and then compare it to the original. Additional features: glitchless program changes, electronically balanced input/output, 40 Hz low cut filter, frequency response 5 Hz to 50 kHz, output noise below -94 dBv, ±6/12 dB cut/boost ranges, software security lock, relay/power-off bypass, and MIDI linking of two or more Autograph IIs. All for the incredible price of £499*! Write on target EQ curves every time with the Autograph II...only from an authorized Peavey dealer.
PEAVEY®
*Suggested retail price (U.K. prices only)
Peavey Electronics (UK) Ltd. Hatton House Hunters Road Corby Northants NN17 5JE United Kingdom Tel: 0536 205520 Fax: 0536 69029

control by acoustic means is not feasible, this available choice of systems is an absolute necessity, even without the added complications introduced by differing personal concepts of what is 'right', and all the other factors discussed both here and elsewhere. A listening test comprising numerous loudspeakers and several respected engineers and producers may well seem to be an attractive proposition to make good reading in a magazine, but it may achieve very little in practice. Even if the problems of positioning, driving, and interference from other units could be overcome, the test would still serve little purpose other than to show which loudspeakers suited best not only the room in which they were auditioned, but the very position in that room. Audition in a different room may produce very different results, as could different music, a different set of producers, different power amplifiers, and a whole host of other variables. In **Fig.2** to **Fig.5**, we can see how the response of one loudspeaker in different positions even in one room could be more different than the responses of different loudspeakers sequentially placed in the same position in the same room, showing how the rooms can dominate to an enormous degree.

One dim light at the end of this long, dark tunnel, is provided by the prospect of digital signal processing using adaptive digital filters and modelling delays. By such means, loudspeakers can be driven with a signal containing the inverse of their response errors, and antiphase drivers can be superimposed to neutralise room-time response problems. Before widespread use can be expected, however, more work needs to be done in terms of the audibility of the inevitable pre-echos which are produced, cost needs to be reduced, and even then, a good acoustic starting point is a necessity. All of the correction inputs are superimposed on the musical signal, so the greater the degree of correction required, the greater needs to be the headroom of the monitor system in order to accommodate both the musical and the correction signals simultaneously. Slightly ironically here, the more linear the room-loudspeaker combination, the more suitable it is for correction, but the less it needs it. Furthermore, while greater improvements can be achieved in the designated listening area, at other points outside this area, the response will be degraded to a greater or lesser degree, as compared to the response outside the designated area without digital correction.

There is still, however, no substitute for a good set of acoustics to begin with. Both Tom Hidley and myself have for some time been pursuing a policy of removing the room acoustic to as great a degree as possible as far as the monitor systems are concerned, and each sticking as far as possible to one set of monitors. This would seem to be the only way of ensuring a high degree of room-to-room compatibility while still allowing for differing shapes and sizes. A growing number of professional studios are opting for this concept, and while Hidley's rooms seem to become larger and larger (his control rooms now typically begin in 35ft x 50ft shells), I have been developing concepts of achieving compatible performances in smaller spaces. The correlation between our approaches is, however, very good. But these are unlikely to become a norm in the expanding number of 'home' facilities, so it would seem that for as long as the wide variability in rooms exists, there is a requirement for an equally wide variability in choice of small loudspeakers. As all loudspeakers and all rooms are 'wrong', it is down to individuals to choose which ones are least wrong for them. This should not preclude or discourage progress in the search for a greater degree of 'rightness', but a little more widespread knowledge by the users of the problems which they may encounter could save a lot of grief both to creativity and pocket, and may also help to save manufacturers from undue, uninformed, negative criticism of their probably quite worthy products. Even for these reasons alone, any listening tests intended to find a generally 'best' monitor would be 90% invalid.

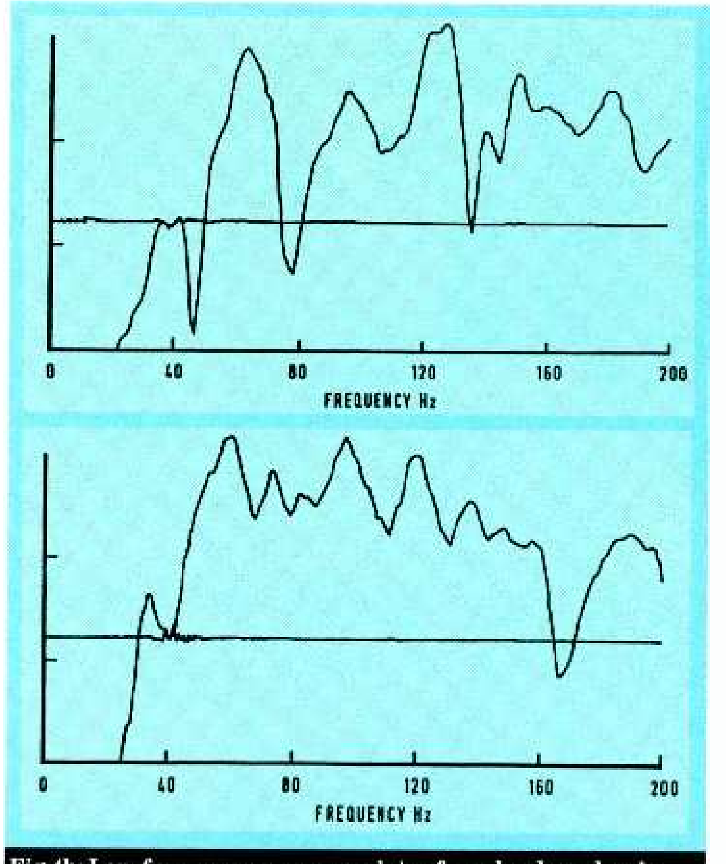


Fig.4b: Low frequency response plots of one loudspeaker in one control room but in two different positions

The sting in the tail

Despite all of the problems in the electromechanical devices, technology is advancing, and as it moves on, some of the seemingly intractable problems will be overcome. The real brick wall that faces us makes the Great Wall of China seem like garden fence. The truly intractable problem is a human one, and we now have excellent evidence of just how great it is. The problem is in human perception, and I fear that to change human beings is way beyond our abilities.

It would to most people seem reasonable to expect two or more human beings to agree in general terms as to when two things sound alike or not. Indeed, when people buy hi-fi magazines to read the equipment reviews, they are expecting this to be largely the case. Doubt soon rears its head, however, in the form of 'favourite' reviewers—the question is are we dealing with matters of taste and preference, or are we dealing with fundamental differences in perception? Two highly controlled experiments carried out at the ISVR in 1989 and 1990 would indicate that actual perception differences play a large part in human assessments of accuracy, neutrality, correctness, or whatever one may wish to call it.

In the previously mentioned listening tests carried out by Keith Holland and myself in 1989 (see *Studio Sound*, March 1994), 16 test sample loudspeakers were individually referenced to four fixed Archetypes labelled A to D. The question was asked for each of nine sounds, 'To which of the four archetypes does the test sample sound most similar?' Let us fix on one individual test of one sound and one sample. As there was no question of preference but simply similarity, we expected the overwhelming majority of listeners to agree. On some sounds and some samples, there was in fact almost uniform agreement, but on others, there was a marked degree of difference from listener to listener. Looking back through my notes, after the first 11 listeners had completed the test on Sound 3, two people thought test sample 14 sounded most similar to Archetype A, one person thought it most similar to B, five, most similar to C, two to D, and one person thought that it did not sound at all like A, B, C, or D.

Let us now take the individual case of the person who thought that Sample 14 on Sound 3 sounded most similar to Archetype B. Let us now also reverse the question: 'Of archetypes A, B, C and D, which one sounds most similar to the test sample, number 14, on Sound 3?'. The answer would, of course, have to be the same, albeit in reverse, that Archetype B sounded most similar to the sample. When we structure the question this way however, we could easily imagine a set of circumstances under which sample 14 was not a test sample, but the real, live, source of a sound. Under those circumstances, we would be asking which of the Archetypes A, B, C or D, was most accurately reproducing the original sound; and while this is never going to be perfect due to microphone imperfections, it would at least be a reasonable test of accuracy. We would then have a likely situation from the results above that two people would think A most accurate, one person B, five C, two D, and one none. Obviously the results above are not an actual test of accuracy, as the sample was not a live sound but a sound carrying the imperfections of the sample, however if eleven experienced people could disagree so widely, then the same could hold true in the hypothetical case stated.

At the other extreme, out of 11 listeners auditioning Sample 6 on Sound 8, all 11 agreed ▶

USED & ENTHUSED

Put yourself in front of a Soundtracs Jade Production console and you know immediately you're in for a treat.

If the clear and concise module layout and big feel don't make your mouth water, turn it on, play some audio and listen.

It's then that you'll discover the sparkling highs, the distortion free lows and the smooth transparency of the patented FdB Parametric Equaliser which has helped make the Jade console 1st choice in so many of the world's most successful recording studios.

The engineers who work in these busy environments enthuse over the exceptional audio quality and time saving features built-in to every Jade.

Features that include the innovative Assignable Dynamics Processor which provides DSP control over gating expansion, compression, limiting and modulation on every channel; and the efficient VCA fader automation available on a combination of up to 64 inputs with mix information stored along with Dynamics settings for speedy recall.

Features that demonstrate the unique understanding Soundtracs have as to the design and manufacture of quality recording consoles for the demanding sessions of today's music industry.

Unparalleled sonic performance with reliability designed in and problems designed out.

The Soundtracs Jade.

JCC Studios Sussex

Ollie J. Roll Over Studios

Steve Mac Skratch Studios,

Stevie V.

LARKING AUDIO LIMITED

For more information please call Larking Audio 0234 772244

SOUNDTRACS PLC

Soundtracs PLC. 91 Ewell Road, Surbiton. Surrey KT6 6AH

081 399 3392

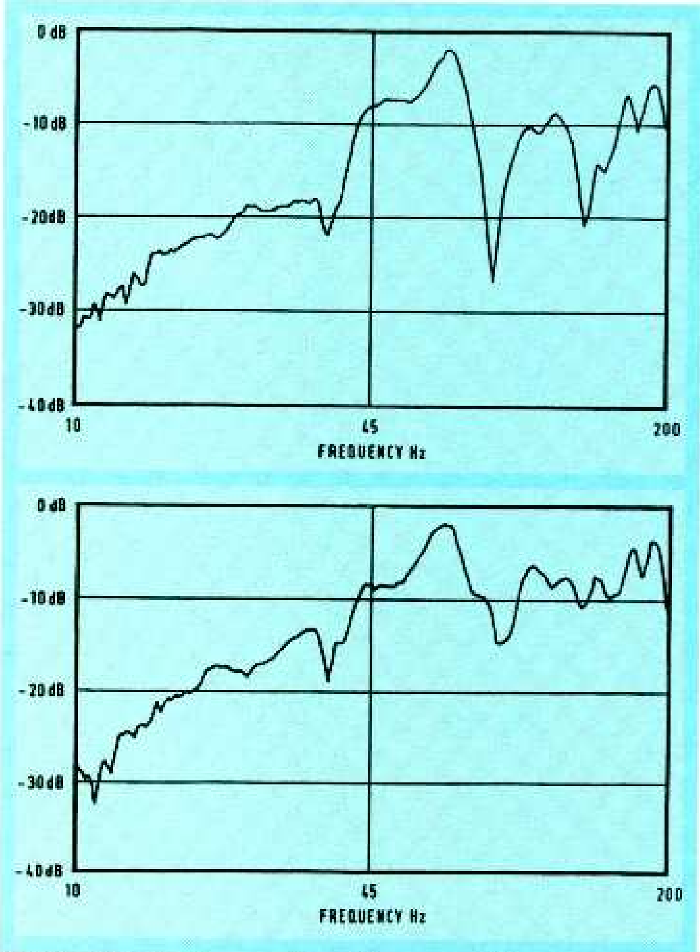


Fig.5: Low frequency response of two very different loudspeaker in the same room. Note the greater similarity than in figures 2, 3 and 4

that it was most similar to Archetype B. What we are therefore faced with is not 11 listeners with totally differing viewpoints or perceptions, but listeners with areas of agreement and areas of difference, so some types of musical sounds may elicit strong agreement on what is 'right', but other musical sounds may fall into areas where no such general agreement exists. In these latter areas, who could possibly say which of the listeners were 'correct'? Certainly any two or more of the listeners who were in general agreement, if talking amongst themselves, would probably be wondering what was wrong with everybody else's ears. This is no doubt what happens when satisfied customers speak to their favourite manufacturers, forming a sort of Mutual Appreciation Society. There is nothing wrong with this, it is a function of human differences and cannot really be avoided, but it does go some way to explaining why certain people have favourite brands of microphones, loudspeakers, amplifiers or whatever. The other aspect of this, of course, is that if a person mainly listens to one type of music, or uses one type of microphone technique, or one type of instrument, there will be a tendency to choose audio products whose characteristics or imperfections are benign or even beneficial on those specific types of music, techniques or instruments, rather than products whose imperfections exacerbate the problem areas.

To further illustrate the inconsistency of human perception, let us consider the results of a test carried out by Dr Andy McKenzie in 1990. My involvement in these tests was a a 'guinea pig'; I was seated in the centre of an array of loudspeakers and asked to switch a rotary switch from position one through to position six, and to say in which position I preferred the sound for each of a selection of musical signals and a series of different test procedures. On numerous occasions I mentioned that I thought the switch was not making contact, but each time I was assured that the switch was OK. The switch box in my hand was not itself switching signal, but was controlling electronic switching in the control cubicle, and I soon became absolutely convinced that there was a problem. I was further reassured that there was no problem and that if we stopped to discuss things now, it would spoil the 'blindness' of the test.

After completing the test, I asked how things were going, to which the experimenter replied that there actually were some problems with the results, but that he would rather not discuss them as there was still some commercial secrecy involved.

The problem was with the 'switch faults' referred to earlier, but they were in fact not switch faults at all. The test involved aspects of Time-Intensity Trading to create spatial effects in wide image stereo, and the switch positions controlled differing amounts of simulated direct and reflected signals, from different directions and with different delays. When I thought there was no change due to a faulty switch, a change was present but I was not hearing it. While the results were highly repeatable, the problem was that effects clearly heard by one listener could be totally undetectable by another. This was not a case of some people being more generally sensitive to the changes than others, but that the areas of insensitivity were highly individual. How, therefore, could a company design a system and market such a system effectively when some people will readily appreciate its assets, but other people would gain no worthwhile sensation whatsoever? In fact, the problems of consistency of results went even deeper, with some people hearing entirely phantom images in positions where other people heard nothing.

That things work for some people and not for others is a fact of life in the perfume or fashion industries, but for something as 'technical' as a loudspeaker, this degree of individual perception difference came as something of a surprise. Differences were expected in terms of preference, as the question asked in each case was, 'Which setting do you prefer?' This was the original reason for the tests but the absolute inability of some people to detect any change whatsoever between some of the settings and others, suggested a new set of test criteria were required. Clearly, the company undertaking these tests prior to the launch of a new product would have to expect very mixed reviews in the press, dependent upon the reviewers own ability to perceive many of the effects. In turn, this could lead to a somewhat hit and miss, 'If it works for you, buy it', marketing philosophy. On reflection, perhaps the only difference between the company in question here, and many others before them, was that this company were aware of the likely perceptual differences of its intended customers, whereas so many other manufacturers have only been aware of preference or ignorance among their targeted market.

In the tests described above, real and highly significant differences were detected in what was heard, even between people with experienced and trained ears. As there are too many variables in loudspeaker 'quality' for any instrumentation readouts to coalesce into a meaningful indication of subjective audible acceptance, where does that leave us? Precisely where we have been for a long time, and precisely where we are likely to stay for as long as any errors exist in loudspeaker performance: different manufacturers and different users will adhere to their own sets of design and performance priorities. Remember, if we do not all perceive the same sensations, we cannot all be expected to prefer the same products.

No commonality

Each of the two tests described dealt with different aspects of auditory perception. In the 1989 test with Dr Keith Holland, the anechoic conditions and relative uniformity of the directions of the different sounds ensured that the differences perceived were in the domains of frequency and phase, with inherent drive unit reflections contributing to the anomalies. No differences in results were apparent if any two drivers were positionally interchanged. In the 1990 tests with Dr. Andy McKenzie, all the loudspeaker units used for the simulated reflections were nominally identical, so all significant differences perceived were due to the timing and positional differences in the simulated reflections.

Loudspeaker system as we know them cannot reproduce an original sound field from an acoustic source. They all therefore produce sound-field distortion, thus when any original sound is reproduced via loudspeakers, sound-field distortion will be an inherent part of the reproduction, varying from unit to unit and room to room. Different people with different pinnae, different heads, different auditory systems, and different brains, will not respond uniformly to these distortions. Consequently, if a live sound is compared to the same sound reproduced via loudspeakers, the degree of difference between the two will almost certainly not be the same for any two listeners; though for identical twins without environmental damage, it may be! As we have already established that many degrees of difference are signal dependent, we must have a situation where the degree of accuracy of any loudspeaker system will be individual to each listener, to his or her choice of music, and to many of the other factors previously mentioned. ■

THE STUFF THAT LEGENDS ARE MADE OF.

Talent. Hard work. A little luck. *And the right equipment.* That's the stuff that legends are made of.

Legendary musician, producer, engineer, arranger and songwriter Alan Parsons knows what makes a legend. So we asked Alan to test the new Audio-Technica AT4050/CM5 multi-pattern, large diaphragm studio capacitor microphone.

"To say that the CM5 is a serious microphone would be an understatement," says Parsons. "It's up there with the very best. It gives me a realistic, warm and true representation in the studio."

"My experience with the CM5 and other mikes in the 40 Series has convinced me that Audio-Technica now ranks as one of the very best manufacturers of high quality microphones. The clincher is that the CM5 offers incredible performance for much less than its competition."

Symmetrical direct-coupled electronics in the AT4050/CM5 provide excellent transient response and low distortion. Featuring three switchable polar patterns (cardioid, omni and figure-of-eight), the CM5 combines warmth and transparency with super-high SPL capability.

Find out for yourself what Alan Parsons has found in the new AT4050/CM5. Call or fax Audio-Technica for more information. U.K.: 0532 771441, Fax 0532 704836. Germany: (069) 80111-12, Fax (069) 812078. U.S.A.: (216) 686-2600 Fax: (216) 686-0719.

audio-technica®

Alan Parsons and the new AT4050/CM5

Alan Parsons will be using Audio-Technica microphones exclusively on his upcoming world tour to promote his latest Arista Records release "Try Anything Once." *(CD 07822-18741-2)*

Martin Polon

Studio hot or studio not?

One of the questions I often field as an audio industry business forecaster is, 'Why are other studios hot while mine is not?' The studio owner in question then explains that his or her studio has a brand new SSL console, four new Sony digital decks and four studios with acoustics by...

The answer to the question is not a simple one relating to one factor or another, but one which reflects a matrix of factors which provide the 'feel' of a studio to the clientele. It is clear that 1994 is not what 1979 was for the recording studio industry. In the US (and this holds essentially true elsewhere), the recording business 15 years ago was an era where failure was almost unachievable—a studio operator would have required a particular 'talent' to fail!

In 1979, the record industry was well on its way to the posting of growth numbers that were not bettered for the another ten years—and doing it with new releases and not back catalogue rereleases. Stereo and surround sound for film and television was just beginning to take hold, and with it came postproduction. There were precious few project studios at that time, and recording studios made their profit on the 80% of their chargeable time that was devoted to the recording of music for release on records for public release. By some measures, there were as many as 15,000 mainstream large-scale recording studio facilities profitable in the US alone.

Nineteen-ninety-four sees (by some of the same measurements) slightly less than 1,000 mainstream large-scale recording studio facilities completely profitable in the US alone. Only 20% of their business, on average, is derived from original recording for musical release. Yet there are in excess of one million personal and project studios extant—of varying size and capability but with some rivalling the larger studios in technical wizardry, if not in total investment.

So, the question again comes to the fore. What traits enhance the attractiveness of a commercial recording studio?

A recent research project identified the following factors that are detailed here:

1. Despite the overwhelming compulsion of an audio 'arms race' that marked the fratricidal studio wars of the mid to late 1980s, a successful studio does not always have to have the newest and most expensive equipment off the designer's drawing board. That certainly remains one of the most attractive options for prospective clients but keeping older equipment is an equally acceptable option 'if'—and this is a big 'if'—the equipment does what the client basically wants (or at least thinks they want).

The point here is that a perfectly acceptable console or tape machine might not have to be replaced every 18 months if the clients are comfortable with the studio's selection of equipment. This brings us back to the one element of studio management that is frequently a mystery to some studio owners. The ability to 'know' one's customers—that is, to really understand their wants in 'your' studio—is paramount to the process of meeting their equipment needs. It should go without saying, if the customers want the very latest equipment—then that is what they must be given.

2. The topic of the conversion of recording studios from analogue to digital technology is never one to be undertaken without lengthy conversation. Firstly, there are those studios which have opted completely for digital (or as much as is possible with today's technology). One of the idiosyncratic problems of digital is that the microphones and loudspeakers in use in any studio, remain analogue devices, coupled to the analogue medium of air and the artist. The second idiosyncrasy is that true digital mixing consoles and other signal processing units dedicated to complete operation in the digital domain exist at certain niches of price and applicability at the very top end and at the lower middle end of the equipment price spectrum. It is very difficult therefore to assemble a digital studio that is totally 'zeros and ones'.

3. If the management of a studio has a strong grasp of the studio's mission and its clientele, then it is relatively easy to configure the studio for that mission. In the case of analogue—or more precisely, noise-reduced analogue—the presence of tools such as complete suites of Dolby SR noise reduction units, vacuum tube [valve] compressors and equalisers, analogue tape machines in all sizes and shapes. There are the several studios that specialise in the '1950s sound,' with the same RCA, Electrovoice, Shure and Turner microphones that were used to record 'Little Richard' Fenniman, 'Fats' Domino, Jerry Lee Lewis and Elvis 40 years ago. And no-one says that an analogue studio cannot use digital devices to edit or record the final musical product.

4. What is heard over and over again from musicians and other studio users is how desirable it is for the 'feeling' of spaciousness both in the actual recording studio environments, in the audio control areas and in the ancillary spaces devoted to nonrecording activities at the studio. This does not mean that there is a universal need for unlimited amounts of real estate, but rather that facility design and utilisation has been the result of collaboration with an architect and-or an interior designer. The best use of available space is made and there is a feeling of spaciousness conveyed to the clients.

> The question again comes to the fore. What traits enhance the attractiveness of a recording studio?

5. Nothing is more important to success in the studio business today than the issue of 'specialisation' or target marketing of, by and for the 'hot' studio. The studio business has, by and large, like the business of medicine (and make no mistake—medicine is a business), gone from being an industry of general practitioners to an era of specialist studios which may focus on postproduction, commercial production, multimedia creation or music recording and so on. The studio business has in many ways become so specialised that (again like to medicine), there has sprung up the curious category of 'specifically general' studios. Needless to say, the success or failure of a particular studio to correctly configure its technology and facilities for a specific marketplace will spell the difference between 'hot' and 'not'.

6. One of the problems that plague the existing studios which have remained successful in most big cities in America, is that the neighbourhoods where they are located may well have changed in more ways than one. The most usual problem is the residential or business 'gentrification' of a neighbourhood housing a studio. The increased business value of a property will reflect itself either as buy-out offers from developers of office skyscrapers or as complaints about noise and after-hours activities by new residents. In either case, such a dynamic complicates the long term perception of the studio by the clients. The clientele will either wonder when the yuppies will succeed in having the city throw out your studio, or when you will sell out to the real estate developers.

7. A difficult studio asset to define is that of ambience. Yet it is an important issue that communicates to a client the 'essence' of the studio. The studio that looks like a garage sale or the window of a used auto parts store does not, as a rule, inspire confidence on the part of musicians, their agents and assorted record company personnel associated with the project. Contrary to the opinion popular with many studio 'tekkies', and upended console with it's guts exposed in the hallway between Studio A and Studio B is not a inducement to most clientele. It suggests failure—like a beached whale.

8. 'Pride goest before a fall,' we are reminded by our friends taken by biblical antecedents. It may well be that 'ego' is really the villain in that sense. To be proud of one's facility for its own sake communicates all of the above factors to the current or potential customer in a way that no amount of money can buy. Having staff who enjoy working at a studio enough to communicate that enthusiasm to the clientele is the other half of the 'pride equation'.

In the research project, each of these factors were identified as being important in retaining existing clients and to prospective studio clients. One problem created by management attitudes at the several studios, is the assumption that existing clients are there to stay. In other words, current clients do not warrant further courting.

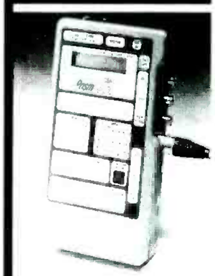
As a footnote to this, in a future column we shall analyse other areas by which studio customers 'rate' a facility and continue the discussion on whether or not current customers are 'locked in'. ■

Does digital audio give you the jitters?

Get the measure of AES/EBU and SPDIF with Prism Sound instrumentation

DSA-1

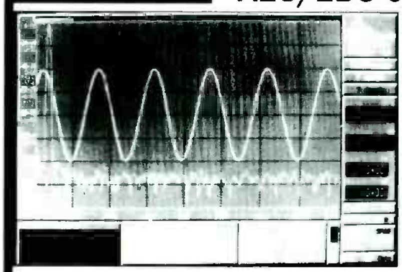
Hand-held Digital Audio (AES/EBU) Interface Analyzer



- Measures jitter and other electrical parameters
- Displays Channel Status
- Monitor loudspeaker and headphone output
- Automatic test programs

DScope

PC-hosted Audio Analyzer for AES/EBU signals

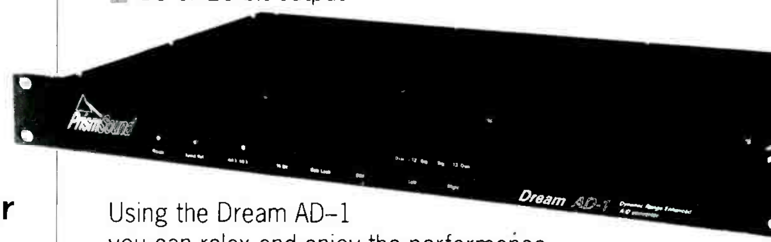


- FFT analysis
- 24 bit signal generation
- Channel Status generation, editing, analysis
- High resolution plotting
- Stores results to disk

Realize your dreams with the Prism Sound AD-1 A/D converter

AD-1

- -115dB dynamic range
- Super Noise Shaping (SNS) or dither
- Dynamic Range Enhancement (DRE) for 16 bit recorders
- 16 or 20 bit output

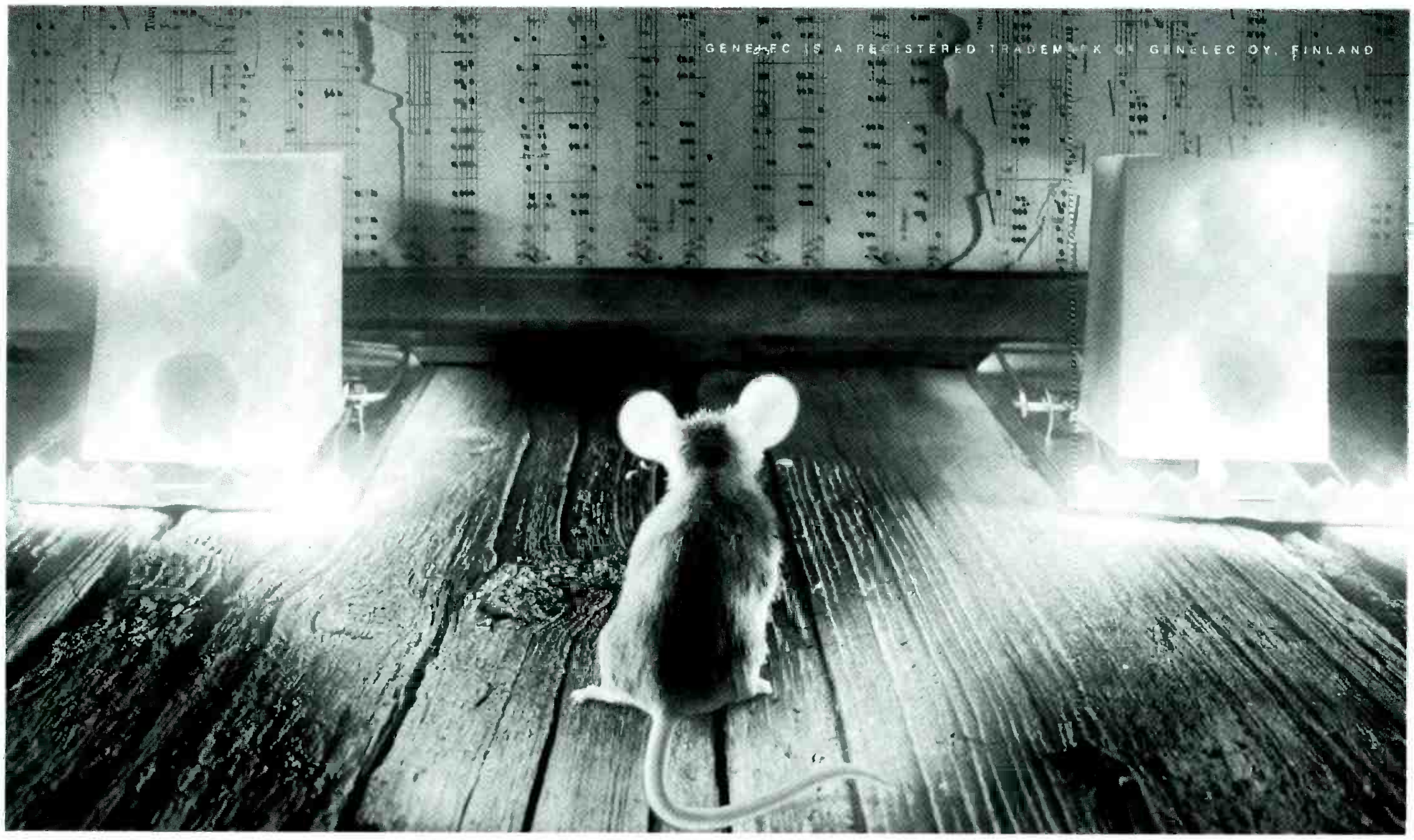


Using the Dream AD-1 you can relax and enjoy the performance. With an extra 24dB of dynamic range yielding a total S/N ratio of -115dB a generous level of headroom can be set without risk of low-level noise or distortion.

Prism Sound

Consultant engineers in DSP · real time control · software

Prism Media Products Ltd.
William James House, Cowley Road, Cambridge CB4 4WX, UK
Telephone +44 (0)223 424988 Fax +44 (0)223 425023



ACTIVE MONITORING

PROJECT AUDIO

PROJECT AUDIO LIMITED
UNIT 1, 321 ESSEX ROAD
LONDON N1 3PS.
TEL: 071-359 0400
FAX: 071-359 3393

GENELEC®

OLVITIE 5
SF-74100 IISALMI, FINLAND
TEL +358-77-13311
FAX +358-77-12267

Kevin Hilton

Licensing new UK frequencies and playing games with the old

It is odd how little things can distract you from the more important issues. For example, I recently called up the Radio Authority, only to get a burst of Virgin 1215 in 'stereo'—one 'channel' from my own radio playing in the background and the other over the phone line from the Authority's press and information office. This threw me so much that I was just about able to ask for the press releases on the recent high jinks that have been going on in the UK radio industry.

That employees of the Radio Authority listen to his radio station is unlikely to console Virgin's Richard Branson, who had it made clear to him that in no way was he going to get the FM frequencies he had campaigned so hard and loudly for. His staff were ungracious in defeat: 'You know when you've been quangoed,' whinged Breakfast Show Presenters Russ Williams and Jonathan Coleman the morning after the Authority announced their decision on the use of the final 3MHz of FM broadcasting spectrum.

It is hard to believe that Virgin could not see what everyone else could, that the Authority would be in a major league firearm-discharge-in-foot situation if they used the 105MHz–108MHz bandwidth for a fourth independent national radio (INR) service or for a second tier of independent local radio (ILR). They were getting enough criticism from some quarters over putting the matter out to consultation at all. The Community Radio Association (CRA) wanted Option C, up to 300 low-power stations, which were promised in the Green Paper of the late 1980s.

What they have got is what CRA Director Steve Buckley called at the time 'the fudge option'—a mixture of new metropolitan services and up to a hundred community stations. Buckley tempered his response after the final announcement, perhaps knowing that they had at least made a start in bringing democracy to the radio industry. Other community radio activists were dismissive of the Authority, saying that they were too concerned with protecting the business interests of established regional groups that own the majority of stations around the country.

'They don't want to let the great unwashed on the air,' said one fervent believer in the right of all to address the nation—or, at least, bits of it. Despite stating that they have to maintain the development of larger-scale radio services, the Authority are initially giving a third of the sub-band (107.0MHz–08.0MHz) to low-powered 'neighbourhood' stations. The lower portion of this spectrum (105MHz–106MHz), will be used for regional or city services offering what is tantalisingly called 'a less populist output' to that of existing ILRs.

It may be a compromise that appeases both big groups and the hugely enthusiastic aspirants, but when the locations for these services are announced in the autumn, UK radio will be seeing the first major change to its make-up since the start of commercial services in 1973. And as experience in both the rest of Europe and the US has shown, it is a very necessary change. Commercial services, or private broadcasters as the mainland Europeans know them, were seen as the great way ahead: more varied services, a chance for new talent on the airwaves, new approaches and an opportunity to make money as well.

Like all good ideas, this never quite happened. The pioneering spirit was there in the beginning but as profit became the thing and some stations grew while others could barely mark time, the industry began to homogenise, up to the situation that we know now. A handful of powerful groups owning several stations around one or more regions, pumping out pretty much the same play-list music, linked by sometimes shared DJs and occasionally interrupted by a stretched news team.

A sign that the radio industry is stagnating is the rotation of some of the presenters. They leave one station, only to be picked up by another a few days later, as happened following Matthew Bannister's much needed weeding out of the BBC's Radio 1. However, his chosen replacements were already well known through TV, newspapers and the larger local stations. It's something that has happened throughout history: whatever is new, exciting and slightly dangerous soon edges towards the establishment as words like 'ratings', 'profit margin' and 'contract' are mentioned.

One of the sadder radio events of this year was the four-week cruise back in time to celebrate the start of Radio Caroline's piratical life. When the station was revived in the mid 1980s, it was redundant and getting it back on air with some of the original jocks and the blessing of the DTI just shows how much a new approach is needed. Okay, some of the new community stations may end up being embarrassing or financial liabilities, maybe even both, but something fresh is needed, both in terms of talent, approach and style.

Since the former IBA's Code of Practice was abandoned, broadcasters have been able to use cheaper (although not necessarily inferior) equipment, enabling smaller operations to get on air. PC-based automation systems have also helped, allowing overnight programming on stations that cannot afford a full staff. It may sound like radio on the cheap, but it depends what you want from it. If all you need are the latest hits and time-checks, then the national and regional pop stations are for you; however, if you want local news, specialist music or issues that relate to your community, whether geographic or ethnic, then small-scale radio could provide that.

> Whatever is new and dangerous soon edges towards establishment

Neighbourhood radio will have its detractors, just as BBC local services do now. Despite the Radio Authority's remit to ensure that the stations can guarantee 'sufficient resources' to sustain the eight-year licence, money will be tight, just as it is with the BBC equivalent. But both are necessary if the radio industry is to grow and remain vital. New talent and ideas have to come from somewhere and comedian Harry Enfield showed us the alternative. Beware of characters like 'Smashy' and 'Nicey'—they exist. Fear them!

More of the same could sum up a sideshow to the current licence action. Last year the news-based London Broadcasting Company (LBC) lost their franchise to broadcast to the capital and is due to be replaced in October by London News Radio (LNR), which is largely managed by ex-LBC executives. Since then LBC have done all they can to carry on, including branding the Radio Authority's decision illegal, applying for the speech based INR3 licence and courting LNR.

While the first of these courses continues in the courts, it may be undermined by the fact that everyone knew what the bidding procedure was from the outset and that LBC's Chairperson at the time was Dame Shirley Porter—an avowed supporter of the then Prime Minister Margaret Thatcher (the mastermind behind the changes to the UK broadcasting industry)—who is also fighting allegations of gerrymandering during her time as Leader of Westminster Council.

Bidding for INR3 seemed a neat way of staying on air and sustaining LBC's costly HQ Crown House, one of the last largely analogue broadcasting centres to be built in the UK. It was all quite promising until the company went into receivership just after submitting their application. However, their cash bid was some way behind that of the eventual winner, Talk Radio UK.

Which left being saved by their replacement. One LBC insider told me that all along most of the staff were pretty confident that LNR would take some proportion of the existing service, although it must have been a worrying time for the employees, something that probably continues for those on short-term contracts.

Then up pops international news agency Reuters, who had been beaten in their bid for the LBC franchise by LNR. On 25th May they announced that they were negotiating to buy-out LNR, a move that the Radio Authority had sanctioned pending the satisfactory completion of a declaration of ownership.

If successful, Reuters could put LNR into either Crown House or their own offices on Grays Inn Road, which they share with Independent Television News (ITN), in which they have an 18% stake. If this happens, any former LBC staff who come along for the ride could find themselves alongside their ex-affiliates Independent Radio News (IRN).

Which just goes to prove two things: if at first you don't succeed, cut out the middle party; and just because there is a change of name, it does not necessarily mean that what is behind it is substantially different to what went before. ■

In art and technology the wildest imagination of the past becomes the reality of today. Sennheiser's 16 channel switchable frequency radio systems bring complete artistic freedom to theatre, stage, and broadcast.

No constraint in performance. No compromise in style. Our transmitters and receivers set new standards in the interaction between computerised mixing and RF technologies. With supreme sound quality and total reliability.

Realistically Sennheiser radio systems are the ultimate choice. Surrealistically of course, they're a fish.

SENNHEISER

Sennheiser UK Ltd, Freepost, Loudwater, High Wycombe, **Bucks HP10 8BR.**
Telephone: 0628 850811. **Fax:** 0628 850958.

SPOTLIGHT ON CMR

Ben Duncan reviews the role of Common Mode Rejection in ensuring that the programme in complex signal chains be kept free from extraneous electrical noises

Common Mode Rejection (CMR) is an equipment and system specification, that describes how well unwanted common mode signals—mainly hum and RF interference—are counteracted. Useful rejection is available once all the equipment in a system makes use of balanced inputs (known technically as 'differential receivers' or 'debalancers'). CMR can be improved and made more rugged when balanced inputs are used in conjunction with balanced outputs ('differential transmitters'), but these are not always essential.

What does CMR achieve?

CMR action prevents the egress and build-up of extraneous hum, buzzes and RFI when analogue signals are conveyed down cables, all the more so in big and-or complex systems. CMR helps make shielding more effective by cancelling the attenuative residue, the bit that any practical shield 'lets through'. The two twisted, parallel signal 'go and return' conductors are essential to ensure this residue is literally coincident and appears 'common mode' —that is, equal to each other in size and polarity on each conductor. A tight enough twist makes the conductors experience interfering fields as if they occupied the same space—which is statistically true when averaged out over a cable's length, provided the signal's frequency is not too high. (In practice this means well above 200MHz with common studio cables.)

In contrast, the wanted, applied signal from both balanced and quasi-balanced sources is distinguished by being no less equal in size but appearing opposite in polarity on each input leg. CMR also makes shielding more effective by freeing it from signal conveyance, enabling it to be connected at one end only, according solely to the dictates of optimum RF suppression and-or individual system practice. Breaking the shield's through connection also prevents (or at least lessens) the build-up of the mesh of loops that causes most hums.

CMR is also required to cancel differences between disparate, physically distant or electrically noisy ground points in a system. **Fig.1** shows the CMV that CMR helps the audio system ignore. Even when connection to mains safety earth is avoided by earth or ground-lifting (ground-lift switch open), or by total isolation (switch open and ground-lift R omitted), considerable capacitance frequently remains, through power transformers and wiring dress.

The one thing CMR cannot do is remove unwanted noises that are already bedded in with the music. It follows that just one piece of equipment with poor CMR and in the wrong place, can determine the hum and RFI level in a complex studio or PA path. Overall, the rejection achieved (which is a ratio, not an absolute amount) is described in minus (-) dB. Often the minus is assumed. A 'CMR of 40dB' simply means 'all extraneous garbage entering this box will be made 100x smaller'.

CMR Measurements CMR (or CMRR) figures expressed in (-) dB are ▶

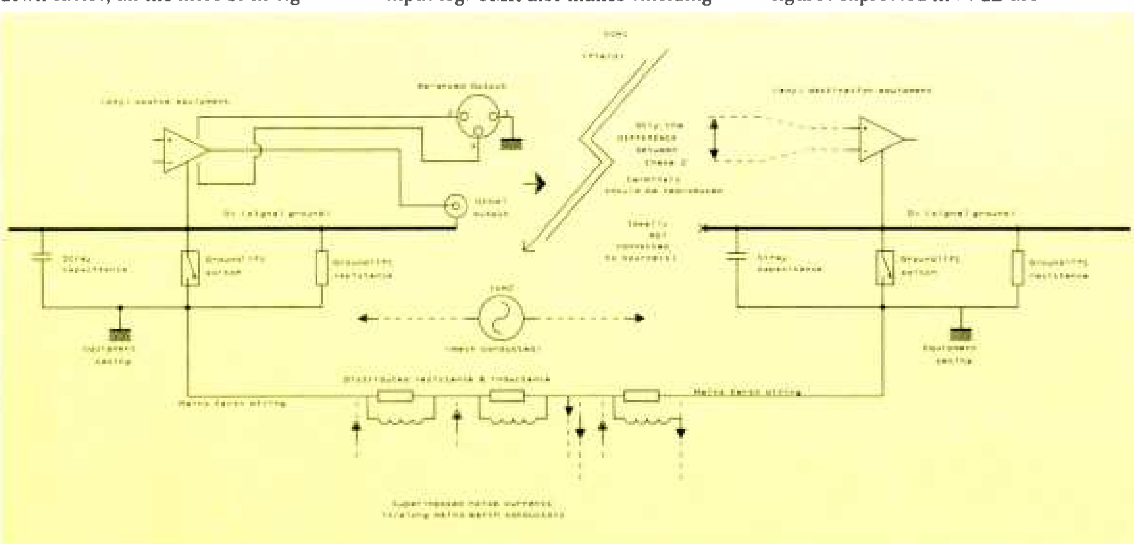


Fig.1: Most of the Common Mode noise that CMR defends against is either RF and 50-60Hz fundamental intercepted in cabling (Vcm1), or 50-60Hz hum plus harmonics caused by magnetic loop, eddy and leakage currents flowing in the safety ground wiring between any two equipment locations (Vcm2)

TERMINOLOGY

CMR: The amount of CM rejection; may be a qualitative statement (high, a lot) or a ratio either vulgar (10,000:1) or referred to 1v (1mV/Volt), or more usually in dB or even dBR. Usually rejection is inferred even when the minus (-) sign is omitted. 'High' in the context of CMR means a big number, albeit negative (-100dB is higher than -80dB).
CMRR: CMR ratio, defined as DMG-CMG, where DMG=(Differential Mode Gain), and CMG=(Common Mode Gain). Also strictly correct when used to describe CMR as a ratio (100:1 or 40dB, for example).
CMV: Common Mode Voltage—the devil itself.
RFI: Radio Frequency Interference.

BALANCING HIGH

CMR and Balancing Common Mode rejection is generally only discussed in the context of balanced connections. The highest, best CMR, better than -90dB, occurs in fully-balanced systems, where both transmitters (that is source outputs) and receivers (destination inputs) are balanced. Systems using balanced inputs alone can give results almost as good, subject to equipment being physically local, and having appropriate grounding and cable connections. ■

somewhat approximate unless you have some clue as to some absolute levels and measurement bandwidth. Why should this be? Because practical CMR measurements include equipment noise. Even if the CMV-to-Noise ratio is favoured by probing inside the equipment to directly read the output of the actual 'debalancer' stage alone, the residue of equipment with a good, high CMR can be seen to be mostly noise across at least part of the spectrum. In effect CMR+N is being measured, where '+N' is noise, and the ratio between the two varies with frequency. CMV residue is favoured over noise at all frequencies by using a high enough drive level when testing. Indeed, CMR testing (CMTST) input signals may be made much larger than normal operating or clipping levels, especially if the rated, differential (regular) input level is low, as with mic inputs having significant gain, or when the CMV rating is high (usual with transformers and a few purely active designs). Even with unity gain, line level equipment, testing can often take place 6dB above the normal input clip level without overloading.

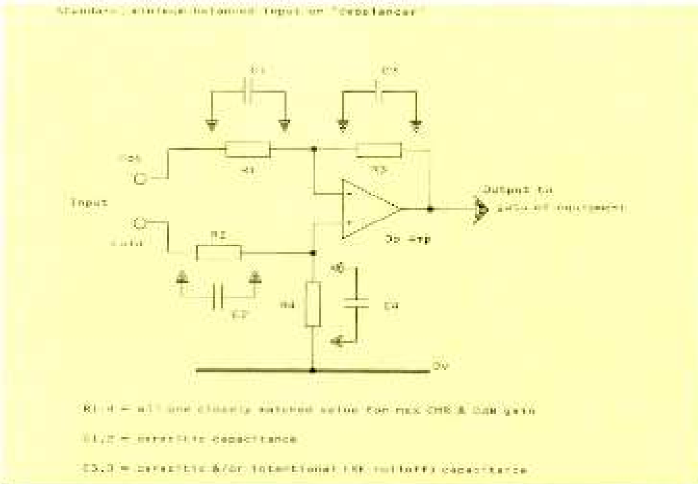


Fig.2: Standard, minimum balanced-input circuit or 'debalancer'. By modelling and varying the values of R1-4 and C1-4 in a realistic fashion, the scope of CMR performance variations across a population can be examined

To be absolutely clear, CMR ideally needs putting into context by using absolute decibels referred (dBr) to a universal test level. In practice +18dBu = 0dBr is one *de facto* standard adopted, being the highest drive level that can be used to sweep up to 200kHz with the industry standard Audio Precision test set. Having elected for a 200kHz sweep, bandwidth is set at 500kHz, although the final detection remains third octave. *Studio Sound* technical reviews contributed by Sam Wise have used the same convention —arrived at independently. With lower or higher reference levels, or with a measurement bandwidth that is narrower or wider, CMR+N can measure 2dB or 10dB or more different, so making fine comparisons dicey. At this stage, equipment manufacturers should take more care to define their CMR specification test conditions. 'CMR = 60dB' is just not good enough in 1994.

Analysing the technology

To look more closely at CMR behaviour, and how far it can be relied upon, **Fig.2** shows the most common, straightforward balanced receiver, for line (not mic) level signals. Although it unbalances the line to which it is connected[1], this has not proven a serious problem in the majority of studio and PA line applications, where the source is has low impedance, is often unbalanced, and usually tied to a single destination. Improvements originally spearheaded by Birt at the BBC[1,2,3] overcome the worst objections while trading some of the simplicity.

A more global problem with this circuit and irrespective of any improvements so far proposed, is that CMR depends absolutely on precise matching between the four resistors (R1–R4) and also the four parasitic (usually rather accidental) capacitances across them (C1–C4). C3 and C4 may include explicit RF filtering-taming capacitors with little or no matching. Surprisingly, most—even the most ancient and lowly—IC op-amps routinely offer potential CMRs better than -100dB to frequencies above audio. But as the **Fig.2** circuit is the mainstay of everything affordable, it is likely that the ultraprecise, 0.005% resistors that need to be fitted to get a decent CMR will be absent. It is also unlikely that CMR will be trimmed with a 25 turn preset to relegate CMR to the sub -90dB realms of which the topology is capable. Instead, in the vast majority of affordable professional equipment, the kind of balanced stage shown in **Fig.2** is built with cheerful 1% tolerance resistors—the kind that end with a brown band. I have to ask: is this colour significant?

In Fig.3, Monte Carlo analysis with *MicroCAP IV* analogue simulation software[4] has been used to demonstrate just how much CMR varies across a population when input stages based on **Fig.2** are manufactured with 1% resistors. The simulation assumes the parasitic (stray) capacitance is a modest 10pF and can vary by 20%, according to lead dress, component type, flux residue, humidity and so on. From the 60 randomly, realistically different units (having values varying linearly within the stated tolerances) plotted in **Fig.3**, it can be seen that the worst case units' CMR is barely better than -35dB. Conversely, one freak is well above average at almost -94dB at 100Hz and below. In some units, capacitance imbalance (between C1–C4) is causing HF rejection degradation to set in as low as 1kHz. This CMR reduction at HF is most evident when the LF CMR is highest, an example of 'peeling back the onion', where one improvement reveals another problem.

Is -40dB enough? If CMR is -40dB (so the residue after rejection is 1/100th of the original), and assuming the hostile CM signal were the same size as the wanted programme, and considering it as noise, then the system's SNR would be no better than 40dB. No good. Fortunately, CM signal levels are mostly smaller than the programme. If we assume they average 40dB smaller (relatively loud hum on a line feed of about -36dBu), then the system's minimum SNR would be 40dB better —80dB. Even this is not too good, considering summation and cumulative occurrence. It follows that the modest CMR in much equipment potentially degrades system SNR unless proper care is taken with cable shielding, placement (away from mains and RF feeds) and correct grounding. You can only get away with spectacular feats like running unshielded (but twisted) mic leads next to dimmer cables with a man-sized CMR. The lowermost curve in **Fig.4** is an example approaching this higher class of CMR (> -80dB) that the more perfectionist professional equipment makers achieve with trimming and-or high precision parts. Here, CMR is almost flat up to 200kHz, a whole decade above audio. While the CMR curve is bound to turn upwards above ▶

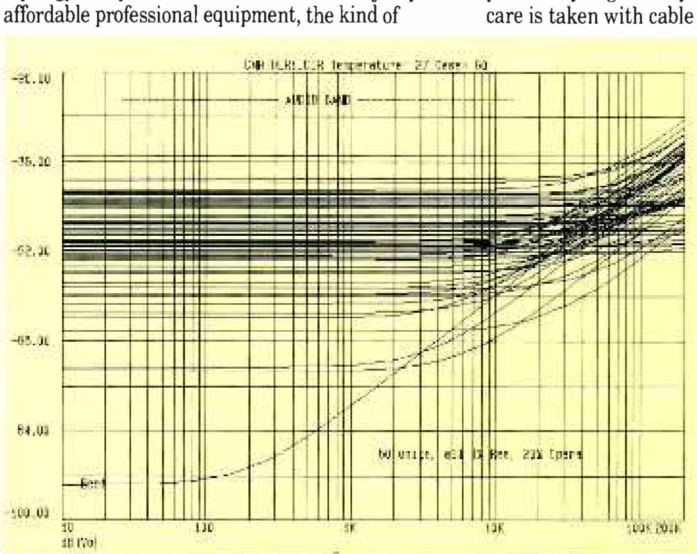


Fig.3: The scope of CMR across a population of 60 units, all with cheerful 1% resistors, showing large, two tier variations at LF (caused by R1–R4) and HF (caused by C1–C4)

DIGITAL AUDIO

Think about fast multitrack editing, go Xtrack audio workstation.

Think about a powerful tool for real-time creativity, immediately operational, without tedious start-up delays, for which everything is fast and easy.
The sound is perfect*, playback is instant. Your manipulations are intuitive, your movements are natural, let your inspiration go free, everything is allowed. Spontaneously, you can forge your sounds on several tracks, make loops, use time-stretching... In the post-production, you edit your audio directly synchronised to the frame. At any moment, you can quickly modify, change or enhance your audio track.

The sound library is immediately accessible and multiple levels of backup let you undo as you like. Production security is total, your mind is free, nothing can stop you... **Think about and go for** an audio workstation easy to install and largely compatible, far from the cost and complexity of dedicated systems.

To discover Xtrack, just return this reply coupon to receive a demonstration cassette or documentation.
Digigram
Parc de Pré Milliet,
38330 Montbonnot France.
Phone: 33/76 52 47 47.
Fax: 33/76 52 18 44.

**Xtrack supports layers 1 and 2 of the ISO/MPEG Audio standard (Musicam).*

Name:................................ Surname:................................
Position:................................
Company:................................
Address:................................
....

Phone:................................
Please send me an Xtrack demo tape.
❑ VHS format ❑ PAL ❑ SECAM ❑ NTSC
Betacam SP format on request
❑ Please send me an Xtrack documentation.

Digigram

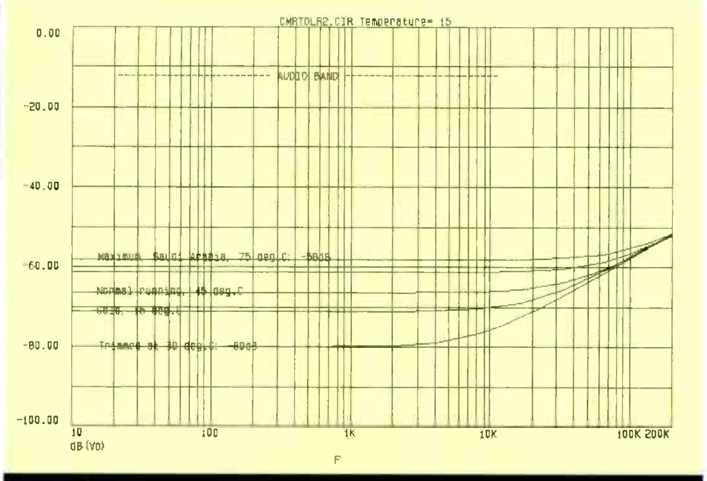


Fig.4: In this case, CMR was finely trimmed before the equipment has fully warmed up (-80dB). It begins at -71dB in the mornings (Cold). After several hours, it degrades to -66dB (Normal running), and would fall as low as -58dB if the gear was taken out to do a job 'like the one we did in Saudi'

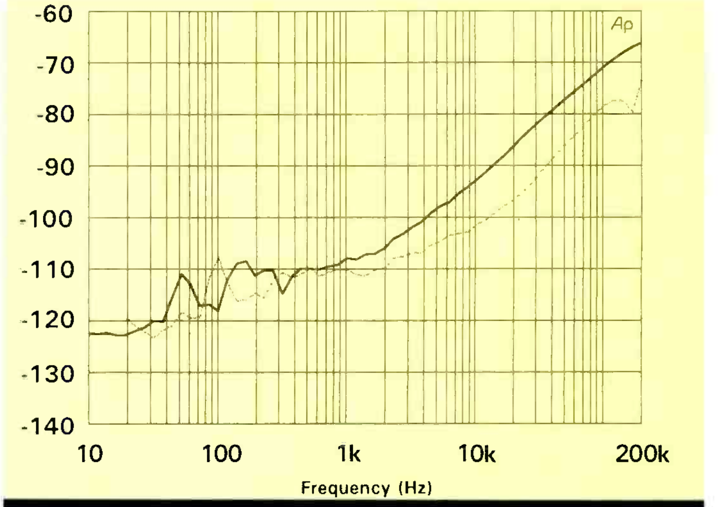


Fig.5: Typical CMR of a high performance mic amplifier showing channel-to-channel variation caused by part tolerances. Bumpy portion at LF indicates noise (not CMV test signals) is being measured

200kHz, the degradation can be cancelled out in both properly designed transformer and active balanced circuitry—and even enhanced—by suitable RF filtering. The main point is that a high CMR corner frequency avoids much of the need to start aggressive filtering immediately above 20kHz.

Refinement

Fig.4 goes on to show how much even the more refined circuitry's CMR can vary, as the equipment (and venue) warms up. Ironically but also fortunately, the less finely tuned equipment with -40dB CMR actually varies far less with temperature. The large variation seen is dependent (at LF) on the temperature coefficient (or 'tempco') of the four resistors, typically 50ppm/°C or less for good quality parts. *MicroCAP IV* can only simulate a global change, while it is any imbalance in temperature change that degrades CMR. Almost inevitably, some resistors get hotter and also get hotter quicker than others, because heat rises in a cabinet and they are higher, for example. *Micro-CAP* partially emulates this effect in **Fig.4** by giving each resistor a different tempco. So, if you are experiencing a residual hum, hiss or radio breakthrough that mysteriously fades in or out when you get round to calling maintenance again, after the gear has warmed up, consider this CMR variation mechanism before dialling for an exorcist.

Just how high can CMR go ? Even if we except temperature effects, the more finely tuned version of **Fig.2** circuit is not a good topology for demonstrating ultimate performance, as the driving equipment's source impedance is assumed nil, but is really finite, interactive and affects CMR more as CMR gets better. Too high a source impedance could detune a -80dB CMR to a mere -40dB, for example. Also, source impedance itself varies with temperature. But with enhanced active topologies which are insensitive to source impedance, CMR at 50–60Hz can be pushed to greater than -140dB, or an attenuation of 10 million-fold. To achieve this meaningfully, requires ultra precision, pre-aged components and very careful layout and assembly; just for once, the IC op-amps (or other active guts) are not usually the limiting factor.

Conclusion

The most salient points about common mode performance are:

The size of common mode ('noise') signals is not fixed or even very predictable; they can range from microvolts to tens of volts. CMR is just a layer of protection.

RF interference is a Common Mode noise, and sources of RF go on increasing. In a competently wired system in premises away from radio transmitters and urban-industrial electrical hash, a modest rejection no better than -40dB has often seemed good enough to make inaudible any induced 50–60Hz hum and harmonics, and the 'glazey' sound of RFI and RF intermodulation artefacts. Unfortunately, RFI artefacts are not blatant, and when mixing, they are the last symptoms you want to be listening out for. Even if there are no blatant noises an inadequate CMR can allow ambient electrical hash to cover up ambient and reverberative detail.

Egress of Common Mode noise is cumulative, as each unit in the chain allows some CM noise leak through. The higher CMR of well-engineered equipment (-80dB or more) provides a safety factor of 100 to over 1000-fold, over the minimum -40dB that is common in more 'cheerful' products. But no amount of high CMR can undo contamination occurring earlier in the chain. So the CMR performance and-or interconnection standards of all the equipment in complex systems (for example, multiroom studios and major live sets) must be either doubly good or highly gated.

In a world where some audio measurements have had their credibility undermined, it is reassuring to know that with CMR, more dB still remains simply better. ■

References

1. Ted Fletcher, 'Balanced or Balanced?', *Studio Sound*, Dec 1981.
2. David Birt, 'Electronically Balanced Analogue Line Interfaces', *Proc.IOA*, Reproduced Sound, 1990.
3. Walt Jung, 'Audio Applications', *Section 8 of System Applications Guide*, Analog Devices, 1993.
4. Micro-CAP analogue electronics SPICE-compatible simulation software is produced by Spectrum Software, 1021 S.Wolfe Rd, Sunnyvale, CA 94086, USA Tel: +1 408 738 4387.

Further Reading

H W Augustadt & W F Kannenberg, 'Longitudinal Noise in Audio Circuits', *Audio Engineering*, 1950, reprinted *J.AES*, July 1968.
Ted Fletcher, 'Balanced or Unbalanced?', *Studio Sound*, Nov 1980.
Manual Huber, 'Conceptual Errors in Microphone Preamplification', *Studio Sound*, April 1993.
Cal Perkins, *Measurement techniques for debugging systems and their interconnection*, 11th AES conference, Oregon, May 1992.

MEASUREMENT TECHNOLOGY

Originally, CMR was a simple, spot measurement. Today's Audio Precision *System One* and similar modern test instruments can routinely plot CMR Vs. frequency, and also overlay curves, so unit-unit variations can be seen (both as demonstrated in **Fig.5**). The System One measures CMR via a third-octave filter, which can be swept up to 200kHz. Compulsory -18dB/octave low-pass filtering can be set to 22kHz, 30kHz, 80kHz or 500kHz, and system EQ or rolloffs may be subtracted, within limits of SNR. Any test set's CMR measuring capability is practically limited by its generator. In the case of the Audio Precision *System One* test set, it is typically -110dB and still -100dB above audio frequencies. ■

Ben Duncan is an international consultant in analogue audio, including power amplification. He has been involved in the design of over 70 products for PA, studio and high-end domestic use. Since his first article in 1976 he has published over 1 million words on electronics and audio quality

MIDEM

Monday, 30 January – Friday, 3 February 1995 • Palais des Festivals, Cannes, France.

IF YOU'VE GOT IT, SHOW IT AT MIDEM

The Premier International Music Market
In this uncertain world one thing is certain. MIDEM is the industry's greatest one-show for professionals only. Nothing comes close to the prestige and epic scale of MIDEM. Which is why your name on a stand at MIDEM '95 will be like no other statement you can make. But whether you participate at MIDEM as an Exhibitor or Visitor just be sure you get there.

The Ultimate Global Meeting Point
MIDEM is where the key people from all sides of music, the movers and shakers, make the deals that define the industry for the year ahead. And where you get the inside track on vital industry issues.

One Stop For The World
A stand at MIDEM puts your company in the spotlight, it says everything about your image and savoir-faire. And it means you can meet your clients in the seclusion of your own private HQ to optimise your five supercharged days in Cannes.

New Lower Cost Tariff
To make sure the cost of visiting and exhibiting is in reach of smaller companies, we've introduced a new lower cost "individual" tariff for MIDEM'95.

The DTI May Help
If you are a UK company wishing to exhibit at MIDEM'95 you may qualify for a DTI subsidy **providing you book by 30 September 1994.**

A Dazzling Setting
The Côte D'Azur, Cannes. Five glittering days. Great artists. Brilliant concerts. A monster media event. What more needs to be said? Cancel the wedding, postpone the vacation, do whatever you have to but get there.
Call Peter Rhodes now on 0171 528 0086 and he'll give you all the details you need about MIDEM'95, including advertising rates for the Showguide and the MIDEM Daily News. We look forward to hearing from you.

REED MIDEM ORGANISATION LTD – Tel: 0171 528 0086 Fax: 0171 895 0949
METROPOLIS HOUSE, 22 PERCY STREET, LONDON W1P 9FF
A MEMBER OF REED EXHIBITION COMPANIES

AMS DMX1580S
2x1.6sec, dual harmoniser and deglitch £1,250.
LEXICON PCM60 Reverb £600
BEL BF20 Stereo analogue flanger £350.
DRAWMER DS201 Dual noise gate £250
2 × DOLBY 361 N/R units
with cat#22 cards £150 each
2 × DBX 160X compressors £200 each
AKG C414EB mic £350.
Prices are exclusive of VAT. All in excellent condition, used in private studio from new and available with original packaging and manuals.
Call John (0737) 764742 (Redhill)

GIGANTIC STUDIO EQUIPMENT SALE
NEVE, TELEFUNKEN, AKG, NEWMANN, ETC
during the
AES in SAN FRANCISCO
NOV.110-13 at COAST RECORDERS
1340 Mission (Bet. 9th and 10th)
SAN FRANCISCO
Fax: (415) 252 7524 Tel: (415) 252 7102

SERVICES

Make it with us . . .
Sound Recording TECHNOLOGY
•D•I•R•E•C•T•
- COMPACT DISCS
- FULL MASTERING
- LATEST 32-BIT DSP
- SUPER BIT MAPPING
- 20 BIT DIGITAL RECORDING STUDIO
- 20 BIT EDITING
- SOUND RESTORATION, DE-CLICK etc
- COPY MASTERS
- DIGITALLY DUPLICATED CASSETTES
- PRINT/REPROGRAPHICS

MARKET LEADERS
☎ 081 446 3218 LONDON
☎ 0480 461880 CAMBRIDGE

REAL RECORDINGS
THE CASSETTE DUPLICATING SPECIALISTS
Real time & high speed loop bin duplication, printing & packaging. Blanks wound to length
TEL: 061-973 1884

RENTAL

[EUROPE AUDIO RENT]
the no. 1 pro audio rent on the continent
We rent out analog and digital multitracks (4-8-16-24 tracks), consoles, mics and all modern outboard equipment. Also samplers, soundmodules, DAT (with timecode), U-matic, synchronizers.
New: Sony 3324S - ring for our competitive prices
PHONE HOLLAND (31) 3465.70670 • OR FAX (31) 3465.72707

SERVICES

BABEL
NEW & USED EQUIPMENT SALES WORLDWIDE

Otari MTR90 MkII	**Lexicon** 224xl
remote & locator	**Lexicon** PCM70
MCI 2 inch 24 track	**Eventide** H3000 harmonizer
Fostex G24sS	**Roland** RSS system
Studer A800 MkIII	Maasenburg GML eq
Roland DM80 4 track system	**AMS** 1580S
Roland DM80 8 track system	**Fostex** 4050 auto locator
Roland DM80S Mac software	**TAC** matchless 32 frame
Akai ADAM digital recorder	**AMEK** Mozart 56 frame,
Alesis adat 8 trackdigital	supertrue automation
Alesis BRC remote	**SSL** 4000e 56 frame 48 mono
Lexicon LARC	channels Producer desk
(for 480L or 224)	**Soundtracs** Jade 40 frame
Lexicon 480L	36 inputs

Tel: +44(0)81 749 8222 Fax: +44(0)81 749 8767
7 Goldhawk Mews, London W12 8PA

CASSETTE DUPLICATION. Superior to Digital. Simon Stable Promotions. Lyrec 32x's loopbin. On-body. Shrink-wrapping. Blanks wound to length. Sample tape available. Tel: 0869 252831.

SOUND MAESTRO HARD DISC EDITING SYSTEM, complete, under a year old, £3,750. 0379 854458.

CRYSTAL! **WHY PAY MORE FOR DATS?**
ALL DATS ARE NOT THE SAME - SOME COST TWICE AS MUCH!
DAT STORAGE RACK-£3.50+VAT!
• HOLDS 10 DATS • LOCKS TOGETHER•
•FREE STANDING OR WALL MOUNTED•
COMPACT 1U PATCHBAYS
ONLY 35.99+VAT!
32 WAY JACK/JACK
24 WAY JACK / 8 WAY MIDI
EASILY REVERSIBLE NORMALISING
DATR30 - FROM £2.50+VAT!
DATR46 - FROM £2.69+VAT!
DATR61 - FROM £2.90+VAT!
DATR92 - FROM £3.39+VAT!
DATR122 - FROM £3.81+VAT!
TEL/FAX 0223 208937
CREDIT CARDS ACCEPTED
NORTH ROAD WENDY ROYSTON HERTS SG8 0AB

jbs records
MUSIC and SPEECH
REAL-TIME/HIGHER-SPEED Quality Cassette Duplication and Blanks from 1-1000. Computer printed Labels
Solo, ¼" reel, Sony Betamax or R-DAT recording.
Fast Security Delivery service.
FILTERBOND LTD, jbs records div, FREEPOST
19 SADLERS WAY, HERTFORD, SG14 2BR 0992-500101

FAIRLIGHT SERIES THREES
Sold, repaired, serviced, hired. Stock constantly changing, please phone or fax for list.
Tel: +44 (0)71-700-1852. Fax: +44 (0)71-607-1410
23a Benwell Road, London N7 7BL
HORIZONTAL PRODUCTIONS

CD AND CASSETTE DUPLICATION. Also Digital and Analogue Editing and Mastering. Mail order is our speciality. **Tam Studio,** Odense. Phone (010)45 6615 0430 or Fax (010)45 6615 0887.

SITS VAC

PRODUCT SPECIALIST
BURNLEY

AMS Neve has an international reputation for pioneering innovative ideas in audio technology. In recent years the company has developed products including automated digital mixing consoles, hard disc based digital editors/recorders, and digital audio processors.

Working within the Sales and Marketing function, you will be providing pre-sales operational support, demonstating equipment to potential customers and aiding the flow of product information between the market place and the company.

You will need to have a good understanding of the impact of hard disc editors on post production methods, excellent communication skills, commercial awareness and a willingness to travel extensively.

A competitive benefits package is available for the right person.

Please telephone for an application form or apply with details of your experience to:

Tony Smith - Personnel and Administration Manager
AMS Neve plc
Billington Road, Burnley, Lancs BB11 5ES
Telephone 0282 417207

A SPECIALIST WITH EXELLENT QUALIFICATIONS

Experienced Microphone Engineer

My name is HANS ROSANDER and I have years of experience building and developing microphones firstly at the Pearl Microphone company and later as founder and owner of the MILAB microphone company.

Now is the time for a new challenge. With positive feedback from you in myself, skills and products we will create the very best product for your demanding market.

Relocation is not a problem.

Please answer to: HANS ROSANDER - Box 1039 - S 262 21 Engelholm - Sweden-or to Fax +46 431 18121

Buying and Selling quality professional equipment throughout Europe

nick ryan ☎ +44 (0)892 861099 Fax. +44 (0)892 863485

Sounds incorporated

'A'UDIO TOYSHOP

PHONE
+44 (0) 272 467711

FAX
+44 (0) 272 730503

Used Equipment List

Amek 2520 40 frame fitted 36, Mastermix automation £26,000
Neve VR 60 fully fitted, flying faders, 2 available ..... £call
Neve V1 48/48 updated, submixer, excellent £55,000
Neve 8068 custom, call for details .... £call
Neve 8108 56 inputs, 48 groups bar-graphs, in line £call
DDA AMR 24 28/24 with optifile automation, excellent condition £17,500
SSL SL 683E programmable eq, pan module £3,500
SSL 6000 Stereo modules 1986 vintage, quantity available £1,600
Midas 36/8/2 TR console PRO §A+, 3 psu, ext p/b, 8×8 matrix, immac £4,500
Studer A 800 rem/loc call for details .... £call
Mitsubishi X-850 Remote, meter bridge 2000 hours £17,500
Mackie 8 buss 24/8/2 as new, warranty £1,700
Studer A 80 ¼", good condition £575
Dolby SP 24 A Fully loaded, immaculate £1,250
ATC SCM 100 A Active versions, finished in black £2,500

Westlake BBSM 12 excellent 2×12, 3 way monitors £1,500
Klark Teknik DN 27 Single 27 band graphic £272
Klark Teknik DN 300 single 30 band graphic £350
Yamaha REV 1 with outrageous remote!, superb condition £1,350
Yamaha REV 5 nice condition £550
Yamaha REV 7 .... £450
Eventide H 949 classic harmonizer £295
Eventide H 910 classic harmonizer £275
Amek/Neve 9098 Eq in stock, on demo and sounding lovely .... £call
EMT 240 Gold foil with remote very nice £625
Neve/BBC compressors type am6/17 4 avail ..... £275 ea
Lexicon 224 white remote £1,350
Lexicon PCM 70 immaculate £1,200
Yamaha D 1500digital delay £250
Aphex Compeller dual channel compressor £450
Audio & Design Compex 2 quality compressor/expander £350

DBX 160 RM the originals, pair rack mounted £675
DBX 162 stereo version of the above £550
DBX 164 over easy, stereo compressor £175
Neumann U 48 valve, excellent, original, as U-47, but uni/fig 8 .... £2,850
Neumann U 67 valve, very good condition £1,250
Neumann U 89 immaculate, boxed £700
Neumann SM 69 stereo, no pattern box, MINT £950
AKG C 12A valve version of 414 .... £675
AKG C 414 ULS MINT condition, boxed £550
AKG C 28A valve version, not the nuvista £350
Quad 521 BBC spec 520, 2 available .... **£425 ea.**
Drawmmer DS 201 Dual gate 2 available £225
Studer A 725 rack mount, pro CD player, XLR's £375
Valley International Keepex 2 10, in powered rack, black fronts £1,550
Valley People Gatex quad gate, excellent £325
Audio Kinetlcs Pacer 2 machine synchroniser with Pacer Pad £1,300

WE WANT: Studer A 800, 827; Lexicon 480L, 224XL, PCM 70; Fairchild, teletronix compressors, all valve mics and anything you have in the studio closet.

WE P/X, BUY, SELL AND BROKER ALL STUDIO EQUIPMENT AND COMPLETE STUDIOS THROUGHOUT THE WORLD, SPECIALISING IN ESOTERIC AND TUBE EQUIPMENT – CALL US WITH YOUR REQUIREMENTS.

OTHER SERVICES: Studio design, installation and servicing. Custom modifications, racks for Neve eq's, compressors and mic amps; phone for client list and details. New equipment supplied, please phone for a competitive quote on any new equipment, packages tailored to your requirements. Clumsy Columbian footballers assassinated.

TONY LARKING

PROFESSIONAL SALES LIMITED

WORLDWIDE DELIVERY

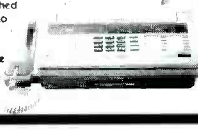
ENGLAND'S LARGEST STOCKIST OF
NEW & USED
PRO-AUDIO EQUIPMENT

CALL NOW! TEL: 0462 490600 FAX: 0462 490700

DIAL A FAX

FOR A COMPLETE UP TO DATE EQUIPMENT LIST ON YOUR FAX

Dial 0336 413 733 on your fax machine & press start when instructed. Some machines may need to be switched to polling mode to use this service. Calls @ 39p per minute cheap rate and 49p per minute all other times.



USED GEAR WANTED

CONSOLES . 16 TRACK . 24 TRACK . ADAT . OUTBOARD MONITORS . STUDIO MICROPHONES . PRO VIDEO

Why continue on with your old equipment when Tony Larking Professional Sales can give you a cracking good part exchange deal - or purchase your unwanted equipment for cash?

PLUS: FINANCE AVAILABLE - USE YOUR EXISTING EQUIPMENT AS A DEPOSIT!

ILLUSTRATION: CARL FLINT

Barry Fox

Digital audio meets visual comedy and the British launch 'Controversy FM' radio

There is a nice background to the recent news that DAR are supplying *SoundStation Delta* workstations to the French studio Son Pour Son. The *Delta*s will be used to lift the soundtrack from an old black and white movie and synchronise it with a new colour version. The new material is not artificially 'colourised', instead it has been recreated from original film shot 50 years ago but never released. And thereby hangs the tale.

Son Pour Son was founded by Sophie Tattischef, who is the daughter of French film-maker Jacques Tati. And the film is Tati's first full length feature, *Jour de Fete*, which he made in 1947.

A few years before he died, Tati gave a memorable lecture at the National Film Theatre in London. He let slip a fascinating fact: his early films were made in two versions, one black and white, and one colour. The colour system relied on complicated optics and three colour filters which did not work well enough to use in cinemas, so only the black and white versions were released. *Jour de Fete* was later 'tricked up' with a little spot colour (so that the back light on a bike was 'painted' red, for example) but there was never a full-colour version. The colour equipment was lost, and Tati died without ever seeing his films in colour.

Son Pour Son are now using digital video technology with computer software that can mimic the original colour hardware. The plan us to have *Jour de Fete* ready by 1995, for celebrations to mark the centenary of film.

What interests me, though, is what happens next. At his NFT lecture Tati told how sad he was never to save seen his next film, *Monsieur Hulot's Holiday*, which he made in 1952, in colour. It contained a complicated visual joke that relied entirely on colour.

As the holiday-makers arrive at the seaside resort, newcomers look white, those who have been there a few days look bright red, and those who are just ready to go home look a golden brown. In long shots, the beach takes on a carefully choreographed colour pattern which is completely lost in the black and white version of the film so far seen.

The Radio Authority have now given their decision on what to do with the last available chunk of VHF FM radio spectrum, right at the top end of the band (105MHz–108MHz). This follows a discussion document which the Authority put out earlier this year. Although the decision has been interpreted as 'one in the eye' for Richard Branson, there is a lot more to it.

In any case, save your tears and sympathy. Having failed to win any of the ITV franchises on offer, Virgin applied for a franchise to run a rock station on the medium wave band. They won this, and encountered all the difficulties anyone with any experience of AM broadcasting should have predicted. None of which gives Richard Branson any 'special' right to an FM frequency allocation.

Whereas the full FM band, from 87.6MHz to 108MHz, has been available for entertainment broadcasters in much of Europe, large chunks have historically been used in Britain by mobile radio operators such as the police, ambulance, fire, gas and water devices.

The British government pushed the BBC into ending the wasteful practice of simulcasting the same programme on both VHF and medium wave. Radio 3 gave up its MW frequency in February 1992 and this is the 1215kHz slot that went to Virgin, Radio 1 quits the medium wave band at the end of 1994 and its allocated slots (1053kHz, 1089kHz and 1107kHz) have been won by Talk Radio UK, to run an international news and talk station.

The BBC continue to own the slice of the FM band that runs from close to the bottom (88MHz) up to 94.6MHz. This currently houses Radio 2, Radio 3 and Radio 4.

Some of the local BBC stations use the band from 94.6MHz–96.1MHz, and some of the local commercial stations (like LBC and BRMB) use the band 96.1MHz–97.6MHz.

Because it came later, BBC Radio 1 FM is oddly placed—away from the other BBC national stations, in the band 97.6MHz–99.8MHz. The only national commercial FM radio station, Classic FM, has the 99.9MHz–101.9MHz pitch.

The gas, electricity and water mobiles move out of the top slot (105MHz–108MHz) at the end of this year. This is what Virgin had their eyes on. But the Radio Authority proposed four options and threw them open to public debate.

The new band could be used to create a fourth national commercial station. This is what Virgin wanted, although there was no guarantee that Virgin would win the franchise. Alternatively, it could be used to provide one new local station in each area already served by an independent local station. Or it could be used for a rash of many new, very local stations.

What the Authority chose was the fourth option, a mix of the second and third, with very local stations operating at very low power at the top of the band where there is greatest risk of interference with aircraft systems (which operate just above the FM band).

This leaves Virgin with no FM frequency. My bet is that Branson will now target the frequencies used by the BBC's local radio stations. If so, the BBC will have nobody to blame except themselves.

I have previously written about the plight of the BBC's local stations currently being starved of the budget needed to compete with ILR stations. In an attempt to open the issue up, I wrote to *Ariel*—the BBC's own newspaper. A lot of people read *Ariel*, if only because the BBC long since gave up buying 'real' newspapers for the consumption of their many visitors who have to wait for the understaffed reception desk to catch terrorists by asking them to sign their names in the visitors' book.

Ariel is not a rag—it costs £50 a year for a UK subscription and £60 for an overseas subscription—but is still known as '*Pravda*' to the BBC staff. So few people take it seriously that the Board of Managers decreed that the paper should win back some credibility by carrying at least a few critical points of view.

I wrote suggesting that, if the BBC's accountants are forcing stations to rely solely on 'puffer' guests (who have a book or product to plug), there cannot be much of a long-term future for BBC local radio.

It took *Ariel* three months and numerous polite reminders to decide that the point 'is not really an issue for the staff newspaper'.

But it will when Richard Branson starts lobbying the RA for reallocation of the BBC's slots 94.6MHz–96.1MHz and 103.5MHz–105MHz. ■

Better Gate Finally Found, Aphex Forced to Pay $10K!

APHEX SYSTEMS 11068 Randall St. Sun Valley, CA 91352 (818) 767-2929

1001

10/09/93

PAY TO THE ORDER OF Aphex Systems Ltd. $ 10,000.00

*****TEN THOUSAND AND 00/00***** DOLLARS

Bank of Credit & Commerce Inc.
DEFUNCT IN USA
ACCT: QATAR

MEMO A BETTER GATE THAN THE MODEL 612

⑈001001⑈ ⑆123456780⑆ 901⑈23456⑈

In 1987 Aphex offered to pay $2,500 to anyone who could find a better gate than their Model 612. When no challengers appeared, Aphex upped the offer to $10,000! Still, no one met the challenge! The Aphex Model 612 Expander/Gate became the industry reference standard in studios and in sound reinforcement around the world. Until now.

Introducing the all new **Model 622 *Logic Assisted Expander/Gate***™.

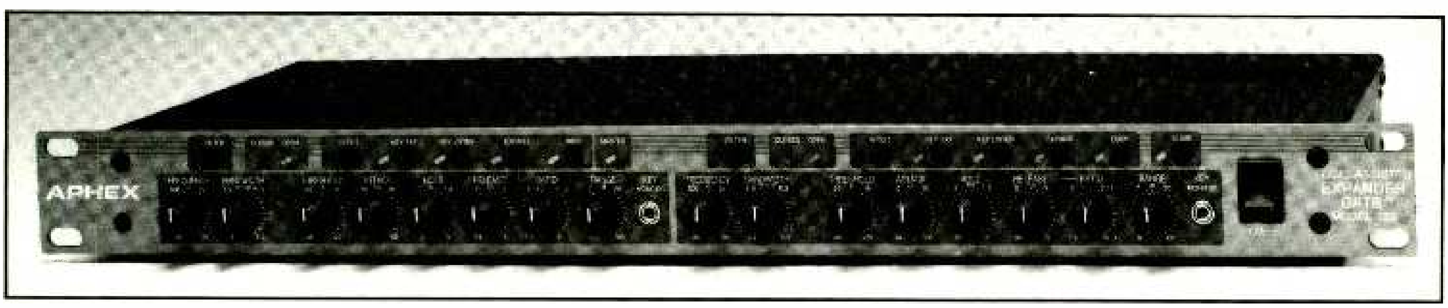
The Logic Assisted Gate circuitry of the Model 622 assures positive, accurate and reliable triggering, patents pending.

The ultra pure audio path features the exclusive Aphex VCA 1001 offering unmatched speed without clicks. Combined with enhanced servo-balanced I/O, the Model 622 boasts a dynamic range in excess of 20-bit digital!

Other special features include:

- Key input monitor via front panel headphone jacks
- 24dB/octave parametric Key input filters
- Dedicated Expander mode
- Relay bypass with remote control
- Switchable ducking mode
- 5-Year Limited Warranty
- Made in the U.S.A.

Listen to the Model 622 Logic Assisted Expander/Gate at your Aphex dealer today. You will agree it was worth every dollar Aphex had to pay ... itself.



APHEX SYSTEMS ***Improving the way the world sounds***℠

11068 Randall Street, Sun Valley, CA 91352 U.S.A ▮ (818) 767-2929
100% owned, engineered and manufactured in the U.S.A.