Two Channel Sound in Sync.

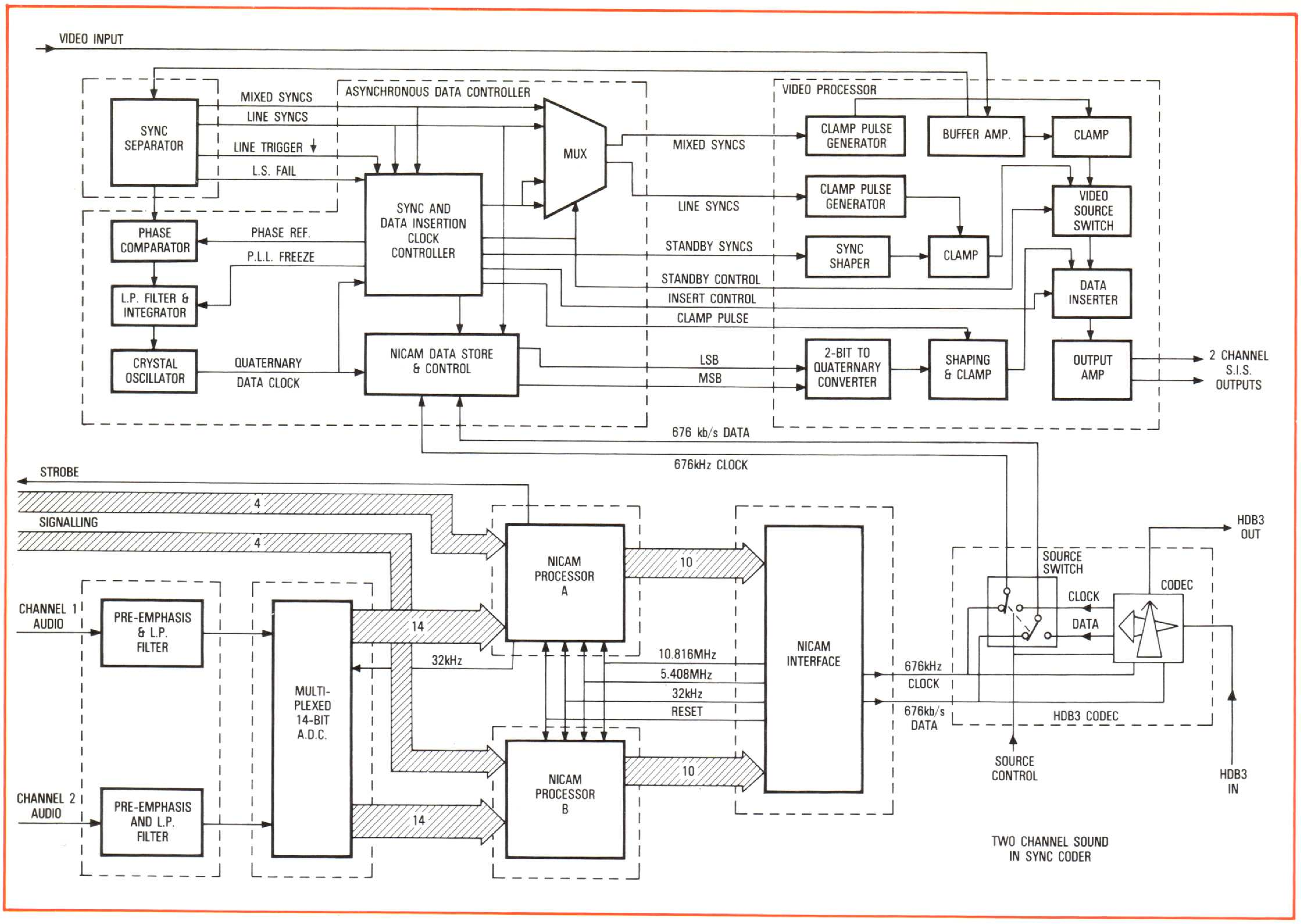
Coder LDM 1901
Decoder LDM 1902

FEATURES:

- **Video and 2 broadcast quality 15kHz audio channels on a single video link.**
- **2 channel or stereo operation — >85dB channel isolation allows use of second channel for second language, or commentary circuits.**
- **NICAM-3. 14 bit to 10 bit digital companding for digital audio quality.**
- **High tolerance to link distortions.**
- **Compatible with satellite transmission circuits.**

varian

Varian TVT Limited



GENERAL

The concept of Sound-in-Syncs has been in use in Europe, and other parts of the World, for monophonic programme transmissions for more than a decade. It has enabled the transmission of combined sound and vision programme on a single video only link, by inserting the sound information in the picture line synchronising periods, thus eliminating the need for additional high quality audio circuits, and guaranteeing the reception of sound with vision at the end of the transmission path. The latest development, based on the design by the British Broadcasting Corporation, extends this same concept to two audio channels, thus affording full 15kHz broadcast quality audio for dual language or stereophonic programmes or any other user requirement.

CONSTRUCTION

The LDM1901/1902 two-channel coder and decoder respectively each comprise a number of standard Eurocard modules in a 480mm (19inch) frame assembly which is 133mm (3RU) high and 381mm deep. The frame assembly houses the power supply transformer and rectifier, and provides, via a mother board, the interface between all modules. All connections to and from the unit are made at the rear, and a hinged front panel allows access to the modules for maintenance. The units are completely self-contained and are designed to be completely compatible "plug-in" replacements for the existing LDM1801/1803 single-channel requirements.

AUDIO PROCESSING

Each audio channel is band-limited to 15kHz, pre-emphasised and sampled at 32kHz. The sampling is carried out sequentially by a dual-channel sample-and-hold module with built-in multiplexer and automatic gain-ranging amplifier, having gains of x1 or x2. The samples are digitized in sign plus magnitude form, the 12 magnitude bits being positioned in a 14-bit word according to the gain range of the sample-and-hold circuit.

AUDIO COMPRESSION

The digitized samples for the two channels are now dealt with separately in a process of near instantaneous compression (NICAM 3) from 14 to 10 bits, in which each set of 32 audio samples is coded into one of five ranges, according to the greatest magnitude found within the 32 samples. For each sample the sign bit is retained; then the leading zeros (up to four) as determined by the greatest magnitude sample are omitted from the word; the nine bits following this are retained for each of these 32 samples. The range code produced for each set of 32 samples informs the decoder of the number of zeros to be inserted after the sign bit. Three range codes are combined to produce one 11-bit word including four Hamming code bits for error correction in the decoder. Sample words are protected by parity bits which enable error concealment by interpolation to be performed in the decoder. With the addition of a framing sequence every 3ms, the data is serialised, taking bits alternately from the two audio channels, and yielding a bitstream at 676kbits/s. The compression process in the coder is done using a custom gate array controlled by Z80 microprocessors. In addition to the normal two-channel audio facilities, each channel is provided with a data signalling input of 4bits/frame at a rate of 1.3kbits/s per channel. The unit can also accept 676kbits/s HDB3 data as an alternative to two-channel base-band audio: or one channel can be used for data transmission (with data processed for insertion into the audio band) whilst the other channel transmits programme.

QUATERNARY DATA IN SYNC

To insert the digitized audio information in the line sync period of the video signal, pairs of bits are combined to form quaternary (i.e. four-level) 11Mbits/s rate to appear at only half that rate, which is within the limits of video circuits. Since the NICAM data rate and the TV line rate are not interrelated, the insertion process has to be asynchronous. Most sync pulses contain a marker plus 22 quits, but every 5 or 6 lines, to justify the data rate, a marker and only 20 quits are inserted, the marker pulse serving as a timing reference for the following data. The asynchronous data insertion is controlled on a board which monitors the depth of data in a FIFO memory, and also provides standby syncs if the video source should fail. The data insertion is done in mainly discrete component analogue circuitry which introduces minimal distortion of the video signal.

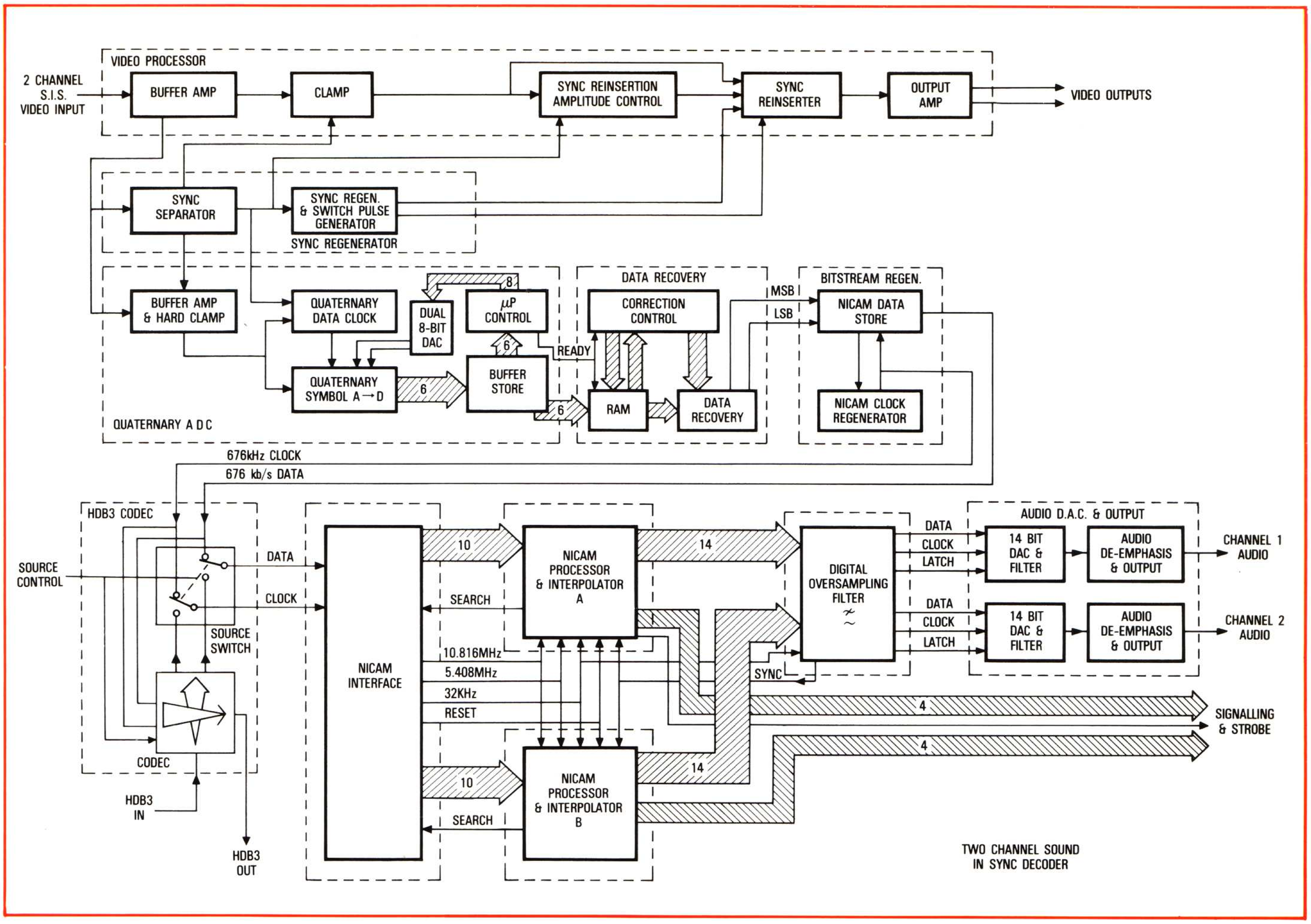
DATA RECOVERY

In the decoder the quits are decoded by a sophisticated process designed to achieve reliable results even after moderate distortion of the signal during transmission. The timing of the marker pulse after line sync leading edge is examined to decide whether there are 22 or 20 following quits. Each quit is sampled and converted by a fast ADC to a 6-bit number. By examining the range of numbers produced, a microprocessor controls (via an 8-bit DAC) the reference voltages used by the ADC, thus compensating for any loss or gain in the transmission path.

The microprocessor program cycle time is approximately ¼ sec, during which the data in 20 sync pulses is examined. As the results are applied to the ADC only during the next ¼ sec, further immediate processing is done on each set of 22 (or 20) 6-bit numbers, to compensate for shorter-term changes in quit amplitude or level; each number in a range considered likely to correspond to a quit level 0 is compared with the ideal value, and used to produce an average error within the set for quit level 0. The same is done for quit level 3, and the two error values used to select one of 256 look-up tables (held in EPROM) for conversion of the whole set of 6-bit numbers to the original pairs of bits which they represent.

BITSTREAM REGENERATION

Each 2 bits recovered are fed into a FIFO memory and fed out at a 676kbits/s bitstream equivalent to



that of the coder. The bitstream clock regeneration is under the control of FPLS (with supporting PROM) which maintains a steady clock rate in spite of the erratic rate of arrival of the recovered data.

CHANNEL SEPARATION AND EXPANSION

The bitstream is searched for the occurrence of the framing sequence, and on this basis, alternate bits corresponding to the two audio channels are fed to the separate processing curcuits for expansion and, when there are parity errors, interpolation. If the parity error rate is too high, the audio is muted and the framing sequence search is repeated.

DAC AND AUDIO FILTERING

A digital oversampling filter IC, as used in Compact Disc players enables a considerable reduction in the complexity of the filter needed after digital to analogue conversion. It increases the sample rate four-fold while band-limiting the audio to 15kHz and applying Sin.x/x correction suitable for the following DAC. A simple analogue active filter suppresses the unwanted lobes in the spectrum of the DAC's output, while another applies de-emphasis before the output amplifier.

RESTORATION OF VIDEO

Composite syncs are full regenerated and switched into the video waveform in place of the syncs with sound data. An oscillator, in a phase-locked loop controlled by incoming line sync edges, clocks an FPLS (Field Programmable Logic Sequencer I.C.) producing line syncs, equalising pulses, and broad pulses, while another FPLS selects these in turn to form the correct field sequence. A broad pulse detector ensures that the regenerated field sequence occurs at the same time as the incoming field sequence and triggers a sync amplitude sampling circuit which automatically adjusts the amplitude of the regenerated syncs.

MONITORING

In addition to the simple front panel LED status indication, each basic rack provides functional monitoring data from each module, via the motherboard, to two module positions, which are available to accept system monitoring modules. These modules process the monitoring data and provide a "statement" as to the operating condition of the equipment for Local (front panel) or Remote indication via telemetry links..

Monitoring is available in three formats, allowing the user the choice of complexity relevant to his operational needs, as follows:

a) Full Facilities Monitoring;
   providing maximum status indication.
b) Main Facilities Monitoring;
   providing status indication of main parameters.
c) Reduced Facilities Monitoring;
   providing essential status information only.

These monitoring facilities are provided by a selection from modules which fit into the two available module positions in the coder/decoder mainframe.

Variant (a) above requires both monitor 1 and monitor 2 modules.

Variant (b) above requires only Monitor 1 module.

Variant (c) above requires Monitor 3 module.

Upgrading can therefore be simply achieved by module addition or replacement.

Full Facilities Monitoring

This is an on-board microcomputer using a Z80, monitoring all the fault/warning indications from the individual boards.

Depending on the software routine, 8 indications can be outputed in relay (2) and open collector (6) form.

A second board compares the rms energy of the audio signal before and after the Analogue to Digital (or Digital to Analoque) conversion in the Coder (or Decoder). It also provides fault information to the computor.

The relay outputs are controlled by timers, but the open collector outputs react immediately to the change in fault status.

Main Facilities Monitoring

This provides identical facilities to those mentioned above, except that the second board monitoring the status of the ADC (or DAC) is not provided.

Reduced Facilities Monitoring

This is the simplest of the monitoring facilities, and provides a single "system good" indication from hard-wired logic circuits.

GENERAL SPECIFICATION

TV Systems:
PAL/625 lines, or NTSC/525 lines.

Remote Controls:
(selected internally)
Analogue/Digital (HDB3) audio source selection on Coder.
SIS/HDB3 audio source selection on Decoder.

Remote Indications:
Dependent on monitoring option supplied.

Monitor Points:

Video	— >1kOhm
Audio	— for 600Ohm headphones
676kbits/s	— >1kOhm
Composite	— >1kOhm

Connectors:

Video and digital audio	— type BNC
Audio input/output	— 5-pin type AXR
Monitoring and remote control	— 37-way 'D' type
Signalling	— 25-way 'D' type
Mains supply	— type CEE22

Ambient Temperature:
0-45 degrees C.

Relative Humidity:
Up to 90%.

Altitude:
Up to 2,000 metres.

Power Supply Requirements:
110/117/220/234 volts ±10%, 47-63Hz, single phase.

Power Consumption:
Coder — 85VA (approx).
Decoder — 125VA (approx).

Cooling Requirements:
Convection cooling, with free airflow above and below unit.

Dimensions:
Height: 133mm (5.25 inches)
Width: 480mm (19 inches)
Depth: 381mm (15 inches)

Weight:
Coder — 7.5kg (17lb)
Decoder — 8.0kg (18lb)

Finish:
Light grey.

TECHNICAL SPECIFICATION

VIDEO PERFORMANCE

Return Loss:
36dB at 5MHz.

Insertion Gain:
0db ±0.2dB.

Random Noise (Unweighted):
Better than −60dB.

Luminance Distortion:
<1% at reference level.

Differential Gain:
<0.5% at reference level.

Differential Phase:
<0.5% at reference level.

Differential Phase:
<0.5 degrees at reference level.

Linear Distortion:
Field-time waveform — <1% at reference level.
Line-time waveform — <1% at reference level.
2T pulse and bar — <0.5%K at reference level.
Chrominance/luminance gain <0.2dB.
Chrominance/luminance delay <10ns.

Spurious in Blanking Period:
<25mV p-p.

AUDIO PERFORMANCE

Input Impedance:
>10kOhms balanced.

Output Impedance:
Typically 30 ohms balanced.

Maximum Signal Level:
Maximum digital coding level +14.8dBm at 2kHz.
Other frequencies as related to 2kHz by CCITT J17 pre-emphasis curve.

Gain:
0dB nominal.

Frequency Response:
40Hz to 125Hz +0.4/−1.0dB
125Hz to 10kHz ±0.4dB
10kHz to 14kHz +0.4/−1.1dB
14kHz to 15kHz +0.4/−1.8dB

Harmonic Distortion:
Less than 0.1% at 1kHz and +9dBm input level.

Crosstalk:

at 40Hz:	>−80dB
500Hz to 15kHz:	>−85dB
at 15kHz:	>−75dB

Noise:
<−57dBq0ps

TYPE NUMBERS

Coders: LDM1901/-
Decoders: LDM1902/-

/00	Audio + PAL/625 lines (basic unit)
/01	As /00 but with reduced facilities monitoring
/02	As /00 but with main facilities monitoring
/03	As /00 but with full facilities monitoring
/10	Audio/HDB3 + PAL/625 lines, (basic unit)
/11	As /10 but with reduced facilities monitoring
/12	As /10 but with main facilities monitoring
/13	As /10 but with full facilities monitoring
/20	HDB3 only + PAL/625 lines (basic unit)
/21	As /20 but with reduced facilities monitoring
/22	As /20 but with main facilities monitoring
/50	Audio + NTSC/525 lines (basic unit)
/51	As /50 but with reduced facilities monitoring
/52	As /50 but with main facilities monitoring
/53	As /50 but with full facilities monitoring

Specifications details subject to change without notice.

varian

Varian TVT Limited
PO Box 41 Coldhams Lane Cambridge England CB1 3JU
Tel: Cambridge (0223) 245115. Telex: 81103 VAR TVT G Fax: (0223) 214632

Printed in England

LPL 5/87