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Television Systems;
Specification of the D2-HDMAC/Packet system

ETSI

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Foreword

This European Telecommunication Standard (ETS) has been produced under the authority of the Joint Technical Committee (JTC) of the European Broadcasting Union (EBU) and the European Telecommunications Standards Institute (ETSI).

This ETS is based upon ETS 300 250 (D2-MAC), it describes the baseband structure, multiplexing and modulation of the D2-HDMAC/packet system for High Definition Television (HDTV) broadcasting. This draft ETS was produced with the co-operation of the EUREKA 95 project.

In view of the urgency for presenting the technical content of this ETS and for historic reasons a number of the figures within this ETS, are in their original form as received from the EBU. These figures have not therefore, undergone the normal ETSI editing procedures or quality control procedures relating to their presentation.

NOTE: The EBU/ETSI Joint Technical Committee was established in 1990 to co-ordinate the drafting of European Telecommunication Standards in the specific field of radio, television and data broadcasting.

The European Broadcasting Union (EBU) is a professional association of broadcasting organisations whose work includes the co-ordination of its Members' activities in the technical, legal, programme-making and programme-exchange domains. The EBU has Active Members in about 50 countries in the European Broadcasting Area; its headquarters is in Geneva *.

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1 Scope

This ETS defines the D2-HDMAC/packet system for High Definition Television (HDTV) broadcasting. It covers the baseband structure, multiplexing and modulation for the High Definition Multiplexed Analogue Components (HDMAC) vision signal and the accompanying digital services.

As a member of the MAC/packet family this system incorporates the following features:

- time division multiplexing;
- HDMAC picture coding;
- packet multiplexing for sound and data;
- digital high and medium quality sound coding;
- service identification;
- conditional access with video, sound and data scrambling.

1.1 Field of application

This ETS is applicable to:

- satellite transmission with FM modulation;
- cable distribution with AM/VSB modulation;
- possible future applications such as:
    - terrestrial broadcasting;
    - point to point transmissions;
    - video cassettes, laser vision discs etc.

2 Normative references

This ETS incorporates, by dated or undated reference, provision from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred applies.

[1] ETS 300 250: "Television systems; D2-MAC/packet specification".

[2] CCIR 1096-1: "Transmission of television signals using Multiplexed Analogue Components".

[3] CCIR Recommendation 653-1 (1990): "Teletext System A".

[4] CCIR Recommendation 653-1 (1990): "Teletext System B".

[5] CCIR Recommendation 656: "Interfaces for digital component video signals in 525-line and 625-line television systems".

[6] CCIR Recommendation 601-2: "Encoding parameters of digital television for studios".

[7] EBU Technical document 3232 (2nd edition 1982): "Displayable character sets for broadcast teletext".

[8] EBU Technical document 3244: "EBU Radio Data System (RDS)".

[9] CCIR Recommendation 457-1: "Use of the modified Julian date by the standard-frequency and time-signal services".

[10] CCIR Recommendation 460-4: "Standard-frequency and time-signal emissions".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of this ETS the following definitions apply:

network: the ensemble of signals broadcast in the transmission channel at a given time.

subframe: any rectangular shaped area within the television frame.

3.2 Symbols

For the purposes of this ETS, the following symbols apply:

α	roll-off factor of half-Nyquist filter
$\oplus$	bitwise Exclusive-OR operation
—	bitwise logic inversion
'x	indicating a hexadecimal value x

3.3 Abbreviations

For the purposes of this ETS, the following abbreviations apply:

ACS	Access Control System
AM	Amplitude Modulation
ANC	Ancillary
ARD	Arbeitsgemeinschaft der Rundfunkanstalten Deutschlands
ASI	Additional Service Identification
BER	Bit Error Rate
BRD	Bandwidth Restoration Decoder
BRE	Bandwidth Reduction Encoder
BSS	Broadcasting Satellite Service
CA	Conditional Access
CC	Continuity Check
CCIR	Comite Consultatif International des Radio-communications
CCITT	Comite Consultatif International Telegraphique et Telephonique
CI	Command Identifier
CI	Continuity Index
CIB	Control Information Bit
CRC	Cyclic Redundancy Check
CRI	Clock Run In
CSN	Coding Sequence Number
CW	Control Word
DATV	Digital Assistance for TeleVision
DCINF	Digital Component Information
DGH	Data Group Header

DK	Distribution Key
EBU	European Broadcasting Union
ECL	Emitter Coupled Logic
ECM	Entitlement Checking Message
EIRP	Equivalent Isotropic Radiated Power
EMM	Entitlement Management Message
ETS	European Telecommunication Standard
FBI	Field Blanking Interval
FD	Format Descriptor
FEC	Forward Error Correction
FM	Frequency Modulation
FSD	Frame Synchronization Data
FSS	Fixed Satellite Service
FSW	Frame Synchronization Word
GPD	General Purpose Data
IF	Intermediate Frequency
IP	Image Parity
ISO	International Standardization Organization
IW	Initialization Word
LBI	Line Blanking Interval
LI	Length Identifier
LSB	Least Significant Bit
LSW	Line Synchronization Word
MAC	Multiplexed Analogue Components
MCCI	Motion Compensation Compatibility Improvement
MD	Mode Decision
MI	Message Index
MJD	Modified Julian Date
MPX	Multiplex
MSB	Most Significant Bit
MV	Motion Vector
NSI	Network Specific Identity
OAA	Over-Air Addressing
OSI	Open Systems Interconnection
PG	Parameter Group
PGI	Parameter Group Identifier
PI	Parameter Identifier
PRBS	Pseudo Random Binary Sequence
PS	Packet Suffix
PT	Packet Type
PX	Primary Index
RDF	Repeated Data Frame
RDS	Radio Data System
RF	Radio Frequency
SDF	Static Data Frame
SI	Service Identification
SX	Secondary Index
TCI	Temporal Compatibility Improvement
TDM	Time Division Multiplex
TDMCTL	TDM Control
TL	Type List
TXT	Teletext
UDT	Unified Date and Time
UI	Universal Identity
UK	Unique Key
UTC	Coordinated Universal Time
VCI	Vertical Compatibility Improvement
Vp-p	Volt peak to peak
VPS	Video Programming Service
VSB	Vestigial Side Band

VTR	Video Tape Recorder
ZDF	Zweites Deutsches Fernsehen
ZVEI	Zentralverband Elektrotechnik- und Elektronik-industrie

4 Specification of the time-division multiplex

4.1 Introduction

This part of the ETS contains the specification of the Time Division Multiplex (TDM) system for the picture, sound and data used for D2-HDMAC transmission. This system is suited for use in satellite broadcasting and any transmission medium which guarantees a baseband bandwidth equal to or greater than 11,14 MHz. In particular, it may be used within an AM/VSB modulation network with a channel spacing not less than 12 MHz.

If the available bandwidth is less than those values this signal can not be processed by a D2-HDMAC/packet decoder. Such a signal could be still processed by a D2-MAC/packet decoder (See Annex H).

4.2 Basic hypotheses

4.2.1 Vision standard

The present specification is compatible with the HDMAC vision standard, as described in Clause 5. However, through the TDM control information (see subclause 4.5.4) provision is made for other uses of the channel such as normal MAC (see ETS 300 250 [1]).

In baseband the signal polarity shall be such that the maximum brightness corresponds to maximum positive amplitude.

4.2.2 Capacity for the sound/Data multiplex accompanying the HDMAC vision standard

The available capacity in the Line Blanking Interval (LBI) is equivalent to four high-quality sound channels compatible with MAC/packet (see Clause 6).

4.2.3 Capacity for DATV/Data accompanying the HDMAC vision standard

A capacity of about 1,1 Mbit/s is available in the Field Blanking Interval (FBI). This capacity is primarily used for transmission of the Digital Assistance for Television (DATV) data (see Clause 5). The extra capacity is free for other applications such as teletext either in data lines or in packets or other data services (as described in Clause 7).

4.3 Transmission multiplex structure

The baseband multiplex structure for the D2-HDMAC/packet signal is shown, for each line, in figure 1.

4.3.1 Instantaneous bit rate

The instantaneous bit-rate of all the data bursts shall be 10,125 Mbit/s ± 2,5 parts in 10 million, except for the field blanking data burst devoted to DATV/data for which the instantaneous bit-rate shall be exactly twice this value.

4.3.2 Data bursts format

For high definition television (HDMAC) transmissions the baseband signal includes two different data multiplexes, one in the LBI referred to as the sound/data multiplex and one in the FBI referred to as the DATV/data multiplex. In addition, special use of the data burst of line 624 and line 625 is defined, and optional teletext data lines as specified in subclause 7.2 (see figure 2a).

4.3.2.1 Sound/Data multiplex

Each LBI data burst shall contain a total of 105 bits. The first six bits shall be allocated as a line synchronising word. The remaining 99 bits shall be used for sound and data. See figure 2b.

The sound/data multiplex shall occupy 623 LBI data bursts per video frame, leaving line 624 free for insertion of a clamp marker (see NOTE 1) and reference signals, and line 625 (see NOTE 2) free for the insertion of a frame synchronization word and the special data burst described in subclause 4.5.

NOTE 1: Provisionally a clock run in is transmitted in the spare bits of line 624, just preceding the clamp marker word (see figure 2b).

NOTE 2: The whole of line 625 is available for transmission of special data (see subclause 4.5).

4.3.2.2 DATV/Data multiplex

Each FBI data burst shall contain at most 1 063 bits. The exact position of the DATV/data multiplex shall be signalled in specific TDMS data in Line 625 (see subclause 4.5.5). The FBI data burst (see figure 2c) shall consist of:

- the bits of the DATV/data multiplex as signalled in the specific TDMS data. These bits shall be allocated for DATV and data. See also subclause 4.5.5;

- the bit directly before the first bit of the signalled multiplex. This bit shall be allocated as a run-in bit;

- the bit directly following the last bit of the signalled multiplex. This bit shall be allocated as a spare bit.

The DATV/data multiplex shall occupy a number of FBI data bursts corresponding to a capacity of at least 56 packets of FBI1 and FBI2 combined (see subclause 4.3.6.3), where:

- FBI 1 Lines 1 to 22 inclusive;
- FBI 2 Lines 311 to 334 inclusive but excluding line 312.

4.3.3 Synchronization

Synchronization can be achieved by two independent methods:

- Line Synchronization Words (LSW) provide both line and frame information;
- Frame Synchronization Words (FSW) provide frame synchronization directly and line synchronization is derived by counting.

Synchronization for the conditional access descrambling process, where applicable, for vision, sound and data is described in Clause 9 (see also the definition of FCNT and CAFCNT in subclauses 4.5.4 and 4.5.3).

4.3.3.1 Line synchronization

Each line blanking data burst shall contain one of two 6-bit Line Synchronization Words (LSWs).

The LSWs are defined in their transmission order as:

$$W1 = 001011$$
$$W2 = \overline{W1} = 110100$$

The LSW shall be transmitted in its true (W1) or inverted form (W2) according to the pattern shown in table 1.

The pattern of line sync words at the frame boundaries provides the video frame synchronization, determines the video frame parity and defines the 80 ms period associated with the video coding (see Clause 5). This is shown in table 1.

The convention used for numbering the lines and fields is described in subclause 5.4.2.

Table 1: Line sync words and video frame parity

Frame parity	Line number	Sync word	HDMAC coding period
Frame boundary	-----------------	--------------	Start of HDMAC coding period
	1	W2	Beginning of first field
	2	W1	
	3	W2	
	4	W1	First field
	5	W2	
Odd frame	...	...	
	620	W1	
	621	W2	Second field
	622	W1	
	623	W1	
	624	W2	
	625	W2	End of second field
Frame boundary	-----------------	--------------	
	1	W1	Beginning of third field
	2	W2	
	3	W1	
	4	W2	Third field
	5	W1	
Even frame	...	...	
	620	W2	
	621	W1	Fourth field
	622	W2	
	623	W2	
	624	W1	
	625	W1	End of fourth field
Frame boundary	-----------------	--------------	Start of next HDMAC coding period
	1	W2	Beginning of first field
	2	W1	
	3	W2	
	4	W1	
	5	W2	
Odd frame	...	...	

4.3.3.2 Frame synchronization

The data burst in line 625 shall contain a frame synchronizing sequence which immediately follows the line sync word and which is defined by the following 96 bits in hexadecimal notation.

Clock run-in　　　　Frame sync word

55 55 55 55　　　　65 AE F3 15 3F 41 C2 46
|
0101
|
First transmitted bit

The frame synchronizing sequence contains a Clock Run-In (CRI) period of 32 bits followed by a 64-bit frame synchronizing word.

The 96-bit frame synchronizing sequence shall be transmitted in its true form preceding even-numbered frames and in its inverted form preceding odd-numbered frames.

4.3.3.3 Colour sequence identification

See subclause 5.4.1.

4.3.3.4 Packet synchronization

The sound and data in the sound/data multiplex system shall be transmitted in packets.

The DATV and data in the DATV/data multiplex system shall be transmitted in packets.

The start of the first packet of a digital TDM component shall be the first bit of the first line of the TDM component. The positions of the subsequent data packets can be obtained by counting (see subclause 4.3.6).

4.3.3.5 Clamping

A clamp period shall be present on each line except line 625, and is defined as following the line blanking data burst. A clamp marker shall be provided in line 624 (see figures 2b and 2c). The clamp marker is a 32-bit word which is fixed in relationship to the end of the LBI burst, independent of the burst duration. The 32-bit word, in hexadecimal notation, shall be EAF3927F. E is read out first, with the MSB first.

4.3.4 Reference and test signals

Three test signals shall be inserted in lines 312, 623 and 624. Lines 312 and 624 shall be transmitted in the basic mode as indicated by signalling of TDMCID code '30 in line 625 (see subclause 4.5.4). Line 623 shall be transmitted in either the basic mode (identified by TDMCID code '30) or in the cyclical mode (identified by TDMCID code '31).

Test signals shall consist of samples series transmitted at a 20,25 MHz rate. Only the same Nyquist filter as for HDMAC video signal samples shall be applied to these signals (no additional shaping or non-linear pre-emphasis filter). The HDMAC test signal bandwidth is therefore equal to 10,125 MHz (-3 dB).

High frequency signal amplitudes are restricted to ± 250 mV (except for the pulses on line 312) to avoid non-linear distortion.

For the following subclauses the sample number ranges are defined according to the following convention: "k = x to y". This defines a range for the sample number k, for which the first sample of the range is "x" and that the last sample of the range is "y" (therefore inclusive sample "y").

4.3.4.1 Reference and test line 624

The first part of line 624 shall contain grey, white and black references which are specified as follows (see figure 3a):

k = 210 to 371	Grey level;
k = 372 to 533	White level;
k = 534 to 695	Black level.

The black to white amplitude shall correspond to a nominal level of 1 Volt peak-peak.

The second part shall contain a test signal devoted for amplitude/phase channel frequency response evaluation. It consists of a complex wobulation constituted by two signals defined over a period of 512 T according to the following relations:

Real part:

$$k = 0 \text{ to } 512 : y_k = \cos\left(\frac{\pi.(k-256)^2}{512}\right)$$

Imaginary part:

$$k = 0 \text{ to } 512 : y_k = \sin\left(\frac{\pi.(k-256)^2}{512}\right)$$

These signals shall be transmitted in sequence in four consecutive frames as follows:

- Even frame : non inverted real part;
- Odd frame : non inverted imaginary part;
- Even frame : inverted real part;
- Odd frame : inverted imaginary part.

The signal wave forms are given in figures 3b and 3c.

The allocation of line 624 samples is given below:

k = 255 to 369	level 0 mV;
k = 370 to 374	transition from 0 mV to + 500 mV;
k = 375 to 531	level + 500 mV;
k = 532 to 536	transition from + 500 mV to - 500 mV;
k = 537 to 693	level - 500 mV;
k = 694 to 698	transition from - 500 mV to 0 mV;
k = 699 to 738	level 0 mV;
k = 739 to 1251	complex wobulation with the levels ± 250 mV;
k = 1252 to 1292	level 0 mV.

4.3.4.2 Reference and test line 312

The first part of line 312 shall be filled over 2 frames by a pseudo random sequence of 511+1 = 512 bits with the levels - 250 mV and + 250 mV corresponding to the bit values "0" and "1" respectively, the first function of which is to help the equaliser process.

The polynomial generator shall be $x^9 + x^4 + 1$ and gives a pseudo random sequence of 511 bits clocked at 20,25 MHz. The first half of the sequence (256 bits) shall be transmitted in the even frame and the second part shall be transmitted in the odd frame. The last bit (256th bit) of the odd frame shall be identical to the first bit of the next sequence.

The Pseudo Random Binary Sequence (PRBS) generator shall be initialised at the beginning of each even frame with the binary word 111111111 (see figure 7).

The first bit of the sequence generated by the PRBS generator on the even frame shall be the value present at the output after it has been loaded and before any shift operations have taken place.

In addition, two inverse half amplitude pulses and a half amplitude transition shall be inserted in even frames, the first function of which is to distinguish between linear and non linear perturbations. The full amplitude pulses and transitions in the D2-HDMAC testline 312 shall not be weighted by Blackman and Hamming windows (in contrast to D2MAC, as described in CCIR 1096-1 [2]).

The signal wave forms are shown in figures 4a and 4b. The pseudo-random sequence is defined in figure 7.

The allocation of sample levels (in mV) are given on the next page.

Even frame:

k = 225 to 233	0
k = 234 to 489	- 250 or 250 (pseudo-random sequence)
k = 490 to 499	0
k = 500 to 524	- 250
k = 525	250
k = 526 to 550	- 250
k = 551 to 575	250
k = 576	- 250
k = 577 to 601	250
k = 602 to 614	0
k = 615 to 775	- 500
k = 776	500
k = 777 to 938	- 500
k = 939 to 1099	500
k = 1100	- 500
k = 1101 to 1262	500
k = 1263 to 1292	0

Odd frame:

k = 225 to 233	0
k = 234 to 489	- 250 or 250 (pseudo random sequence)
k = 490 to 499	0
k = 500 to 550	250
k = 551 to 601	- 250
k = 602 to 614	0
k = 615 to 938	+500
k = 939 to 1262	- 500
k = 1263 to 1292	0

4.3.4.3 Reference and test line 623

Line 623 can be transmitted in different modes as indicated by signalling of TDMCID in line 625 (see subclause 4.5.4). Subclause 4.3.4.3.1 describes the format of line 623 in the basic mode. Subclause 4.3.4.3.2 describes the format of line 623 in the cyclical mode.

4.3.4.3.1 Basic mode

The basic mode test signal attributed to the line 623 is devoted for static linearity and noise measurements and shall consist of a ramp defined over a period of 1 000 T (1/T = 20,25 MHz) from - 500 mV to 500 mV in even frame and 500 mV to - 500 mV in odd frame (see figures 5a and 5b).

A ramp is a signal for a duration nT in the equation:

$$k = 0 \text{ to } n : y_k = k/n \quad \text{for a rising ramp;}$$

$$k = 0 \text{ to } n : y_k = 1 - k/n \text{ for a falling ramp.}$$

The allocation of sample levels are given on the next page.

Even frames:

k = 225 to 243	level 0 mV
k = 244 to 248	transition from 0 mV to - 500 mV
k = 249 to 267	level - 500 mV
k = 268 to 1267	ramp from - 500 mV to + 500 mV
k = 1268 to 1273	level + 500 mV
k = 1274 to 1278	transition from + 500 mV to 0 mV
k = 1279 to 1292	level 0 mV

In the odd frames, the levels are inverted.

4.3.4.3.2 Cyclical mode

The cyclical mode test signal inserted into line 623 may consist of a random sequence of predefined test patterns. The exact definition of these predefined test signals and their individual identification is still subject of investigation and is not further specified here.

The sequence of test signals in the cyclical mode may be free running, i.e. not locked to either the odd/even frame synchronization or to the line 625 RDF frame counter FCNT (see subclause 4.5.4).

4.3.5 Signalling of RF channel, TDM configuration and service identification information

The Radio Frequency (RF) channel and TDM configuration and service identification information is transmitted in the special data burst in line 625, following the frame synchronization sequence (see subclause 4.5).

4.3.6 Packet transmission

A packet multiplexing system shall be used that provides flexible use of the transmission capacity in the available TDM components. Packets of different services with different addresses may be multiplexed into a single continuous bit stream forming a packet multiplex. An integer number of packets shall be contained in each TDM component in the 625 lines, 40 ms frame.

In the case that the required capacity in a packet multiplex is smaller than the available capacity, the spare capacity shall be filled with dummy packets (see subclause 4.3.6.1.1).

Each packet shall have a constant length and shall consist of a string of 751 bits (see figure 6). The packet is divided into two parts:

- packet header 23 bits (described in subclause 4.3.6.1);

- data area 728 bits (described in subclauses 4.3.6.2 and 4.3.6.3).

4.3.6.1 Packet header

The packet header shall consist of 23 bits, divided into three parts:

- an address field 10 bits (least significant bit is transmitted first);

- a continuity index 2 bits (least significant bit is transmitted first);

- a protection suffix 11 bits.

The transmission order shall be thus:

address field	Continuity index	Protection suffix
a b c d e f g h j k	l m	p q r s t u v w x y z

the address field LSB is "a" the address field MSB is "k"
the continuity index LSB is "l" the continuity index MSB is "m"

4.3.6.1.1 Address field

A unique address code shall be allocated to each service component. It is used in the receiver to select by recognition the packets of the required service and to reject other packets. This 10-bit field allows up to 1 024 different simultaneous service components.

The packet address "0" shall be permanently allocated to the service identification system (see Clause 8).

The packet address "1 023" shall be permanently allocated to dummy packets which are inserted to fill a packet multiplex.

4.3.6.1.2 Continuity index

The length of this index shall be two bits. It assures the link between successive packets of a same service component and can be used to detect possible packet loss (see figure 6 and subclauses 5.8.3.1, 6.5.4, 8.3.2 and 9.5.2).

4.3.6.1.3 Protection suffix

This field shall have a length of 11 bits and provides high protection to the address and index information. It is based on a systematic Golay cyclic code (23,12) which can correct up to three errors among 23 bits. The description of the code used is as follows:

- the 10-bit address is followed by two continuity index bits and the 11-bit protection suffix (see figure 6). The code is fully defined by its generator polynomial:

  - $G(x) = x^{11} + x^{10} + x^{6} + x^{5} + x^{4} + x^{2} + 1$.

- the specified configuration constitutes a systematic code; i.e. the code word M(x) formed from 23 bits is made up from 12 bits of useful information:

  - (polynomial m(x): address + continuity index), followed by eleven redundancy bits determined as the remainder of the division of the polynomial $x^{11}m(x)$ is by the generator polynomial G(x).

- let:

  - (abcdefghjk) be the ten address configuration bit;

  - (lm) be the two continuity index configuration bits;

  - (pqrs tuvw xyz) be the eleven redundancy bits.

then:

- $m(x) = ax^{11} + bx^{10} + cx^{9} + \ldots + jx^{3} + kx^{2} + lx + m;$

$$M(x) = ax^{22} + bx^{21} + cx^{20} + \ldots + jx^{14} + kx^{13} + lx^{12} + mx^{11} + px^{10} + qx^{9} + \ldots + yx + z;$$

and $x^{11}.m(x) = q(x).G(x) + r(x)$

- the transmitted code $M(x) = x^{11}.m(x) + r(x) = q(x).G(x)$,is, therefore, a polynomial which is a multiple of the generator polynomial G(x).

4.3.6.2 Useful data

The total length of the useful data area per packet is 728 bits (91 bytes). For many services the first 8 bits are used as a Packet Type (PT) byte which can be used to identify various packet usage's within a single service component comprising packets of one address. In the case where the PT byte is present then the remaining useful data area becomes 720 bits (90 bytes).

In general the PT byte is used as an (8,2) code, which can correct up to 2 errors among 8 bits. The 4 possible PT bytes shall be '00, '3F, 'C7 and 'F8. The LSB of the PT byte shall be transmitted first.

The meaning of the PT byte, if defined, is given in the specification of the related sound or data packet service.

4.3.6.3 Packets per subframe

The transmission multiplex structure for the LBI data burst described in subclause 4.3.2.1 allows:

- 99 useful bits per LBI data burst;
- 623 LBI data bursts per video frame for the sound/data multiplex;
- consequently, 61 677 bits are available in the sound/data multiplex per video frame;
- hence each video frame contains 82 packets of length 751 bits (as shown in figure 2b).

The transmission multiplex structure of the FBI data burst described in subclause 4.3.2.2 provides:

- a maximum of 1 061 bits per FBI data burst, depending on the actual values transmitted in specific TDMS data in line 625 (see subclause 4.5.5);
- a capacity that corresponds to not less than 56 packets per frame (in FBI1 and FBI2), equivalent to a capacity of not less than 42 056 bits per FBI. Figure 2c shows an example of the FBI multiplex structure organised as two TDM components, one in each field.

4.3.6.4 Bit-interleaving

Bit-interleaving shall be applied to all data transmitted in packet format over a block length of 751 bits in order to minimize the effect of multiple bit errors. The bits of each packet shall be transmitted in the following order:

$$f(n) = (((n - 1) * 94) \text{ modulo } 751) + 1 \qquad n = 1 \ldots 751$$

where:

n the bit location of n^{th} bit of a packet before bit-interleaving, ranging from 1 to 751;
f(n) the bit location of the n^{th} bit of a packet after bit-interleaving, also ranging from 1 to 751.

The interleaving shall not be applied to the special data burst on line 625, to the data burst in line 624, and to the teletext data lines in the field blanking interval (see subclauses 7.2.2.2 and 7.2.2.3).

The following illustrates the above formula (read from left to right and from top to bottom):

1	95	189	...	565	659
2	96	190	...	566	660
...					
...					
93	187	281	...	657	751
94	188	282	...	658	

4.3.7 Data scrambling for spectrum shaping purpose

After bit interleaving, energy dispersal shall be achieved by adding (modulo 2) a scrambling sequence to the data applied to the modulator, with the following exceptions:

- the first 6 bits of each line blanking data burst (LSW);
- the special data burst in line 625 and the data on line 624;
- teletext data transmitted according to subclause 7.2.

The sequence used for scrambling a data burst shall be derived from a PRBS generator which is clocked at a rate equal to the instantaneous bit rate of that data burst. One PRBS generator shall operate continuously at a clock rate of 10,125 MHz and a second PRBS generator shall operate continuously at a clock rate of 20,25 MHz. Both PRBS generators have an identical structure. This structure is shown in figure 8. During periods when scrambling is not applied, the scramblers continue to generate the PRBS sequences, but its actions shall be inhibited by means of gating signals.

Each PRBS generator shall be initialized every 625 lines.

The first bit from the 20,25 MHz PRBS generator shall be produced on sample position 7 of line 1. The first bit from the 10,125 MHz PRBS generator shall be produced on sample position 10 of line 1 (and added to bit 7).

The first bit of the sequence generated by each PRBS generator is the value present at the output after it has been loaded and before any shift operations have taken place.

4.3.8 Scrambling for the free-access and controlled-access modes of the conditional-access system

When scrambling is used, the digital sound and data signals shall be subjected first to scrambling for the conditional-access system and then to bit interleaving and transmission scrambling (for spectrum shaping purposes). The conditional-access scrambling process is specified in subclauses 5.8 and 9.3.

4.4 Data coding and multiplexing with the picture

4.4.1 Digital components

The multiplexing of D2-HDMAC/packet shall be carried out at baseband. The digital components shall be coded in duobinary form at rates of 10,125 Mbit/s and at 20,25 Mbit/s. The coding is described in subclauses 4.4.2.4 and 4.4.2.3 respectively.

4.4.2 Digital component coding

4.4.2.1 Digital coding

The digital components shall be coded in duobinary form. The application of the basic coding method is illustrated in figure 9a for data at 20,25 Mbit/s and in figure 9b for data at 10,125 Mbit/s. With these coding systems the signal has three characteristic levels where the extreme levels represent logic "1" and the intermediate level represents logic "0".

4.4.2.2 Relationship between bits of data and the sampling structure

Although the data can be transmitted at a bit rate of 10,125 Mbit/s and 20,25 Mbit/s in D2-HDMAC, the TDM multiplex is defined in terms of samples at 20,25 MHz. Whenever in the following subclauses the correspondence between a bit number and a sample number results in a number that is negative or zero the sample number can be derived by adding 1 296.

NOTE: A D2-HDMAC multiplexer may use a 40,5 Mbit/s sample rate. The sample numbering and coding for this case are given in Annex D.

4.4.2.3 20,25 Mbit/s data

The sample number of the first bit of the DATV/data multiplex (see subclause 4.3.2.2) is signalled in the FCP parameter of the TDMS field (see subclause 4.5.5) related to TDMCIDs '40 and '41. The FCP field defines the sample of the first bit number (bit number 1) of each DATV/data burst, not including the run-in bit (FCP is coded such that sample 1 is coded 0).

Let FCP be coded as F (corresponding to the sample number F+1), then data bit y corresponds to sample number y + F.

Example: Let the FCP parameter equal 229 (corresponding to 20,25 MHz sample number 230). Bit 1 then corresponds to sample number (1 + 229) = 230.

4.4.2.3.1 Data precoding of 20,25 Mbit/s data

The binary data stream A_k shall be precoded to the binary data stream B_k to avoid the propagation of errors using the following relationship:

$$B_k = \overline{A_k} \oplus B_{k-1}$$

For data at 20,25 Mbit/s the index k is the sample number.

During the periods where no digital component is transmitted, the data sequence shall be $A_k = 0$, resulting in a corresponding B_k sequence of alternately 1 and 0.

Associated with the sequence B_k shall be the sequence C_k which takes values of - 1 and + 1 according to:

- $C_k = 2.B_k - 1.$

4.4.2.3.2 Duobinary coding of 20,25 Mbit/s data

The signal coded in duobinary form for data at 20,25 Mbit/s shall be written as:

- $$D(t) = \sum_j C_j . h(t - jT)$$

T represents the sampling period of the picture signal, which is about 49,4 ns. h(t) represents the pulse response of the filter H corresponding to duobinary coding, of which the frequency response H(f) for 20,25 Mbit/s data shall be given by:

- $$|f| \leq \frac{1}{2T} \quad : H(f) = \left(\frac{MT}{2}\right) . \cos(\pi f T)$$

- $$|f| \geq \frac{1}{2T} \quad : H(f) = 0$$

where M represents the amplitude of data signal between extreme levels logic "1".

4.4.2.3.3 Sample coding of 20,25 Mbit/s data

A sampled representation will be taken for D(t).

For data transmitted at 20,25 Mbit/s duobinary coded samples shall have the following value for all values of k:

- $$D_k = \frac{M}{4}(C_k + C_{k-1})$$

The duobinary burst amplitude M shall correspond to 80% of that of the nominal black to white amplitude as referenced in line 624, disregarding overshoots.

The value "0" of samples D_k shall be aligned to the clamp level.

4.4.2.4 10,125 Mbit/s data

Data transmitted at 10,125 Mbit/s shall be carried in the even samples. Odd samples shall correspond to the data transitions.

Data bit y shall correspond to sample number (2y-4). See also subclause 4.4.2.2.

4.4.2.4.1 Data precoding of 10,125 Mbit/s data

The relationship is written so that the index corresponds to the sample number:

$$B_{2k} = \overline{A_{2k}} \oplus B_{2k-2}$$

The first transmitted bit in the LBI multiplex of each line shall correspond to A_{1294} (see figure 10a).

During the periods where no digital component is transmitted, the data sequence shall be $A_{2k} = 0$, resulting in a corresponding B_{2k} sequence of alternately 1 and 0.

Associated with the sequence B_{2k} shall be the sequence C_{2k} which takes values of -1 and +1 according to:

- $$C_{2k} = 2.B_{2k} - 1$$

4.4.2.4.2 Duobinary coding of 10,125 Mbit/s data

The signal coded in duobinary form for data 10,125 Mbit/s shall be written as:

- $$D(t) = \sum_j C_{2j}.h(t-(2j+1)T)$$

T represents the sampling period of the picture signal, which is about 49.4 ns. h(t) represents the pulse response of the filter H corresponding to duobinary coding, of which the frequency response H(f) for 10,125 Mbit/s data shall be given by:

$$|f| \leq \frac{1}{4T} \quad : \ H(f) = MT.\cos(2\pi f T)$$

$$|f| \geq \frac{1}{4T} \quad : \ H(f) = 0$$

where M represents the amplitude of data signal between extreme levels logic "1".

4.4.2.4.3 Sample coding of 10,125 Mbit/s data

A sampled representation will be taken for D(t).

For data transmitted at 10,125 Mbit/s even numbered duobinary coded samples shall have the following values:

- $$D_{2k} = \frac{M}{4}(C_{2k} + C_{2k-2})$$

For data transmitted at 10,125 Mbit/s odd numbered duobinary coded samples (transitions) shall have the following values:

- $$D_{2k+1} = \frac{M}{\pi}\left[C_{2k} + \frac{1}{3}(C_{2k-2} + C_{2k+2}) - \frac{1}{15}(C_{2k-4} + C_{2k+4}) + \frac{1}{35}(C_{2k-6} + C_{2k+6})\right]$$

This coding constitutes an excellent approximation to the theoretical value of the duobinary odd samples, obtained by truncation of the pulse response of the theoretical filter for 10,125 Mbit/s data as described in subclause 4.4.2.4.2.

The duobinary burst amplitude M shall correspond to 80 % of that of the nominal black to white amplitude as referenced in line 624, disregarding overshoots.

The value "0" of samples D_{2k} shall be aligned to the clamp level.

4.4.3 Picture/Data multiplex

The picture/data multiplex (figures 10a and 10b) shall be obtained by adding the picture signal as defined in Clause 5 and the duobinary data signal after having forced the data signal to zero between samples 208 and 1 292 for the lines having a data burst in the line blanking interval only (see figures 11a and 11b), between samples 208 and the run-in bit of the FBI data burst for lines having a FBI data burst, and between samples 208 and 228 for the other lines except line 625.

4.5 Specification of data transmitted in line 625

Within this subclause the least significant bits are transmitted first for all sequences of bits representing magnitudes, for all bytes, whatever they are representing, and for all sequences of numbered code bits, whereby transmission is in increasing numerical order. This rule for bit transmission order does not apply for bit sequences used for line or frame synchronization or used for Cyclic Redundancy Checking (CRC) purposes.

4.5.1 Overall structure of line 625

The data in line 625 shall be organized according figure 12. The allocated bits shall not be interleaved in the way described in subclause 4.3.6.4, nor scrambled for spectrum shaping.

Line 625 is divided into:

- one Frame Synchronization Data (FSD) part;
- one Static Data Frame (SDF) data block;
- five other data blocks, that are used for what is called the Repeated Data Frames (RDF).

4.5.1.1 Frame synchronization data

The first 102 bits shall contain a fixed pattern which shall be complemented on alternate frames, consisting of:

- the line synchronization word (LSW - 6 bits, which is common to all LBI data bursts);
- a 32-bit Clock Run-In (CRI);
- a 64-bit Frame Synchronization Word (FSW) (see figure 14).

The remainder of the burst provides an easily-accessed signalling capacity independent of the packet multiplex and is, therefore, capable of controlling the multiplex structure itself.

4.5.1.2 Data blocks

The data block of the static data frame shall contain a data field of 57 bits followed by a 14-bit error control group which can be used to detect most error patterns, and to correct one or two errors in the 71-bit group.

Each data block used as a repeated data frame shall contain a data field of 80 bits, also followed by a 14-bit error control group with the same protection properties.

BCH (71,57) and BCH (94,80) codes are specified, and their properties are the following:

- both BCH codes are derived from the (127,113) primitive BCH code. It shall be generated by the polynomial:

$$\left(x^7+x^3+1\right).\left(x^7+x^3+x^2+x+1\right)=$$

$$x^{14}+x^9+x^8+x^6+x^5+x^4+x^2+x+1$$

- the message to be sent shall be composed, in the sending order, of bits m_{56} or m_{79} to m_0, followed by check bits r_{13} to r_0 in the same order. The check bits shall be such that the polynomials of;

$$m_{56}x^{70}+m_{55}x^{69}+\dots+m_0x^{14}+r_{13}x^{13}+\dots+r_1x+r_0$$

and

$$m_{79}x^{93}+m_{78}x^{92}+\dots+m_0x^{14}+r_{13}x^{13}+\dots+r_1x+r_0$$

are multiples, modulo 2, of the polynomial:

$$x^{14}+x^9+x^8+x^6+x^5+x^4+x^2+x+1$$

4.5.2 Unified Date and Time (UDT)

The following five bits contains data which changes from frame to frame and which, over a 25-frame sequence, shall contain a statement of the unified date and time according to CCIR Recommendations 457 [9] and 460 [10], together with provision for signalling a local offset.

The sequence is detailed in table 2. The Modified Julian Date (MJD) is a five-digit decimal day count incremented at 0000 hour UTC (Coordinated Universal Time). The local offset from UTC, when provided, shall be specified relative to UTC in multiples of 1/2 hour with an inclusive range +15 to -12 hours.

The first bit of the sequence shall be an element of a chain code such that five consecutive correct bits are sufficient to define the position within the 25-bit sequence. Figure 13 indicates schematically the logic required to synchronize to, and reproduce, this chain code.

The boundary between the 25-frame sequence marks the time in seconds, and all the information in each sequence shall relate to the next following second.

This information is not protected against errors as it is a deterministic sequence which can reliably be anticipated by a local clock acting as a 'flywheel'.

4.5.3 Static Data Frame (SDF)

The static data frame shall be organized according to figure 14.

The SDF is used for information about the channel and about the time division multiplex format, together with information to assist the normal operation of the sound and television decoders and to preserve their use with future options as far as possible. In general, all of this information is repeated many times, so majority logic can be used to assist the correct recovery of the data under high error-rate conditions.

The following items of information are included:

(CHID) Satellite channel identification:

- a 16-bit code, each unique world-wide, which shall give satellite position, channel number and polarisation, country of origin, responsible administration, in accordance with a published table (see Clause A.2).

(SDFSCR) Services configuration reference:

- currently specified only for sound services transmitted in the subframes defined by TDMCID code '01 and '02, see subclause 4.5.4, and in the free-access mode.

- eight bits, of which the last seven form an arbitrary reference code allocated by the broadcaster to identify a commonly-used configuration of services within the channel. If non-zero, it can be taken with CHID to recall and/or store information transmitted by the service identification system that can configure the sound decoders for the television main sound and the main component of the major radio sound services.

- information can be identified by the SDFSCR code for the television main sound and for the first seven radio sound services listed in LISTX (see subclause 8.2.1). However, if bit 2 of byte 1 of CI 1 of the interpretation block data (see subclause 6.7.3) is set at 1 for any of these services, then the details for that service are not intended for storage.

  The details that can be stored for each service are as follows:

-	index number	8 bits	see subclause 8.2.1
	packet address	10 bits	
	digital component location	2 bits	
-	audio configuration	3 bits	see table 20.
	scrambling	1 bit	
	coding law	1 bit	
	error protection level	1 bit	

- if any of this information changes for any of the services defined above (except those flagged as not intended for storage) then a different SDFSCR code shall be allocated.

- the first bit of SDFSCR shall be changed to indicate that the configuration identified by the following SDFSCR code has been redefined and all stored information should be updated.

- if the broadcaster does not wish to use the SDFSCR facility, then all bits shall be set to "0". The use of the SDFSCR code in the receiver is explained in Clause A 4.

(MVSCG) Multiplex and video scrambling control group

gives information on the physical signal organization within the satellite channel. The eight bits are subdivided into two subgroups and allocated as follows:

Bit 1
Bit 2
Bit 3 } this time division multiplex configuration (TDMC) sub-group is defined as follows:
Bit 4
Bit 5

- Bit 1: bv = video configuration;

  - if bv = 1, the vision signal shall be compatible with decoders intended for the time compressed components system defined in this ETS;

  - if bv = 0, the vision signal is not compatible.

- Bit 2: bm = sound/data multiplex format;

  - if bm = 1, the sound/data multiplex shall be compatible with decoders intended for the normal burst multiplex, defined in this ETS;

  - if bm = 0, the sound/data multiplex is not compatible;

  - extension of this information, for example for the use of an extended sound/data multiplex or a field-blanking video/data multiplex, is provided by the repeated data frames described in subclause 4.5.4.

- Bit 3: unallocated;

- Bit 4 and Bit 5: vision coding indication;

  - bit 4 and bit 5 shall be set to 0. This bit combination indicates that the HDMAC coding system is applicable to the transmitted signal. See also the NOTE for compatibility with the D2-MAC/packet system.

- Bits 6, 7 and 8: VSAM subgroup;

  - this vision scrambling and access mode (VSAM) subgroup indicates whether the HDMAC video signal is scrambled by one of two techniques (see subclause 5.2), and whether access is controlled or free;

- five combinations are at present allocated:

$$\left.\begin{array}{l}\text{bit } 6 = 0\\ \text{bit } 7 = 0\\ \text{bit } 8 = 0\end{array}\right\}\text{ free access, double-cut component rotation scrambling}$$

$$\left.\begin{array}{l}\text{bit } 6 = 0\\ \text{bit } 7 = 0\\ \text{bit } 8 = 1\end{array}\right\}\text{ controlled access, double-cut component rotation scrambling}$$

$$\left.\begin{array}{l}\text{bit } 6 = 0\\ \text{bit } 7 = 1\\ \text{bit } 8 = 0\end{array}\right\}\text{ free access, single-cut line rotation scrambling}$$

$$\left.\begin{array}{l}\text{bit } 6 = 0\\ \text{bit } 7 = 1\\ \text{bit } 8 = 1\end{array}\right\}\text{ controlled access, single-cut line rotation scrambling}$$

$$\left.\begin{array}{l}\text{bit } 6 = 1\\ \text{bit } 7 = 0\\ \text{bit } 8 = 0\end{array}\right\}\text{ free access, unscrambled}$$

- A change to the transmitted MVSCG (bits 4 to 8 only) shall be made exactly 16 frames before the resultant modification takes effect at the start of the frame following the one in which FCNT = 0 modulo 16, i.e. in frame 1, 17, 33, 49, 65 etc. This timing allows majority logic to be used to assist correct recovery of the data under high error-rate conditions.

- All receivers should decode and use the TDMCTL information defined in subclause 4.5.4.

NOTE: In the D2-MAC/packet system (see ETS 300 250 [1]) bit 4 indicates the aspect ratio and bit 5 is reserved and set to 1. For correct compatible reception of a D2-HDMAC signal by a D2-MAC receiver, such a receiver should ignore bit 5. The combinations of bit 4 and bit 5 that are at present allocated are:

bit 4	bit 5	aspect ratio	coding system
1	1	4:3	MAC
1	0	unallocated	unallocated
0	1	16:9	MAC
0	0	16:9	HDMAC

(CAFCNT) Conditional access frame count

- the twenty most significant bits of a 28-bit frame count (see subclause 9.5.1). This information shall change regularly every 256 frames. The eight least significant bits shall be sent in the Repeated Data Frame (see FCNT in subclause 4.5.4).

(Rp, Fp) Replacement and Fingerprint bits:

- these bits are allocated in the conditional-access systems and are described in the separate specifications.

(SIFT) Service identification channel format:

the bit adjacent to the 14 bit error control group in the static data frame shall indicate that Golay coded packets containing the complete SI data are available in the dedicated packet channel of address '0'. The bit is coded as follows:

- SIFT = 1
    - Golay coded packets can be available within the SI dedicated packet channel but do not contain the complete SI data.
- SIFT = 0
    - Golay coded packets containing the complete SI data are available within the dedicated packet channel.

4.5.4 Repeated Data Frame (RDF) for Time Division Multiplex Control (TDMCTL)

The repeated data frame transmits Time Division Multiplex Control (TDMCTL) information that describes the individual components of the time division multiplex. It shall consist of five successive identical 94-bit data blocks, as shown in figure 12. The coding structure of each RDF block is shown in figure 15.

(FCNT) Frame counter:

an 8-bit counter which gives a cumulative frame count, modulo 256.

The value 0 of the frame counter modulo 128 (seven least significant bits) shall be used to synchronize changes in the TDM configuration and to delimit groups of television frames, as specified below.

The frame counter modulo 256 (all 8 bits) shall be used in conjunction with the conditional-access system specified in Clause 9. See also (CAFCNT) in subclause 4.5.3.

NOTE: The origin of the count is arbitrary, and that there are no discontinuities; it therefore bears no particular relationship to the UDT bits.

(UDF) Update flag:

if non-zero, this bit shall indicate that the RDF contains new information describing the structure of the corresponding TDM component after the next change of TDM configuration (see below). If zero, the information shall describe the present structure.

(TDMCID) TDM Component Identification:

an 8-bit field which shall carry a unique code for every type of TDM component (the size and position of which are defined by TDMS), in accordance with the following table. This specification defines codes for the following components (hexadecimal notation):

'00	TDM component containing no significant signal.
'01 or '02	The area within the television frame reserved for sound/data organized according to figure 2b.
'03-'0F	Other areas within the television frame reserved for sound/data.
'10-'11	Vision signal which is compatible with decoders intended for time compressed components (MAC) system. Code '10 identifies the colour-difference signal and code '11 the luminance signal, in accordance with subclause 5.3.6. (Black level reference is not allocated a code because, as specified in subclause 5.3.6, it shall be transmitted every field following the end of the first colour-difference signal. For normal video conditions as defined in Clause 5, this is on lines 23 and 335.)
'12-'1E	Reserved for future vision applications.
'1F	VPS (Video Programme System). Biphase signal inserted in compressed form in the luminance part of line 16 (temporary national option ; see also subclause 8.2.2).
'20	Field blanking teletext signal according to the fixed-format system (CCIR system B).
'21	Field blanking teletext signal according to the variable-format system (CCIR system A).
'30	Basic field-blanking interval insertion test signals (see subclause 4.3.4).
'31	Cyclical field-blanking interval insertion test signals (see subclause 4.3.4.3.2), which identifies insertion test signals that have a sequence of test line patterns.
'32-'3F	Reserved for other field-blanking interval insertion test signals.

'40-'41 the TDMCID codes '40 and '41 identify the DATV/data multiplex; code '40 relates to the DATV/data multiplex inserted within FBI 1 and code '41 relates to the DATV/data multiplex inserted within FBI 2 (FBI1 and FBI2 as defined in subclause 4.3.2.2). The exact allocation of lines and of bits per data burst within each FBI is described in line 625 by specific TDMS data (see below) introduced by these codes; see the NOTE. Note that the DATV/data TDM component may use non-contiguous TV lines in each FBI. However, the DATV/data TDM component shall be limited to a maximum of 2 subframes per FBI, each of them having the same vertical boundaries.

NOTE: The TDMS data describing TDMCIDs '40 and '41 does not include the leading run-in bit and the trailing spare bit (as described in subclause 4.3.2.2).

'42-'4F allocated to areas reserved for data bursts not divided in two related subframes.

'81 used for panning vectors, see subclause 4.5.6.

(TDMS) Time Division Multiplex Structure

shall define the horizontal and vertical boundaries of subframes allocated to a TDM component in terms of line numbers and clock periods, respectively. One TDM component may comprise one or more subframes, and each TDMS field can define two separate subframes, if required. These shall occupy identical clock periods (e.g. in the definition of the luminance component in fields 1 and 2 of the television frame). The format of the TDMS field is a follows:

- (FLN1) 10 bits: first line number of TDM component subframe 1;
- (LLN1) 10 bits: last line number of TDM component subframe 1;
- (FLN2) 10 bits: first line number of TDM component subframe 2;
- (LLN2) 10 bits: last line number of TDM component subframe 2;
- (FCP) 11 bits: first clock period of TDM component subframe(s);
- (LCP) 11 bits: last clock period of TDM component subframe(s).

Line number 1 shall be coded as binary 0, clock period 1 shall be coded as binary 0; higher numbers are coded correspondingly. All 1's in FLN1, FLN2, etc. represent invalid codes and shall be used to signal undefined subframes. Thus, a TDMS field defining only one subframe has all 1's in FLN2 and LLN2.

The TDMS field of different TDM components shall be defined in such a way that overlapping of existing TDM components does not occur. The one exception to this rule is described in subclause 4.5.5.

NOTE: For scrambled video the boundaries of the subframe shall refer to the format before scrambling and after correct descrambling (see subclause 5.3.6).

(LINKS) Linked structure:

one-bit switch used to link the group of TDMS field(s) needed to fully define one TDM component. This bit shall change on each repetition of the linked TDMS field(s).

TDMCTL data for different TDM components can be sent in any order in successive television frames. Linked structures shall be described in increasing order of FLN1. TDMS fields having the same value of FLN1 shall be transmitted in increasing order of FCP. The number of different TDM components in a D2-HDMAC signal shall never exceed 128.

Any change of the TDM configuration shall be synchronized by the frame counter. New TDMS data, which is flagged by the UDF bit, shall be transmitted prior to the change. New and old TDMS data can be interleaved in any order in successive data frames. The actual change of configuration shall start from the beginning of line 1 of the second frame following the frame in which FCNT code 0 (modulo 128) is sent.

A TDM component that is to be deleted is flagged by the UDF bit, and the TDMS data shall be set to all 1's. The component is deleted after the next change of configuration as described above. The procedure can be repeated several times to increase the probability that no receiver has failed to recognize the deletion.

New TDMCTL data should be sent shortly before any change of configuration in order to minimize the acquisition delay for those receivers which are turned on during this process.

4.5.5 DATV/Data multiplex and FBI teletext multiplex overlapping

The DATV/data multiplex shall have a capacity of not less than 56 packets per video frame, divided over FBI1 and FBI2 (see subclause 4.3.2.2). Much of the time the actually required capacity in the DATV/data multiplex may be less.

By, exceptionally, defining overlapping TDM components for DATV/data and FBI teletext (as defined in subclause 7.2) the unoccupied lines of the DATV/data multiplex can be made available to the FBI teletext service. The end of the active part of the DATV/data multiplex per FBI shall be indicated by a special marker. This marker shall be carried by the PT byte (see subclause 5.8.3.2) of the last packet of the DATV component in the particular FBI. The DATV/data multiplex is effectively terminated at the end of that packet and any spare bits on the remainder of this line shall be set to "0" (before scrambling for spectrum shaping purposes).

This means that in case after the DATV/data multiplex has been terminated some remaining area is not allocated by any other TDM component, this area is set to clamping level (as defined in subclause 4.4.3).

In the case of overlapping TDM components for DATV/data and FBI teletext each FBI shall have at least one DATV packet (containing the special marker). See subclause 5.8.2 for further rules concerning multiplexing of DATV packets.

In the same FBI the FBI teletext may occupy in its allocated window the lines left free by the DATV/data multiplex.

4.5.6 Panning vectors

Panning of 4:3 displays from 16:9 transmissions can be controlled by panning vectors sent in the repeated data frame. Each panning vector shall be transmitted as a 2's complement byte, giving the offset from the centre position. The centre position of the compatible 4:3 picture is defined as starting from unscrambled colour-difference component sample number 48 and luminance component sample number 91. The offset vector shall be an 8-bit number which, when added to the value 48 gives the colour difference sample number of the start of the 4:3 picture. Thus a panning vector having all bits equal to 1 will set the start of the 4:3 picture at colour-difference component sample number 47 (and luminance component sample number 89). The range of panning vectors shall be from - 43 to + 44 inclusive.

Each panning vector shall apply to one frame. The vectors shall be transmitted in groups of 7 for consecutive frames within the TDMS coding space of the RDF, and shall be indicated by the TDMCID value '81. The TDMCID is followed by the 7 vectors, each of 8 bits, sent LSB first. The remaining 7 bits in TDMS and LINKS in each of the TDMCTL frames used for panning information shall be unallocated and set to 1. Their values shall be ignored except for calculation of the error control group.

The first vector after TDMCID = '81 shall apply to the second frame after the frame containing the panning vectors. The second vector shall apply to the following frame, and so on. Thus if the vectors are transmitted in frame n, the first vector applies to frame n + 2, and the last vector applies to frame n + 8.

In normal operation, when panning is in effect, every seventh frame shall contain panning vector information in line 625. In exception, for occasional events such as programme item changes, a new set of panning vectors may be sent in an earlier frame than the seventh frame. In this case, the new set of vectors takes priority over the previously received set.

NOTE: D2-HDMAC receivers have by definition a 16:9 display aspect ratio, and therefore do not require panning vectors. Panning vectors are supported by the D2-HDMAC/packet system for compatibility with the D2-MAC/packet system. D2-MAC receivers may have either a 16:9 or a 4:3 display aspect ratio. Those receivers with a 4:3 aspect ratio display may make use of the panning vector information as specified above. When a transmission changes from 4:3 to 16:9 aspect ratio, the initial panning position should be central, and should remain so in default of panning vectors. If, after some time, panning vectors fail to be available for any reason, the decoder should initially maintain the last panning position received. After a period of between 256 and 512 frames, the decoder should make a smooth return to the central position. The effect of any panning control should be cancelled when a transmission changes from 16:9 to 4:3 aspect ratio.

4.5.7 Unallocated bits

Any unallocated bits in the static data frame are set to "1".

Table 2: Details of the UDT coding

Sequence repeats over 25 frames (one second)

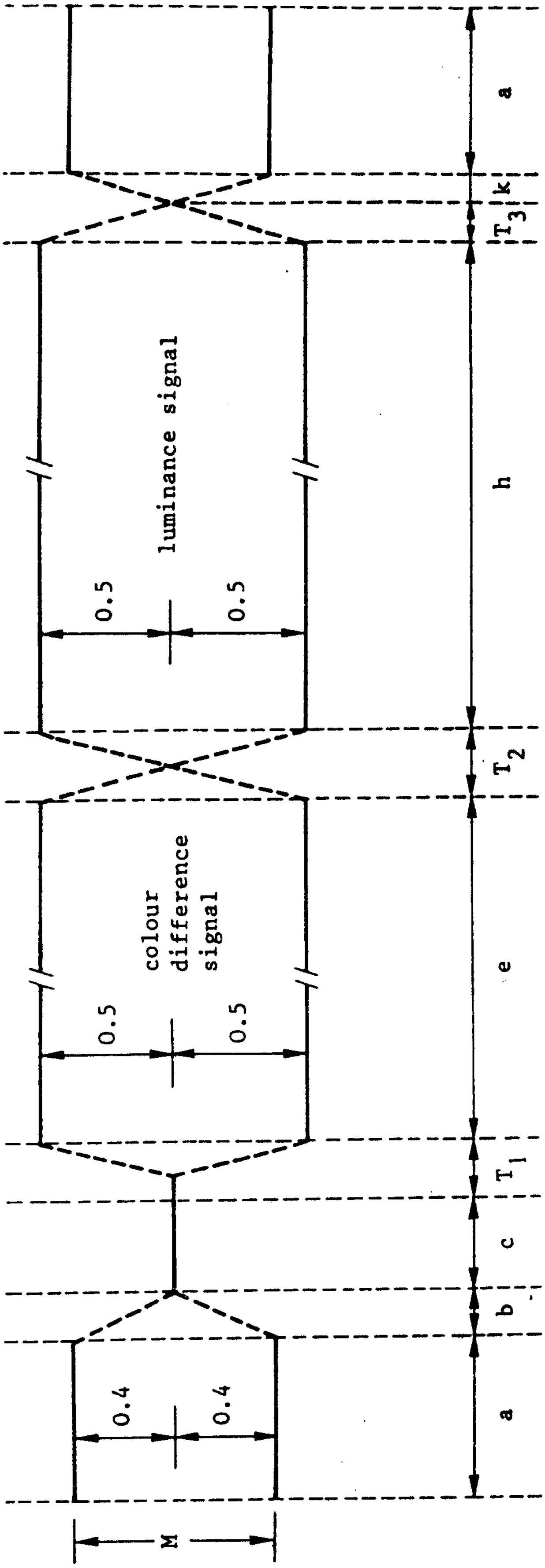
Information relates to next second

	Chain code sequence	Example data	Meaning of 4-bit data
Seconds Marker	0	4	MJD digit weight 10^4
	0	5	" " " 10^3
	0	4	" " " 10^2
	0	4	" " " 10^1
	1	3	" " " 10^0
	0	0	UTC hours tens
	1	7	" " units
	0	3	" minutes tens
	1	1	" " units
	1	1	" seconds tens
	1	8	" " units
	0	x	Reserved for future use
	1	x	
	1	x	
	0	x	
	0	0	0 0 signifies this facility is used; LOCAL OFFSET (hours): 0 0 ± 2^3 — Sign (0 = positive, Eastern hemisphere)
	0	4	2^2 2^1 2^0 2^{-1} — Binary weights
	1	x	Reserved for future use
	1	x	
	1	x	
	1	x	
	1	x	
	0	x	
	0	x	
	1	x	

The example corresponds to 1983 April 19   09.31.17 French time

MJD Modified Julian Date

UTC Coordinated Universal Time



a = 209 clock periods for 105 bits of data and sync.

b = 4 clock periods for transition from end of data

c = 15 clock periods - clamp period (0.5 V)

T_1 = 10 clock periods for weighted transition to colour-difference signal

e = 349 clock periods for colour-difference component

T_2 = 5 clock periods for weighted transition between colour-difference signal and luminance signal

h = 697 clock periods for luminance component

T_3 = 6 clock periods for weighted transition from luminance signal

k = 1 clock period for data transition

Figure 1: Approximate baseband signal wave form for normal unscrambled picture transmission (not to scale)
Clock frequency: 20,25 MHz (see figure 25 for more details)

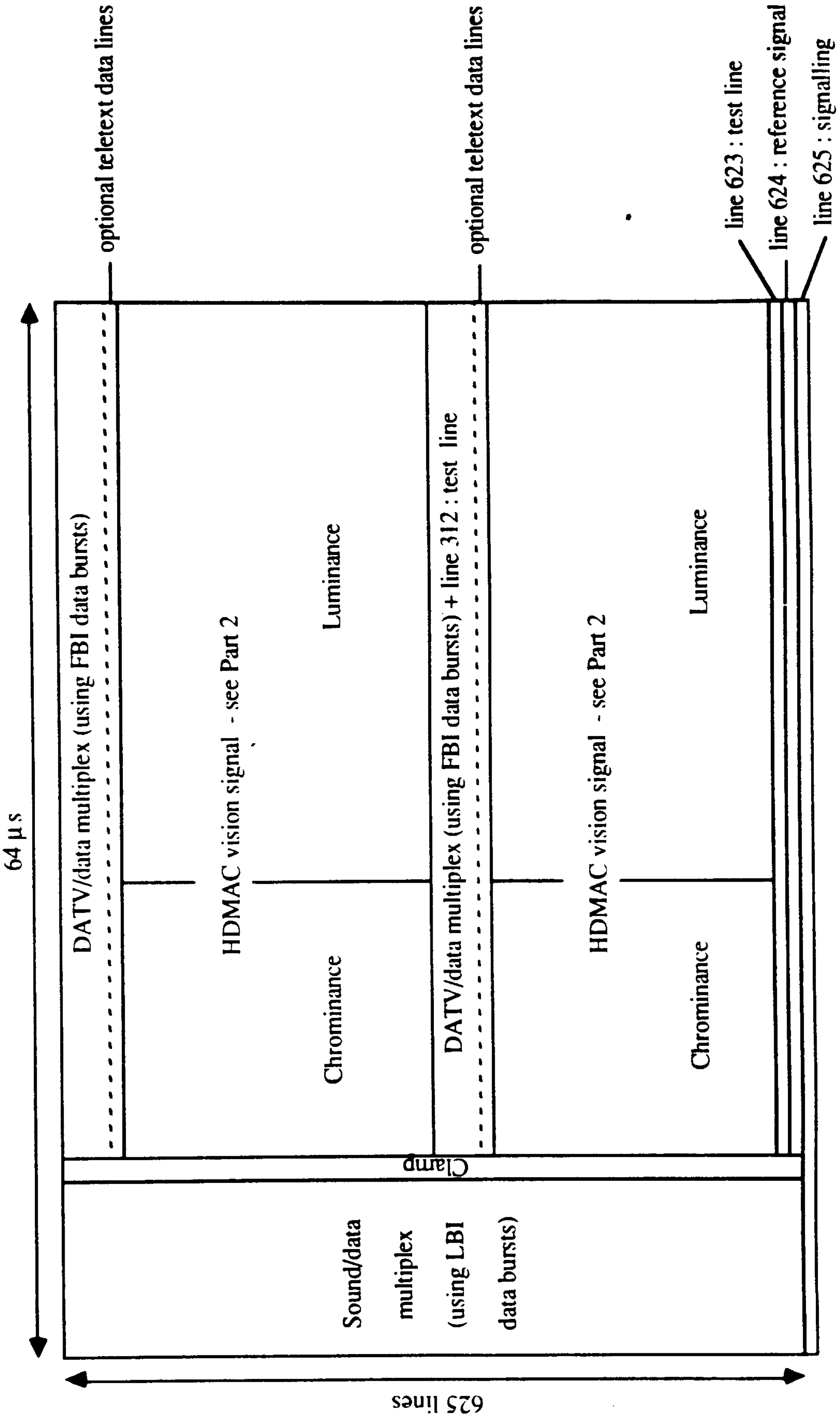


Figure 2a: General HDMAC/packet TDM structure

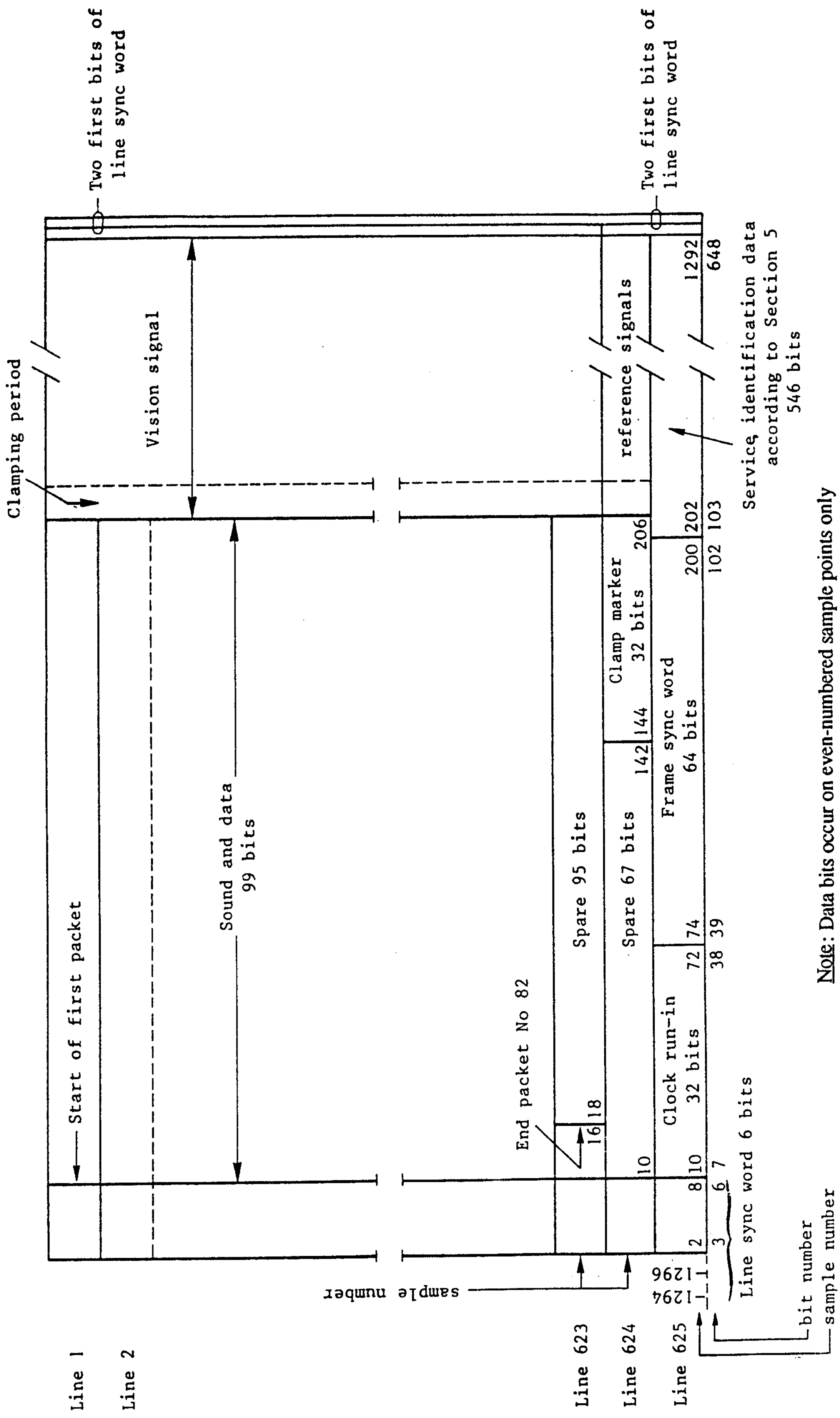


Figure 2b: D2-HDMAC/packet system
details of the TDM structure ; LBI, line 624 and line 625

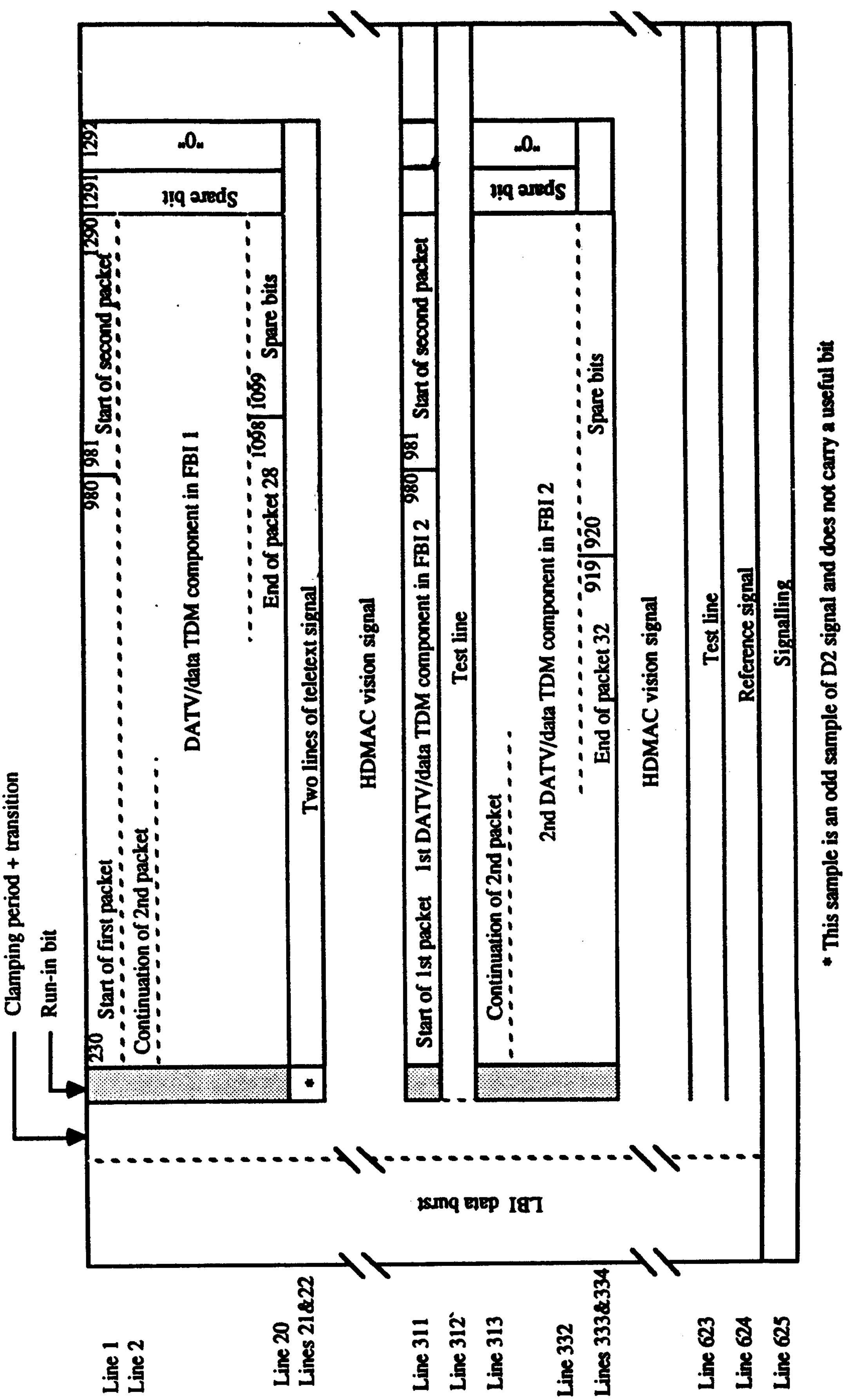
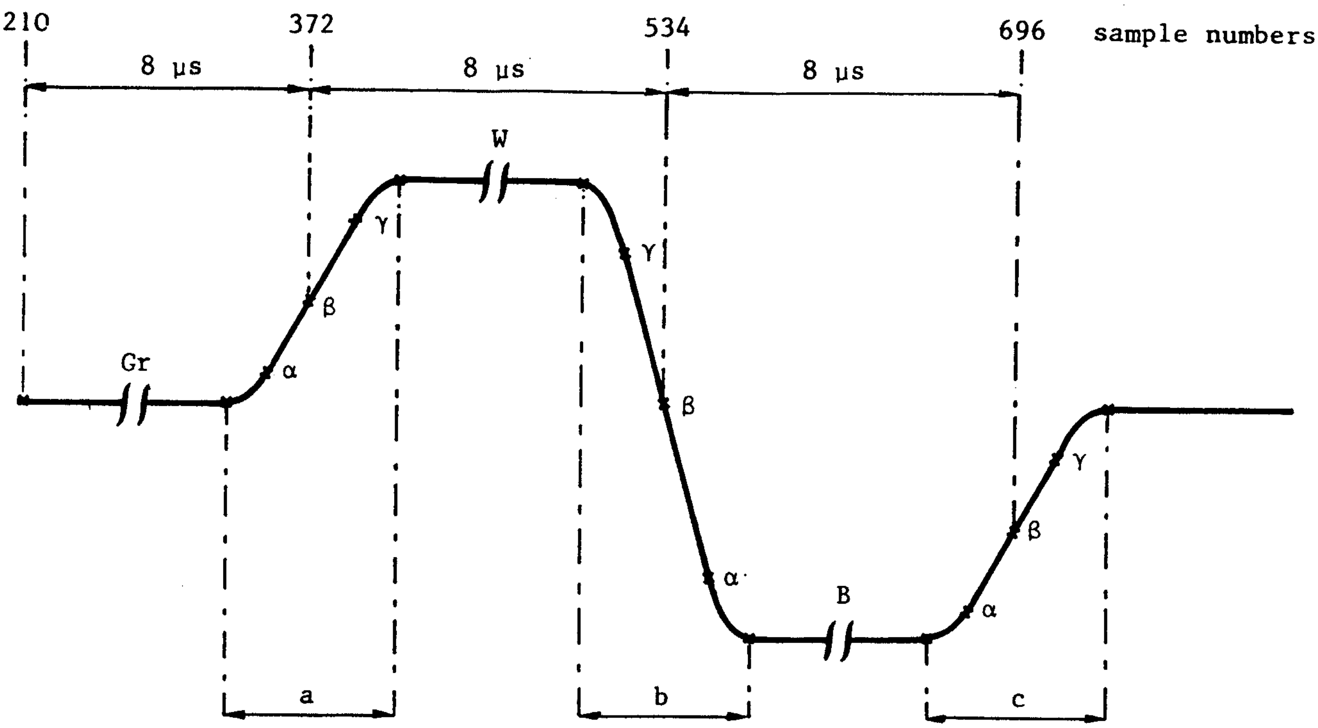


Figure 2c: D2-HDMAC/packet system
details of the TDM structure ; FBI possible configuration



__Transitions__

a:

$\alpha = 7/8\ (Gr) + 1/8\ (W)$
$\beta = 1/2\ (Gr) + 1/2\ (W)$
$\gamma = 1/8\ (Gr) + 7/8\ (W)$

b:

$\alpha = 7/8\ (B) + 1/8\ (W)$
$\beta = 1/2\ (B) + 1/2\ (W)$
$\gamma = 1/8\ (B) + 7/8\ (W)$

c:

$\alpha = 7/8\ (B) + 1/8\ (Gr)$
$\beta = 1/2\ (B) + 1/2\ (Gr)$
$\gamma = 1/8\ (B) + 7/8\ (Gr)$

Figure 3a: Wave form of black, grey and white reference signal in line 624

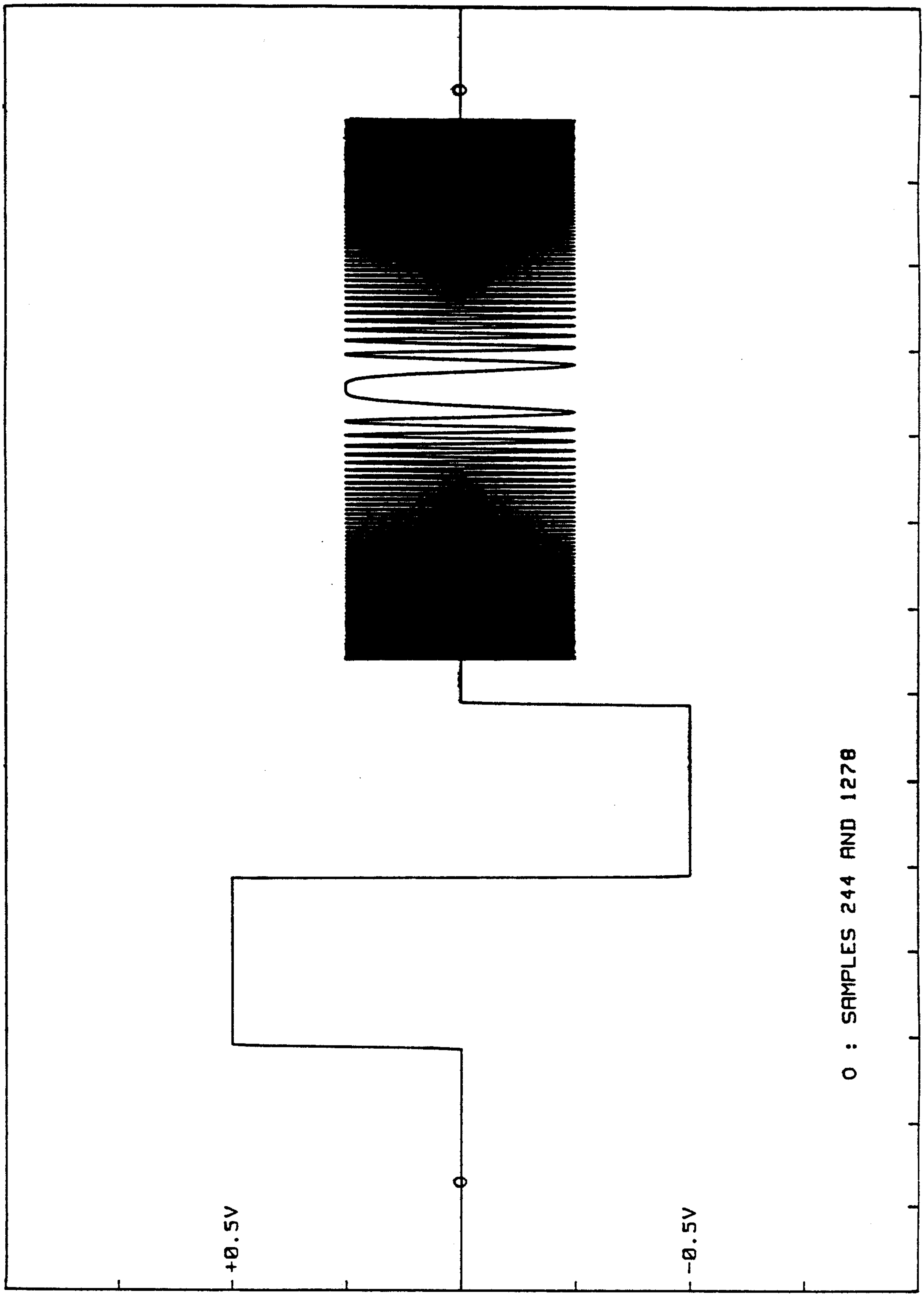


Figure 3b: Line 624 even frame

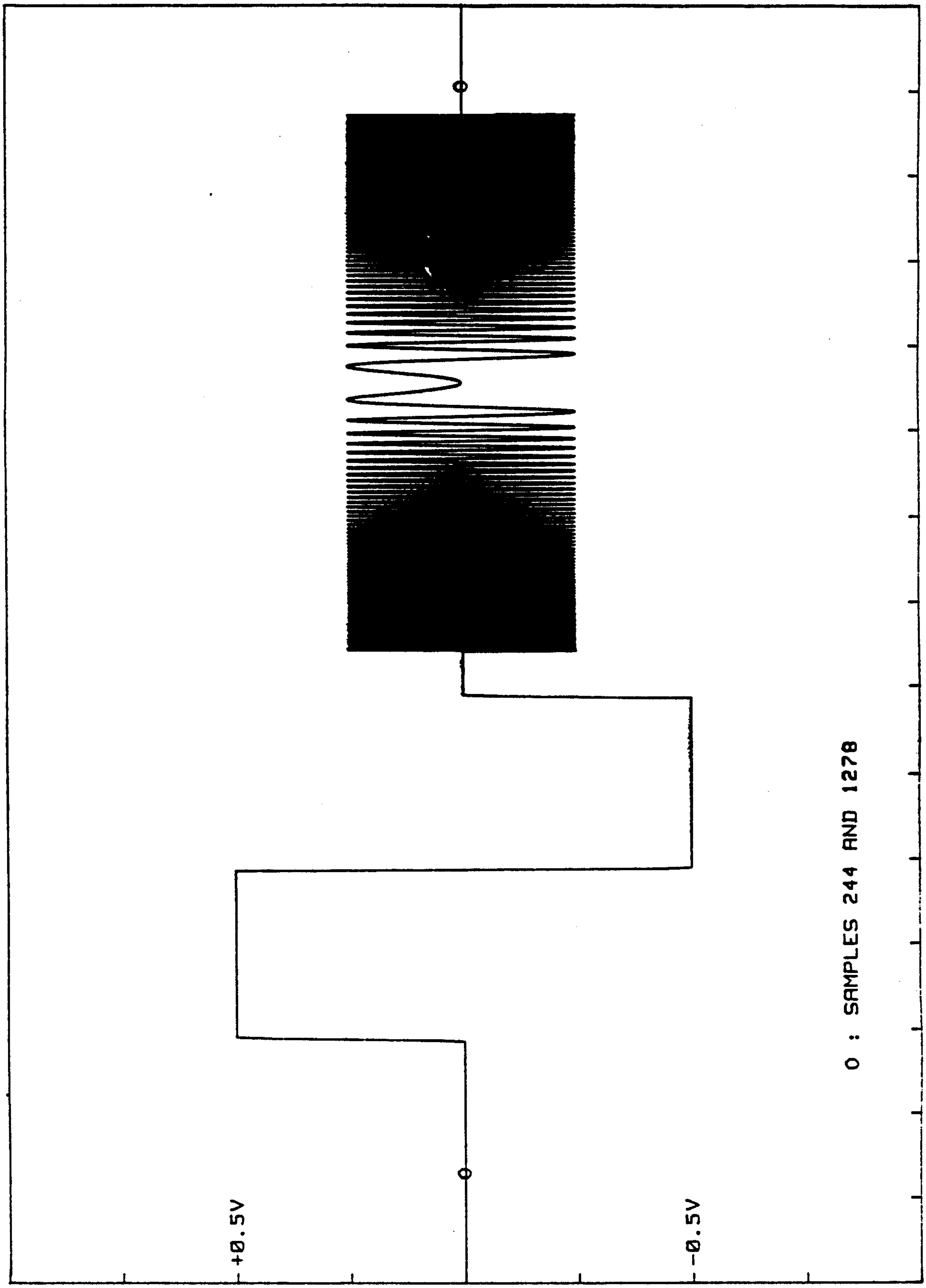


Figure 3c: Line 624 odd frame

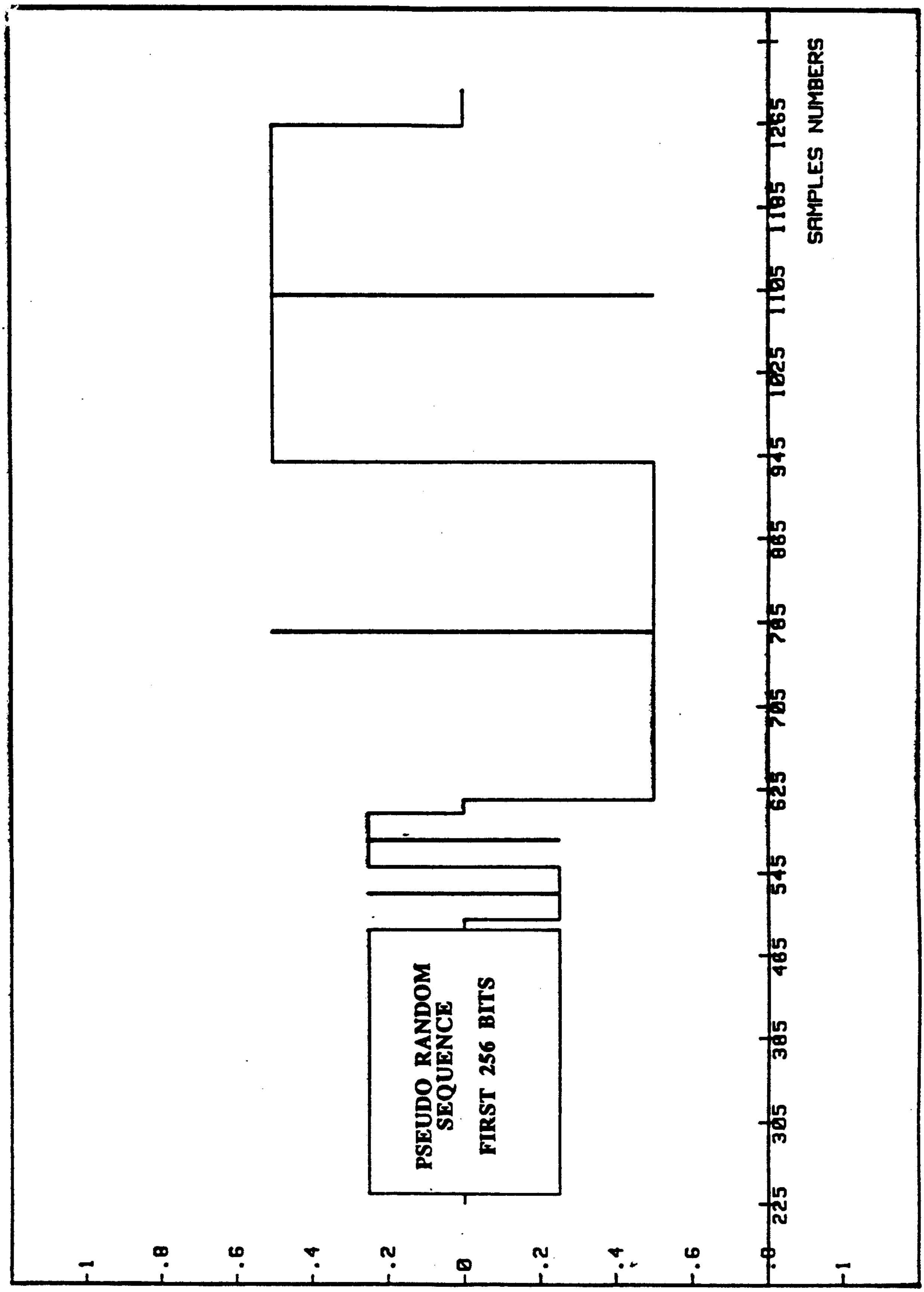


Figure 4a: Line 312 even frame

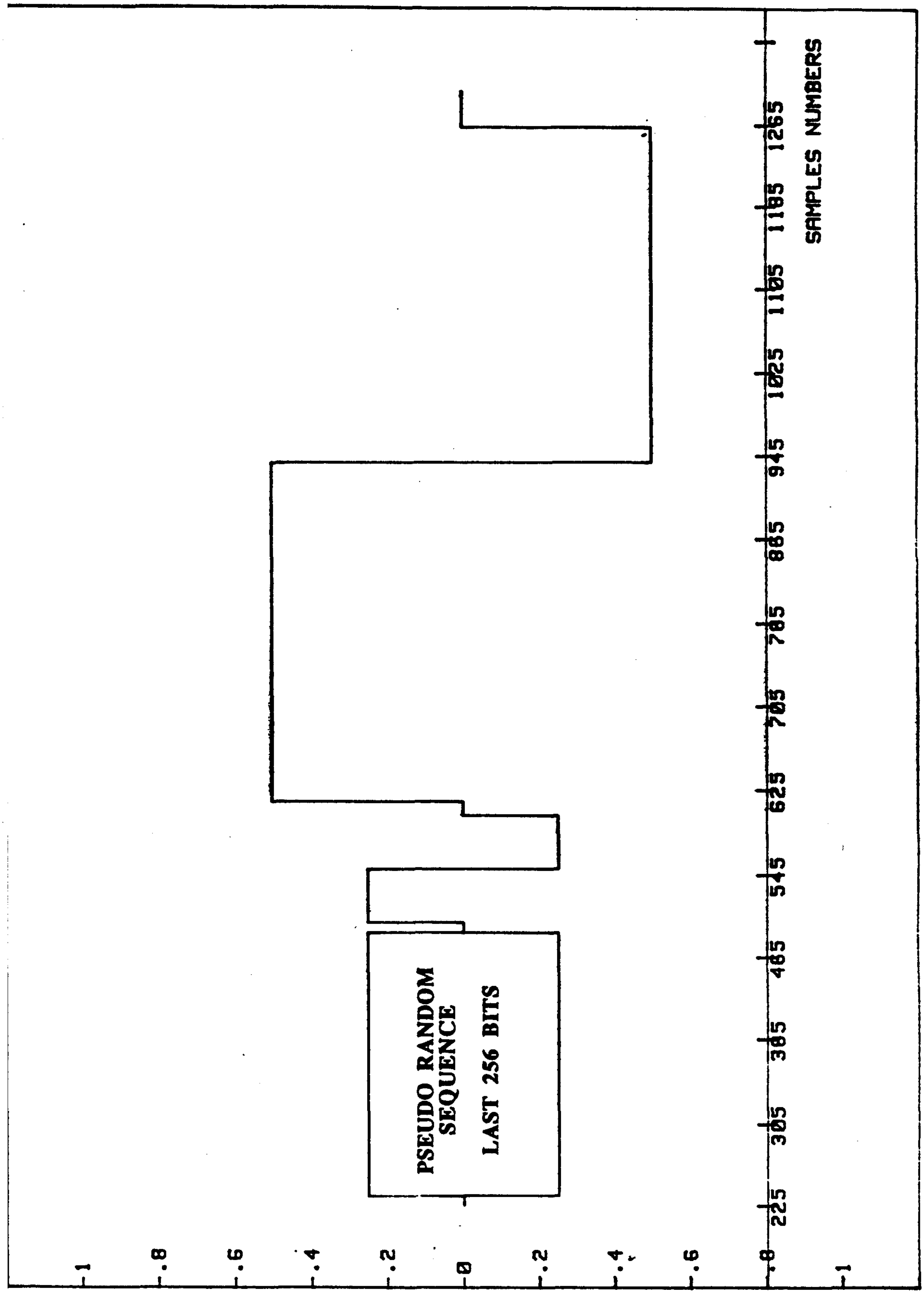


Figure 4b: Line 312 odd frame

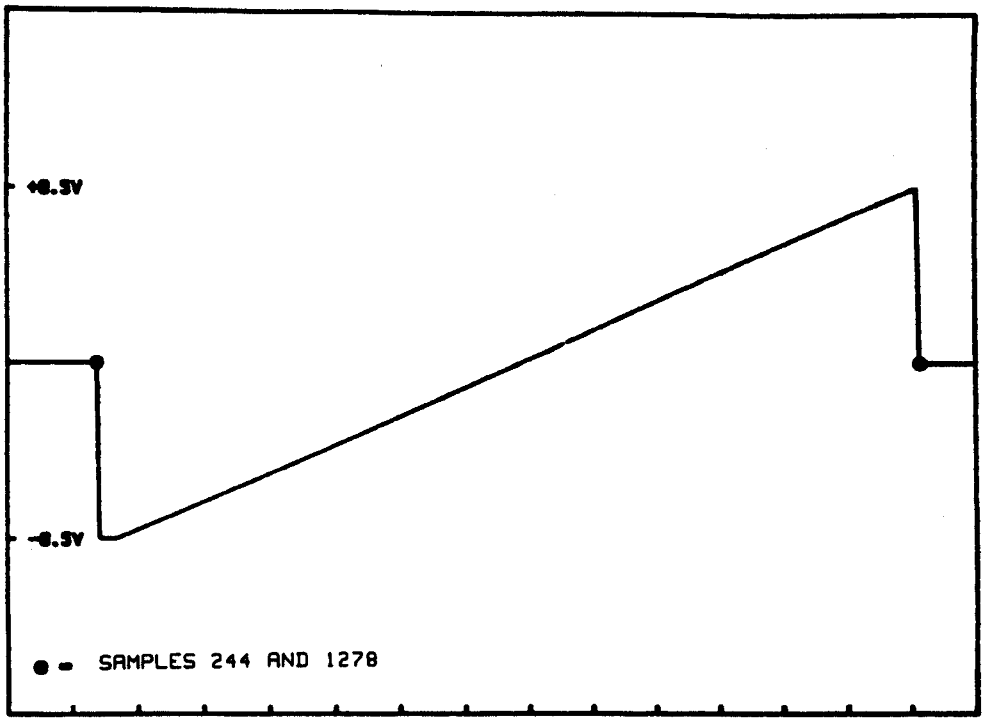


Figure 5a: Line 623 even frame

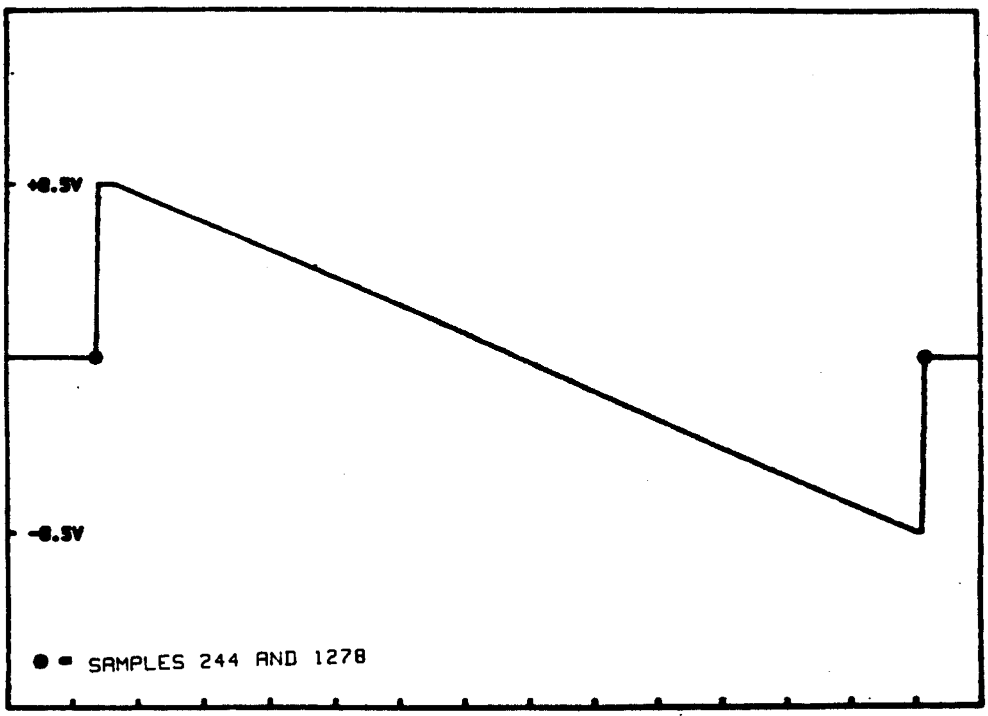


Figure 5b: Line 623 odd frame

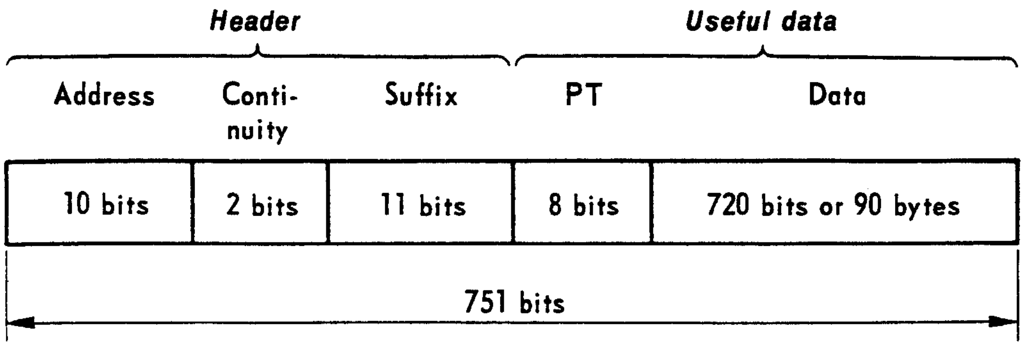


Figure 6: Packet structure

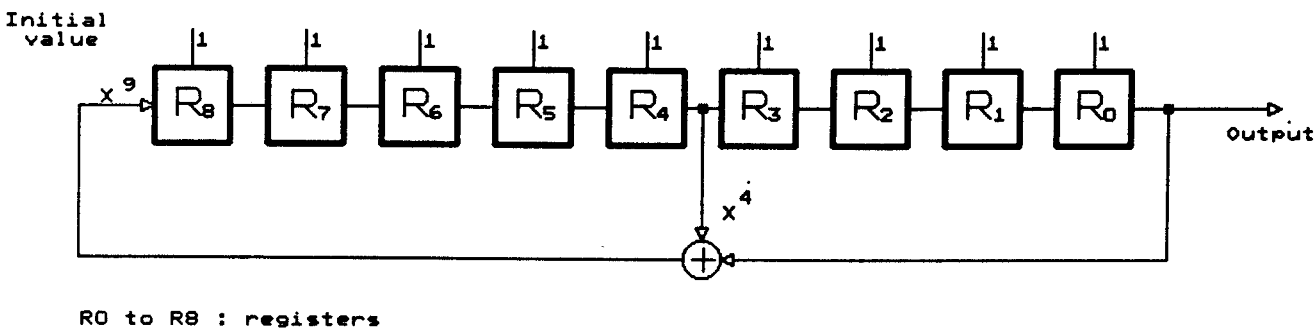


Figure 7: Pseudo random generator for equalization

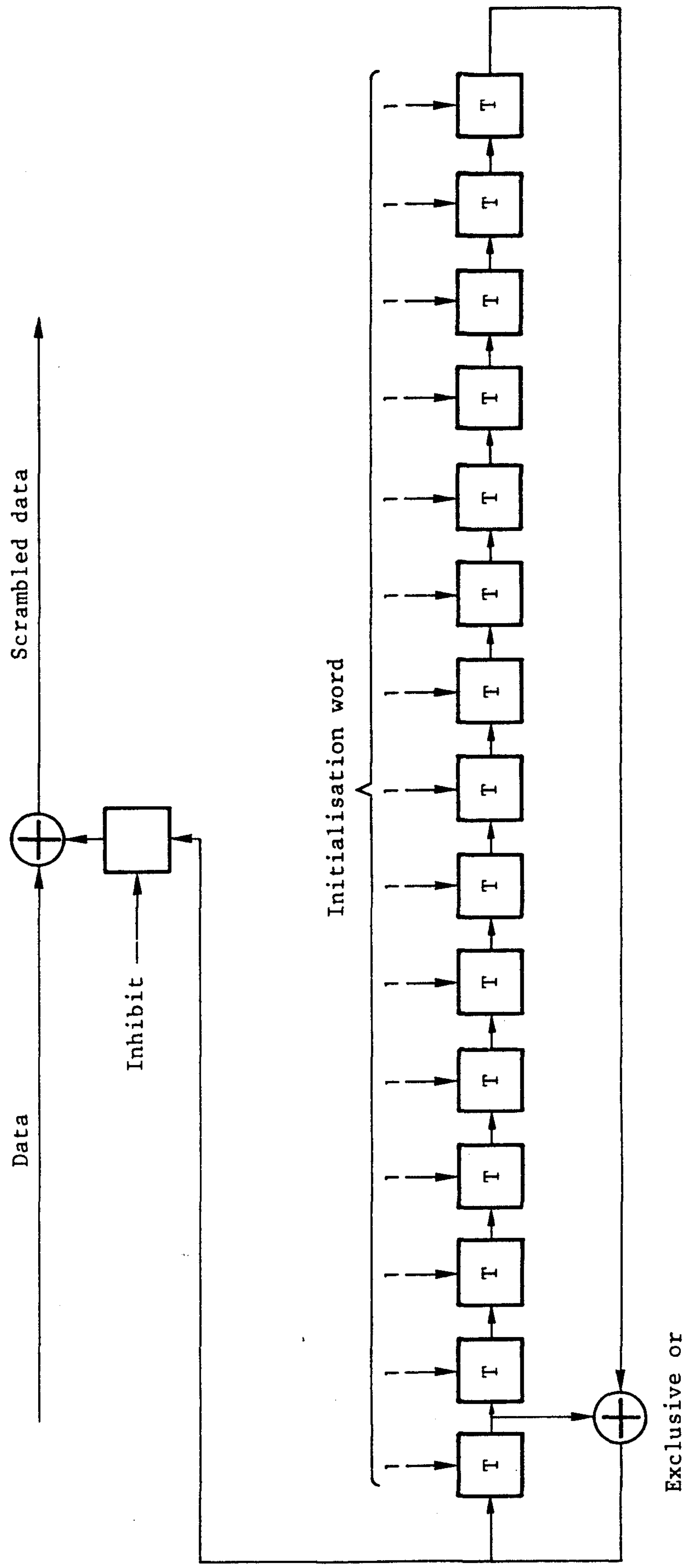


Figure 8: Pseudo-random generator for scrambling and descrambling
(for spectrum shaping purpose)

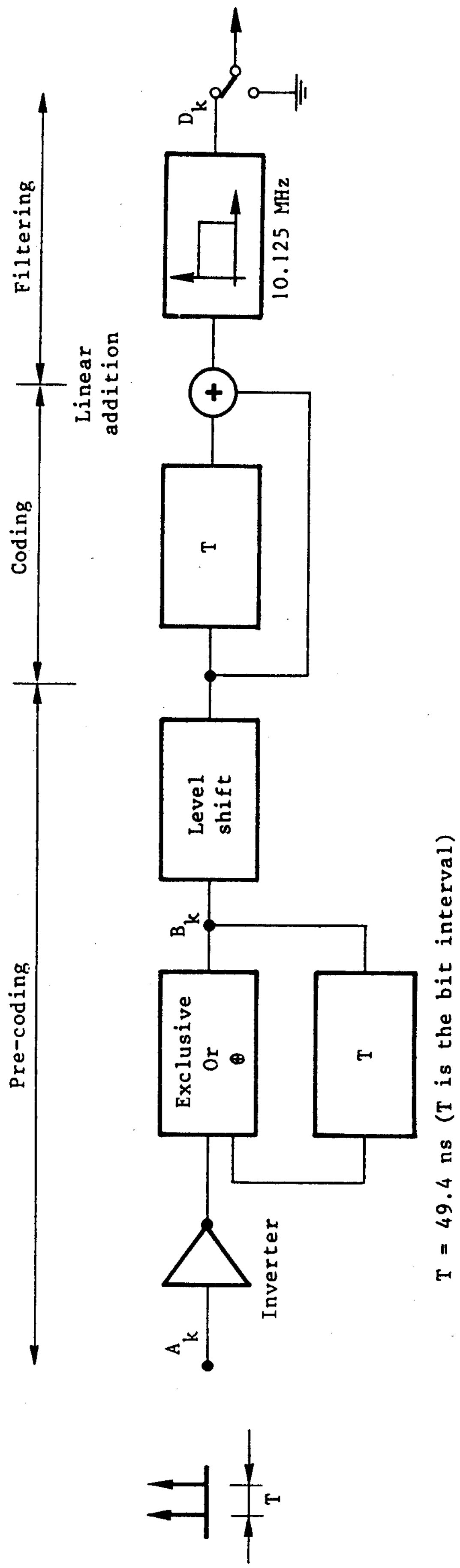


Figure 9a: Coding method of a duobinary signal at 20,25 Mbit/s

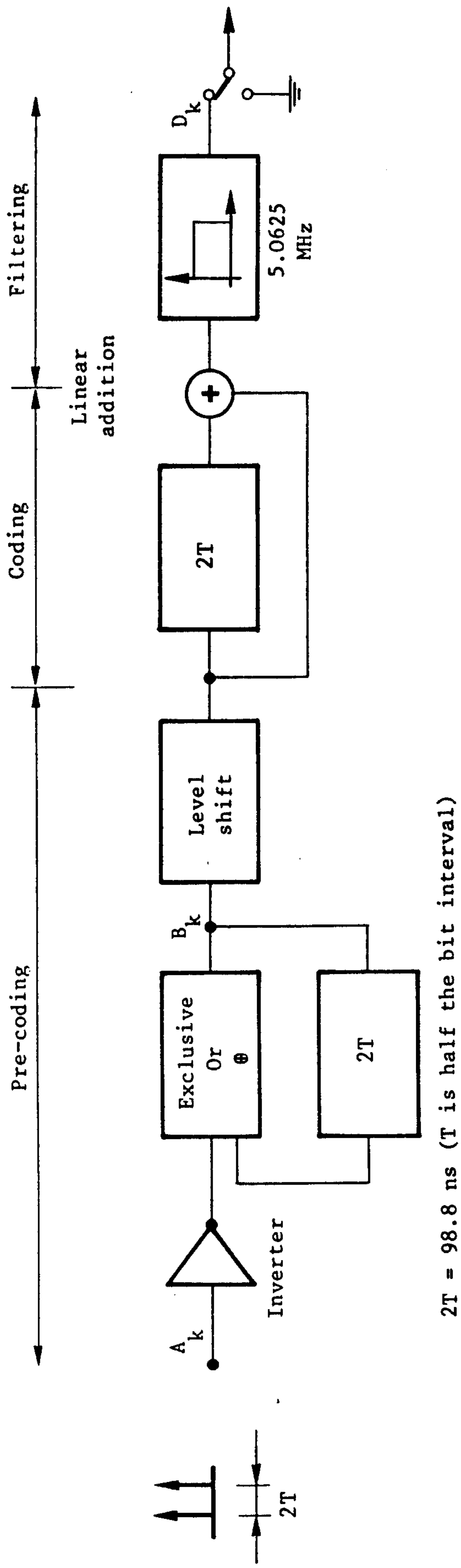


Figure 9b: Coding method of a duobinary signal at 10,125 Mbit/s

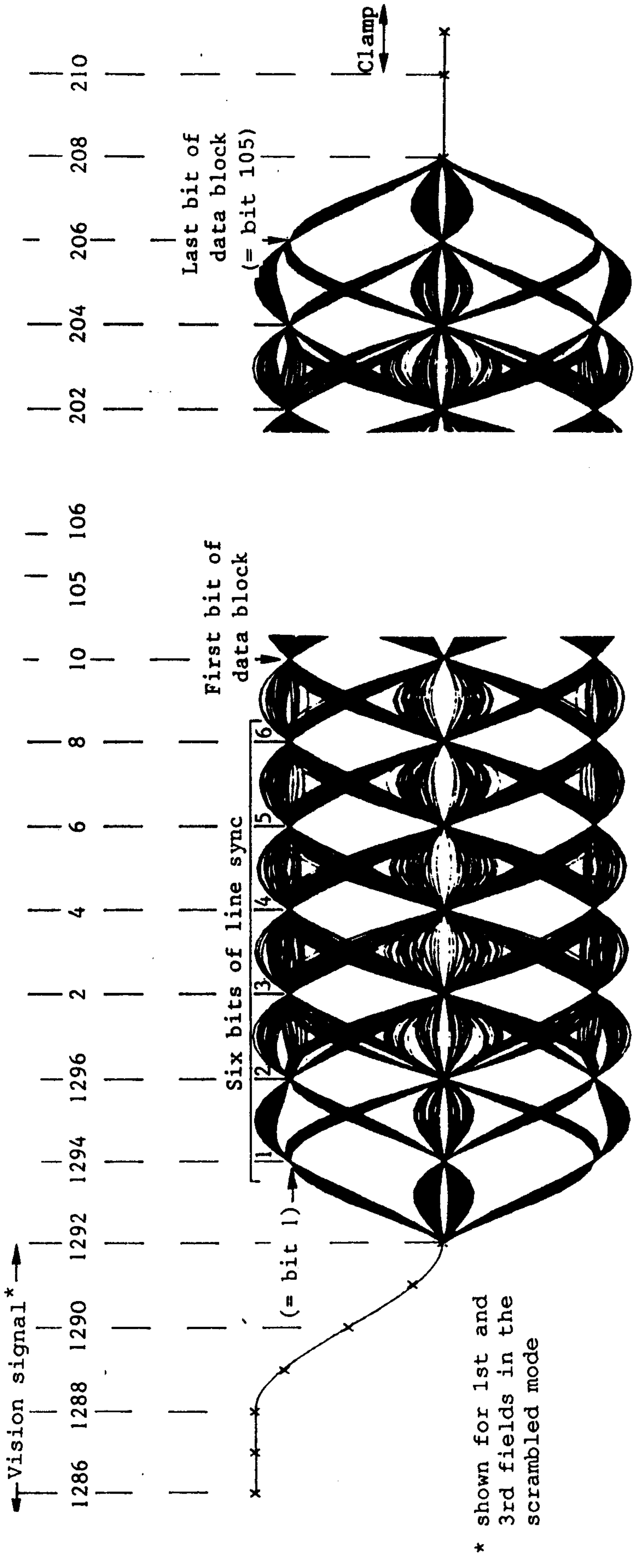


Figure 10a: LBI data period and transition from video

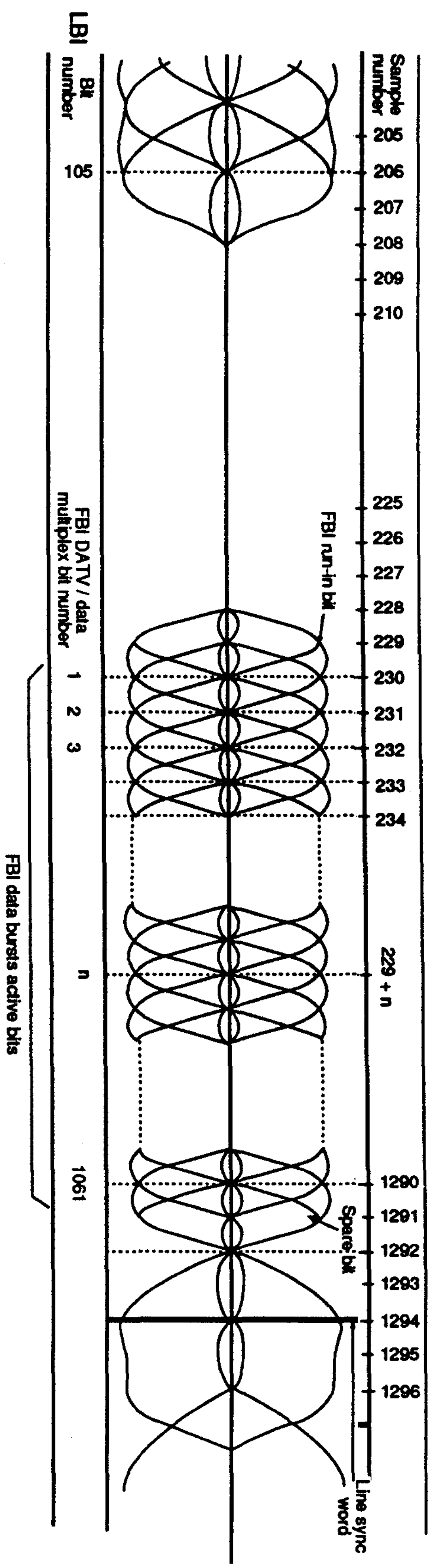


Figure 10b: FBI period for DATV/data multiplex

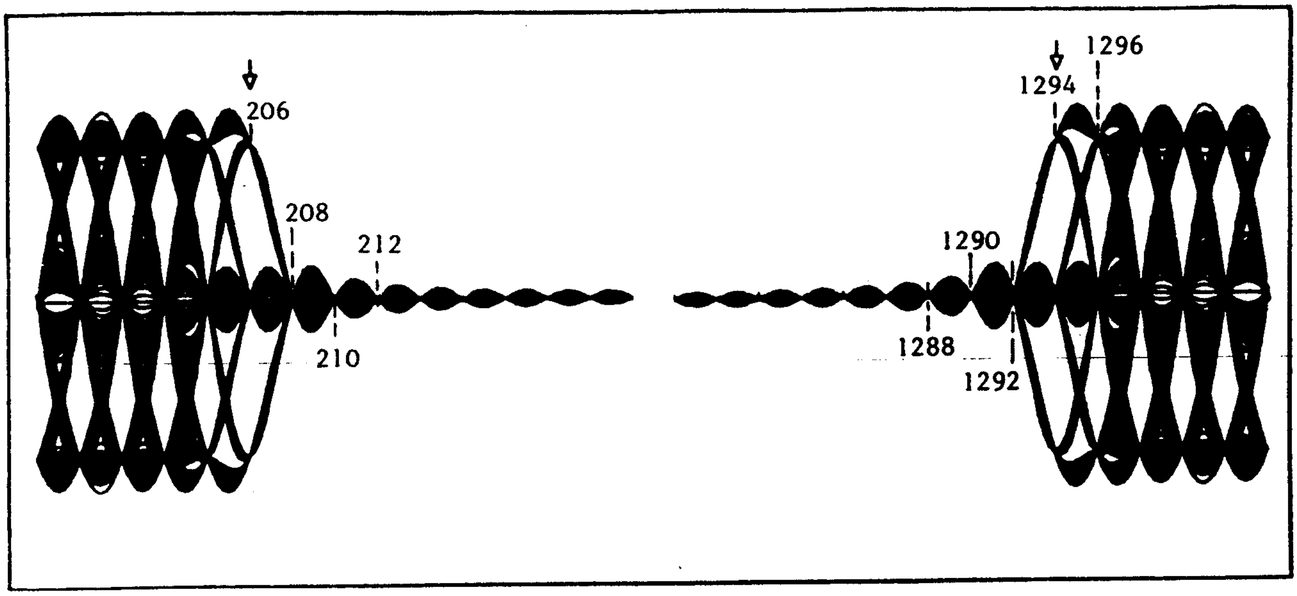


Figure 11a

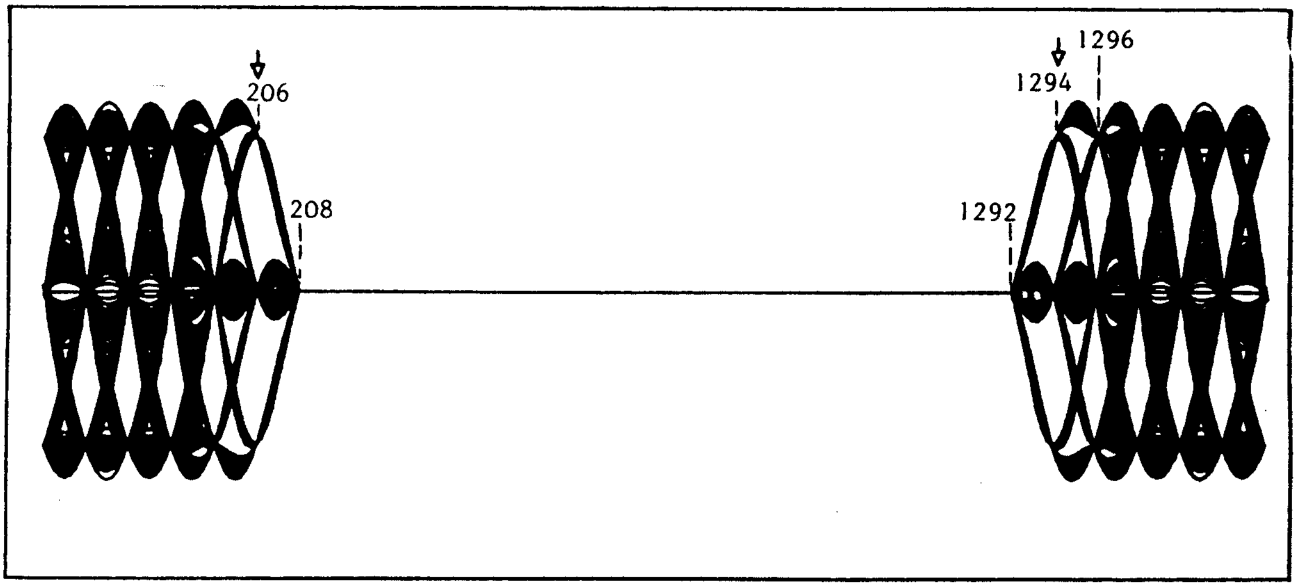


Figure 11b

Data burst transitions obtained by interpolation with a half bit-rate rectangular filter (shown for those lines having a data burst in the line blanking only).

Figure 11a: before forcing to zero clamping and vision periods.

Figure 11b: after forcing to zero clamping and vision periods

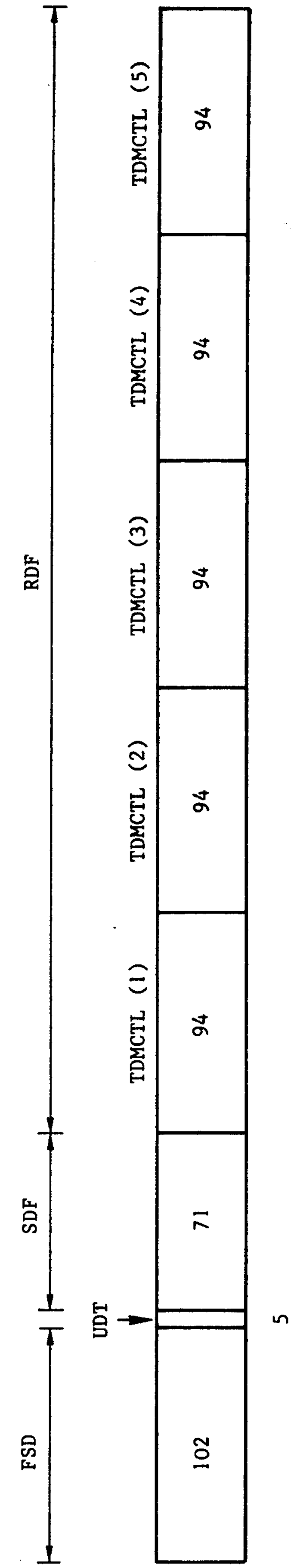


Figure 12: Overall structure of line 625

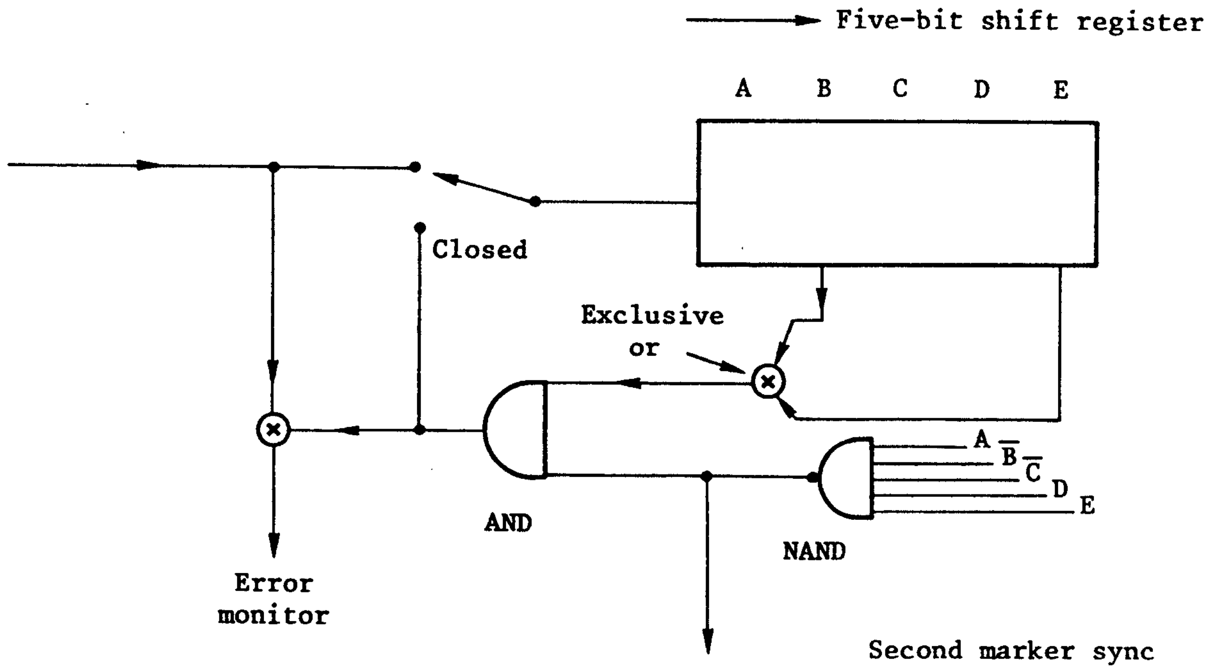


Figure 13: Schematic diagram of sequence generator for the UDT chain mode

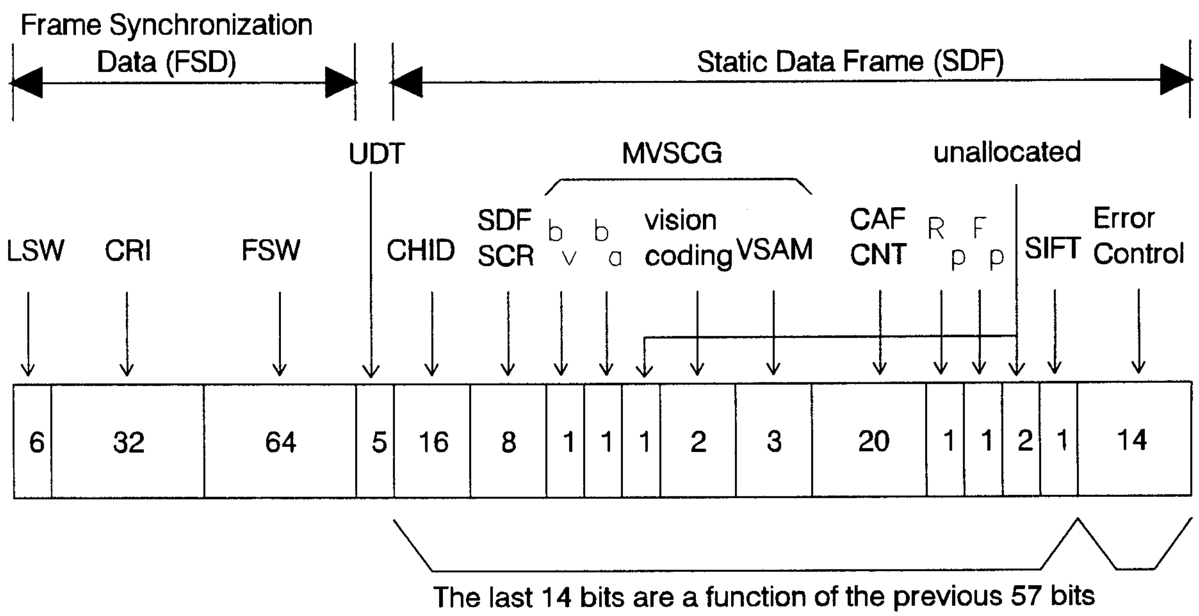


Figure 14: Frame synchronization data, universal date and time part and static data frame block (line 625)

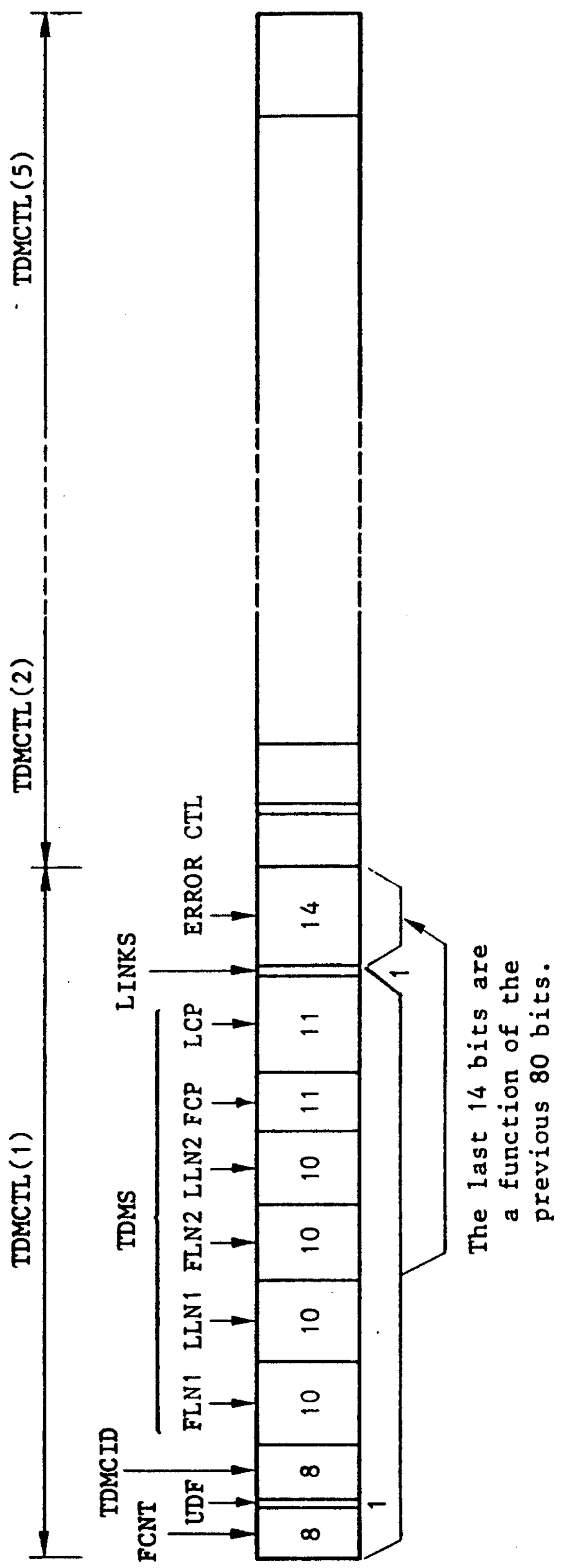


Figure 15: Coding structure of the repeated data frame

5 Specification for the HDMAC vision signal

5.1 Introduction

Clause 5 gives the specification of the HDMAC vision processing.

In order to specify the HDMAC signal as it is emitted, the distinction is made between HDTV source signal characteristics, the bandwidth reduced signal characteristics and the signal characteristics after MAC compression and time division multiplexing.

Subclause 5.2 describes the HDMAC picture coding, which transforms an HDTV source picture into the HDMAC bandwidth reduced vision components and its related DATV component.

Subclause 5.3 discusses the general video characteristics of the HDMAC compressed vision signal.

Subclause 5.4 describes the synchronization of the HDMAC bandwidth reduced component signals and the TDM.

Subclause 5.5 specifies the characteristics of the HDMAC time compressed vision components.

Subclause 5.6 specifies the colour equations used for HDMAC.

Subclause 5.7 specifies the picture scrambling methods.

Subclause 5.8 specifies DATV related subjects. It defines the DATV coding and error protection schemes. Furthermore the DATV packet structure and conditional access scrambling for the DATV component are discussed here.

5.2 Description of the HDMAC Bandwidth Reduction Encoding (BRE) system

The part of the HDMAC picture coding system that transforms the HDTV source picture into a CCIR recommendation 601 [6] compatible component format is generally referred to as the Bandwidth Reduction Encoding (BRE) system. Its inverse operation is performed in the HDMAC receiver and is called the Bandwidth Restoration Decoder (BRD).

A notional block diagram of the BRE system is depicted in figure 16. The following subclauses will discuss this block diagram in more detail. For some of the sub-systems only a notional transfer function is given, since these sub-systems can be implemented in several ways and with several levels of optimization. Other sub-systems are defined in detail.

The luminance and each of the colour difference components of the HDTV source picture are processed by the BRE system in separate branches.

5.2.1 HDTV source signal characteristics

The HDTV source signal shall have a total number of 1 250 lines per frame, of which 1 152 shall be active lines. The field frequency shall be 50 Hz. Two interlace ratios may be applied: 2:1 and 1:1. These two formats are denoted as 1 250/50/2 and 1 250/50/1 respectively. See table 3.

The nominal sampling frequency for the luminance signal at the input of the BRE system shall be 72 MHz for the 1 250/50/2 format and 144 MHz for the 1 250/50/1 format. The sampling frequency shall be line locked, resulting in an orthogonal sampling grid.

The nominal sampling frequency for the colour difference signals at the input of the BRE system shall be 36 MHz for the 1 250/50/2 format and 72 MHz for the 1 250/50/1 format. The sampling frequency shall be line locked, resulting in an orthogonal sampling grid. This sampling frequency is exactly half the sampling frequency of that used for the corresponding luminance signal.

For this ETS all luminance sample positions of the BRE system are referenced to a nominal 54 Msample/s orthogonal sampling structure, which is further referred to as the HD sampling grid. The 72 or 144 Msample/s luminance input format is transformed by a Sample Rate Converter (SRC) (see figure 16) to the HD sampling grid. This grid constitutes 1 728 samples per line with 1 440 active samples per line and 1152 active lines. However, note that the HDMAC transmission does not allow all active samples to be emitted (see subclause 5.3.6).

Furthermore all colour difference sample positions of the BRE system are referenced to a nominal orthogonal sampling structure having exactly half the sample rate as used for the luminance samples. The 36 or 72 Msample/s colour difference input format is transformed by a sample rate converter (see figure 16) to the HD colour difference sampling grid. This grid constitutes 864 samples per line with 720 active samples per line and 1 152 active lines and is a subset of the HD sampling grid for luminance. However, note that the HDMAC transmission does not allow all active samples to be emitted (see subclause 5.3.6).

The odd numbered luminance samples and the colour difference samples shall be spatially co-incident.

The numbering convention for lines and samples on the HD luminance and colour difference grids is such that the first line or sample is assigned number 1.

Table 3: Basic video characteristics of the BRE system

Scanning rate of source signal	1 250 lines / 50 fields / 1:1 interlace 1 250 lines / 50 fields / 2:1 interlace
Sampling structure of the luminance signal (HD sampling grid)	orthogonal with 1440 active samples per 1 152 active lines
Sampling structure of the colour difference signal (HD sampling grid)	orthogonal with 720 active samples per 1 152 active lines
Clock rate HD sampling grid	54 Msample/s
Maximum reproduced signal bandwidth	Luminance : 24 MHz Chrominance : 12 MHz
Synchronizing signal	as in MAC/Packet family

5.2.2 General description of BRE system

5.2.2.1 Introduction

The bandwidth reduction system is based on the principle of sub-Nyquist sampling. The idea behind this principle is that not all information needs to be transmitted in one temporal instant, but that, depending on the resolution and amount of movement, the information can be transmitted in a number of temporal instants which combined have all the information of the source picture. This operation transforms high spatial resolution of a low temporal frequency into a lower spatial resolution with a higher temporal frequency. The information can subsequently be restored to its original spatial resolution by storing and re-assembling in a memory device. The samples to be transmitted in each of the temporal instants are chosen in such a manner that each of these instants represents a subsample phase of the source picture. This ensures that D2-MAC/packet decoders receive compatible vision information resulting in a displayable picture of good quality.

5.2.2.2 Compatibility improvement

The compatibility of the picture with D2-MAC/packet receivers is improved from the original raw processing by a combination of special filters. They are applied only on the luminance component of the signals. These filters largely suppress the visibility of the HDMAC coding artefacts within the signal before the HDMAC vision restoration process in an HDMAC receiver. In the BRD the inverse of the compatibility improvement filtering is performed.

5.2.2.3 BRE coding modes

The BRE system uses three different luminance coding modes, each of which is associated with a given velocity range (see table 4a). These modes are termed the 80 ms mode, the 40 ms mode and the 20 ms mode. The 80 ms mode is meant for stationary parts of the picture to be reproduced with high spatial resolution. The 40 ms mode is associated with limited velocities and will reproduce with medium resolution. The 20 ms mode is associated with low spatial resolution and high velocities. An outline of the process for deriving mode decisions is described in the following subclauses. This process is not defined here normatively since several ways of deriving the decision information are possible.

The colour difference signals of the BRE system are also coded using three possible modes; the choice of mode is derived from the luminance decisions. These modes are also termed the 80 ms mode, the 40 ms mode and the 20 ms mode.

For the 80 ms mode, for stationary information and high spatial resolution, the information is transmitted during 4 consecutive fields (80 ms). This gives a movement portrayal of 12,5 pictures per second. Similarly, for slowly moving parts of the picture (40 ms mode), where only medium resolution is required, sub-Nyquist sampling is applied. The picture is transmitted during two consecutive fields (40 ms). This gives a movement portrayal of 25 pictures per second. The 20 ms mode is used for fast moving parts of the pictures. These parts do not need to be restored with high spatial resolution. Here a movement portrayal of 50 pictures per second can be achieved.

Table 4a: Properties of block motion estimation

Mode	velocity indication	nominal velocity rang
80 ms	stationary + slowly moving	0 - 0,5 samples / 40 ms
40 ms	moving	≤ 12 samples / 40 ms
20 ms	rapid movement + sudden changes of picture content	> 12 samples / 40 ms

Table 4b: Properties of luminance coding modes

Mode	spatial resolution limits		pass band shape of spatial frequency domain
	horizontal	vertical	
80 ms	720 cpw	576 cph	diamond (see figure 17)
40 ms	720 cpw	288 cph	diamond (see figure 17)
20 ms	360 cpw	288 cph	diamond (see figure 17)

Table 4c: Properties of colour difference coding modes

Mode	spatial resolution limits		pass band shape of spatial frequency domain
	horizontal	vertical	
80 ms	360 cpw	288 cph	diamond (see figure 18)
40 ms	180 cpw	144 cph	rectangular (see figure 18)
20 ms	180 cpw	144 cph	diamond (see figure 18)

NOTE: In tables 4b and 4c cph is the abbreviation of cycles per picture height, and cpw is the abbreviation of cycles per picture width.

The BRE system subdivides the samples of the luminance HD grid into blocks of 16 samples by 16 lines. The blocks of the HD colour difference grid are spatially co-incident with the luminance blocks and have 8 samples by 16 lines. The samples belonging to a specific block are all treated in an identical manner. They form the basic elements for which mode decisions and motion vectors are estimated.

5.2.2.4 BRE modes of operation

The BRE system has two modes of operation, which depend on the temporal resolution of the source signal. The "film" mode is intended for input material (e.g. film) which has 25 pictures per second (i.e. 25 movement phases per second). The "camera" mode is intended for input material (e.g. electronic camera) which has 50 pictures (i.e. movement phases) per second.

The BRE has a further fixed colour mode. When the fixed colour mode is active the luminance processing shall be unaffected, but the colour difference processing shall be forced to the 20 ms mode irrespective of the choice of the processing mode and the block decisions.

In the case of film-mode only the 80 ms and 40 ms modes shall be active for both the luminance and the colour difference branches and the 40 ms motion vectors shall be forced to zero (resulting in temporal interpolation between spatially co-incident samples). For the camera-mode all three modes shall be active for the luminance processing branch. For the colour difference branches in the case of camera-mode only the 80 ms and 20 ms modes shall be active. Table 5 shows the derivation of the colour difference mode decision signal from the luminance mode decision signal for each of the modes of operation.

Table 5: Derivation of colour difference modes from luminance modes

	luminance mode				
	film-mode		camera-mode		
	80 ms	40 ms	80 ms	40 ms	20 ms
fixed colour mode	20 ms	20 ms	20 ms	20 ms	20 ms
non fixed colour mode	80 ms	40 ms	80 ms	20 ms	20 ms

5.2.2.5 Block decision and motion estimation

The block decision and motion estimation sub-system, as depicted in figure 16, estimates for each individual block which of the processing branches will optimally reconstruct the source picture after BRE coding and BRD decoding. See also subclause 5.2.4.2. It estimates the speed of movement and its direction. For the 40 ms luminance processing branch for each block a motion vector is also estimated. The block decisions and motion vectors may be post-processed in order to improve their spatial and temporal consistency.

The mode decision signal shall be encoded using five possible temporal routes per 80 ms period. These routes shall be as given in table 7.

The motion vectors have a horizontal and vertical range from -6 to +6 samples of the luminance HD grid per 20 ms (see table 6). This results in 169 possible motion vectors for each block of the first frame in a basic 80 ms period (the odd frame). In the second (even) frame only 9 different vectors are possible per block. These 9 vectors shall be selected from the vector of the same block in the previous frame or one of the vectors of its 8 spatially neighbouring blocks (see subclause 5.8.7.3.4).

The block decision and motion vector information constitutes a digital control signal. The DATV data is transmitted in the FBI as described in subclause 4.3.2.2 and contains all the information required for the decoding process. It allows the decoder to be a complete slave of the encoder.

Table 6: Characteristics of motion vectors

Horizontal range of motion vectors	-12 up to +12 samples / 40 ms
Vertical range of motion vectors	-12 up to +12 picture lines / 40 ms
Precision of motion vectors	*integer per field*
Block size on luminance HD-grid	16 samples by 16 lines
Number of blocks per picture generated in the BRE for an 80 ms coding period	(1 440/16) x (1 152/16) = 90 x 72 = 6 480

Table 7: allowed routes for mode decisions of the BRE system

Mode decision per field					Route	Number of possibilities per route
Field	1	2	3	4		
	80	80	80	80	1	1
	40	40	40	40	2	169 x 9
	40	40	20	20	3	169
	20	20	40	40	4	8
	20	20	20	20	5	1
Total number of possibilities						1 700

5.2.2.6 Motion compensation in the 40 ms mode

Motion compensation should be employed in the HDMAC decoder for the 40 ms luminance restoration process to render a higher spatial resolution during movement. The BRE system shall employ motion compensation in order to accurately predict the quality of the picture after BRE coding and BRD restoration for this processing mode. This information enhances the quality of the BRE block decision and motion estimation process.

The motion compensation process is described in more detail in subclause 5.2.4.2.4.

Motion compensation shall not be employed for the colour difference signals.

5.2.3 Shuffling

In the BRE system the source signal has twice the number of lines and twice the number of samples per line compared to the 625 line output signal. Sample shuffling is applied in order to convert the source sampling structure to the 625 line output structure, while maintaining compatibility of the vision signal with D2-MAC/packet receivers. This results in a field quincunx sampling structure for the emitted HDMAC signal.

The sub-systems of the BRE that perform the shuffling operation for each processing branch are depicted in figure 16 and are named SH_80, SH_40 and SH_20. Shuffling is applied to both the luminance and the colour difference components. The reverse action is performed in the HDMAC receiver.

5.2.4 *Luminance signal processing of the BRE system*

5.2.4.1 Luminance 80 ms processing branch

The luminance 80 ms processing branch as depicted in figure 16 consists of the following sub-systems *(the abbreviated names as used in figure 16 are given within brackets)*:

- 80 ms branch spatial and temporal low pass filter (LPF80);

- sub-Nyquist subsampler (SS_80);

- intra-field shuffler (SH_80);

- temporal compatibility improvement filter (TCI+).

Each of these sub-systems is described in more detail in the following subclauses.

5.2.4.1.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 17. This characteristic gives the notional transfer function of the luminance low-pass filter LPF80 of figure 16. See also table 4b.

5.2.4.1.2 Sub-Nyquist sampling

In the 80 ms mode the BRE system shall use a four phase sub-Nyquist sampling (sub-system SS_80 in figure 16), which takes four emission fields to fully transmit all the samples (see figure 19a). In the encoder the sub-sampling shall be done continuously; this means that all four phases are generated from different temporal instants with 20 ms spacing after the spatio-temporal low-pass filter LPF80 (see figure 16).

5.2.4.1.3 80 ms shuffling

In the 80 ms mode a movement portrayal of 12,5 movement phases per second can be supported. All samples of a high definition input frame are coded for emission after a temporal low-pass filtering during four field periods. Pairs of source lines, spaced by one line on an HD sampling grid, shall be transformed to one line of 64 µs and double the number of samples as shown in (i.e. intra-field shuffling). This 80 ms shuffling has a four phase periodicity, although figures 21a and 21b show shuffling for two fields only. In figure 16 the 80 ms shuffling is performed by the block named SH_80.

5.2.4.1.4 TCI+ Temporal Compatibility Improvement filter

This subclause describes the "TCI+" temporal compatibility improvement filter for both the transmitter and the receiver side. The transmitter TCI+ filter is shown in figure 31 and the receiver TCI+ filter is shown in figure 32.

The transmitter TCI+ filter, as depicted in figure 31, shall be applied every field to the output of the 80 ms shuffler of the luminance coding branch (see figure 1). The n^{th} field of the input signal is termed S(nT), while the n^{th} field of the resulting TCI+ filtered signal is termed S*(nT), with $T = 20$ ms and $D = 2 \times T = 40$ ms.

Block T_F delays S(nT) for a period D (2 fields) giving S(nT-D). The operation:

- $S_{HF}(nT) = S(nT) - S(nT-D)$

constitutes a temporal high-pass filter. $S_{HF}(nT)$ shall be filtered by a symmetrical horizontal low-pass filter F_1, and subsequently fed into the non-linear network N_1. After this first non-linear filtering a pre-correction filter shall be applied. This pre-correction filter shall consist of a 7-tap horizontal low-pass filter F_2 and the non-linear network N_2.

Blocks T_{1a} and T_{1b} constitute a delays, which compensate for any processing delays introduced by the subtractor, F_1, N_1, F_2 and N_2.

The receiver TCI+ filter is depicted in figure 32 and is given for information. This filter has the reciprocal operation of the transmitter TCI+ and restores the signal to its original state as before transmitter TCI+ filtering. It has a recursive structure, since the output of the receiver TCI+ filter is delayed by T_F and then used as a feed-back signal. The delay T_F has a length of two fields, and can be shared with other BRD functions. The input signal is derived by the D2-HDMAC demultiplexer. F_2 is a horizontal low-pass filter and N_2 is a non-linear network (nominally identical to the functions of the same name in the transmitter TCI+ filter). The switch S is closed only if the DATV information corresponding to received signal indicates that the signal has been encoded using the 80 ms mode. Delay block T_2 compensates for any processing delays introduced by the adder, the filter F_2, the non-linearity N_2 and the switch S. In the case where the signal was not encoded using the 80 ms mode, the switch S is kept open and the signal is passed without processing.

The filters F_1 and F_2 are defined in table 8. The transmitter filter F_1 shall have 11 taps, the receiver filter F_2 has 7 taps.

Table 8: Filter coefficients of filters F_1 and F_2

coefficient	F1 (transmitter)	F2 (receiver)
$C_{-5} = C_5$	1/256	-
$C_{-4} = C_4$	-2/256	-
$C_{-3} = C_3$	10/256	7/256
$C_{-2} = C_2$	-29/256	-29/256
$C_{-1} = C_1$	50/256	55/256
C_0	196/256	190/256

The non-linearity N_1 shall be defined by the following formula:

- $$N_1(V_{in}) = p_2 \times \log_e\left(\frac{V_{in} + \sqrt{V_{in}^2 + p_1^2}}{p_1}\right) + (p_3 - 1) \times V_{in}$$

where $V_{in} = S_{F1}(nT) / 2$ (scaling to nominal 1 Vp-p range);

and with parameters p_1, p_2 and p_3 set to

p_1	0,0201
p_2	0,00678
p_3	0,0935

The receiver non-linearity N_2 can be derived via inversion of the transmitter non-linearity N_1.

5.2.4.2 Luminance 40 ms processing branch

The luminance 40 ms processing branch as depicted in figure 16 consists of the following sub-systems (the abbreviated names as used in figure 16 are given within brackets):

- 40 ms branch spatial low pass filter (LPF40);
- vertical compatibility improvement filter (VCI);
- sub-Nyquist subsampler (SS_40);
- inter-field shuffler (SH_40);
- motion compensation compatibility improvement filter (MCCI+).

Each of these sub-systems is described in more detail in the following subclauses.

5.2.4.2.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 17. This characteristic gives the notional transfer function of the luminance low-pass filter LPF40 of figure 16. See also table 4b.

5.2.4.2.2 Vertical Compatibility Improvement (VCI) filter

In the 40 ms luminance branch a 3-tap vertical intra-field low-pass filter shall attenuate the vertical frequency components around 288 cph by 3,5 dB before the bandwidth reduction process. The filter coefficients of the transmitter VCI filter shall be as given in table 9.

At the decoder side, after reconstruction, the frequency components are restored by a reciprocal 3-tap vertical filter. The filter coefficients of the receiver VCI are given in table 9.

Table 9: VCI filter coefficients

	transmitter	receiver
C_0	1 706 / 2 048	10 / 8
$C_{-1} = C_1$	171 / 2 048	-1 / 8

5.2.4.2.3 Sub-Nyquist sampling

In the 40 ms mode the BRE system shall use a two phase sub-Nyquist sampling (sub-system SS_40 in figure 16), which takes two emission fields to fully transmit all the samples (see figure 19b).

5.2.4.2.4 40 ms shuffling

According to the subsampling pattern of figures 21c and 21d the 40 ms mode generates for each frame 625 lines of 720 active samples per active line. The lines to be transmitted are vertically equidistant and two field periods of 20 ms are available for emission. A temporal subsampling shall be performed in the BRE, by using the odd fields for emission and discarding the even fields ; this is known as the field skip method. The lines of these odd fields shall be emitted alternately in the odd and even emission fields (i.e. inter-field shuffling). This results in a movement portrayal for the compatible picture of 25 movement phases per second (see figure 22 and figure 23). In figure 16 the 40 ms shuffling is performed by the block named SH_40.

To render a higher spatial resolution during movement the BRE system shall employ motion compensation. This is done in order to accurately predict the quality of the picture after BRE coding and BRD restoration for this processing mode. As earlier specified in subclause 5.2.2.6, the HDMAC decoder (the BRD specifically) should also employ motion compensation for the 40 ms luminance coding branch for an optimal result.

The even field shall be reconstructed by the BRE motion compensation by interpolating the information received during the two fields of a motion phase. Because of the field skip method the even field is not transmitted and this field shall be reconstructed by means of a temporal interpolation in the direction of the motion vectors between the two adjacent reconstructed odd fields (see figure 22). The principle of the motion compensated interpolation is given in figure 23. The orientation of the motion vectors shall be as defined in figure 23. Note that the Y-axis for the motion vectors is defined differently compared to the spatial subsampling definitions in the other figures. The motion vectors are specified on the HD luminance sampling grid with integer pixel accuracy both horizontally and vertically. Table 6 gives the characteristics of the motion vectors.

5.2.4.2.5 MCCI+ Motion Compensated Compatibility Improvement filter

The "MCCI+" motion compensated compatibility improvement filter shall be applied only to signals of 40 ms mode luminance coding branch (as depicted in figure 16).

The inter-field shuffling, which is used for the 40 ms encoding mode, transmits two consecutive lines of the same odd HD field, on two consecutive transmission fields. This technique would create artefacts known as "judder" in the D2-MAC compatible picture.

The visibility of the judder effect is reduced by transmitting a prediction error of alternate lines of the odd HD field (instead of the lines themselves) during the even transmission fields. The preceding odd transmitted field is projected along the motion trajectory and added to the prediction signal so that the compatible picture exhibits smooth motion. In an HDMAC decoder the alternate lines of the odd HD field are recovered from the transmitted even field by applying inverse processing.

The transmitter MCCI+ filter is depicted in figure 33. The samples transmitted during odd fields shall be passed without processing. Samples of even fields shall be MCCI+ filtered. The block T_{field} in figure 33 constitutes a delay with a duration of one field (20 ms).

A projection of the delayed odd field samples along the motion vector shall be derived by the block PRJ in figure 33. This process is further illustrated by figure 35. If the horizontal position of the projection along the motion vector has a sample position co-incident with the horizontal position of transmitted even field samples (i.e. the motion vector has an odd horizontal displacement), a horizontal 5-tap low-pass filter F_1 shall be applied. Otherwise the sample shall be interpolated using the horizontal 4-tap filter F_2. The filter coefficients of F_1 and F_2 shall be as defined in tables 10 and 11.

NOTE: The horizontal position of required samples is by definition either co-incident with an existing sample or half-way between existing samples, because motion vectors define an integer displacement on the HD sampling grid.

If the vertical position of the projection along the motion trajectory has a sample position co-incident with the vertical position of transmitted even field samples (i.e. the motion vector has a vertical displacement of ± 2 or ± 6 picture lines), then no processing is performed. Otherwise a vertical two tap interpolation filter shall be applied, having coefficients of (0,25 ; 0,75), (0,5 ; 0,5) or (0,75 ; 0,25) depending on the vertical position of the required sample. For example, if the vertical vector component is zero, an interpolation filter of (0,5 ; 0,5) is used, since the position of the sample being generated lies on a line in field 2 half way between two lines in field 1.

The value of the projected field samples shall be clipped by the limiter LIM1 so that the signal amplitude after MCCI+ filtering remains within the nominal range of 1 Vp-p.

Table 10: coefficients of horizontal low-pass filter F_1

coefficients	F_1
$C_{-2} = C_2$	-1/32
$C_{-1} = C_1$	3/32
C_0	28/32

Table 11: coefficients of horizontal interpolation filter F_2

coefficients	F_2
$C_2 = C_{-2}$	-1/8
$C_1 = C_{-1}$	5/8

The block INT performs a bilinear interpolation in order to estimate samples from the even field from the transmitted odd field samples.

The estimation of the samples in the even field from samples in the odd field shall be performed using an equally weighted bilinear interpolation from the four spatially neighbouring odd field samples, as illustrated below.

A	•	B	•	C	•	D	•
•	•	•	•	•	•	•	•
•	E	•	F	•	G	•	H
•	•	•	•	•	•	•	•
J	•	K	•	L	•	M	•

where:

- A, B, C, D, J, K, L and M represent samples of the 40 ms branch transmitted in field 1.
- E, F G and H represent interpolated samples. Sample F is e.g. determined by the equation F = (B + C + K + L) / 4;
- • represents samples on the HD sampling grid that are discarded.

The samples estimated in this way are subtracted from the actual samples in the corresponding positions in the odd HD field, giving the prediction error ε_p. The prediction error ε_p is passed through the non-linear network N, giving the scaled prediction error ε_{sp}. N shall be described by the formula below:

$$\varepsilon_{sp} = \begin{cases} \varepsilon_p & \\ \frac{3}{4} \times \varepsilon_p \pm \frac{1}{112} & \text{for} \\ \frac{\varepsilon_p}{3} \pm \frac{1}{12} & \end{cases} \qquad \begin{array}{l} \left|\varepsilon_p\right| < \frac{1}{28} \\ \frac{1}{28} \le \left|\varepsilon_p\right| < \frac{5}{28} \\ \frac{5}{28} \le \left|\varepsilon_p\right| \end{array}$$

where ± takes the sign of ε_p and ε_p is in units of the coding range.

The output of the MCCI+ filter process for even field samples shall be determined by the addition of the projected odd field samples and the scaled prediction error ε_{sp}, which is subsequently limited to a 1,14 Vp-p range in block LIM2.

The receiver MCCI+ filter is given in figure 34. The receiver filter is given here for information. It operates in a similar manner as the transmitter MCCI+ filter. Again the odd field samples are passed without processing. The filtering of the samples of the even fields is reversed by recovering the scaled prediction error and applying the inverse of the transmitter non-linearity N. The difference between the actually transmitted samples of the even field and the calculated projection along the motion vector derived from the transmitted odd field samples gives the recovered prediction error ε_{rp}, which after passing through the non-linearity N_{inv} gives the correction signal ε_{cor}. The correction signal ε_{cor} is added to the bilinearly interpolated even field sample estimates (from block INT), as in the transmitter MCCI+ filter.

The receiver non-linearity N_{inv} is determined by the formula below:

$$\varepsilon_{cor} = \begin{cases} \varepsilon_{rp} & \left|\varepsilon_{rp}\right| < \frac{1}{28} \\ \frac{4}{3} \times \left(\varepsilon_{rp} \pm \frac{1}{112}\right) \quad \text{for} & \frac{1}{28} \leq \left|\varepsilon_{rp}\right| < \frac{1}{7} \\ \varepsilon_{rp} \times 3 \pm \frac{1}{4} & \frac{1}{7} \leq \left|\varepsilon_{rp}\right| \end{cases}$$

where ± takes the sign of ε_{rp}.

5.2.4.3 Luminance 20 ms processing branch

The luminance 20 ms processing branch as depicted in figure 16 consists of the following sub-systems (the abbreviated names as used in figure 16 are given within brackets):

- 20 ms branch spatial and temporal low pass filter (LPF20);
- sub-Nyquist subsampler (SS_20);
- intra-field shuffler (SH_20).

Each of these sub-systems is described in more detail in the following subclauses.

5.2.4.3.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 17. This characteristic gives the notional transfer function of the luminance low-pass filter LPF20 of figure 16. See also table 4b.

5.2.4.3.2 Sub-Nyquist sampling

In the 20 ms mode the BRE system shall use a one phase sub-Nyquist sampling (sub-system SS_20 in figure 16), which takes one emission field to fully transmit all the samples (see figure 19c).

5.2.4.3.3 20 ms shuffling

The samples of the 20 ms mode to be transmitted are contained in 576 lines of 32 µs, having 360 active luminance samples per line. A transformation to lines of 64 µs shall be performed by horizontally interleaving the samples of a pair of lines of 32 µs (i.e. intra-field shuffling). Figures. 21a and 21e depict the 20 ms shuffling operation. In figure 16 the 20 ms shuffling is performed by the block named SH_20.

5.2.5 Colour difference signal processing of the BRE system

The colour difference branches as depicted in figure 16 consist of the following sub-systems for each of the processing branches (the abbreviated names as used in figure 16 are given within brackets):

- low pass filters (LPF80, LPF40, LPF20);
- sub-Nyquist subsamplers (SS_80, SS_40, SS_20);
- shufflers (SH_80, SH_40, SH_20).

Each of these sub-systems is described in more detail in the following subclauses.

5.2.5.1 Colour difference 80 ms processing branch

5.2.5.1.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 18. This characteristic gives the notional transfer function of the colour difference branch low-pass filter LPF80 of figure 16. See also table 4c.

5.2.5.1.2 Sub-Nyquist sampling

In the 80 ms mode the BRE system shall use for the colour difference signals a four phase sub-Nyquist sampling, which takes four emission fields to fully transmit all the samples. (See figure 20a).

5.2.5.1.3 80 ms shuffling

The shuffling of the colour difference signals for the 80 ms mode (intra-field shuffling) shall be as defined in figures 24a, 24b, 24c, 24d and 24e.

5.2.5.2 Colour difference 40 ms processing branch

5.2.5.2.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 18. This characteristic gives the notional transfer function of the colour difference branch low-pass filter LPF40 of figure 16. See also table 4c.

5.2.5.2.2 Sub-Nyquist sampling

The 40 ms mode shall use two emission fields to transmit all samples (see figure 20b).

5.2.5.2.3 40 ms shuffling

In the 40 ms mode the inter-field shuffling shall be used as defined in figures 24f, 24g, 24h and 24j.

5.2.5.3 Colour difference 20 ms processing branch

5.2.5.3.1 Transmissible range of the spatial frequency spectrum

The transmissible range of the spatial frequency spectrum is given in figure 18. This characteristic gives the notional transfer function of the colour difference branch low-pass filter LPF20 of figure 16. See also table 4c.

5.2.5.3.2 Sub-Nyquist sampling

In the 20 ms mode the BRE system shall use a one phase sub-Nyquist sampling (sub-system SS_20 in figure 16), which takes one emission field to fully transmit all the samples (see figure 20a).

The 20 ms mode is a subset of the 80 ms mode and has a quincunx subsampling structure.

5.2.5.3.3 20 ms shuffling

The shuffling of the colour difference signals for the 20 ms mode (intra-field shuffling) shall be as shown in figures 24a, 24b, 24c, 24d and 24e.

5.3 General video characteristics

5.3.1 Picture signal

The picture signal shall correspond to the scanning of the image at uniform velocities from left to right and from top to bottom.

5.3.2 Number of lines per picture

The number of emitted lines per picture shall be 625.

5.3.3 Interlace

The interlace ratio of the emitted signal shall be 2:1.

5.3.4 Aspect ratio

The ratio of the image width to image height within the picture area shall be 16:9.

5.3.5 Gamma

The gamma of the transmitted signal shall be related to a display gamma of 2,8 ± 0,3. The overall gamma is approximately 1,2.

5.3.6 Baseband format

Within the normal D2-HDMAC video multiplex only 697 of the available 720 active samples of the luminance component, and 349 of the 360 available active samples of the two colour-difference components can be transmitted. Figure 27 shows which samples of the active video samples shall be actually transmitted. The video multiplex shall further apply line alternation for the two colour difference components (see also subclause 5.4.1).

The luminance and the colour-difference components shall be compressed in time (in this document further referred to as MAC compression) so that each transmitted line can contain:

- the compressed luminance signal, consisting of 697 transmissible samples;
- one of the compressed colour-difference signals (alternating from line to line), consisting of 349 transmissible samples;
- a multiplex of digital sound and data.

The DATV data signal, for assisting the HDMAC BRD in its reconstruction of the HDTV picture, shall be transmitted in the FBI DATV/data multiplex.

This format gives complete separation of luminance, colour-difference, digital sound, data and DATV data components as shown in figure 2.

Video signals shall be transmitted corresponding to lines 24 to 310 inclusive and lines 336 to 622 inclusive. There shall be colour-difference information on lines 23 and 335 as transmitted ; on these lines the luminance shall be set to black level to provide a black reference.

The non-scrambled wave form is shown in figure 25. The scrambled wave forms are shown in figures 26a and 26b.

5.3.6.1 Sub-Nyquist sampling phases

As can be seen in figure 19 and figure 20 the samples transmitted in the even fields have a spatial offset of half a period with respect to the samples transmitted in the odd fields. It is necessary to distinguish between the sub-Nyquist sampling phases. In every second and fourth field all the video samples shall be delayed by $T/2 = \tilde{}\ 24{,}7$ ns (see figure 25). This shift improves the quality of the picture received by a D2-MAC/packet receiver, since it corrects the spatial position of the samples in the even fields for the quincunxial subsampling structures.

5.3.7 Sampling clock frequencies

The transmission clock is defined as:

- $f_{ck} = 20{,}25$ MHz ± 2,5 parts in 10 million.

f_{ck} shall be equal to exactly twice the frequency given in subclause 4.3.1 for the instantaneous bit-rate of the sound/data multiplex.

5.3.8 Relationship between HDMAC clock frequency and line frequency

- $f_h = 2/3 * 1/864 * f_{ck}$.

5.3.9 Field frequency

- $f_{field} = 2/625 * f_h$.

5.3.10 Relationship with CCIR Recommendation 656

The relationship of the component sample numbers from CCIR Recommendation 656 [5] to the transmitted HDMAC component sample numbers shall be as indicated in figure 27.

5.4 Synchronization and TDM control

Details of line synchronization, frame synchronization and image parity with respect to the HDMAC 80 ms coding period and overall TDM control are given in Clause 4.

5.4.1 Colour sequence identification

The colour sequence identification is derived directly by a line count from the frame synchronization. The odd lines of the frame shall carry the U component of the colour sequence information and the even lines shall carry the V component, with U and V as defined in subclause 5.6.4.

5.4.2 Line numbering

Figure 28 shows the line numbering system for the HDMAC system. Some wave form details are included for clarity.

5.4.3 Frame synchronization

The HDMAC vision coding system has a basic two-frame (80 ms) sequence. This sequence is synchronized to the frame parity identification as defined in table 1. The first frame of the two frame sequence shall be transmitted in the odd frames, the second frame shall be transmitted in the even frames.

5.5 The HDMAC vision signal

5.5.1 General specification

The HDMAC coded signal shall comprise a luminance component and a pair of colour-difference components, bandwidth reduced according to subclause 5.2, time-compressed, optionally scrambled and formatted as a time-division multiplex according to Clause 4, and non-linear pre-emphasised and half-Nyquist filtered according to Clause 10. The luminance component shall be transmitted on each vision line with colour-difference components being transmitted on alternate lines (see figure 28). The scrambling of the signal is specified in subclause 5.7.

5.5.2 Luminance component

An increase in the incident light intensity shall correspond to an increase in the amplitude of the video signal. For a luminance signal, the relationship between the spectral composition of the incident light and the luminance amplitude shall be as defined in subclause 5.6.2.

5.5.2.1 Luminance amplitude range

The maximum amplitude of the luminance signal shall be 1 Vp-p (from black level to white level), disregarding overshoots . Black level shall be at -0,5 V with reference to the clamping level. The nominal black, grey and white levels are referenced in the signals of line 624 (see subclause 4.3.4.1).

5.5.2.2 Luminance time compression

The luminance compression ratio applied to the bandwidth reduced signal shall be 3:2.

5.5.2.3 Amplitude/Frequency characteristic after MAC compression

After MAC compression a non-linear pre-emphasis (E7E) shall be applied to the signal; this is specified in subclause 10.2.1.2.

Thereafter half-Nyquist filtering shall be performed around the Nyquist frequency of 10,125 MHz. This filtering is specified in subclause 10.2.2.

5.5.3 Colour-difference components

The two colour-difference components are defined in subclause 5.6.3. These are scaled according to subclause 5.6.4 to form the uncompressed colour-difference signal voltages $E'U_m$ and $E'V_m$.

5.5.3.1 Colour-difference amplitude range

The maximum amplitude of the colour-difference signals shall be 1 Vp-p, disregarding overshoots (corresponding to 77% electrical saturation, or 98% displayed saturation, at 100% amplitude).

5.5.3.2 The colour-difference time compression

The colour-difference compression ratio applied to the bandwidth reduced signal shall be 3:1.

5.5.3.3 Amplitude/Frequency characteristic after MAC compression

After MAC compression a non-linear pre-emphasis (E7E) shall be applied to the signal; this is specified in subclause 10.2.1.2.

Thereafter half-Nyquist filtering shall be performed around the Nyquist frequency of 10,125 MHz. This filtering is specified in subclause 10.2.2.

5.5.3.4 Colour-difference line sequence

The colour-difference signal voltages shall be transmitted within alternate lines as $E'U_m$ and $E'V_m$. The colour-difference signal $E'U_m$ or $E'V_m$ selected for transmission shall be spatially co-incident with the corresponding luminance signal, but shall be transmitted one line in advance. $E'U_m$ shall be sent on odd lines of the picture. This arrangement is chosen to maintain compatibility with the D2-MAC/packet signal.

5.5.4 Transcoding from PAL or SECAM to HDMAC

PAL or SECAM will not normally be transcoded to HDMAC, since D2-MAC can fully support this requirement. However, if this were necessary, the odd frame of the HDMAC signal shall be derived from the frame in which the colour burst is at +135° relative to the +U axis on the odd-numbered lines in the case of PAL and from the frame in which the D_b signal is transmitted on odd-numbered lines in the case of SECAM.

5.5.5 DATV data - vision timing

DATV data for an HDMAC 80 ms coding period shall be transmitted during the previous 80 ms period (see subclause 5.8.5).

5.6 Colour signal equations

5.6.1 Primary colour chromaticities

The primary colour chromaticities are those of the EBU phosphors as defined in the digital studio standard CCIR Recommendation 601-2 [6] and in CCIR Report 624-2, Table II, for PAL and SECAM. The colour equations are based on the NTSC primaries as specified in CCIR Recommendation 601-2 [6] and in CCIR Report 624-2, Table II, for NTSC.

5.6.2 Luminance signal voltage

The luminance signal voltage E'_Y shall be related to the colour separation signal voltages E'_R, E'_G, E'_B, by the following equation:

- $E'_Y = 0{,}299\,E'_R + 0{,}587\,E'_G + 0{,}114\,E'_B$

5.6.3 Unscaled colour-difference signal voltage

The colour-difference signal voltage shall be defined by the equations:

- $E'_R - E'_Y = 0{,}701\,E'_R - 0{,}587\,E'_G - 0{,}114\,E'_B$
- $E'_B - E'_Y = -\,0{,}299\,E'_R - 0{,}587\,E'_G + 0{,}886\,E'_B$

5.6.4 Transmitted colour-difference voltages

The two colour-difference signal voltages $E'U_m$ and $E'V_m$ shall be derived from the equations:

- $E'U_m = 0{,}733\,(E'_B - E'_Y)$
- $E'V_m = 0{,}927\,(E'_R - E'_Y)$

These equations are subject to the limits in subclause 5.5.3.1.

5.7 Picture scrambling

5.7.1 Introduction

The general principles of conditional access are described in Clause 9. This subclause gives only a description of the scrambled picture wave forms used for conditional access.

5.7.2 Description of the picture signal scrambling process

The non-scrambled video wave form is described in figure 25. Scrambling may be applied in two different ways which have been designed to permit, as far as possible, a common receiver implementation. The principles used are:

a) double-cut component rotation;

b) single-cut line rotation.

The component rotation principle is more secure but it is more sensitive to impairments arising from vestigial sideband distortion in cable systems and AM/VSB receivers. The single-cut line rotation system is less secure but is less sensitive to AM/VSB distortion. The scrambling scheme used is chosen by the service operator on the basis of the security and transmission requirements. The signalling which indicated which scheme is being used is described in subclause 4.5.4 and in subclause 8.2.3.

Vision scrambling shall only be applied to lines containing vision signals. These are defined in signalling given in the RDF of line 625, see subclause 4.5.4.

5.7.2.1 Double-cut component rotation

In double-cut component rotation, each component on each line is cut into two segments. The two segments of each component are transposed to scramble the signal. The cut points are varied in a pseudo-random way from line to line. To restore the signal to its intelligible form within the receiver the cut points are regenerated and the segments re-transposed.

Figure 26a illustrates the general principles of the double-cut component rotation process. Within the active part of each component, a flight range of 256 equally-spaced positions is defined, providing a choice of 256 cut points. For each component on each line a cut point CP shall be chosen within the defined range, using numbers derived from a pseudo-random sequence. Independent cut-points shall be chosen for the luminance and colour-difference components. The two segments of each component are then transposed. The spacing between adjacent cut-points for the compressed luminance signal shall be twice the spacing between the cut points for the compressed colour-difference signal.

The range of cut points is approximately evenly disposed about the centre of each component: the exact disposition shall be as shown in figures 26c and 26d. For each line, the numbers of the two cut points shall be obtained from the 16-bit binary word generated by the PRBS system described in subclause 9.3.2. The eight most significant bits shall represent the colour-difference cut-point number and the eight least significant bits shall represent the luminance cut-point number.

The scrambling process introduces additional transitions into the wave form. Edge effects at these transitions could impair the descrambled picture. To reduce the influence of edge effects, overlap samples shall be included on both sides of each cut and the transitions shall be shaped by a raised cosine function. The overlap samples are shown as P and P+1 in the colour-difference signal and as Q and Q+1 in the luminance signal in figure 26a, which describes the scrambled wave form in detail. Two possible shapings of the transitions are shown in table 12. The shaping ensures that the spectrum of the transitions is kept well within the signal bandwidth to minimise ringing.

To conceal landmarks in the wave form that could help piracy, the transitions between the segments of each component shall be replaced by smooth cross-fades as shown in figure 26a and as defined in table 12.

5.7.2.2 Single-cut line rotation

In single-cut line rotation, the colour-difference component of each line is cut into two segments. The first segment is then moved to the end of the line. The cut point is varied in a pseudo-random way from line to line. To restore the signal to its intelligible form within the receiver the cut point is regenerated and the segments moved back to their correct positions.

Table 12: Transitions definition for the double cut system

Transition	Method A (continuing samples)	Method B (held samples)
d	α = 1/8 (P-3) + 7/8 Gr	α = 1/8 (P) + 7/8 Gr
	β = 1/2 (P-2) + 1/2 Gr	β = 1/2 (P) + 1/2 Gr
	γ = 7/8 (P-1) + 1/8 Gr	γ = 7/8 (P) + 1/8 Gr
f	α = 7/8 (354) + 1/8 (2)	α = 7/8 (353) + 1/8 (5)
	β = 1/2 (355) + 1/2 (3)	β = 1/2 (353) + 1/2 (5)
	γ = 1/8 (356) + 7/8 (4)	γ = 1/8 (353) + 7/8 (5)
g	α = 7/8(P+2) + 1/8 (Q-3)	α = 7/8 (P+1) + 1/8 (Q)
	β = 1/2(P+3) + 1/2 (Q-2)	β = 1/2 (P+1) + 1/2 (Q)
	γ = 1/8(P+4) + 7/8 (Q-1)	γ = 1/8 (P+1) + 7/8 (Q)
i	α = 7/8 (702) + 1/8 (2)	α = 7/8 (701) + 1/8 (5)
	β = 1/2 (703) + 1/2 (3)	β = 1/2 (701) + 1/2 (5)
	γ = 1/8 (704) + 7/8 (4)	γ = 1/8 (701) + 7/8 (5)
j	α = 7/8 (Q+2) + 1/8 Gr	α = 7/8 (Q+1) + 1/8 Gr
	β = 1/2 (Q+3) + 1/2 Gr	β = 1/2 (Q+1) + 1/2 Gr
	γ = 1/8 (Q+4) + 7/8 Gr	γ = 1/8 (Q+1) + 7/8 Gr

NOTE: Gr is the reference clamping level.

Table 13: Transitions definition for the single-cut system

Transition	Method A (continuing samples)	Method B (held samples)
d	α = 1/8 (P-4) + 7/8 Gr	α = 1/8 (P-1) + 7/8 Gr
	β = 1/2 (P-3) + 1/2 Gr	β = 1/2 (P-1) + 1/2 Gr
	γ = 7/8 (P-2) + 1/8 Gr	γ = 7/8 (P-1) + 1/8 Gr
f	α = 7/8 (355) + 1/8 (1)	α = 7/8 (354) + 1/8 (4)
	β = 1/2 (356) + 1/2 (2)	β = 1/2 (354) + 1/2 (4)
	γ = 1/8 (357) + 7/8 (3)	γ = 1/8 (354) + 7/8 (4)
i	α = 7/8 (703) + 1/8 (1)	α = 7/8 (702) + 1/8 (4)
	β = 1/2 (704) + 1/2 (2)	β = 1/2 (702) + 1/2 (4)
	γ = 1/8 (705) + 7/8 (3)	γ = 1/8 (702) + 7/8 (4)
j	α = 7/8 (P+3) + 1/8 Gr	α = 7/8 (P+2) + 1/8 Gr
	β = 1/2 (P+4) + 1/2 Gr	β = 1/2 (P+2) + 1/2 Gr
	γ = 1/8 (P+5) + 7/8 Gr	γ = 1/8 (P+2) + 7/8 Gr

NOTE: Gr is the reference clamping level.

Figure 26b illustrates the general principles of the line rotation process. Within the active part of the colour-difference component, a flight range of 256 equally-spaced positions is defined, providing a choice of 256 cut points. For each line, a cut point CP shall be chosen within the defined range, using a number derived from a pseudo-random sequence. The first of the two segments of the colour-difference component is then moved to the end of the active line: this has the effect of rotating the whole line by the length of this segment.

The range of cut points is approximately evenly disposed about the centre of the colour-difference component: the exact disposition shall be as shown in figure 26c For each line the number of the cut points shall be given by the eight most significant bits of the 16-bit binary word generated by the PRBS system described in subclause 9.3.2.

The scrambling process introduces additional transitions into the wave form. Edge effects at these transitions could impair the descrambled picture. To reduce the influence of edge effects, overlap samples shall be included on both sides of the cut. The extra transitions shall be shaped by a raised-cosine function as shown in table 13. The overlap samples are shown as P-1, P, P+1 and P+2 in figure 26b, which describes the scrambled wave form in detail. Two possible shapings of the transitions are shown in table 13. The shaping ensures that the spectrum of the transitions is kept well within the signal bandwidth to minimise ringing.

To conceal landmarks in the wave form that could help piracy, the transitions between the components shall be replaced by a smooth cross-fade as shown in figure 26b and defined in table 13.

5.8 DATV data capacity and coding

5.8.1 DATV data

The DATV data signal shall include the following elements:

a) the Mode Decision (MD) signal;

b) the Motion Vectors (MV) for the 40 ms mode;

c) the film-mode indication bit;

d) the fixed colour mode indication bit.

5.8.2 Capacity for DATV

The DATV/data multiplex, as specified in subclause 4.3.2.2, shall permit a minimum of 56 DATV packets per frame, corresponding to a net useful DATV data capacity of 1,008 Mbit/s.

The DATV/data multiplex shall be defined in such a manner that FBI1 has a capacity of not less than 27 packets and that FBI2 has a capacity of not less than 29 packets.

The exact number of DATV packets per individual FBI is defined by the BRE system (also see Annex B). The D2-HDMAC multiplexer shall allocate the DATV packets in the exact order as supplied and in the same FBI as defined by the BRE. The BRE shall comply to the rules as given in table 14 for the minimum and maximum number of DATV packets per FBI.

Table 14: DATV packet allocation rules for the DATV/data multiplex

	minimum	maximum
FBI1	1	27
FBI2	1	29

If spare capacity is available in the DATV/data multiplex, packets of other services may be introduced at any packet location (see also subclause 4.5.5). Furthermore spare capacity can be made available to FBI teletext using the TDM overlapping mechanism as defined in subclause 4.5.5.

5.8.3 Structure of DATV packets

The DATV packet structure shall be as defined in subclause 4.3.6. In subclause 5.8.3.1 the coding of the continuity index is defined. The PT byte shall be used to transmit specific information, as defined in subclause 5.8.3.2, and is therefore not available for data transmission. This results in a data capacity of 720 data bits per packet. The coding of these data bits is described in subclause 5.8.7.

5.8.3.1 Continuity index

The Continuity Index (CI) of a DATV packet shall be incremented from that of the previous DATV packet, regardless of its PT byte value. The CI of the first DATV packet of a subframe shall be incremented from that of the CI of the last DATV packet of the previous subframe containing DATV packets. This means that CI of DATV packets is incremented as though the FBI1 and FBI2 subframes are defined as a single combined subframe. Note that FBI1 and FBI2 are defined as independent subframes (see subclause 4.5.5).

5.8.3.2 PT byte allocation

The PT byte of each individual DATV packet shall convey:

- the access control status (see subclause 5.8.4);
- the special marker, as defined in subclause 4.5.5.

The PT byte can have four possible values and shall be allocated as follows:

	normal PT	special PT
scrambled	'F8	'C7
non scrambled	'00	'3F

The PT values related to the special marker shall be used in conjunction with TDM overlapping as defined in subclause 4.5.5. The two normal PT values shall be used when no TDM overlapping is defined and also for packets not having the special marker in the case of TDM overlapping.

5.8.3.3 Scrambling for conditional access

The scrambling of DATV data for conditional access shall be identical to that for other data services. The control word used to feed the scramblers may be the same as that used for the vision scrambling itself, or could be different where a separate subscription facility is desired for reception of the DATV data for picture enhancement. The scrambling of data within the DATV/data multiplex is further specified in subclauses 4.3.8 and 9.3.2.

5.8.3.4 Procedure for changing the DATV packet address

The change of the packet address of the DATV packets shall be signalled by an SI update message (see subclause 8.2.4) during 16 frames in advance of the transition, and shall effectively occur at line 1 of the frame after FCNT modulo 16 equals 0.

5.8.4 Service identification data

Access coordinates of the DATV component transmitted in the DATV/data multiplex shall be given by the parameter DCINF of the service identification channel (see subclause 8.2.3) characterized by the parameter DATVD with parameter identifier 'FE.

The parameter DATVD shall be part of the Medium Priority TV service and shall be coded as follows:

- language used (one byte):
  - set to 00 (language not applicable);
- access coordinates (16 bit sequence), where:
  - the ten LSBs represent the packet address (PA) used for the DATV component within the DATV/data multiplex;
  - bits 11, 12 are set to 0;
  - bits 13, and 14 are reserved and set to 0;
  - bit 15 signals the access control mode, with:
    - 0 no scrambling;
    - 1 scrambling;
  - bit 15 signals whether scrambling may be applied for packets of the DATV component. For each individual DATV packet the information indicating whether this packet is scrambled or not is conveyed within the PT byte, as described in subclause 5.8.3.2;
  - bit 16 signals the access mode (see subclause 9.3.1) in case scrambling is applied (as signalled in bit 15) according to:
    - 0 free-access mode;
    - 1 controlled-access mode.

5.8.5 DATV - correspondence to video

The order of transmission shall be along the scan line, starting in the top-left corner. Moreover, the DATV data of an 80 ms period shall be sent four fields ahead to minimize the video delay required in the receiver BRD.

5.8.6 DATV block structure mapped on the transmitted signal

In figure 29 the DATV block structure is defined on the non-interlaced HD-Grid of 1250 lines. The DATV block structure map on the 625 lines emitted signal can be deduced by combining the subsampling and shuffling as shown in figures 21 and 24.

5.8.7 DATV data coding system

The general block diagram of the DATV data coding system in figure 30 depicts the main modules. The block diagram also shows the error protection system applied for DATV data.

5.8.7.1 General format

The input information to the compression and error-protection system of the DATV encoder per 80 ms period shall consist of:

- film-mode indication;

- fixed colour mode indication (contained in the feature byte);

and for each video block:

- odd and even frame Mode Decision (MD);

- odd frame Motion Vector (MV);

- even frame Relative Address (RAD).

5.8.7.2 DATV data compression

The compression system contains a two step reduction process.

The mode-decision and motion-vector information from the BRE system (mode decisions, odd frame 40 ms vectors and relative addresses) are first encoded into variable length block code words by the MENU compression system.

These block code words are fed into the run-length compression system. If the run-length system performs no bit-rate reduction at all, the block code words at the output of the MENU system shall be transmitted directly, i.e. the run-length encoder is bypassed.

Together with additional information from the MENU and run-length compression system (CL, RL, MVSTR), SYNC words, film-bit, feature byte and stuffing bits, the CWD forms the output of the compression system. For error protection coding the output of the compression system is fed into the Fire encoder, where the information stream is chopped up into 82-bit sequences. After every sequence the 8 check-bits of the Fire (90, 82) code shall be inserted. See subclause 5.8.8 for the specification of the Fire (90, 82) code.

Detailed specification of the various signals in figure 30 are given in subclause 5.8.7.3.

5.8.7.3 DATV data format

In the following subclauses the information stream between MUX and Fire encoder (see figure 30) in an 80 ms period is specified.

The information stream between MUX and Fire encoder (see figure 30) in an 80 ms period shall consist of the following data, listed in the order of transmission:

Film mode	1 bit	(first transmitted)
MENU 1	173 bits	
MENU 2	173 bits	
.....	....	
MENU 9	173 bits	
CWD	variable length	
Stuffing bits	variable length	
Feature byte	8 bits	(last transmitted)

The individual parts of the list are further specified in the following subclauses.

5.8.7.3.1 Film bit

This 1-bit parameter contains the film/camera mode indication as described in subclause 5.2.2.4, which shall be coded as follows:

film-mode bit	mode
1	film
0	camera

5.8.7.3.2 Feature byte

The feature byte shall contain two information bits b_1 and b_0, which are appended by six protection bits p_5 to p_0 forming a Hamming protected byte. The protection scheme is derived from the PT byte values by permutation to make the code systematic. The four possible combinations shall be coded as follows:

information bits		code (b_1, b_0, p_5, p_4, p_3, p_2, p_1, p_0)	
b_1	b_0	binary	hexadecimal
0	0	00 000000	'00
1	0	10 001111	'8F
0	1	01 111110	'7E
1	1	11 110001	'F1

where p_0 shall be transmitted first.

Information bit b_1 shall be used to transmit the fixed colour mode information as described in subclause 5.2.2.4, where:

b_1	mode
1	fixed colour
0	non fixed colour

Information bit b_0 is reserved for future use and shall be set to 0.

5.8.7.3.3 MENU 1..9

The MENU's, numbered from 1 to 9, shall contain the CL, RL, and MVSTR information resulting from the MENU compression system (see figure 30).

The MENU compression system shall divide the picture into 9 MENU areas as defined in table 15. For each MENU area the CL, RL and MVSTR shall be determined.

Table 15: Subdivision in 9 MENU's (dimensions in number of video blocks)

	30	28	30
24	MENU 1	MENU 2	MENU 3
24	MENU 4	MENU 5	MENU 6
24	MENU 7	MENU 8	MENU 9

Since the D2-HDMAC vision signal contains 697 out of 720 horizontal pixels (as defined in subclause 5.3.6) the DATV information of only 88 horizontal video blocks shall be transmitted. The DATV information of the outermost video block columns are not transmitted. The exact location of the video blocks on the HD grid is defined in subclause 5.8.6.

Each of the 9 MENU's shall have a fixed length of 173 bits.

In a MENU the transmission order shall be as follows:

- block Code word Length (CL) 3 bits;
- Run-Length encoding indicator (RL) 1 bit;
- Motion Vector String (MVSTR) 169 bits.

The block Code word Length (CL) shall be directly related to the number of different odd frame 40 ms vectors (N) in a MENU area. The LSB shall be transmitted first.

CL MSB LSB	N	block code word length
000	0	4
001	2	5
010	5	6
011	11	7
100	24	8
101	50	9
110	101	10
111	169	11

If, in the encoder, the number of different odd frame 40 ms vectors in a MENU area is not equal to one of the values in the table above, the number shall be increased to the nearest value in the table by artificially adding dummy 40 ms vectors. These dummy vectors shall be signalled in the MVSTR of that MENU and shall be taken into account in the block code word table.

The 1-bit Run-Length encoding indicator (RL) indicates if the block code words in a MENU area are run-length encoded or not and shall be coded as follows:

RL	run-length coding
1	applied
0	not applied

The Motion Vector String (MVSTR) is a 169-bit string, that shall indicate the presence ("1") or absence ("0") of a 40 ms vector in an odd frame MENU area.

The mapping between a motion vector (h, v) and the MVSTR bit position shall be defined by:

- $m = 7 + h + 13 \times (v + 6)$

where m represents the bit position in MVSTR corresponding to motion vector (h, v) (h represents the horizontal displacement and v represents the vertical displacement).

Table 16 illustrates the correspondence between the 40 ms motion vector and the bit position in MVSTR. The first bit position in MVSTR shall indicate if motion vector (-6, -6) is present in the MENU area or not. Bit position 1 (vector = (-6, -6)) is the LSB of MVSTR and shall be transmitted first and bit position 169 (vector = (+6, +6)) is the MSB of MVSTR and shall be transmitted last.

In table 16 the horizontal displacement is depicted along the horizontal axis of the table.

5.8.7.3.4 Code words (CWD)

The CWD part of the total DATV information stream shall contain the mode-decision and motion-vector information of the 6 336 (88 × 72) video blocks. The CWD part has a variable length.

The transmission order of the DATV information of the individual video blocks shall be from left to right and from top to bottom of the picture, i.e. video block rows shall be transmitted one after the other.

Each video block row shall be split up in three parts in accordance to the MENU area boundaries.

Table 16: Correspondence between the 40 ms motion vector and the bit position in MVSTR

+6	157	158	159	160	161	162	163	164	165	166	167	168	169
+5	144	145	146	147	148	149	150	151	152	153	154	155	156
+4	131	132	133	134	135	136	137	138	139	140	141	142	143
.....	.....	.....	.....	.....	.....	.....	.....	.....	.....	.....	.....	.....	.....
-4	27	28	29	30	31	32	33	34	35	36	37	38	39
-5	14	15	16	17	18	19	20	21	22	23	24	25	26
-6	1	2	3	4	5	6	7	8	9	10	11	12	13
	-6	-5	-4	-3	-2	-1	0	1	2	3	4	5	6

Table 17: Block code word definition

block code word	odd frame decision	even frame decision	odd frame motion vector	even frame relative address
0	80	80	-	-
1	20	40	-	9
2	20	40	-	8
3	20	40	-	7
4	20	40	-	6
5	20	40	-	5
6	20	40	-	4
7	20	40	-	3
8	20	40	-	2
9	--------------------------- not used ---------------------------			
10	20	20	-	-
11 to N+10	40	20	P1 to PN	-
N+11 to 2N+10	40	40	P1 to PN	1
2N+11 to 3N+10	40	40	P1 to PN	2
3N+11 to 4N+10	40	40	P1 to PN	3
4N+11 to 5N+10	40	40	P1 to PN	4
5N+11 to 6N+10	40	40	P1 to PN	5
6N+11 to 7N+10	40	40	P1 to PN	6
7N+11 to 8N+10	40	40	P1 to PN	7
8N+11 to 9N+10	40	40	P1 to PN	8
9N+11 to 10N+10	40	40	P1 to PN	9

where "to" = "up to and including".

Depending on the RL parameter of the corresponding MENU the mode-decision and motion-vector information in that part shall be transmitted either block code word by block code word, where a block code word contains the DATV information (from both odd and even frame) of one video block (RL="0"), or as a run-length encoded sequence of block code words (RL="1").

The block code words shall be as defined in table 17.

The actual transmitted block code word shall be the binary representation of the decimal number given in table 17. The number of bits in the block code word shall be determined by CL or identically by N (see subclause 5.8.7.3.3) of the corresponding MENU area.

The odd frame 40 ms motion vectors which are present in a MENU area, shall be numbered from P1 up to and including PN.

P1 shall correspond to the 40 ms motion vector indicated by the first "1" in the MVSTR, when starting to search from the LSB position. P2 shall correspond to the second "1" in the MVSTR etc....

The relative address of an even frame 40 ms video block X and its reference to the motion vector of a video block in the odd frame shall be defined by:

odd frame

	6	2	7
vertical	5	1	3
	9	4	8

horizontal

even frame

X

Block 1 has the same spatial position as block X, while the other 8 blocks in the odd frame are the spatial neighbours of block 1.

If on a video block row between two MENU boundaries the RL = "1", the block code words shall not be transmitted straightforward, but first be run-length encoded.

From left to right the neighbouring block code words, that are identical, shall be grouped together into runs. The spatial order from left to right shall not be changed during the grouping and that a run shall not exceed a MENU boundary.

Runs of length 1 (the neighbour video blocks have different block code words) shall be transmitted in the following format:

MSB	LSB
block code word	0

<--------------- CL --------------->

A run of length greater than 1 shall be coded in the format:

MSB		LSB
block code word	runl. indicator	1
<--------------- CL --------------->	<------------ 5 bits ------------>	

The 5-bit run-length shall be defined as follows:

length of the run	run-length indicator MSB LSB
not used	00000
not used	00001
2	00010
3	00011
.....	.....
30	11110
not used	11111

After the transmission of the mode-decision and motion-vector information of one complete video block row (88 blocks) a unique 21-bit SYNC word shall be inserted, having the following bit-pattern:

0 1111 1111 1111 1111 1110.

5.8.7.3.5 Stuffing bits

Depending on the length of the stream film-bit, MENU 1,2, ...,9, CWD and the feature byte dummy zeros ("0") shall be inserted. These dummy bits are called stuffing bits and are used to fill the spare capacity within the DATV packets.

The total length of the stream between MUX and Fire encoder in an 80 ms period shall be equal to the nearest multiple of 656 greater than or equal to the length of the sequence film-bit, MENU 1..9, CWD and feature byte, however without stuffing bits ("0"). The resulting data stream including the stuffing bits after Fire (90, 82) encoding then occupies a multiple of 720 bits, which can be inserted directly into the data fields of the DATV packets.

The feature byte shall replace the last eight stuffing bits. If the last DATV packet of a 80 ms period contains less than eight stuffing bits (before the replacement as described here), the DATV data stream is extended by an additional 656 stuffing bits, of which the last eight bits are replaced by the eight bits of the feature byte.

5.8.8 Fire code (90, 82)

The compressed DATV data shall be sliced into blocks of 82 information bits, which are appended by a complementary set of eight check bits, resulting in data blocks of 90 bits. Each DATV packet consists of eight 90-bit data blocks, having 8 x 82 = 656 information bits per packet. The eight check bits shall be coded according to Fire,(90, 82) code as defined below.

The Fire (90, 82) is fully described by its generator polynomial:

- $G(x) = (x^3 + 1)(x^5 + x^2 + 1) = x^8 + x^3 + x^2 + 1$

where the " + " sign indicates a modulo-2 addition operation (exclusive-or logic function).

NOTE: The Fire (90, 82) code is derived from the Fire(93,85) code by shortening. The Fire (93, 85) code can be found in most of the text books on error-correcting codes.

The specified configuration constitutes a systematic code; i.e. the code word M(x) formed from 90 bits is made up from 82 bits of original information (polynomial m(x)), followed by eight check bits determined as the remainder when the polynomial $x^8m(x)$ is divided by the generator polynomial G(x).

Let m(x), M(x) and r(x) be denoted by:

- $m(x) = t_{81}x^{81} + t_{80}x^{80} + t_{79}x^{79} + \ldots + t_1x + t_0$
- $r(x) = r_7x^7 + r_6x^6 + \ldots + r_0$
- $M(x) = t_{81}x^{89} + t_{80}x^{88} + \ldots + t_0x^8 + r_7x^7 + r_6x^6 + \ldots + r_0$

where t_{81}, t_{80}, t_{79}, ..., t_0 are the 82 original information bits (t_{81} is transmitted first) and r_7,r_6, ... r_0 are the 8 check bits (r_7 is transmitted first).

By definition;

- $x^8m(x) = q(x).G(x) + r(x)$

Then it follows that;

- $M(x) = x^8m(x) + r(x) = q(x).G(x).$

The transmitted code M(x) is, therefore, a polynomial, which is a multiple of the generator polynomial.

The Fire (90, 82) code, as defined above, can correct any single error or any two adjacent errors in a code word.

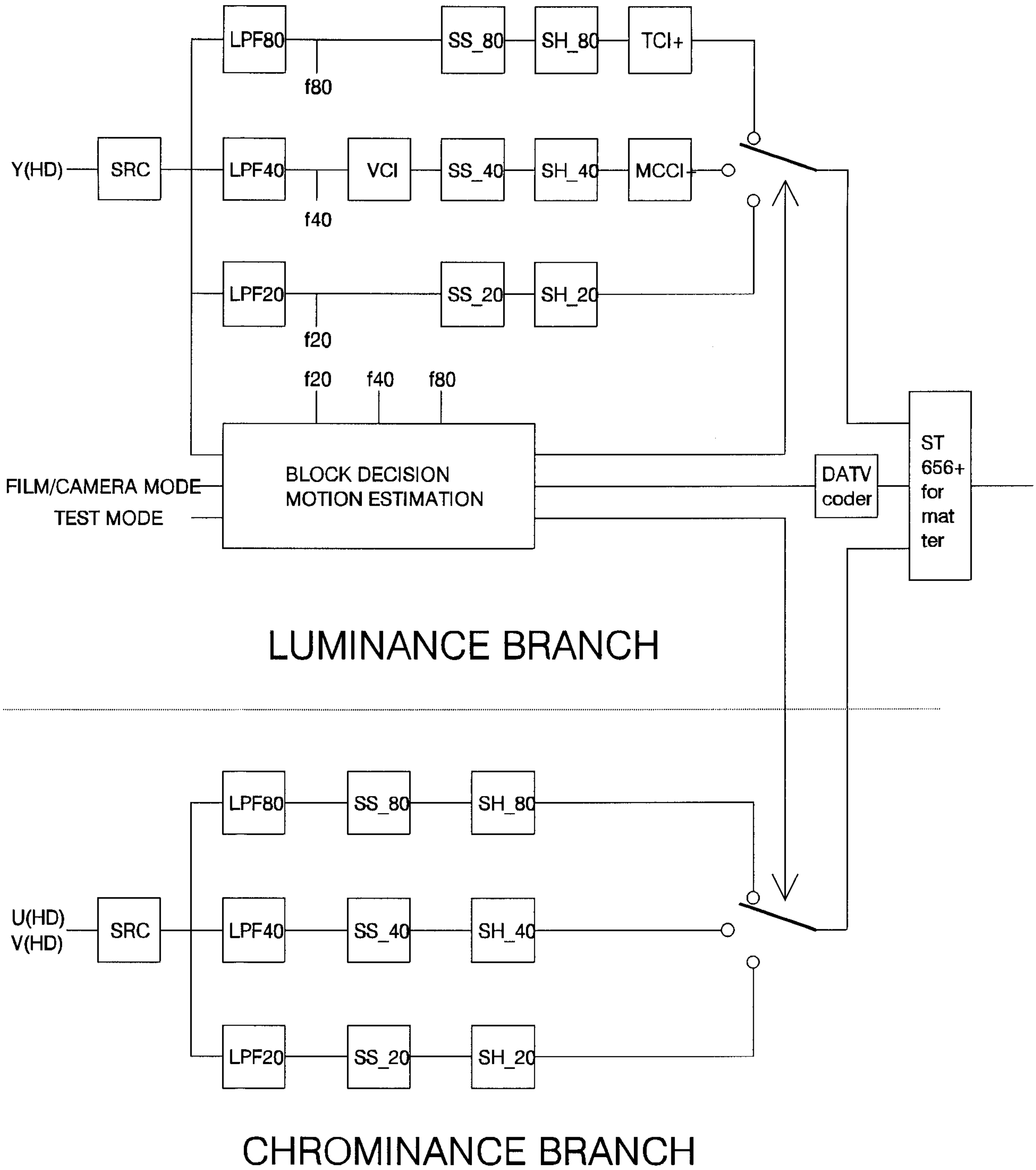


Figure 16: Notional block diagram of Bandwidth Reduction Encoding (BRE) system

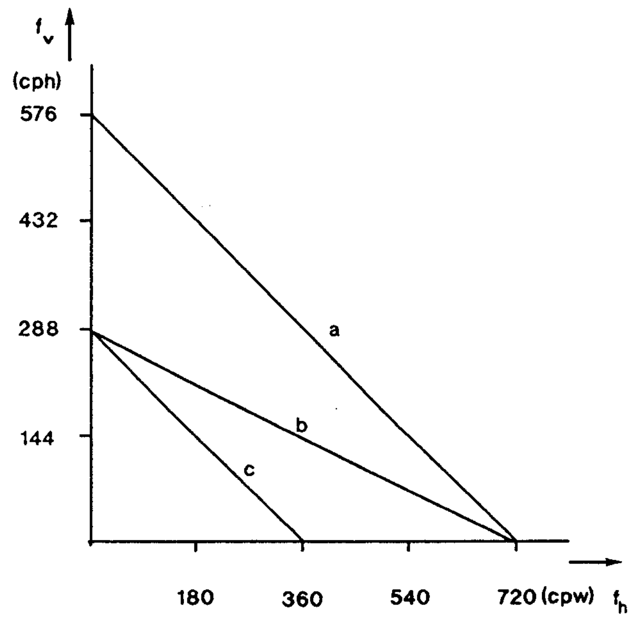


Figure 17: Transmissible range in spatial frequency domain for the luminance sampling patterns (as depicted in figure 19)

a) 80 ms mode;
b) 40 ms mode;
c) 20 ms mode.

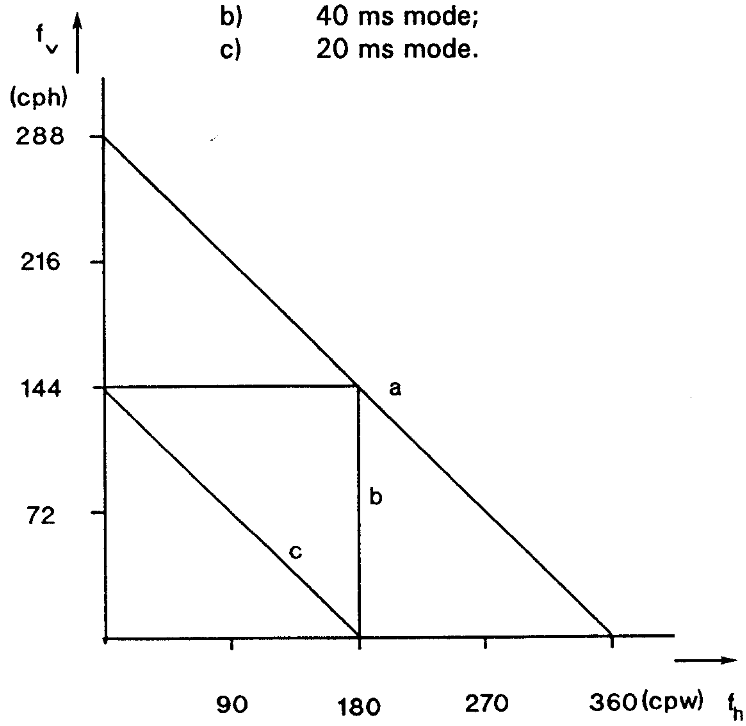


Figure 18: Transmissible range in spatial frequency domain for the colour difference sampling patterns (as depicted in figure 20)

a) 80 ms mode;
b) 40 ms mode;
c) 20 ms mode.

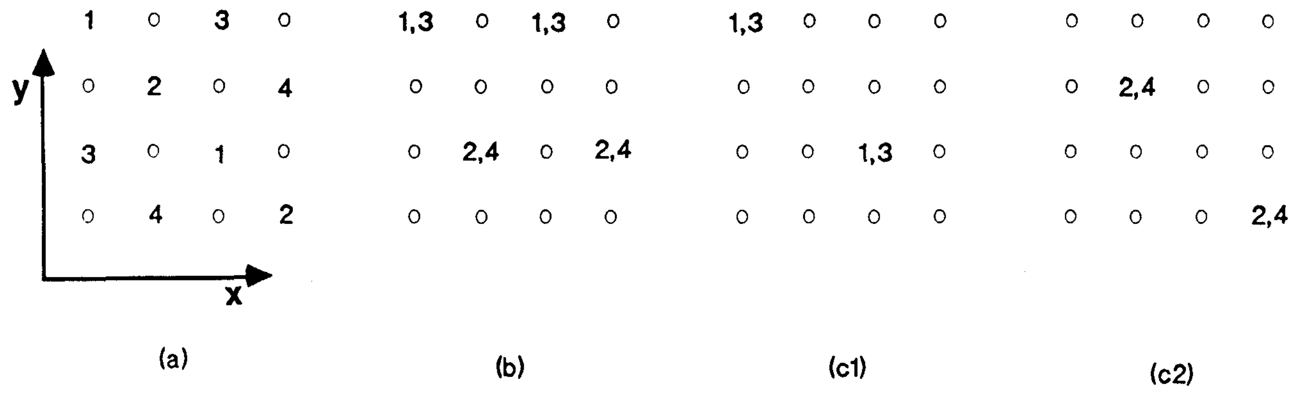


Figure 19: Luminance subsampling patterns on 1 250/50/2 sampling grid with 54 MHz sampling frequency

a) 80 ms spatial temporal subsampling pattern;
b) 40 ms spatial temporal subsampling pattern;
c1) 20 ms spatial temporal subsampling pattern (odd fields);
c2) 20 ms spatial temporal subsampling pattern (even fields);
with "x" and "y" denoting the horizontal and vertical directions respectively.

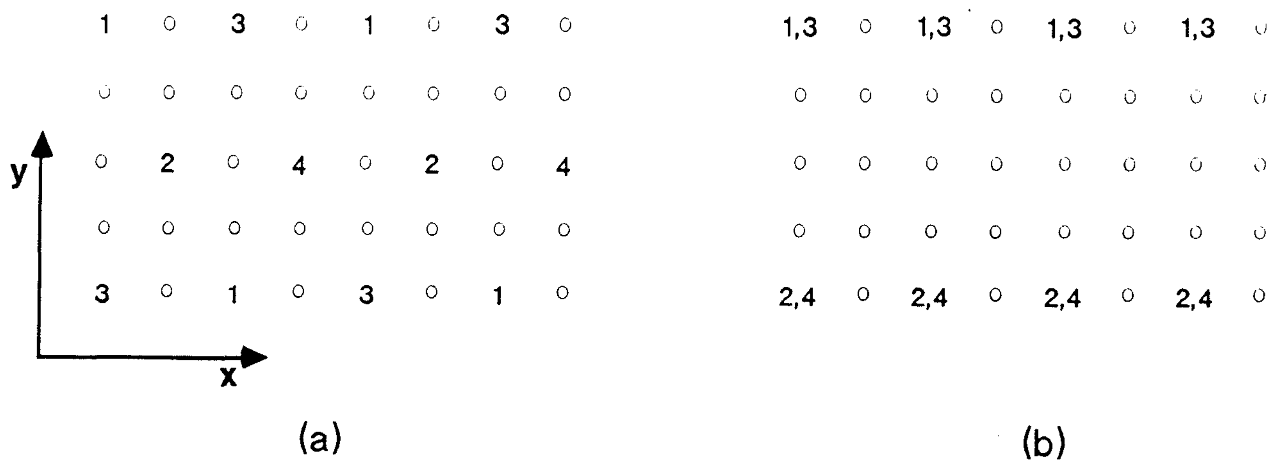


Figure 20: Colour difference subsampling patterns on 1 250/50/2 sampling grid with 27 MHz sampling frequency

a) 80 ms and 20 ms spatial temporal subsampling pattern;
b) 40 ms spatial temporal subsampling pattern;
with "x" and "y" denoting the horizontal and vertical directions respectively.

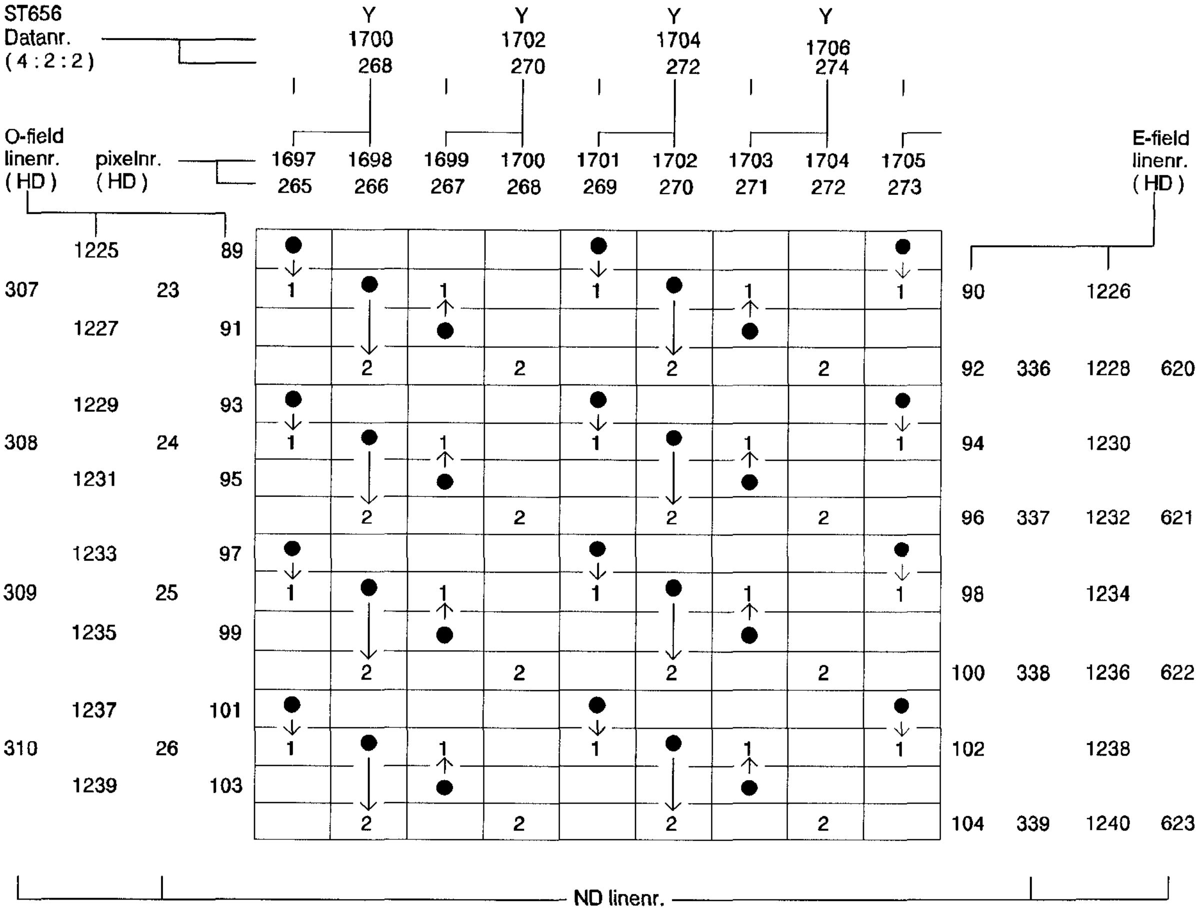


Figure 21a: Y shuffling field 1, 2 (80 ms and 20 ms modes)

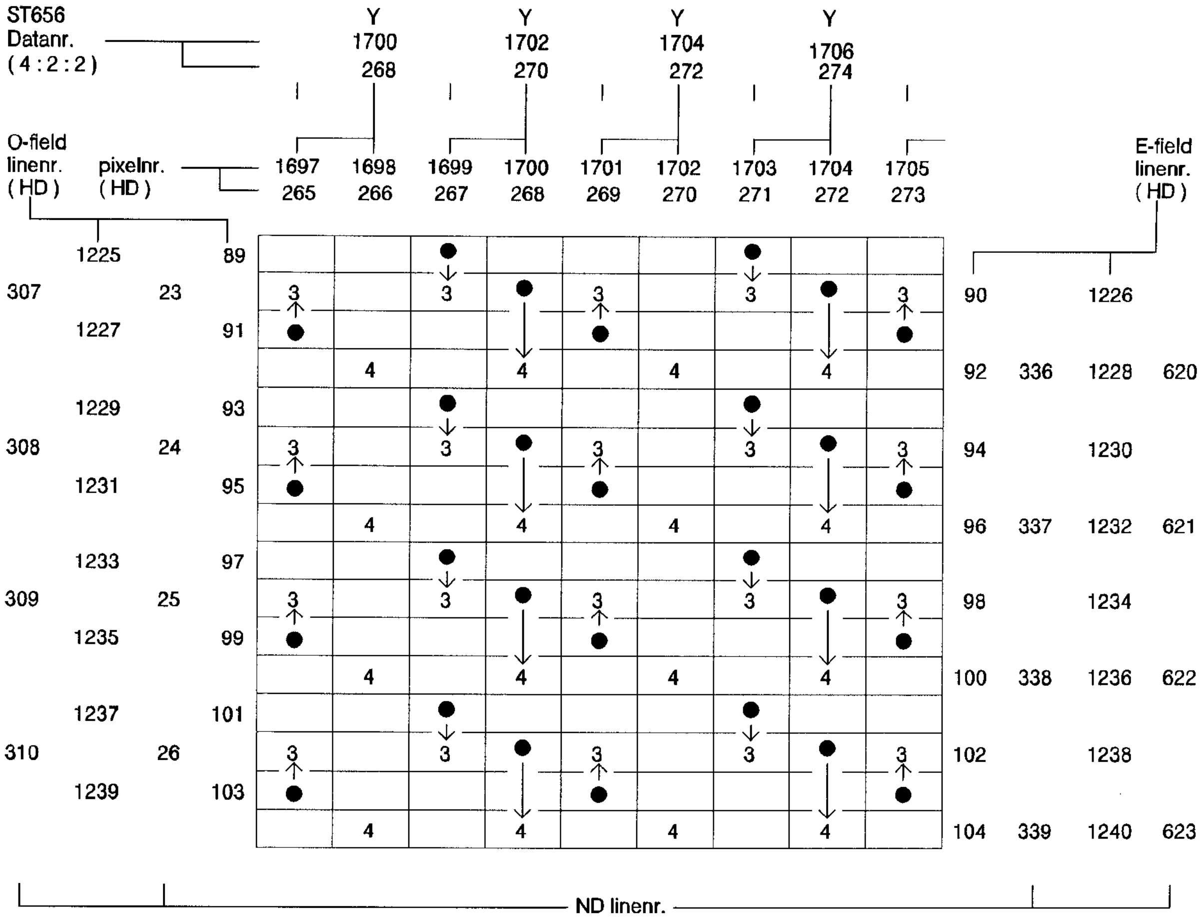


Figure 21b: Y shuffling field 3, 4 (80 ms mode)

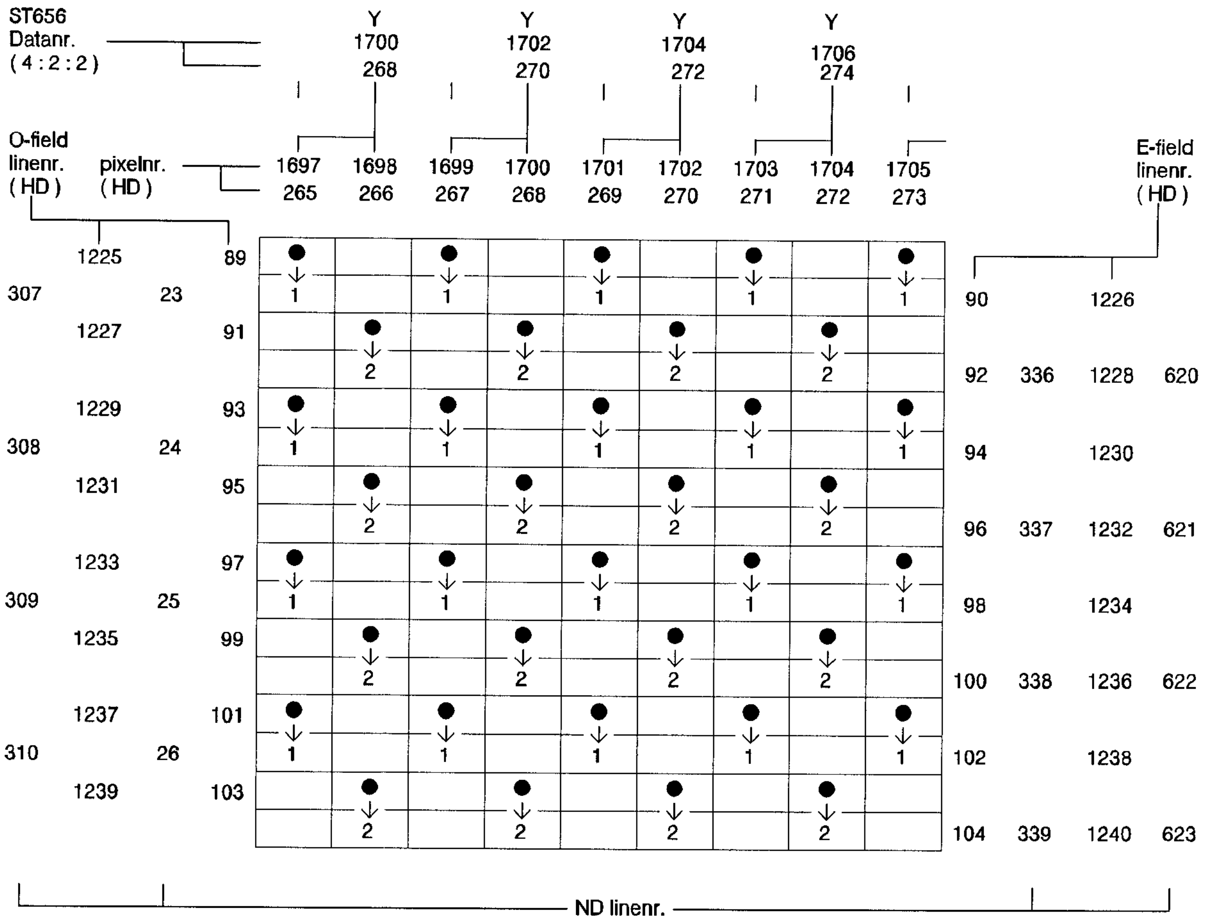


Figure 21c: Y reallocation 40 ms mode (no shuffling required)

O-field linenr. (HD)	pixelnr. (HD)	1697 265	1698 266	1699 267	1700 268	1701 269	1702 270	1703 271	1704 272	1705 273		E-field linenr. (HD)
1225	89	1		1		1		1		1		
											90	1226
1227	91		1		1		1		1			
											92	1228
1229	93	1		1		1		1		1		
											94	1230
1231	95		1		1		1		1			
											96	1232
1233	97	1		1		1		1		1		
											98	1234
1235	99		1		1		1		1			
											100	1236
1237	101	1		1		1		1		1		
											102	1238
1239	103		1		1		1		1			
											104	1240

Figure 21d: Y sampling 40 ms mode

O-field linenr. (HD)	pixelnr. (HD)	1697 265	1698 266	1699 267	1700 268	1701 269	1702 270	1703 271	1704 272	1705 273		E-field linenr. (HD)
1225	89	1				1				1		
			2				2				90	1226
1227	91			1				1				
					2				2		92	1228
1229	93	1				1				1		
			2				2				94	1230
1231	95			1				1				
					2				2		96	1232
1233	97	1				1				1		
			2				2				98	1234
1235	99			1				1				
					2				2		100	1236
1237	101	1				1				1		
			2				2				102	1238
1239	103			1				1				
					2				2		104	1240

Figure 21e: Y sampling 20 ms mode (for 20 ms mode shuffling see figure 21a)

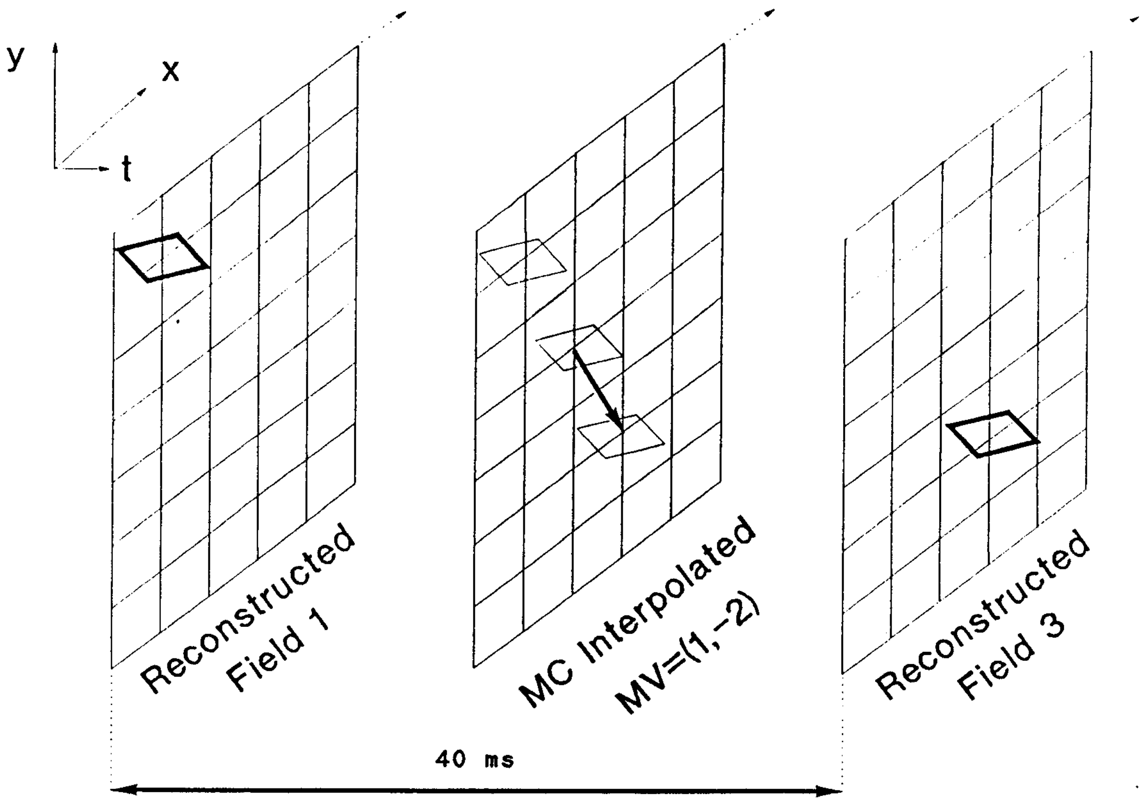


Figure 22: The principle of motion compensated interpolation

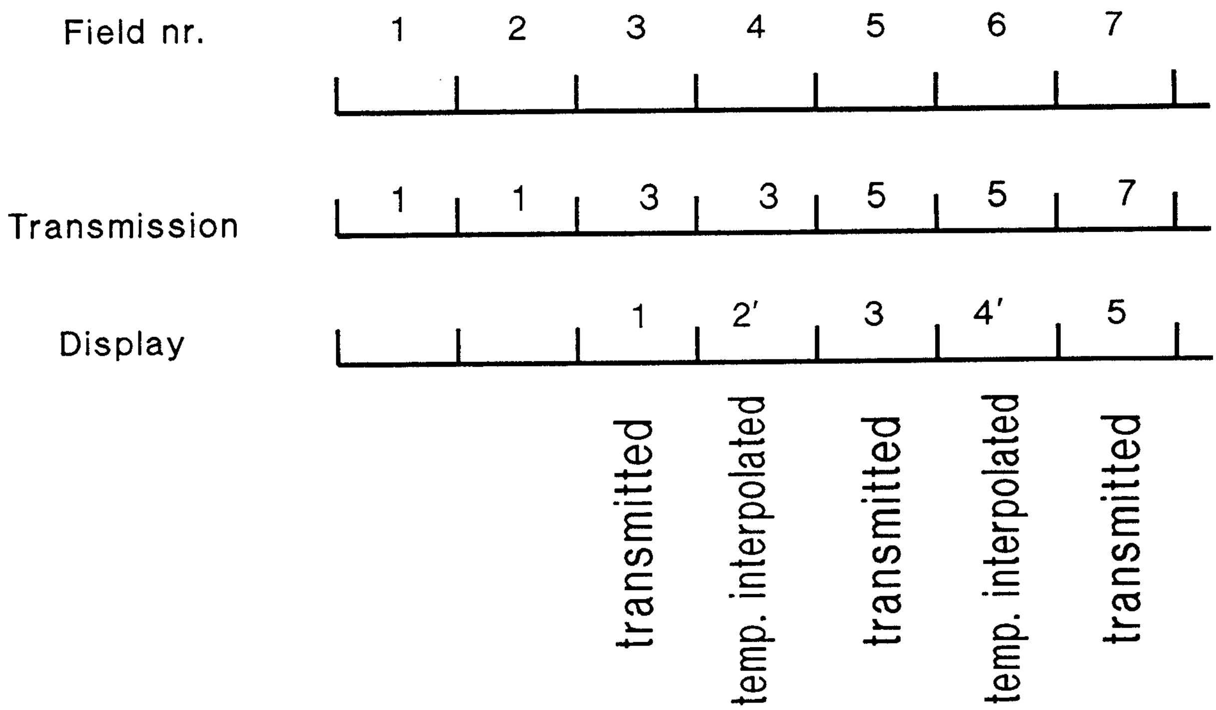


Figure 23: Field skip method

O-field linenr. (HD)	pixelnr. (HD)	1697	1698	1699	1700	1701	1702	1703	1704	1705		E-field linenr. (HD)
		265	266	267	268	269	270	271	272	273		
1225	89	2				4				2		
											90	1226
1227	91			1				3				
											92	1228
1229	93	4				2				4		
											94	1230
1231	95			3				1				
											96	1232
1233	97	2				4				2		
											98	1234
1235	99			1				3				
											100	1236
1237	101	4				2				4		
											102	1238
1239	103			3				1				
											104	1240

Figure 24a: Colour difference subsampling 80 ms and 20 ms mode (Cb and Cr)

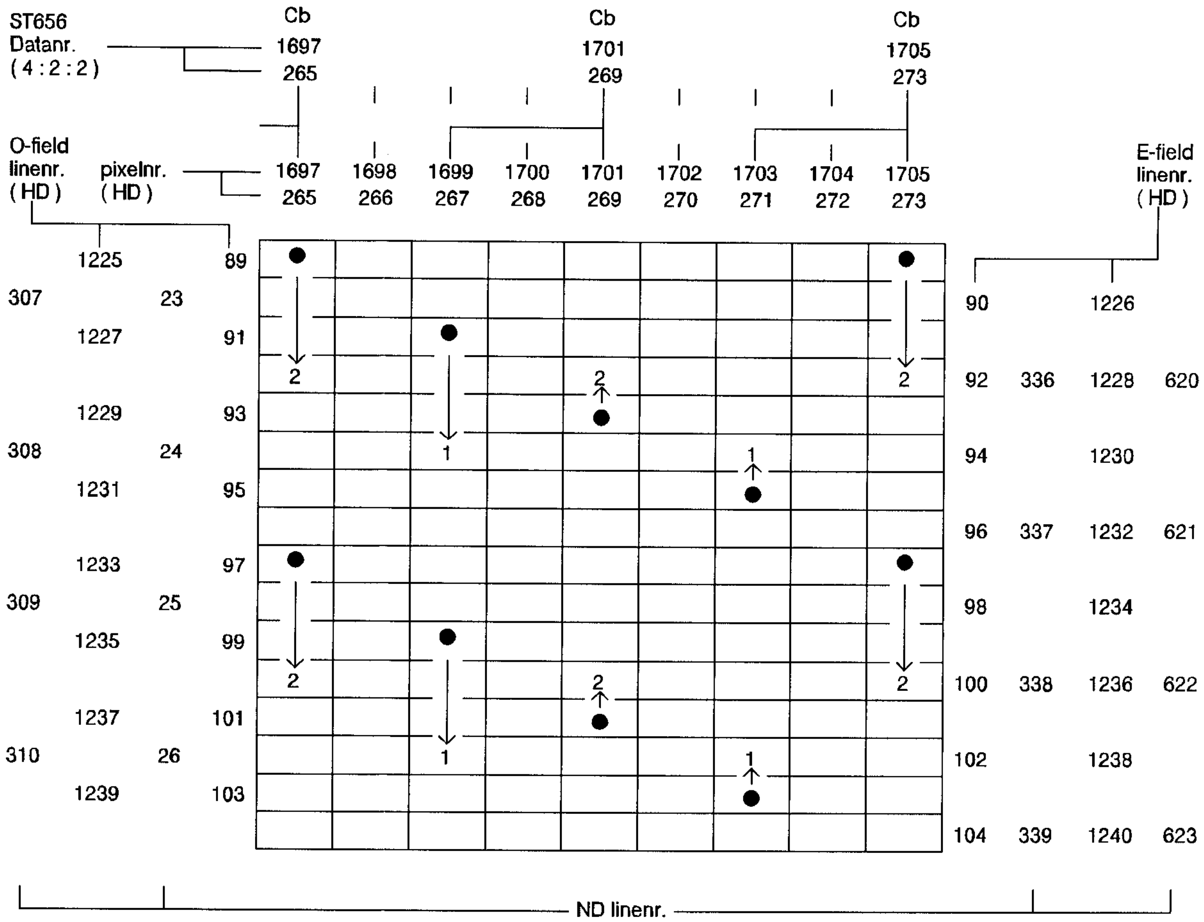


Figure 24b: Cb shuffling field 1, 2 (80 ms and 20 ms modes)

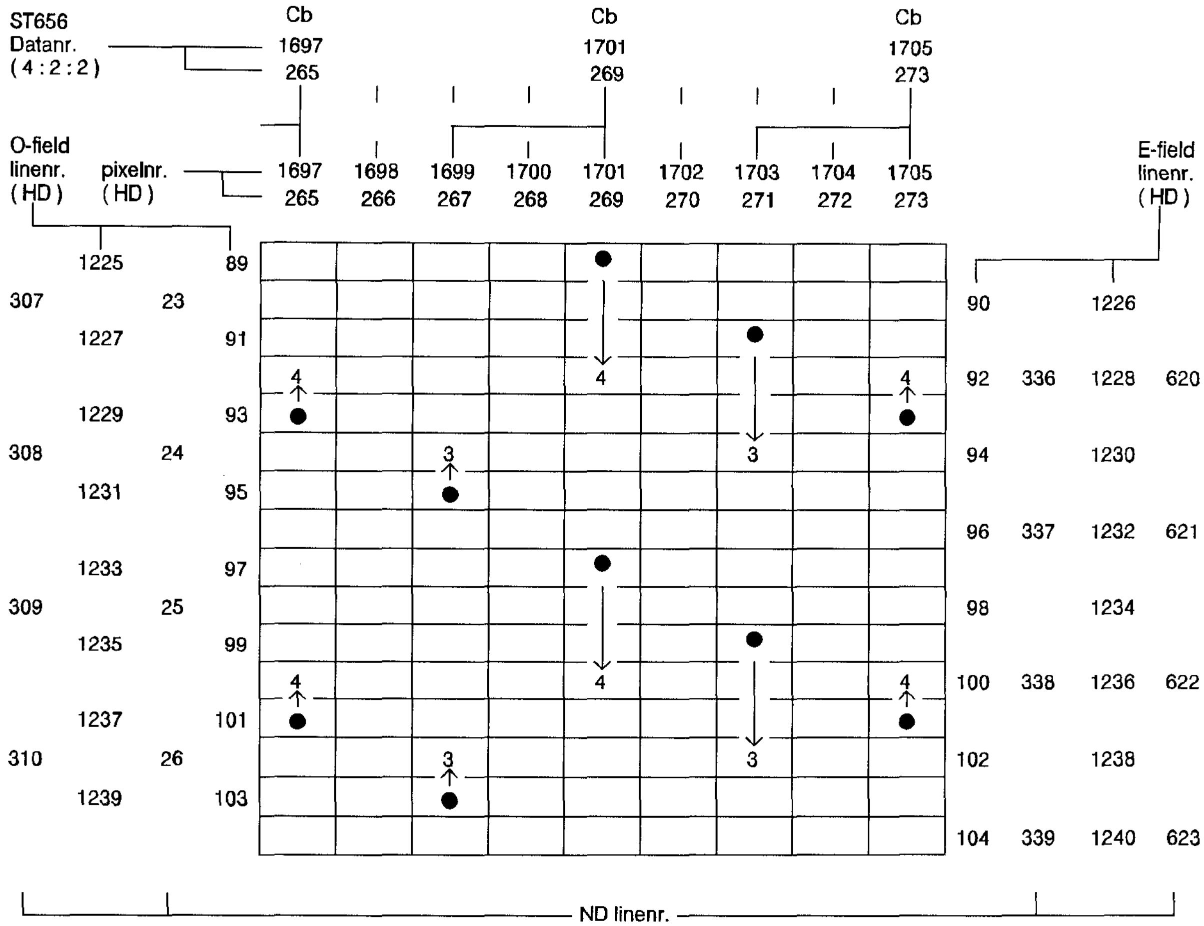


Figure 24c: Cb shuffling field 3, 4 (80 ms and 20 ms modes)

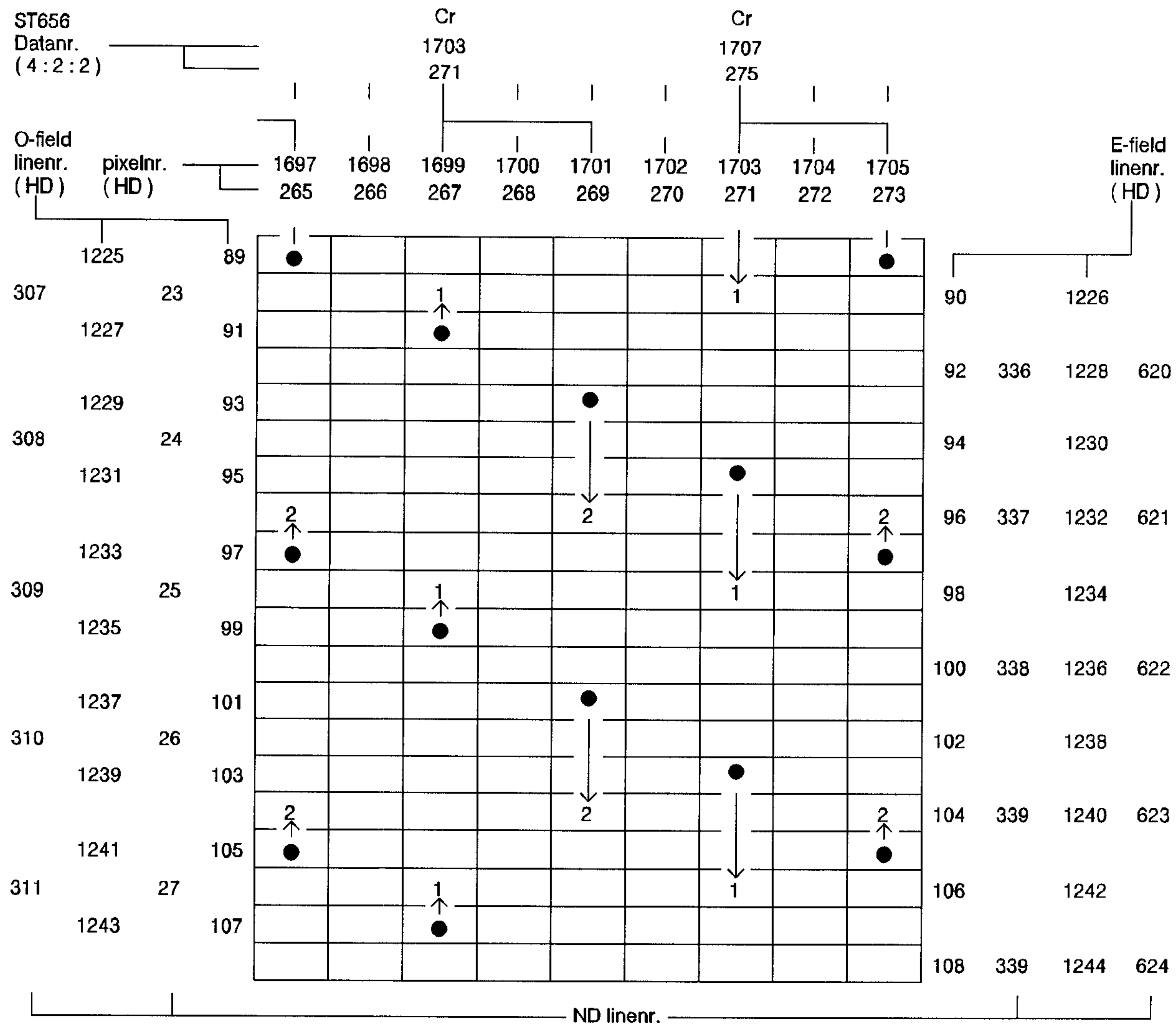


Figure 24d: Cr shuffling field 1, 2 (80 ms and 20 ms modes)

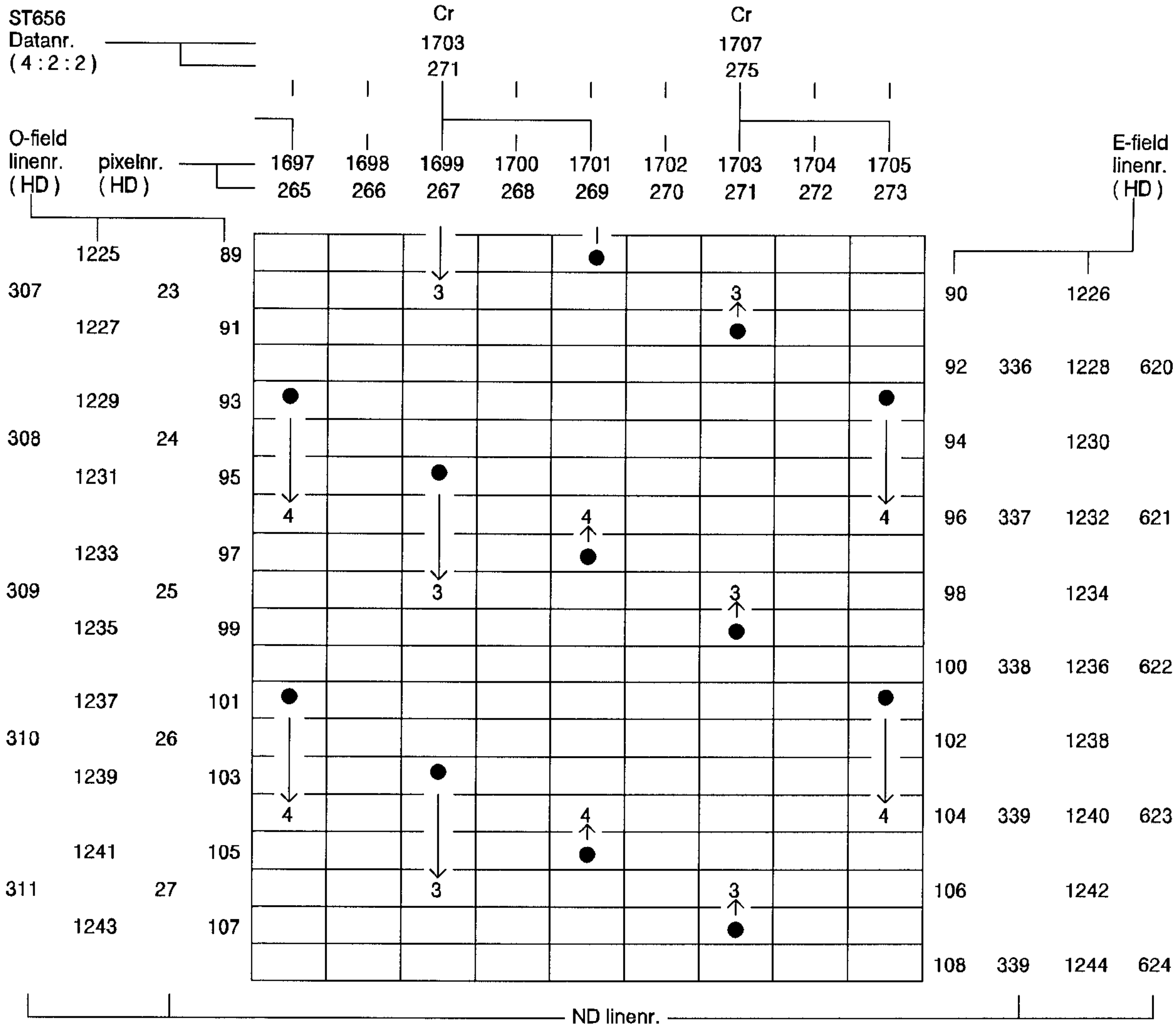


Figure 24e: Cr shuffling field 3, 4 (80 ms and 20 ms modes)

O-field linenr. (HD)	pixelnr. (HD)	1697 265	1698 266	1699 267	1700 268	1701 269	1702 270	1703 271	1704 272	1705 273		E-field linenr. (HD)
1225	89	2				2				2		
											90	1226
1227	91											
											92	1228
1229	93	1				1				1		
											94	1230
1231	95											
											96	1232
1233	97	2				2				2		
											98	1234
1235	99											
											100	1236
1237	101	1				1				1		
											102	1238
1239	103											
											104	1240

Figure 24f: Cb subsampling 40 ms mode

O-field linenr. (HD)	pixelnr. (HD)	1697 265	1698 266	1699 267	1700 268	1701 269	1702 270	1703 271	1704 272	1705 273		E-field linenr. (HD)
1225	89	1				1				1		
											90	1226
1227	91											
											92	1228
1229	93	2				2				2		
											94	1230
1231	95											
											96	1232
1233	97	1				1				1		
											98	1234
1235	99											
											100	1236
1237	101	2				2				2		
											102	1238
1239	103											
											104	1240
1241	105	1				1				1		
											106	1242
1243	107											
											108	1244

Figure 24g: Cr subsampling 40 ms mode

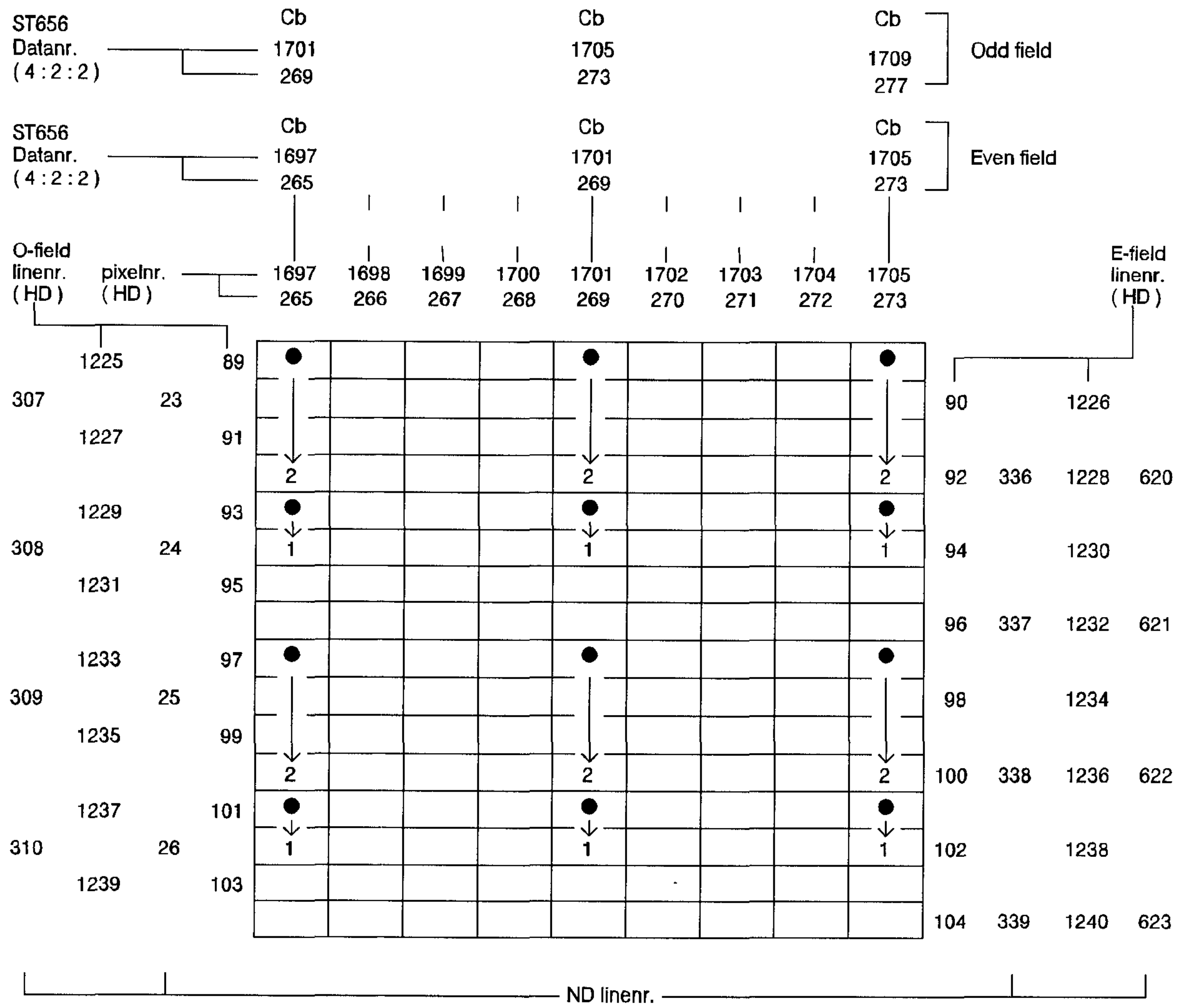


Figure 24h: Cb reallocation 40 ms mode (no shuffling required)

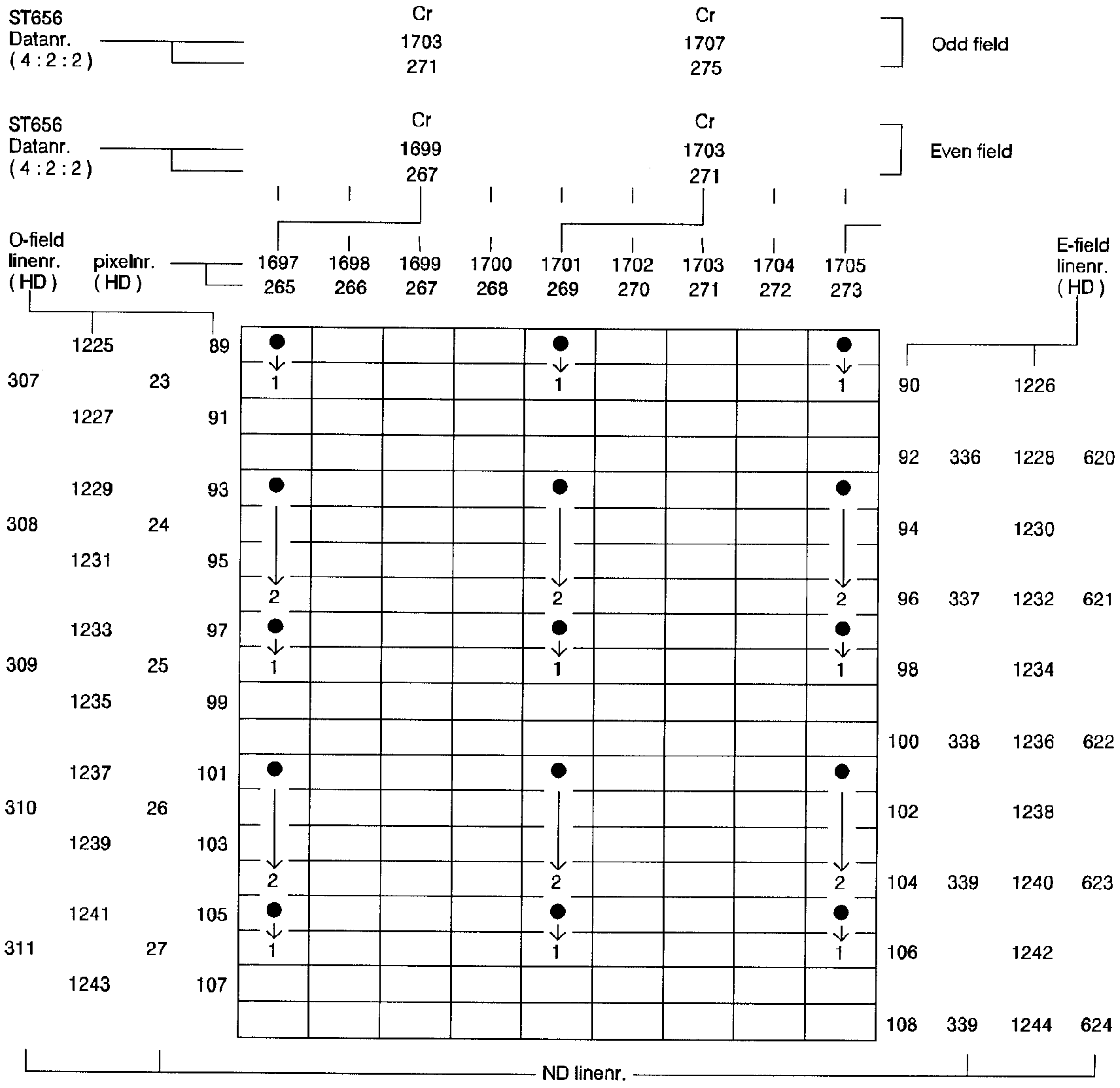


Figure 24j: Cr reallocation 40 ms mode (no shuffling required)

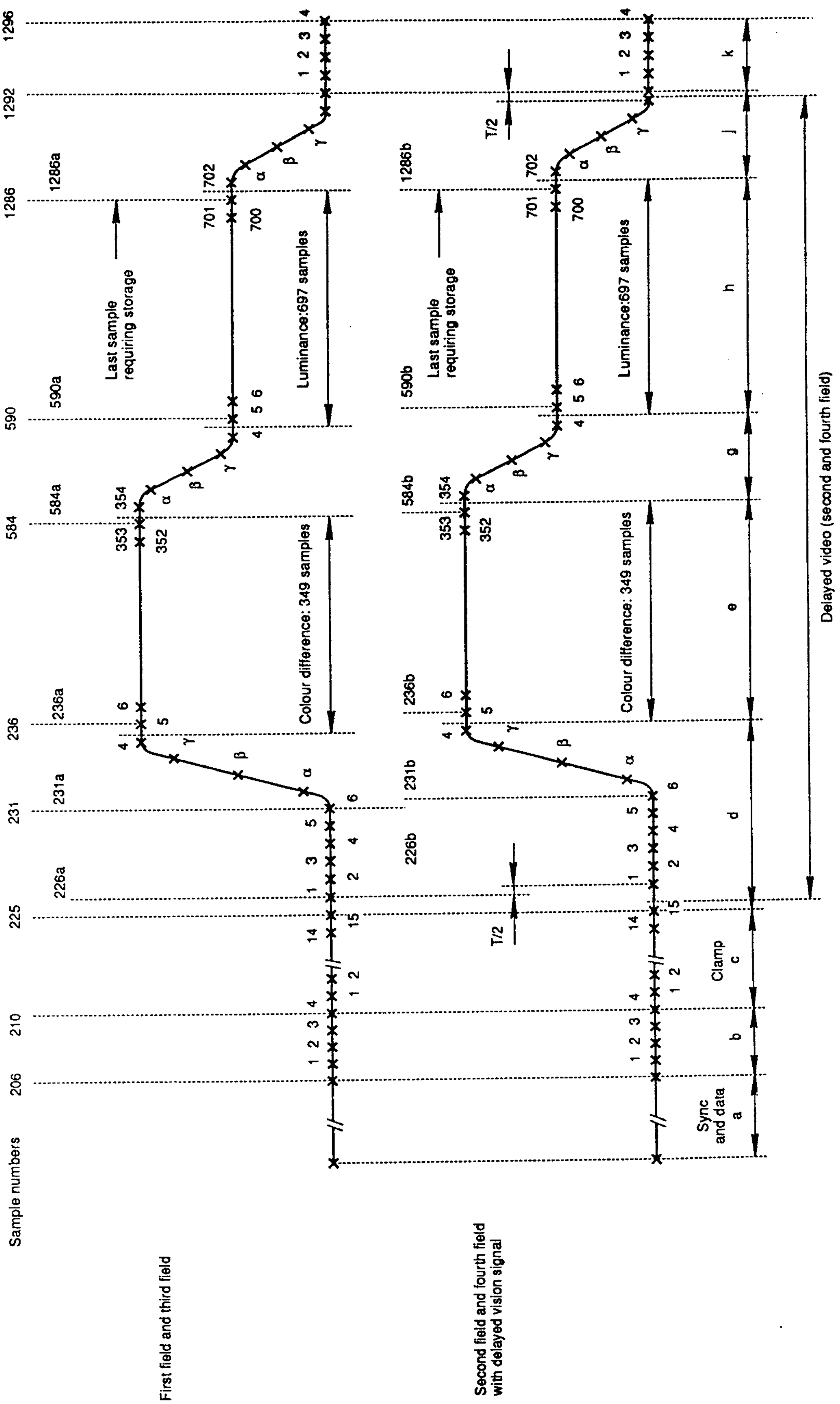


Figure 25: Representation of the D2-HDMAC video wave form (unscrambled) of the transmitter modulating signal (before pre-emphasis)

Remarks to figure 25:

a) (n) is the value of sample n;

b) colour difference and luminance samples correspond at the display position according to C(n) = L(2n-5);

c) Gr = grey level (reference clamping level);

d) for definition of a..k see also figure 1;

e) on even lines the colour difference sample number 5, which is the first active picture sample, is derived from sample 291 from the CCIR. Recommendation 656+ interface. Correspondingly on odd lines derivation of sample number 5 is from CCIR. Recommendation 656+ interface sample number 289;

f) in the first and third field the samples of the transmitted vision signal (indexed "a" samples) are co-incident with the 20,25 MHz sampling grid. In the second and fourth field the samples of the transmitted vision signal (indexed "b" samples) are delayed T/2 = 24,7 ns from the corresponding "a" sample position. See figure 21 for the four field sequence.

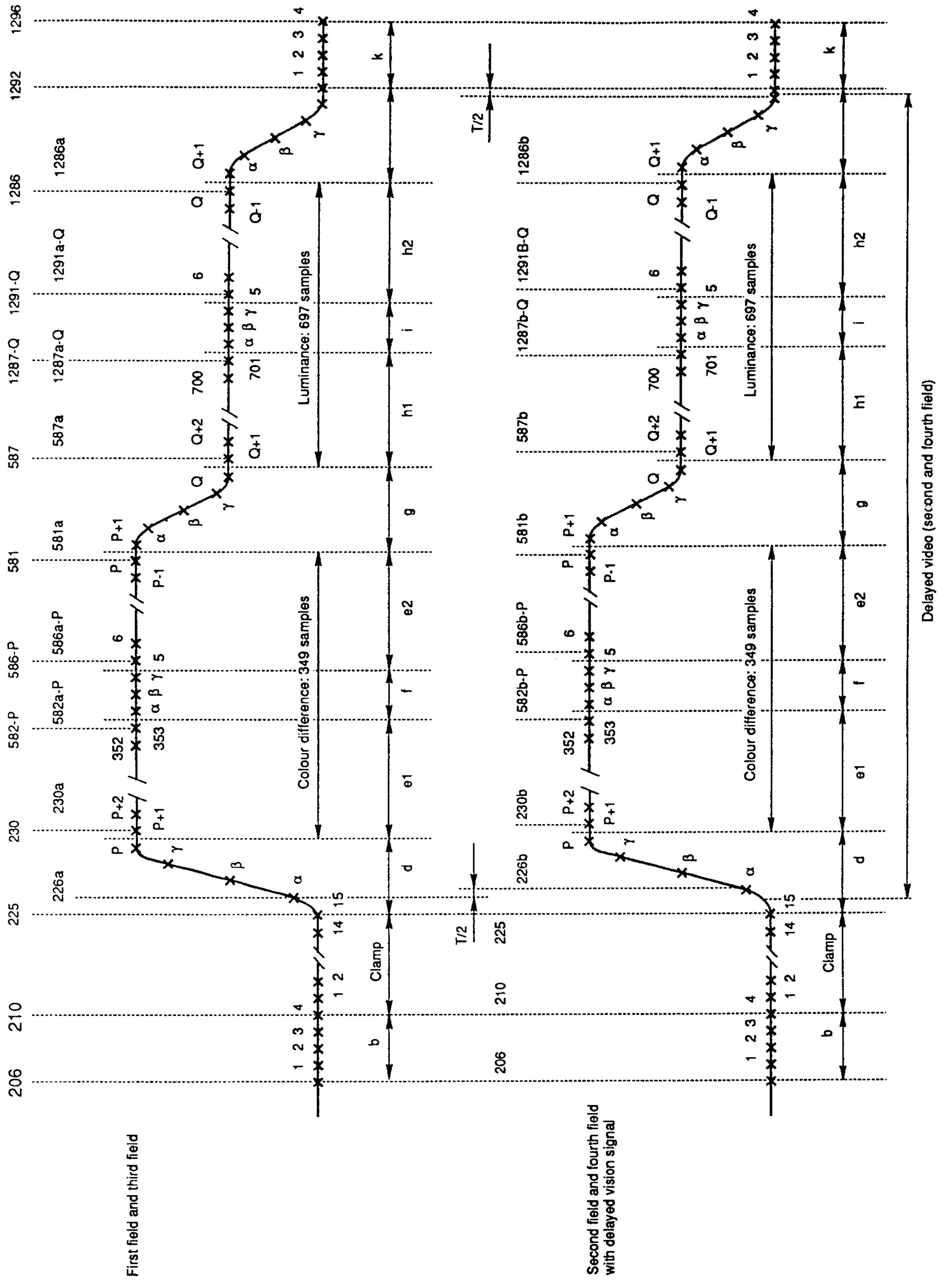


Figure 26a: Representation of the D2-HDMAC video wave form with double cut component rotation scrambling of the transmitter modulating signal (before pre-emphasis)

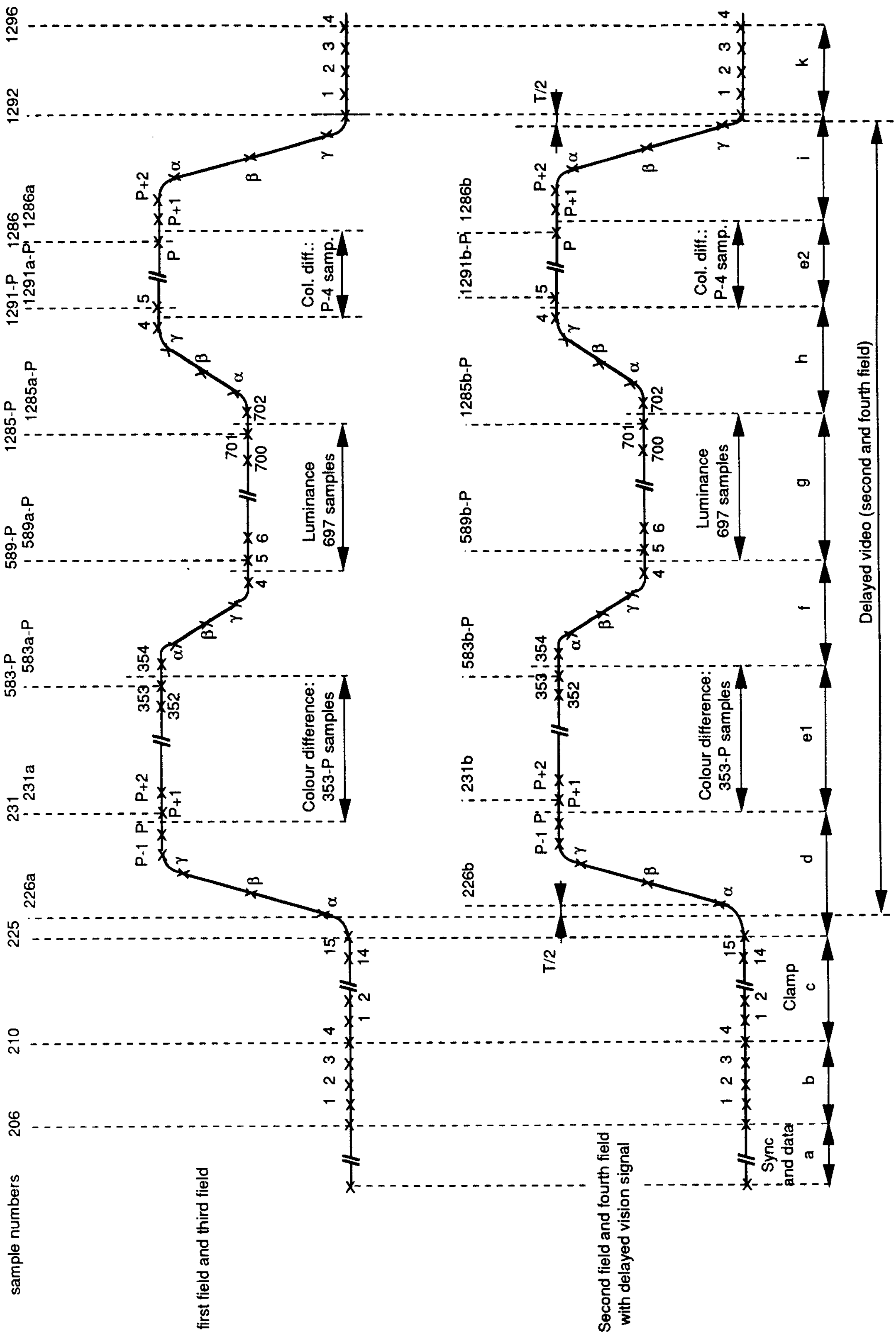


Figure 26b: Representation of the D2-HDMAC video wave form with single cut line rotation scrambling of the transmitter modulating signal (before pre-emphasis)

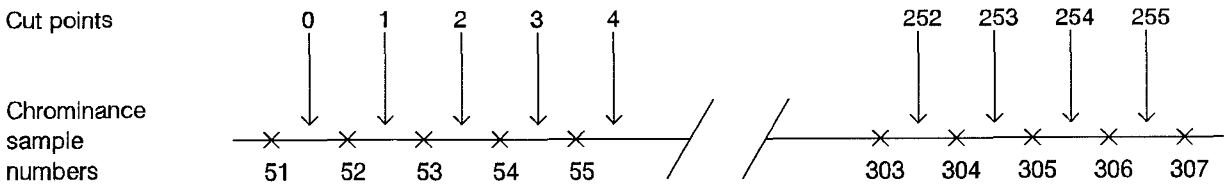


Figure 26c: Permissible cut points for the colour difference signal (double cut component rotation and single cut line rotation)

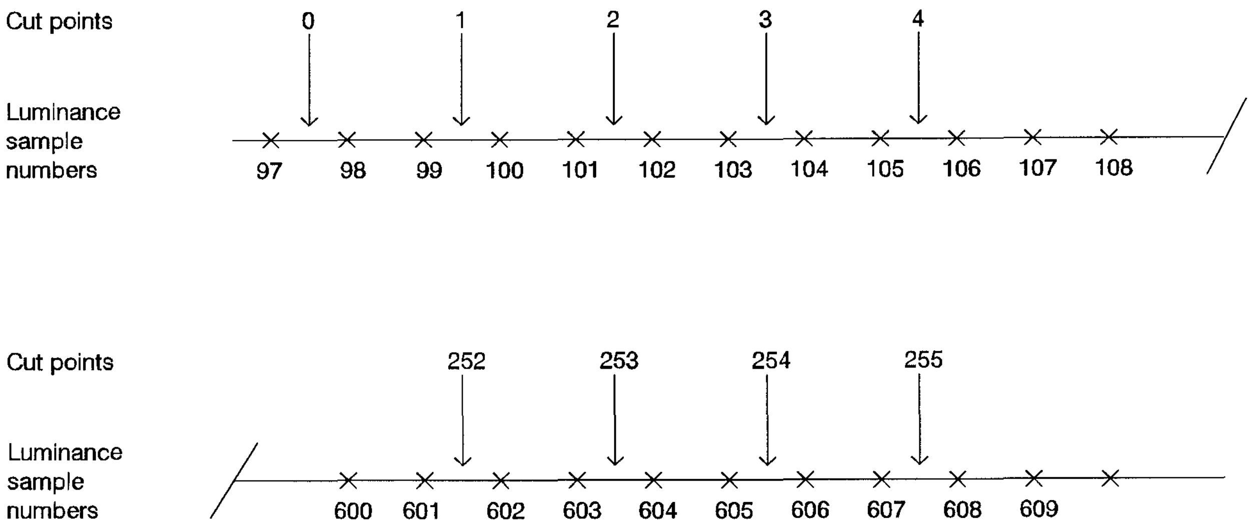


Figure 26d: Permissible cut points for the luminance signal (double cut component rotation)

See figure 25 for luminance and colour difference sample numbering.

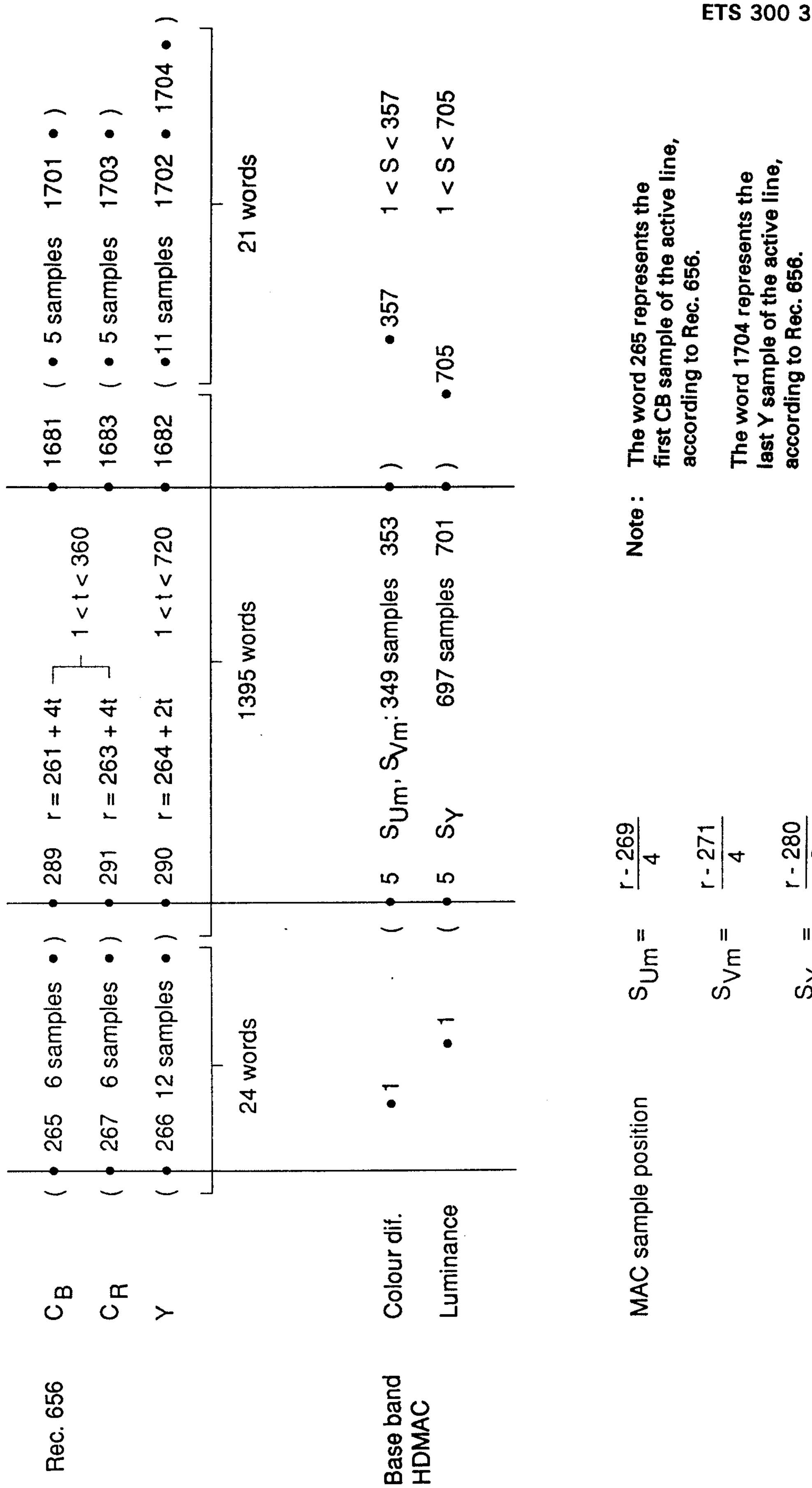


Figure 27: Relationship of component sample numbers from CCIR Recommendation 656 [5] to transmitted component sample numbers

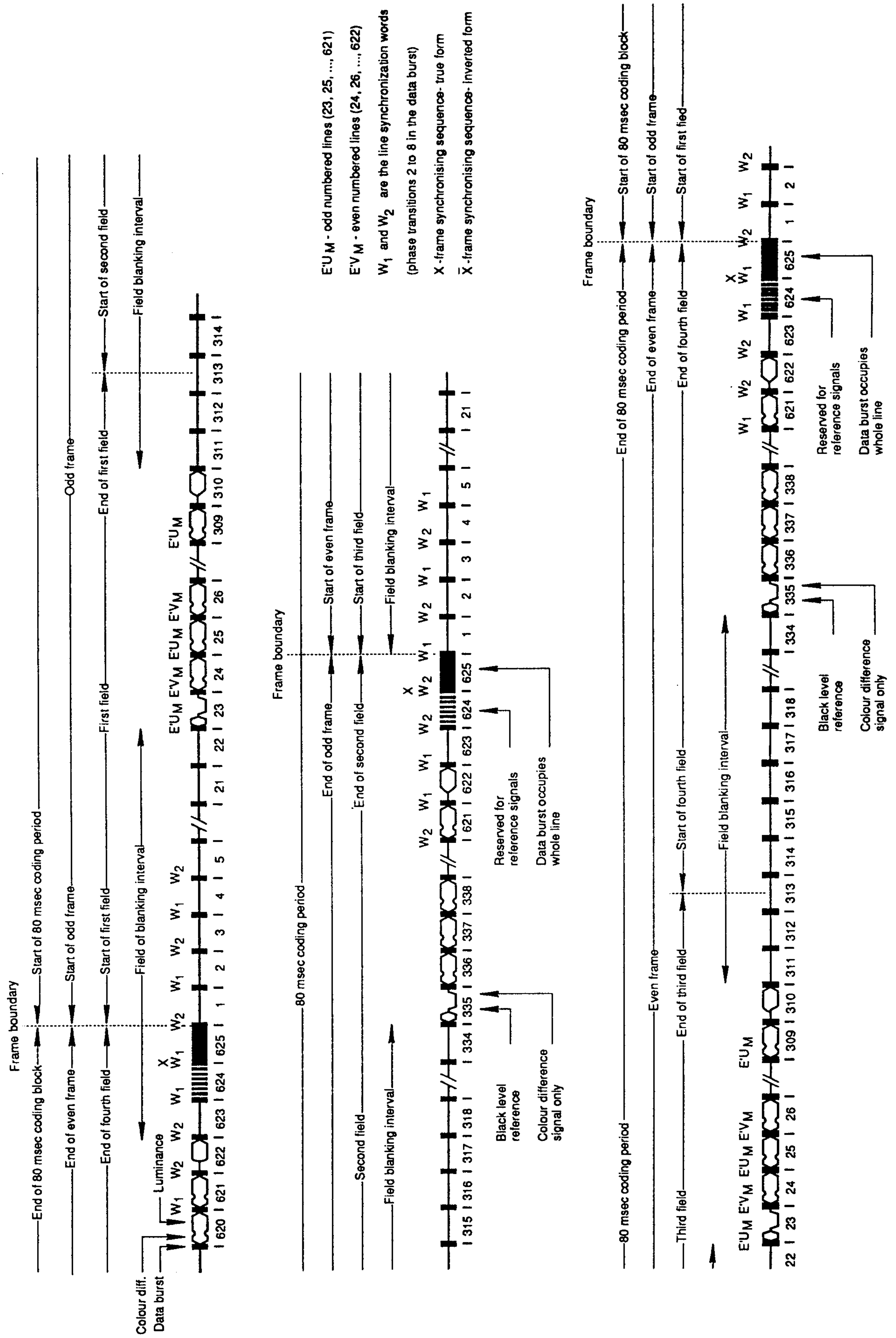


Figure 28: Diagrammatic representation of the transmitted signal showing the two frame sequence and the line numbering sequence (not to scale)

p=90	1689	1690	1691	....	1703	1704				
p=1	265	266	267	....	279	280	Y/Cb	Y/Cb	Cr	Cr
	n+0	n+1	n+2	....	n+14	n+15	q=1	q=72	q=1	q=72
m+0	1	2	3	....	15	16	89	1225	93	1229
m+1	2			....			90	1226	94	1230
m+2	3			....			91	1227	95	1231
m+3	4			....			92	1228	96	1232
....	....	....	....	....	....	....				
....	....	....	....	....	....	....				
m+12	13			....			101	1237	105	1241
m+13	14			....			102	1238	106	1242
m+14	15			....			103	1239	107	1243
m+15	16			....			104	1240	108	1244

Numbering conventions applied in this figure:

- Horizontal direction:
    - $n = p \times 16 + 249$
- Vertical direction (Y and Cb):
    - $m = q \times 16 + 73$
- Vertical direction (Cr only):
- Cr blocks are shifted 4 HD grid lines downwards with respect to Y/Cb blocks;
    - $m = q \times 16 + 77$
- Ranges:

  $q = 1 \ldots 72$
  $m = 92 \ldots 122$

NOTE: During blanking the branch decisions are set to the 20 ms mode. Due to the vertical shift of the Cr blocks the Cr lines 89, 90, 91 and 92 are processed in the 20 ms mode.

Figure 29: Block allocation 16 × 16 on HD luminance sampling grid

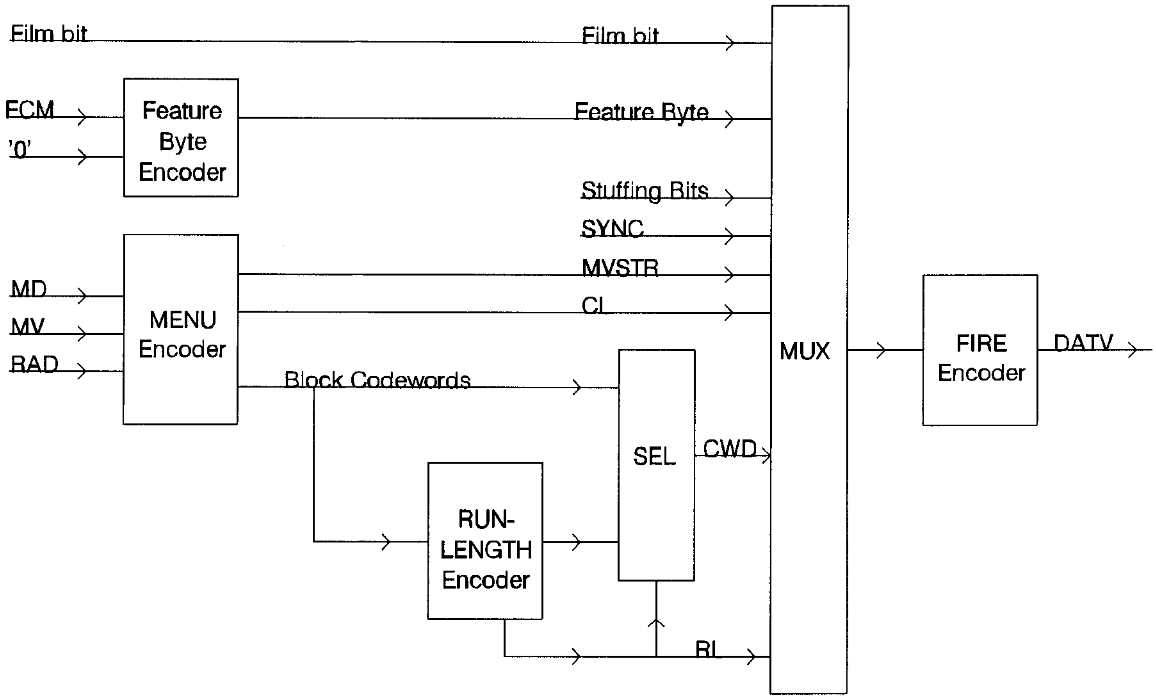


Figure 30: DATV encoder block diagram

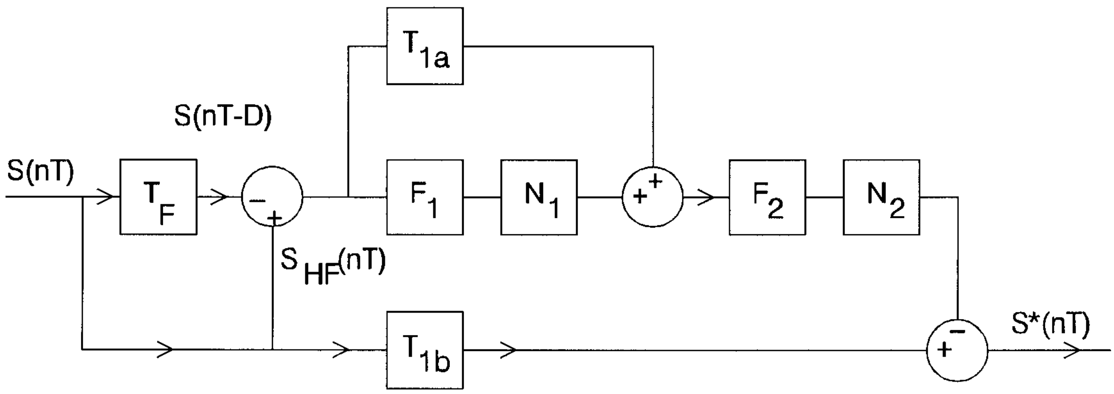


Figure 31: Transmitter TCI+ filter block diagram

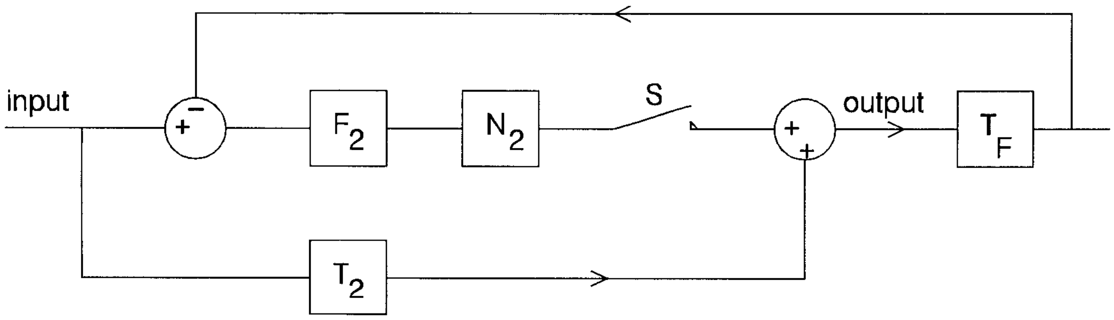


Figure 32: Receiver TCI+ filter block diagram

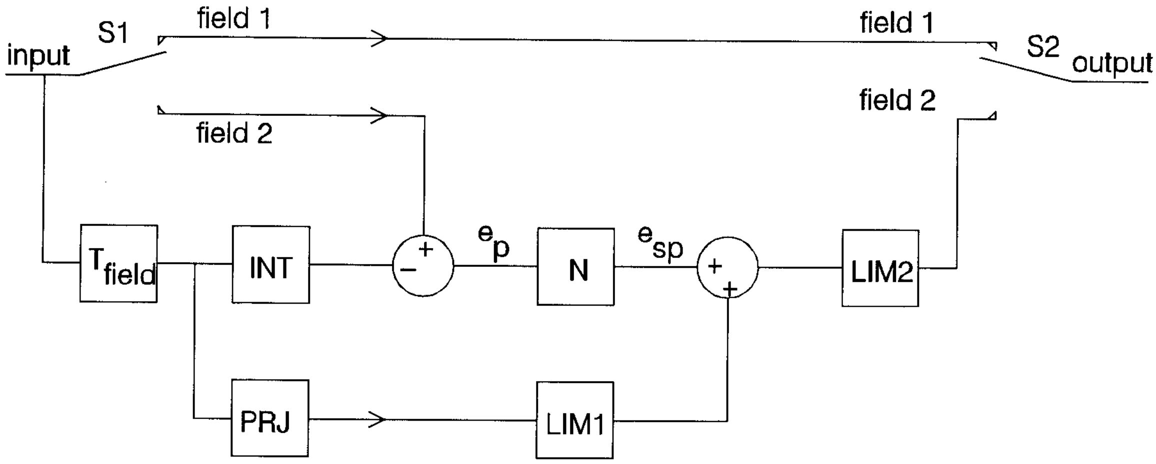


Figure 33: Transmitter MCCI+ filter block diagram

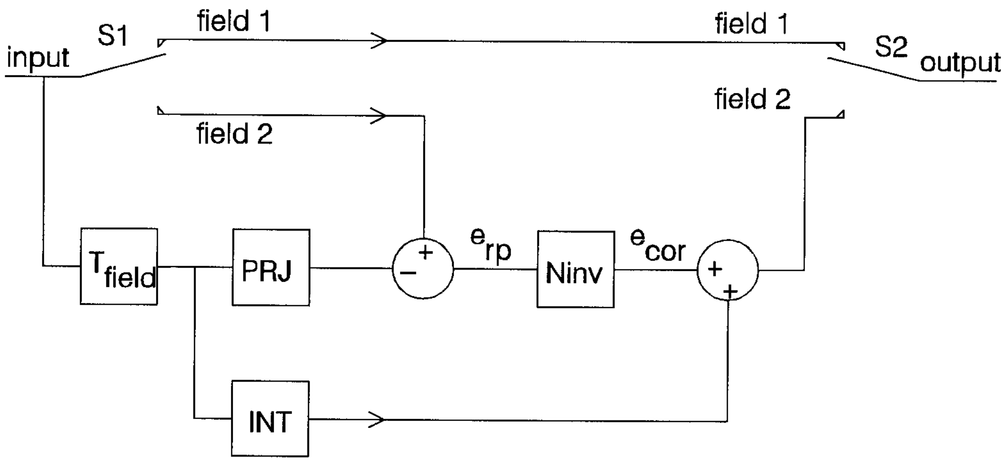


Figure 34: Receiver MCCI+ filter block diagram

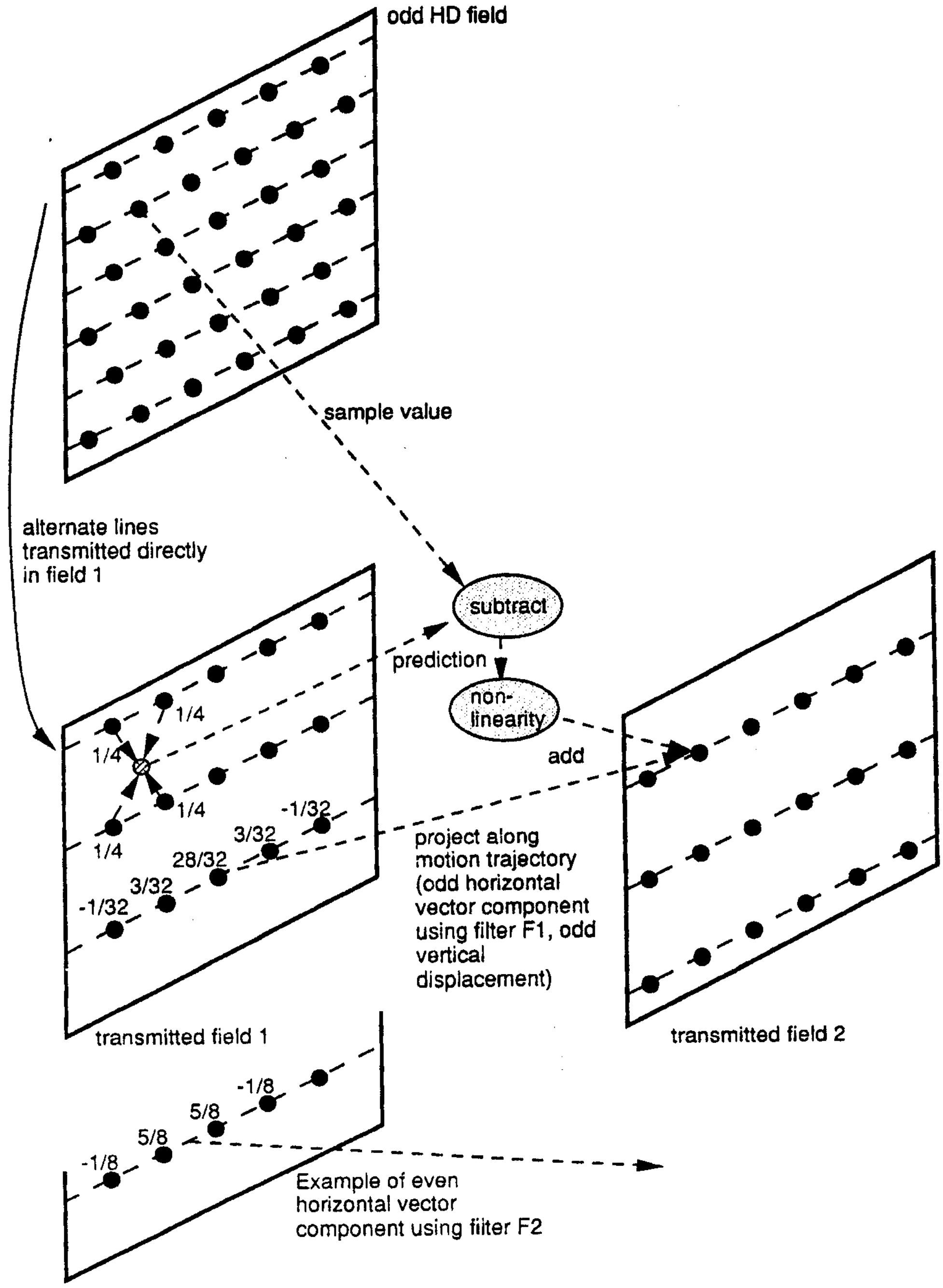


Figure 35: MCCI+ filter diagrammatic operation

6 Specification of sound coding methods

6.1 Introduction

Clause 6 contains the specifications of the sound coding methods (conforming to CCIR Report 953).

The sound components are transmitted in the sound/data packet multiplex (see Clause 4) and can be configured for a combination of different levels of quality and error protection. For example, one configuration can provide, with basic error protection, up to four high-quality companded sound channels, or three high-quality linear sound channels (with capacity for auxiliary data); another can provide, with improved error protection, two high-quality linear sound channels (with capacity for auxiliary data). But any combinations of the sound channel types and data capacity can co-exist in the same transmission providing they require less than the overall bit-rate.

6.2 Sound coding methods

The present specification considers three different types of sound signals:

a) high quality stereophonic sound;

b) high quality monophonic sound;

c) monophonic sound with a reduced bandwidth.

Two coding methods are specified. Their main characteristics are presented in table 18. Figure 36 details the actual codes for linear and companded signals.

Table 18

<table>
<tr><td colspan="2">Bandwidth</td><td>High quality 40-15000 Hz</td><td>Medium quality 40-7000 Hz</td></tr>
<tr><td colspan="2">Type

Sampling frequency
(see NOTE 2)
Accuracy
Conversion
Overload level
Pre-emphasis</td><td>stereophonic
or monophonic
32 kHz

10⁻⁶
14 bit/sample
12 dBm
J.17-CCITT
(-6,5 dB at 800 Hz)</td><td>monophonic (see NOTE 1)

16 kHz

10⁻⁶
14 bit/sample
12 dBm
J.17-CCITT
(-6,5 dB at 800 Hz)</td></tr>
<tr><td>1)</td><td>Linear coding</td><td>Linear code:</td><td>14 bits per sample
2's complement configuration</td></tr>
<tr><td>2)</td><td>Near-instantaneous coding</td><td>Law:

Coding:</td><td>near-instantaneous law with blocks of 32 successive samples
10 bits per sample
2's complement configuration
scale factor on five ranges</td></tr>
<tr><td colspan="4">Figure 36 describes the structure of both linear and companded samples.
In stereophonic mode, the right and left channels are sampled simultaneously and coded separately.</td></tr>
</table>

NOTE 1: The coding methods and their formulation into coding blocks are identical for high quality and medium quality sound channels. It is not envisaged that medium quality sound channels will operate in stereophonic mode.

NOTE 2: For the synchronism between sampling frequencies of channels intended for mixing (see subclause 6.7.5).

6.3 Protection of sound signals against errors

Two specific protection levels have been specified to offer respectively either four companded or three linear sound channels with a basic protection but enough for a national coverage area, or a lower capacity (three companded or two linear) but with an improved protection capability able to satisfy the criterion of sound failure at the same C/N level as the picture failure point.

6.3.1 First level protection

6.3.1.1 Companded mode

For companded 10-bit coding, one parity bit per sample shall be initially applied to the six most significant bits such that the parity group is even (i.e. the modulo-2 sum of the six sample bits and the parity bit is zero). Subsequently, the parity bits corresponding to 32 consecutive samples of a monophonic component or to 32 consecutive samples of each channel of a stereophonic component shall be modified to signal the scale factor and control information (see subclause 6.4.2 and figure 37b).

Table 19 presents the scale factor codes on 3 bits corresponding to the different companding and protection ranges.

a) the coding ranges shall be related to the characteristics of the near-instantaneous law (5 ranges).

b) the protection ranges shall be related to the following protection method:

- when the 32 samples which define the scale factor all have an amplitude lower than - 30 dB relative to the overload level, the corresponding protection range is the 6th: the X_8 bit of the companded sample is equal to the X_9 bit for each of the 32 samples. Consequently, any difference between X_9 bit and X_8 indicates an error;

- a similar procedure can be applied for blocks of 32 samples lower than -36 dB where the X_8 and the X_7 bits are equal to the X_9 bit (7th protection range).

Table 19: Scale factors

Level (dB) relative to overload	Coding ranges	Protection ranges	Silence period	Scale factor value		
				R2	R1	R0
0 to -6	1st range	1st range	no	1	1	1
-6 to -12	2nd range	2nd range	no	1	1	0
-12 to -18	3rd range	3rd range	no	1	0	1
-18 to -24	4th range	4th range	no	0	1	1
-24 to -30	5th range	5th range	no	1	0	0
-30 to -36	5th range	6th range	no	0	1	0
< -36	5th range	7th range	no	0	0	1
< -36	5th range	7th range	yes	0	0	0

6.3.1.2 Linear mode

For linear 14-bit coding, one parity bit per sample shall be initially applied to the eleven most significant bits such that the parity group is even (i.e. the modulo-2 sum of the eleven bits and the parity bit is zero). Subsequently, the parity bits corresponding to 32 consecutive samples of a monophonic component or to 32 consecutive samples of each channel of a stereophonic component shall be modified to signal the scale factor and control information (see subclause 6.4.1 and figure 37a).

The recovery of the scale factor can be used to limit click levels by the following protection method:

- when the 32 samples which define the scale factor all have an amplitude lower than -6 dB relative to the overload level, the corresponding coding range is the 2nd: the X_{12} bit of the sample is equal to the X_{13} bit for each of the 32 samples. Consequently, any difference between X_{13} and X_{12} indicates an error;
- a similar procedure can be applied for blocks of 32 samples lower than -12 dB where the X_{12} and X_{11} bits are equal to the X_{13} bit (3rd protection range);
- and so on for the seven protection ranges.

6.3.2 Second level protection

6.3.2.1 Companded mode

For companded 10-bit coding, the extended Hamming (11,6) code shall be applied to each sample. Subsequently, one parity bit for 32 consecutive samples of a monophonic component or the 32 consecutive samples of each channel of a stereophonic component shall be modified to signal the scale factor and control information (see subclause 6.4.4 and figure 37d).

Figure 38 presents coding and decoding tables.

Table 19 (see subclause 6.3.1.1) presents the 3-bit scale factor codes corresponding to the different companding and protection ranges.

The same procedure as described in subclause 6.3.1.1 can be applied to perform the protection of the samples.

6.3.2.2 Linear mode

For linear 14-bit coding, the extended Hamming (16,11) code shall be applied to the eleven most significant bits of each sample.

Subsequently, one parity bit corresponding to 18 consecutive samples for a monophonic component or to 18 consecutive samples for each channel of a stereophonic component shall be modified to signal part of the scale factor information. The remaining part of the scale factor information shall be transmitted using spare bits available.

NOTE: The control information is sent using also five spare bits available in the coding block (see subclause 6.4.3 and figure 37c).

Figure 38 presents coding and decoding tables.

The same procedure as described in subclause 6.3.1.2 can be applied to perform protection of the samples.

6.4 Structure of the sound coding blocks

Four coding block structures have to be considered.

6.4.1 Linear mode and first level protection

One coding block shall contain 64 samples of 15 bits:

either;

- 64 successive samples in monophonic mode. The scale factor FE1 shall be related to the first 32 samples. The scale factor FE2 shall be related to the second 32 samples; or
- 2 x 32 samples related to the A (left) and B (right) channels in stereophonic mode. The scale factor FE1 shall be related to the 32-A channel samples. The scale factor FE2 shall be related to the 32-B channel samples.

Figure 37a describes the structure of the coding block.

The three-bit information of the scale factor code R_2, R_1, R_0 (see table 19 in subclause 6.3.1.1) shall be transmitted by the use of the original parity bit P_i of sample i which becomes P'_i by the relation:

$P'_i = P_i \oplus R_j$

The values of i and j are given in subclauses 6.4.1.1 and 6.4.1.2.

6.4.1.1 Stereophonic mode

See figure 37a for notations.

Two scale factor values are transmitted:

FE1 = R_{2A}, R_{1A}, R_{0A} and FE2 = R_{2B}, R_{1B}, R_{0B};

and

$P'_i = P_i \oplus R_{2A}$ for i = 1, 7, 13, 19, 25, 31, 37, 43, 49;
$P'_i = P_i \oplus R_{1A}$ for i = 3, 9, 15, 21, 27, 33, 39, 45, 51;
$P'_i = P_i \oplus R_{0A}$ for i = 5, 11, 17, 23, 29, 35, 41, 47, 53;
$P'_i = P_i \oplus R_{2B}$ for i = 2, 8, 14, 20, 26, 32, 38, 44, 50;
$P'_i = P_i \oplus R_{1B}$ for i = 4, 10, 16, 22, 28, 34, 40, 46, 52;
$P'_i = P_i \oplus R_{0B}$ for i = 6, 12, 18, 24, 30, 36, 42, 48, 54.

6.4.1.2 Monophonic mode

See figure 37a for notations.

Two scale factor values are transmitted:

FE1 = R_{2n}, R_{1n}, R_{0n} and FE2 = R_{2n+1}, R_{1n+1}, R_{0n+1}; and

$P'_i = P_i \oplus R_{2n}$ for i = 1, 4, 7, 10, 13, 16, 19, 22, 25;
$P'_i = P_i \oplus R_{1n}$ for i = 2, 5, 8, 11, 14, 17, 20, 23, 26;
$P'_i = P_i \oplus R_{0n}$ for i = 3, 6, 9, 12, 15, 18, 21, 24, 27;
$P'_i = P_i \oplus R_{2n+1}$ for i = 28, 31, 34, 37, 40, 43, 46, 49, 52;
$P'_i = P_i \oplus R_{1n+1}$ for i = 29, 32, 35, 38, 41, 44, 47, 50, 53;
$P'_i = P_i \oplus R_{0n+1}$ for i = 30, 33, 36, 39, 42, 45, 48, 51, 54.

6.4.1.3 Control information

Control information is used to assure high-speed switching for a process which is identified in the interpretation block of the sound component (e.g. music/speech control, see subclause 6.7 and table 20). This binary Control Information Bit (CIB) shall be related to each 32-sample group and shall be transmitted by the use of the original parity bit P_i of samples which becomes P'_i by the relations:

$P'_i = P_i \oplus CIB_1$ for i = 55, 56, 57, 58, 59;
$P'_i = P_i \oplus CIB_2$ for i = 60, 61, 62, 63, 64.

NOTE: Scale factor information and control information can be recovered by majority decision logic. Subsequently, the original parity can be restored.

6.4.2 Companded mode and first level protection

One coding block shall contain 64 samples of 11 bits:

either;

- 64 successive samples in monophonic mode. The scale factor FE1 shall be related to the first 32 samples. The scale factor FE2 shall be related to the second 32 samples; or
- 2 x 32 samples related to the A (left) and B (right) channels in stereophonic mode. The scale factor FE1 shall be related to the 32-A channel samples. The scale factor FE2 shall be related to the 32-B channel samples.

At the beginning of the block, 16 unallocated bits shall be free for future needs.

Figure 37b describes the structure of the coding block.

The scale factor shall be sent by signalling in parity as follows:

- The three-bit information of the binary range code R_2, R_1, R_0 (see table 19 in subclause 6.3.1.1) shall be transmitted by the use of the original parity bit P_i of the i sample which becomes P'_i by the relation:

  $P'_i = P_i \oplus R_j$

The values of i and j are given in subclauses 6.4.2.1 and 6.4.2.2.

6.4.2.1 Stereophonic mode

See figure 37b for notations.

Two scale factor values are transmitted:

FE1 = R_{2A}, R_{1A}, R_{0A} and FE2 = R_{2B}, R_{1B}, R_{0B}; and

$P'_i = P_i \oplus R_{2A}$ for i = 1, 7, 13, 19, 25, 31, 37, 43, 49;
$P'_i = P_i \oplus R_{1A}$ for i = 3, 9, 15, 21, 27, 33, 39, 45, 51;
$P'_i = P_i \oplus R_{0A}$ for i = 5, 11, 17, 23, 29, 35, 41, 47, 53;
$P'_i = P_i \oplus R_{2B}$ for i = 2, 8, 14, 20, 26, 32, 38, 44, 50;
$P'_i = P_i \oplus R_{1B}$ for i = 4, 10, 16, 22, 28, 34, 40, 46, 52;
$P'_i = P_i \oplus R_{0B}$ for i = 6, 12, 18, 24, 30, 36, 42, 48, 54.

6.4.2.2 Monophonic mode

See figure 37b for notations.

Two scale factor values are transmitted:

FE1 = R_{2n}, R_{1n}, R_{0n} and FE2 = R_{2n+1}, R_{1n+1}, R_{0n+1}; and

$P'_i = P_i \oplus R_{2n}$ for i = 1, 4, 7, 10, 13, 16, 19, 22, 25;
$P'_i = P_i \oplus R_{1n}$ for i = 2, 5, 8, 11, 14, 17, 20, 23, 26;
$P'_i = P_i \oplus R_{0n}$ for i = 3, 6, 9, 12, 15, 18, 21, 24, 27;
$P'_i = P_i \oplus R_{2n+1}$ for i = 28, 31, 34, 37, 40, 43, 46, 49, 52;
$P'_i = P_i \oplus R_{1n+1}$ for i = 29, 32, 35, 38, 41, 44, 47, 50, 53;
$P'_i = P_i \oplus R_{0n+1}$ for i = 30, 33, 36, 39, 42, 45, 48, 51, 54.

6.4.2.3 Control information

Control information is used to assure high speed switching for a process which is identified in the interpretation block of the sound component (see subclause 6.7 and table 20). This binary control information bit (CIB) shall be related to each 32-sample group and shall be transmitted by the use of the original parity bit P_i of samples which becomes P'_i by the relations:

$P'_i = P_i \oplus CIB_1$ for i = 55, 56, 57, 58, 59;
$P'_i = P_i \oplus CIB_2$ for i = 60, 61, 62, 63, 64.

NOTE: Scale factor information and control information can be recovered by majority decision logic. Subsequently, the original parity can be restored.

6.4.3 Linear mode and second level protection

One coding block shall contain 36 samples of 19 bits:

either;

- 36 successive samples in monophonic mode. The scale factor FE1 shall be related to the first 18 samples. The scale factor FE2 shall be related to the second 18 samples; or

- 2 x 18 samples related to the A (left) and B (right) channels in stereophonic mode. The scale factor FE1 shall be related to the 18-A channel samples. The scale factor FE2 shall be related to the 18-B channel samples.

At the beginning of the block, a 36-bit field shall be formed of three parts:

- 8 unallocated bits, free for future needs;

- 10 control information bits (CI) used to send control information as described in subclause 6.4.3.3;

- 18 bits (B_i) used to partly send scale factor information as described in subclauses 6.4.3.1 and 6.4.3.2.

Figure 37c describes the structure of the coding block.

The scale factor value shall be sent partly by signalling in parity P^5 and partly in the 18 (B_i) bits reserved for this purpose at the beginning of the block.

6.4.3.1 Stereophonic mode

See figures 38 and 37c for notations.

Two scale factor values are transmitted:

FE1 = R_{2A}, R_{1A}, R_{0A} and FE2 = R_{2B}, R_{1B}, R_{0B}; and

$P'^5_i = P^5_i \oplus R_{2A}$ for i = 1, 7, 13, 19, 25, 31 and $R_{2A} = B_1 = B_4 = B_7$;
$P'^5_i = P^5_i \oplus R_{1A}$ for i = 3, 9, 15, 21, 27, 33 and $R_{1A} = B_2 = B_5 = B_8$;

$P'^5_i = P^5_i \oplus R_{0A}$ for i = 5, 11, 17, 23, 29, 35 and $R_{0A} = B_3 = B_6 = B_9$;
$P'^5_i = P^5_i \oplus R_{2B}$ for i = 2, 8, 14, 20, 26, 32 and $R_{2B} = B_{10} = B_{13} = B_{16}$;
$P'^5_i = P^5_i \oplus R_{1B}$ for i = 4, 10, 16, 22, 28, 34 and $R_{1B} = B_{11} = B_{14} = B_{17}$;
$P'^5_i = P^5_i \oplus R_{0B}$ for i = 6, 12, 18, 24, 30, 36 and $R_{0B} = B_{12} = B_{15} = B_{18}$.

6.4.3.2 Monophonic mode

See figures 38 and 37c for definition of notations.

Two scale factor values are transmitted:

FE1 = R_{2n}, R_{1n}, R_{0n} and FE2 = R_{2n+1}, R_{1n+1}, R_{0n+1}; and

$P'^5_i = P^5_i \oplus R_{2n}$ for i = 1, 4, 7, 10, 13, 16 and R2n = $B_1 = B_4 = B_7$;
$P'^5_i = P^5_i \oplus R_{1n}$ for i = 2, 5, 8, 11, 14, 17 and $R_{1n} = B_2 = B_5 = B_8$;
$P'^5_i = P^5_i \oplus R_{0n}$ for i = 3, 6, 9, 12, 15, 18 and $R_{0n} = B_3 = B_6 = B_9$;
$P'^5_i = P^5_i \oplus R_{2n+1}$ for i = 19, 22, 25, 28, 31, 34 and $R_{2n+1} = B_{10} = B_{13} = B_{16}$;
$P'^5_i = P^5_i \oplus R_{1n+1}$ for i = 20, 23, 26, 29, 32, 35 and $R_{1n+1} = B_{11} = B_{14} = B_{17}$;
$P'^5_i = P^5_i \oplus R_{0n+1}$ for i = 21, 24, 27, 30, 33, 36 and $R_{0n+1} = B_{12} = B_{15} = B_{18}$.

6.4.3.3 Control information

Control information is used to assure high-speed switching for a process which is identified in the interpretation block of the sound component (see subclause 6.7 and table 20). This CIB shall be related to each 18-sample group and shall be transmitted in the 10 (CI_i) bits reserved for this purpose at the beginning of the block.

$CIB_1 = CI_1 = CI_2 = CI_3 = CI_4 = CI_5$

$CIB_2 = CI_6 = CI_7 = CI_8 = CI_9 = CI_{10}$

NOTE: Scale factor information and control information can be recovered by majority-decision logic. Subsequently, the original parity can be restored.

6.4.4 Companded mode and second level protection

One coding block shall contain 64 samples of 15 bits:

either;

- 64 successive samples in monophonic mode. The scale factor value FE1 shall be related to the first 32 samples. The scale factor value FE2 shall be related to the second 32 samples; or

- 2 x 32 interleaved samples related to the A (left) and B (right) channels in stereophonic mode. The scale factor value FE1 shall be related to the 32-A channel samples. The scale factor value FE2 shall be related to the 32-B channel samples.

Scale factor value shall be transmitted in the parity bits as follows:

- The three-bit information of the binary range code R_2, R_1, R_0 shall be transmitted by the use of the original parity bit of the Hamming code ($P_1{}^5$, ..., $P_{64}{}^5$) to become transmitted parity values ($P'_1{}^5$, ..., $P'_{64}{}^5$).

Figure 37d describes the structure of the coding block.

6.4.4.1 Stereophonic mode

See figures 38 and 37d for notations.

Two scale factor values are transmitted:

FE1 = R_{2A}, R_{1A}, R_{0A} and FE2 = R_{2B}, R_{1B}, R_{0B}; and

$P'^5_i = P^5_i \oplus R_{2A}$ for i = 1, 7, 13, 19, 25, 31, 37, 43, 49;
$P'^5_i = P^5_i \oplus R_{1A}$ for i = 3, 9, 15, 21, 27, 33, 39, 45, 51;
$P'^5_i = P^5_i \oplus R_{0A}$ for i = 5, 11, 17, 23, 29, 35,41, 47, 53;
$P'^5_i = P^5_i \oplus R_{2B}$ for i = 2, 8, 14, 20, 26, 32, 38, 44, 50;
$P'^5_i = P^5_i \oplus R_{1B}$ for i = 4, 10, 16, 22, 28, 34, 40, 46, 52;
$P'^5_i = P^5_i \oplus R_{0B}$ for i = 6, 12, 18, 24, 30, 36, 42, 48, 54.

6.4.4.2 Monophonic mode

See figures 38 and 37d for notations.

Two scale factor values are transmitted:

FE1 = R_{2n}, R_{1n}, R_{0n} and FE2 = R_{2n+1}, R_{1n+1}, R_{0n+1}; and

$P'^5_i = P^5_i \oplus R_{2n}$ for i = 1, 4, 7, 10, 13, 16, 19, 22, 25;
$P'^5_i = P^5_i \oplus R_{1n}$ for i = 2, 5, 8, 11, 14, 17, 20, 23, 26;
$P'^5_i = P^5_i \oplus R_{0n}$ for i = 3, 6, 9, 12, 15, 18, 21, 24, 27;
$P'^5_i = P^5_i \oplus R_{2n+1}$ for i = 28, 31, 34, 37, 40, 43, 46, 49, 52;
$P'^5_i = P^5_i \oplus R_{1n+1}$ for i = 29, 32, 35, 38, 41, 44, 47, 50, 53;
$P'^5_i = P^5_i \oplus R_{0n+1}$ for i = 30, 33, 36, 39, 42, 45, 48, 51, 54.

6.4.4.3 Control information

Control information is used to assure high speed switching for a process which is identified in the interpretation block of the sound component (see subclause 6.7 and table 20). This CIB shall be related to each 32-sample group and shall be transmitted by the use of the original parity bit P_i^5 of samples which becomes P'^5_i by the relations:

$P'^5_i = P^5_i \oplus CIB_1$ for i = 55, 56, 57, 58, 59;

$P'^5_i = P^5_i \oplus CIB_2$ for i = 60, 61, 62, 63, 64.

NOTE: Scale factor information and control information can be recovered by majority decision logic. Subsequently, the original parity can be restored.

6.5 Insertion of the coding block into packet structure

6.5.1 General

The length of the sound coding blocks shall depend on the coding law and on the protection level:

- linear law and first protection level : 120 bytes;
- companded law and first protection level : 90 bytes;
- linear law and second protection level : 90 bytes;
- companded law and second protection level : 120 bytes.

The total length of the useful data area in the packet structure as described in subclause 4.3.6.2 is 91 bytes (i.e. 728 bits).

The first byte of the data block shall be used to indicate the type of packet (i.e. sound packet or control packet). This information shall be carried by the PT byte. The remaining 90 bytes shall convey either control or sound information.

Two types of coding blocks can be distinguished:

- sound coding block (BC) as described in subclause 6.4;
- interpretation block (BI) as described in subclause 6.7.

The coding blocks shall immediately follow the PT byte which characterizes their nature.

The PT byte informs the sound decoder of the nature of the sound signal being received. For example, the sound decoder needs to know whether the coding law is linear or near-instantaneously companded, and also whether the audio signal is at full bandwidth (high quality) or reduced bandwidth. This information shall be conveyed by occasional "interpretation" packets. These packets shall have the same address as the packets containing the sound samples and scale-factor information but shall be identified by having different PT bytes at the start of the useful data block. The useful data block shall be designated as an "interpretation block" (BI) when it contains data for setting-up the sound decoder; interpretation blocks shall have a PT byte with either the BI1 configuration or, if it extends beyond a single packet length, the BI2 configuration. When the block contains sound samples it shall be designated as a "sound coding block" (BC), and the PT byte shall have either the BC1 configuration or the BC2 configuration.

The configurations BC1 and BC2 shall characterize sound coding blocks. Their use shall be alternated, which allows the precise switching process from one coding structure to another and optionally the synchronization between different sound components and the television signal, by following the procedure described below.

The configurations BI1 and BI2 shall characterize the interpretation blocks, the structure of which is described in subclause 6.7. These blocks shall be transmitted at a low and regular rate (e.g. between one and three per second) when the nature of the sound signal is unchanged. These blocks can additionally or alternatively contain information about the next configuration to be used on that packet address. The switching process in the receiver decoder is realised when the "PT" byte related to the sound coding blocks changes from one configuration to the other (either BC1 to BC2 or BC2 to BC1), for more than a single packet, and it takes effect from the third packet after the change.

In the optional case where there is a requirement to co-time different sound components (this may for example be required in the case of sound channels intended for automatic mixing, see subclause 6.7.5), a packet carrying sound data and occurring within a given television frame may carry an isolated alternation between BC1 and BC2 for that packet only. The first samples (or sample pairs) of all such selected packets are intended to be applied to the output digital-to-analogue converters simultaneously, at a nominal time corresponding to the start of line 32 of the following television frame. (In some cases, separate decoders for each sound component may then have to operate simultaneously.)

The four configurations for the PT byte shall have the following definition:

PT byte	MSB-----LSB	hexadecimal
BC1	11000111	'C7
BC2	11111000	'F8
BI1	00000000	'00
BI2	00111111	'3F

with the least significant bit transmitted first.

6.5.2 Insertion of a 90-byte coding block

The insertion of a 90-byte coding block into the packet structure as described in subclause 6.4 is realised by placing the start of each coding block immediately after the PT byte. This relation is presented in figure 39.

6.5.3 Insertion of a 120-byte coding block

The insertion of a 120-byte coding block into the packet structure as described in subclause 6.4 shall be realised by inserting three successive coding blocks into four successive packets in the manner shown in figure 40. They shall be understood as successive packets which contain the same sound component address. They may be interleaved in the global bit stream with packets related to other components.

6.5.4 Continuity index (see subclause 4.3.6.1.2)

The continuity index of any BC packet (PT = BC1 or BC2) shall be incremented from that of the previous BC packet of the same address.

The continuity index of any BI packet (PT = BI1 or BI2) (see subclause 6.7) shall be decremented from that of the previous packet of the same address, regardless of its packet type.

6.5.5 Buffer storage

The multiplex shall be assembled by the broadcaster in such a way that the buffer store at the transmitting end shall never exceed 12 packets for each sound component in the multiplex. This figure has been chosen in order to keep capacity available for future applications.

6.5.6 Scrambling for the free-access and the controlled-access modes of the conditional-access system

In addition to bit interleaving and transmission scrambling specified in subclauses 4.3.6.4 and 4.3.7, the sound coding blocks may be scrambled by modulo 2 addition of a pseudo-random sequence. Such scrambling is used in the controlled-access mode and may, at the option of the broadcaster, be used in the free-access mode. See Clause 9 for details.

It should be noted that, where used:

a) scrambling is applied to the sound coding blocks after error protection; in the receiver, error detection or correction should therefore be made after the descrambling process;

b) scrambling is neither applied to the packet headers nor to the PT byte and the BI blocks defined in subclause 6.7;

c) when scrambling is used, the digital sound signals are subjected first to scrambling for the conditional access system and then to bit interleaving and transmission scrambling (for spectrum shaping purposes, see subclause 4.3.8).

6.6 Transmission of silence information

Silence information is concerned with the means of managing the receiver buffer storage in order to obtain a smooth, regular output of sound samples. It indicates that all the transmitted samples inserted in the sound block have a level lower than a threshold in the coder.

The characteristics defining a silence shall be:

- level:
  - variable, but always lower than or equal to - 36 dB relative to the overload level (the level of - 36 dB corresponds to the upper limit of the 7th protection range);
- duration:
  - variable, but always greater than or equal to 8 ms.

The transmission of silence information shall be realised in the scale factor (see table 19 in subclause 6.3.1.1).

In the stereophonic mode, a silence period can only be detected if the silence characteristics are present on the two channels simultaneously.

6.7 Specification of sound interpretation data

6.7.1 General

Several different messages can be multiplexed within one digital sound channel by using a block structure within the useful data zones of the packets.

Two types of block are defined in subclause 6.5.1:

- sound coding blocks (BC); and
- interpretation blocks (BI), (see subclause 6.5.1 for the normal repetition rate).

In each case, the address code shall be that allocated for the particular sound component. As specified in subclause 6.5.4, the continuity index of any BI packet shall be decremented from that of the previous packet of the same address, regardless of its packet type.

The sound interpretation data shall be inserted in interpretation blocks BI in accordance with the coding procedure specified in subclause 6.7.2. At the present time, only those data relating to the automatic control of the sound decoder configuration have been specified; these are necessary for the following principal reasons:

a) during the receiver switch-on period, they enable the sound decoder to be properly configured in order to produce the correct audio output for the selected component;

b) they prepare the decoder for forthcoming changes in the characteristics of the selected sound component.

They shall be grouped in the "commands" defined in subclause 6.7.3, for which the indicators shall have the value "CI = 1" and "CI = 2".

6.7.2 General structure for the coding of sound interpretation data

6.7.2.1 Commands

The data shall be assembled into commands, each of which shall be introduced by one of 16 Command Indicators (CI) as a Hamming-coded byte (see NOTE below). This shall be followed by a Hamming-coded byte Length Identifier (LI) indicating the length of the command in the range 0 - 14 bytes. If the command length is in the range 15-254 bytes, the first LI byte shall indicate length 15 and the next pair of Hamming-coded bytes shall be also LI bytes indicating the length.

NOTE: The Hamming code used for all Hamming-protected bytes defined in this subclause is described in table 21.

Command indicators having values 0, 1 and 2 are defined as follows:

- the command indicator CI = 0 shall introduce a list, in ascending order, of those CI's in the range 0 - 15 (Hamming protected) that include commands which are about to be changed. This procedure is explained in subclause 6.7.4;

- the command indicators CI = 1 and CI = 2 shall introduce a fivefold repetition of the sequence whose significance is detailed in subclause 6.7.3 and which contains the data needed for commanding the configuration of the decoder;

- the sound interpretation data shall be coded in a byte-oriented way. For each byte, the least significant bits shall be transmitted first.

6.7.2.2 Data groups

The commands shall be grouped into data groups which shall be transmitted within the interpretation blocks described below.

6.7.2.2.1 Structure of the interpretation blocks

The sequence of interpretation blocks within one digital sound channel provides a transmission capacity which shall be used to transmit commands. To allow a command to spread over more than one block, a delimitation mechanism is defined to indicate the beginning of a data group. This shall be done using the fourth possible value (BI2) of the packet type (PT) byte (see subclause 6.5.1). Both values BI1 and BI2 shall identify an interpretation block; BI1 shall indicate that the beginning of the block coincides with the beginning of a data group; BI2 shall indicate that the block carries the continuation of a data group. (If the entire data group is shorter than 86 bytes, BI1 is the only value which is used).

6.7.2.2.2 The data group transport

The positioning of the data group in the interpretation block packets is detailed in figure 41. Although this general coding scheme allows a maximum of 22692 bytes in the data group, the data group shall be restricted to a maximum of 264 bytes (i.e. three packets). In the case of four companded sound channels with first-level error protection or three linear sound channels with first-level error protection the length of the data group shall be further restricted to a single packet (86 bytes).

The sequence of BI packets may be interrupted by any number of BC packets having that same address, and any number of packets of other addresses, in any order. The values taken successively by the continuity index for BC and BI packets are defined in subclause 6.5.4.

Commands within a data group shall be transmitted in increasing CI order, but it should be noted that there may be more than one command with the same CI and that this may be a simple repetition or a different command.

Commands with the same CI may be transmitted in any order except that, if current and next versions of a command are sent, the current versions shall be sent first (see in particular an application of this sort in subclause 6.7.4).

The protection against errors shall be achieved by the combination of:

- the continuity check CC, itself Hamming protected (see figure 41);
- Hamming protection of the structuring bytes LI, CI, S1, S2, F1, F2.

6.7.3 Specification of commands having the indicators CI = 1 and CI = 2

The data needed to control the automatic configuring of the sound decoder shall be grouped in the commands having the indicators CI = 1 and CI = 2. The parameter fields of these two commands have the same meaning but CI = 1 shall refer to the existing state and CI = 2 shall refer to the future state. The organization of these parameter fields shall be in accordance with the following description. They shall be formed from two bytes labelled Byte No. 1 and Byte No. 2 in the order in which they shall be transmitted; bit 1 of each byte shall be transmitted first.

Byte No. 1

Bit 1: Parity check;

This bit is chosen such that the modulo-2 addition of all the bits in byte 1 and byte 2 shall be 0. It may used to enable errors in decoding the fivefold repetition of bytes 1 and 2 to be rejected.

Bit 2: SDFSCR flag;

This bit shall be set to 1 to indicate that details of the selected sound channel should not be stored in the receiver in association with the current SDFSCR code (see subclause 4.5.3). In particular, this bit shall be set to 1 in the case of a sound component in the controlled-access mode.

Bit 3: News flash indication;

The value 1 of this bit in the main (or "original") sound, or optionally also in a commentary (or "additional") sound, shall indicate that a digital news flash sound is currently being broadcast (the corresponding channel address is given in channel 0). The value 0 shall indicate that no news flash sound is currently being broadcast. In the event that the user wishes to hear news flash sound messages which are broadcast sporadically, the sound decoder shall, on receipt of the value 1, automatically switch to the channel carrying that component and return to the previous component(s) as soon as the news flash sound component has ended (as indicated by bit 8 of byte 1 in the news flash BI).

Bit 4: Identification of sound coding blocks;

The switching between two configurations of sound channel characteristics shall be achieved by BC1-BC2 alternation (see subclause 6.5.1).

Bit 4 shall ensure the correct phasing of this procedure by indicating to the decoder the value of the PT byte of the coding blocks associated with the configuration specified by the present command and defined by the pair of command indicators CI = 1 and CI = 2:

bit 4 = 1 shall correspond to the configuration BC1;

bit 4 = 0 shall correspond to the configuration BC2.

Bit 5: Temporal characteristic;

The selected sound signal may be continuous or intermittent, i.e. its transmission may not be permanent.

This situation arises in particular in commentary channels (see subclauses 6.7.5 and 6.7.6).

Bit 5 shall signal this characteristic;

bit 5 = 0: continuous;

bit 5 = 1: intermittent.

Bit 6: Function of the command information.

Bit 7

The coding blocks contain command information (CIB) which may be used in a variety of ways:

- Function 1:

  - it enables the user of suitably-equipped receivers to adjust the balance between the volume of reproduction of speech and music as desired:

    bit 6 = 0 selection of function 1;

    bit 7 = 0 CIB bit = 1 and CIB bit = 0 indicate that music and speech are respectively present in the channel.

- Function 2:

  it governs the control of the transition when the sound signal is interrupted or when its transmission is resumed (this function implies that bit 5 = 1) (see subclause 6.7.5 and 6.7.6):

  bit 6 = 0 selection of function 2;

  bit 7 = 1 CIB bit = 1 indicates that the main sound is increasing toward or at full volume; CIB bit = 0 indicates that the main sound is decreasing toward or at minimum volume.

- Configurations (1,0) and (1,1) have not yet been defined.
  In these cases, CIB bit shall be set to 0.

Bit 8: State

This bit shall be used when bit 5 = 1. It shall indicate the periods during which the sound signal is present.

bit 8 = 0: sound signal present;

bit 8 = 1: the broadcast is interrupted.

The use of this bit is described in greater detail in subclause 6.7.5. It shall be set to 0 when bit 5 = 0.

Byte No. 2

$\left.\begin{array}{l}\text{bit } 1\\ \text{bit } 2\\ \text{bit } 3\end{array}\right\}$ Audio configuration

Eight options are possible:

$\left.\begin{array}{l}\text{bit } 1 = 0\\ \text{bit } 2 = 0\\ \text{bit } 3 = 0\end{array}\right\}$ bandwidth 40 Hz - 15 kHz, monophonic

$\left.\begin{array}{l}\text{bit } 1 = 0\\ \text{bit } 2 = 0\\ \text{bit } 3 = 1\end{array}\right\}$ bandwidth 40 Hz - 15 kHz, stereophonic

$\left.\begin{array}{l}\text{bit } 1 = 0\\ \text{bit } 2 = 1\\ \text{bit } 3 = 0\end{array}\right\}$ bandwidth 40 Hz - 7 kHz, monophonic

The five other options have not yet been defined.

Bit 4: Automatic mixing

In the case of an additional sound channel, this bit shall indicate if this sound should be reproduced alone or if it is intended to be mixed in the receiver with the main sound. This bit shall be set to 1 for all additional sound channels for which mixing is intended.

In the case of the main sound channel, the automatic mixing bit shall indicate the relative attenuation to be used in the receiver to mix the main sound and any additional sound intended for mixing.

The mixing procedure is described in subclauses 6.7.5 and 6.7.6.

Additional sound channel:

bit 4 = 0 mixing not intended;

bit 4 = 1 mixing intended.

Main sound channel:

bit 4 = 0 equal attenuations;

bit 4 = 1 significant attenuation of the main sound.

Bit 5: Scrambling

This bit shall indicate whether the sound channel is subject to scrambling (see Clause 9).

bit 5 = 0 no scrambling;

bit 5 = 1 scrambling.

Bit 6: Conditional access

This bit shall indicates whether the sound channel is subject to free-access or to controlled-access (see Clause 9).

bit 6 = 0 free-access mode;

bit 6 = 1 controlled-access mode.

Bit 7: Coding law

Two coding laws are specified. This bit shall indicate which law is in use.

bit 7 = 0 linear law;

bit 7 = 1 companded law.

Bit 8: Level of error protection

Two levels are specified. This bit shall indicate which level is in use.

bit 8 = 0 first level;

bit 8 = 1 second level.

Table 20 lists all these functions and the codes.

Bit 5 of byte 1 and bits 4, 5 and 6 of byte 2 shall be repeated in the SI in the DCINF access coordinates of the digital audio components (see Annex A). The BI data may also be repeated in the SI in the DCINF complementary access coordinates of the digital audio components.

These two bytes shall be repeated five times so that the total length of the command defined by the indicators CI = 1 and CI = 2 are equal to 12 bytes (LI = 10). This repetition permits the use of majority-decision logic in the decoder as a means of providing error protection.

6.7.4 Procedure for updating the parameters of a sound configuration

In a steady state, the command with CI = 0 shall be either not transmitted or transmitted without 1 and 2 in its parameter field, and the command with CI = 1 shall be transmitted (describing the existing configuration).

When one or more of the parameters in a command defined by the indicator CI = 1 require to be modified, the following procedure shall be adopted:

- During the two seconds preceding the transition, the new configuration shall be signalled at least 3 times per second. The structure of the corresponding data group shall include at least three commands:
  - a first command with CI = 0 and first parameter byte equal to 2, signifying that command 2 has been updated;
  - a second command with CI = 1 describing the existing configuration;
  - a third command with CI = 2 describing the future configuration.

In the particular case of a change of bit 8 of byte 1 only (sound signal present/interrupted), the new configuration shall be signalled at least twice during the 160 ms preceding the transition.

After several repetitions of this data group, the BC1-BC2 or BC2-BC1 alternation shall be effected in accordance with the procedure described in subclause 6.5.1.

After the change, the content of the command with CI = 1 shall describe the new configuration and the process comes back to the steady state described at the beginning of this subclause.

6.7.5 Automatic mixing of main sound and commentary in the television service

To allow for the production of multilingual TV programme and in particular a European TV programme, an automatic sound mixing facility is needed in the receiver. The sound channels involved in this processing depend on the choice of the user. In general, two types of sound components have to be considered:

a) main sound which is specified in the command and parameter identifiers as "Digital TV original sound" (CI = '90, PI = 'A4: see table 22). This sound channel may be automatically selected in the sound demultiplexer and decoded in the sound decoder as soon as the television service type is selected (see LISTX in subclause 8.2.1);

b) digital TV additional sound signals which are specified by the command and parameter identifiers '90 and 'A5 (see table 22), and which, in the actual type of programme are of a commentary nature and may be carried by medium-quality channels.

The user is free to choose a sound signal as follows:

1) the main sound alone (type a) above);

2) only one of the available identified commentary channels (type b) above);

3) a mixture of the main sound and one of the available commentary channels, when mixing is intended according to the value of bit 4 of byte 2 in the BI block (the sampling frequencies of channels intended for mixing will be kept in synchronism by the broadcaster).

In cases 1) and 2), the address configuration related to the single selected sound channel is automatically loaded in the sound demultiplexer and the corresponding blocks are processed to provide the analogue signal. In case 3), two address configurations, related to the main sound and the selected commentary channels, are selected from service component description information (in particular in DCINF: see subclause 8.2.3) and are loaded in the demultiplexer; the corresponding coding blocks are processed to provide the two sound signals simultaneously.

At the sound output, these two signals shall be mixed as follows:

output x = α main sound + β commentary sound, α and β values can vary as described as follows.

Three cases have to be considered:

Case 1:

Modulation exists in the selected commentary channel. This state is defined in the associated BI block by the two possible configurations (see table 20):

- Byte 1 Bit 5 Temporal characteristic = 0 (continuous);
  Bit 8 State = X (irrelevant);

or;

- Byte 1 Bit 5 Temporal characteristic = 1 (intermittent);
  Bit 8 State = 0 (active).

Depending on the automatic mixing bit of the main sound channel, the main sound gain (α) shall either be equivalent to that of the commentary (β) or significantly lower. In the latter case, a relative attenuation of the main sound channel of a maximum of 15 dB is appropriate. In both cases the user should be able to select the exact value. Note that this state does not mean that at any moment commentary sound has a high level. The decision as to whether or not modulation is present in the commentary sound shall be the responsibility of the broadcaster and shall be signalled in bits 5 and 8 of Byte 1.

Case 2:

Commentary channel transmission is interrupted. That means that during this period normally, only BI blocks shall be transmitted at a rate of around three per second. In these blocks, Byte 1 is as follows:

- Byte 1 Bit 5 Temporal characteristic = 1 (intermittent);
  Bit 8 State = 1 (inactive).

In this case, the main sound gain shall have its nominal value ($\alpha = 1$ and $\beta = 0$).

Case 3:

This corresponds to the transitions from Case 1 to Case 2 and from Case 2 to Case 1. It involves adapting the relative gains of the two sound signals following a "cross fading" process. The fading control is provided by two mechanisms.
The first, which is always present, is the two-level control provided by the state of bit 8 of Byte 1 of the BI block of the relevant commentary component. This control provides timing information for the cross-fade between the two sources. The characteristics of the cross-fade are fast-attack and slow decay but the precise shape can be determined by the receiver manufacturer.
The second mechanism, which is an option for the broadcaster, is the CIB (see subclause 6.4). It may be used to define the shape of the cross-fade by low-pass filtering of the one-bit CIB sequence which arrives at a regular fast rate. This function is designated by the setting of bit 6 and bit 7 of Byte 1 of the BI block of the relevant commentary component (table 20).
The operation is illustrated by figure 42.

6.7.6 Automatic mixing of main sound and a commentary in sound broadcasting

The same procedure as described in subclause 6.7.5 may be applied to the automatic mixing of a main sound channel and a commentary in a sound broadcasting service. Only the specifications of the two signals at the service identification level are different.

a) The main sound shall be specified in the command and parameter identifiers under the heading "Digital radio sound" ('A8: see table 22).

b) The additional digital sounds shall be specified by the identifier: "Digital radio additional sound" ('A9: see table 22).

Table 20: Function and code for each bit of the bytes defined in subclause 6.7.3

Byte 1	Function	Code	
b1	Parity check	to give modulo-2 sum of all bits in bytes 1 and 2 equal to 0	
b2	SDFSCR flag	1 0	don't store
b3	News flash indication	1 0	yes no
b4	Identification of sound coding blocks	1 0	BC1 BC2
b5	Timing characteristic	0 1	continuous intermittent
b6 b7	Function of command information (CIB)	0 0	command music/speech ON
		0 1	cross-fade sound ON
		1 0	not defined
		1 1	not defined
b8	State	0 1	signal present interrupted
Byte 2	**Function**	**Code**	
b1 b2 b3	Audio configuration	0 0 0	bandwidth 40 Hz-15 kHz monophonic
		0 0 1	bandwidth 40 Hz-15 kHz stereophonic
		0 1 0	bandwidth 40 Hz-7 kHz monophonic
		0 1 1	not yet defined
		1 0 0	not yet defined
		1 0 1	not yet defined
		1 1 0	not yet defined
		1 1 1	not yet defined
b4	Automatic mixing	0 1	mixing not intended mixing intended
b5	Scrambling	0 1	no yes
b6	Controlled access	0 1	no yes
b7	Coding law	0 1	linear companded
b8	Level of error protection	0 1	first level second level

Table 21: Hamming code used for the Hamming protected bytes (Defined in subclause 6.7)

ENCODING										
Hexadecimal number	Decimal number	b 1	b 2	b3	b 4	b 5	b 6	b 7	b 8
0	0	1	0	1	0	1	0	0	0
1	1	0	1	0	0	0	0	0	0
2	2	1	0	0	1	0	0	1	0
3	3	0	1	1	1	1	0	1	0
4	4	0	0	1	0	0	1	1	0
5	5	1	1	0	0	1	1	1	0
6	6	0	0	0	1	1	1	0	0
7	7	1	1	1	1	0	1	0	0
8	8	0	0	0	0	1	0	1	1
9	9	1	1	1	0	0	0	1	1
A	10	0	0	1	1	0	0	0	1
B	11	1	1	0	1	1	0	0	1
C	12	1	0	0	0	0	1	0	1
D	13	0	1	1	0	1	1	0	1
E	14	1	0	1	1	1	1	1	1
F	15	0	1	0	1	0	1	1	1

$b7 = b8 \oplus b6 \oplus b4$
$b5 = b6 \oplus b4 \oplus \overline{b2}$
$b3 = b4 \oplus \overline{b2} \oplus b8$
$b1 = \overline{b2} \oplus b8 \oplus b6$

DECODING
$A = b8 \oplus b6 \oplus b2 \oplus b1$
$B = b8 \oplus b4 \oplus b3 \oplus b2$
$C = b6 \oplus b5 \oplus b4 \oplus b2$
$D = b8 \oplus b7 \oplus b6 \oplus b5 \oplus b4 \oplus b3 \oplus b2 \oplus b1$

A	B	C	D	Interpretation	Information
1	1	1	1	no error	accepted
0	0	1	0	error in b8	corrected
1	1	1	0	error in b7	accepted
0	1	0	0	error in b6	corrected
1	1	0	0	error in b5	accepted
1	0	0	0	error in b4	corrected
1	0	1	0	error in b3	accepted
0	0	0	0	error in b2	corrected
0	1	1	0	error in b1	accepted
A·B·C = 0			1	multiple errors	rejected

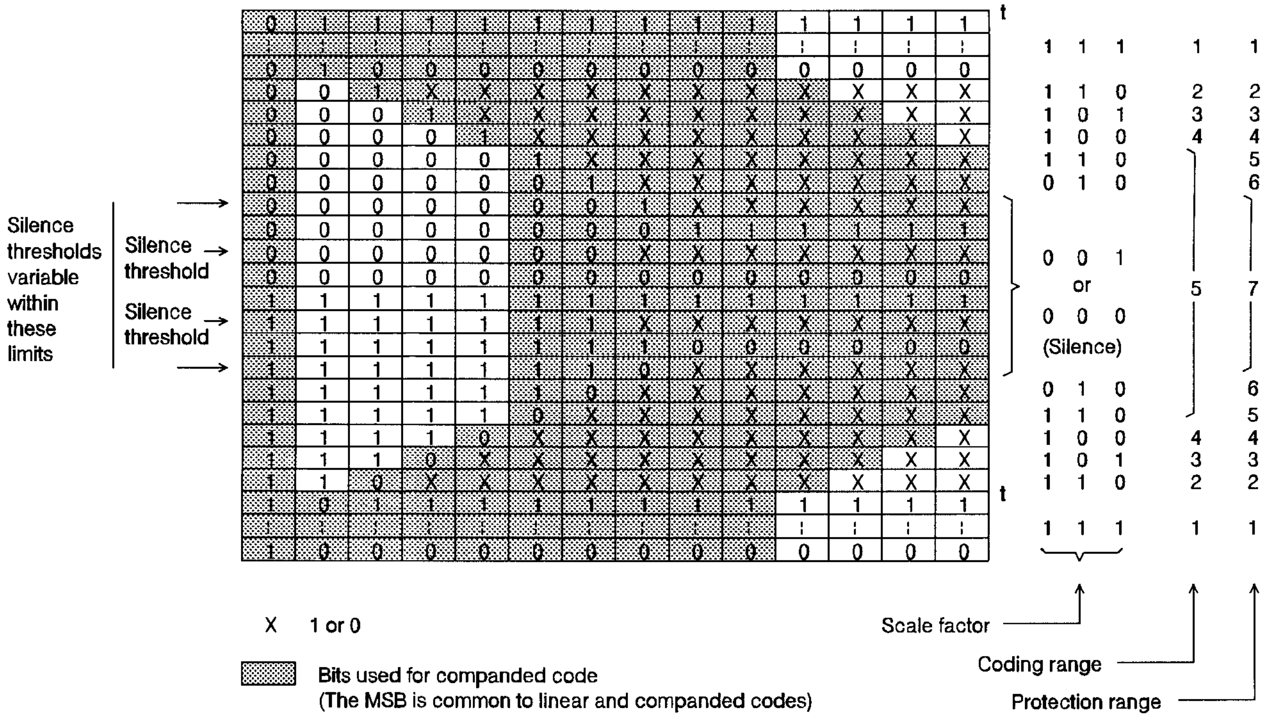
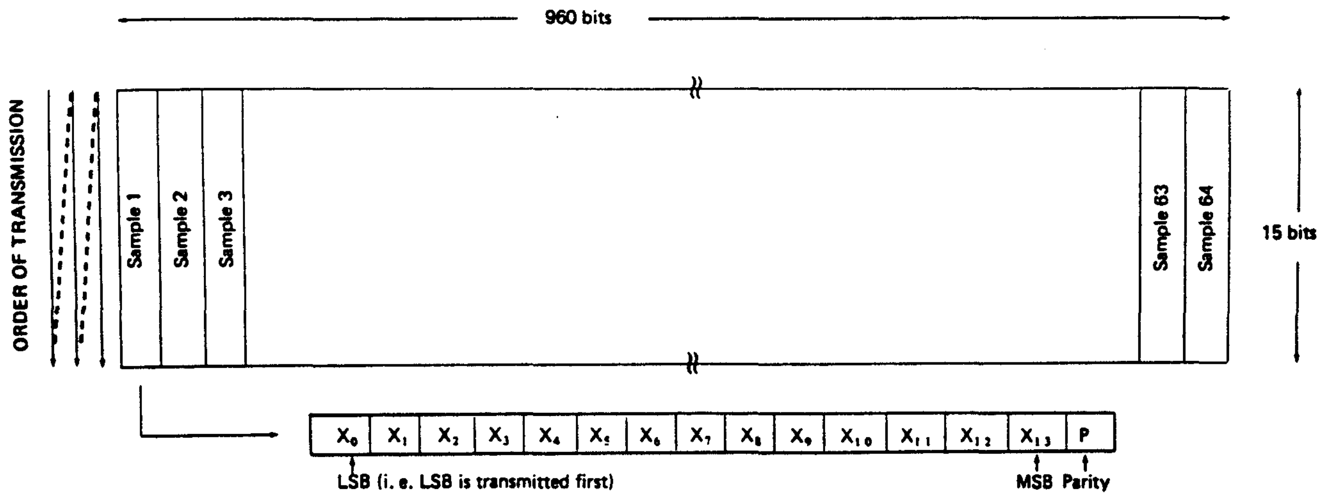
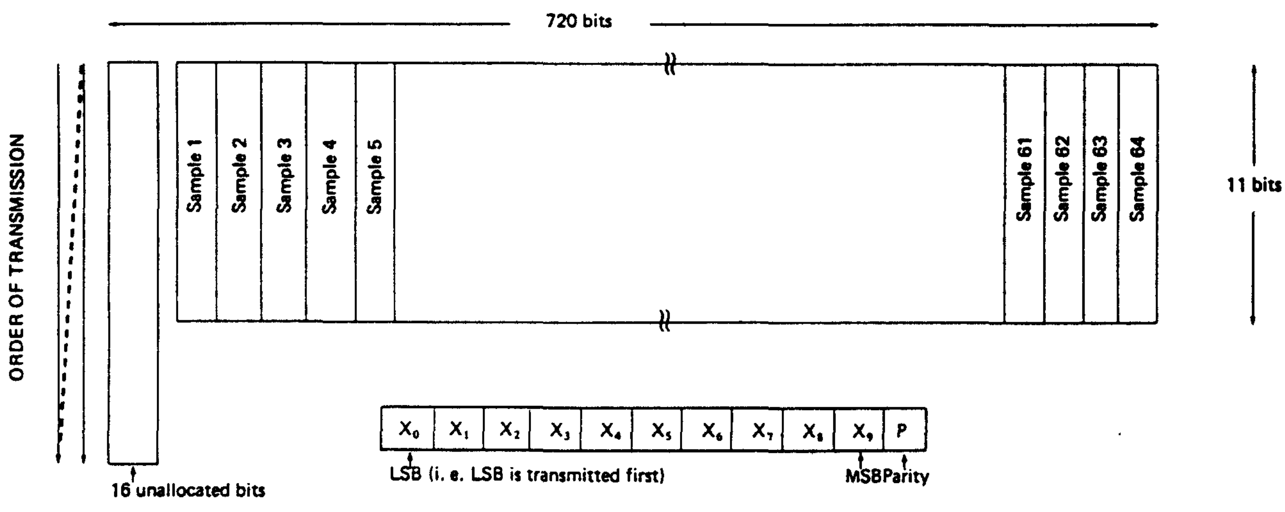


Figure 36: Coding for linear and companded signals (2's complement)



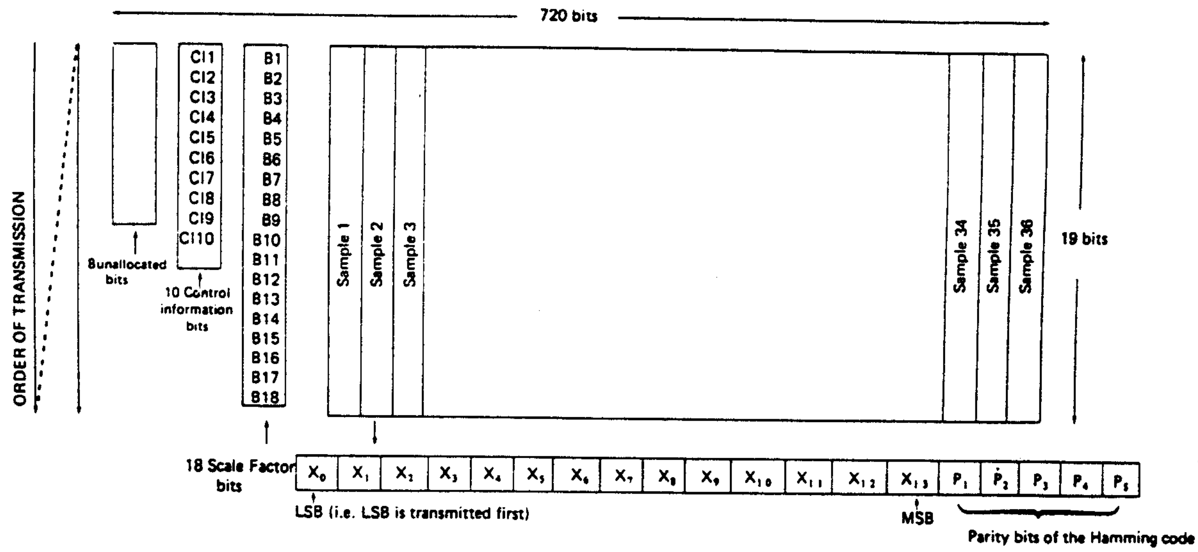
	SAMPLES																		
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ Stereophonic Linear sound	A_1	B_1	A_2	B_2	....	A_{15}	B_{15}	A_{16}	B_{16}	A_{17}	B_{17}	A_{18}	B_{18}	A_{19}	...	A_{31}	B_{31}	A_{32}	B_{32}
	Block n																		
HQ/MQ monophonic Linear Sound	A_1	A_2	A_3	A_4	....	A_{29}	A_{30}	A_{31}	A_{32}	A_1	A_2	A_3	A_4	A_5	...	A_{29}	A_{30}	A_{31}	A_{32}
	Block n									Block n + 1									

Figure 37a: Arrangement of 120-byte linear sound coding blocks
in relation with the first level protection



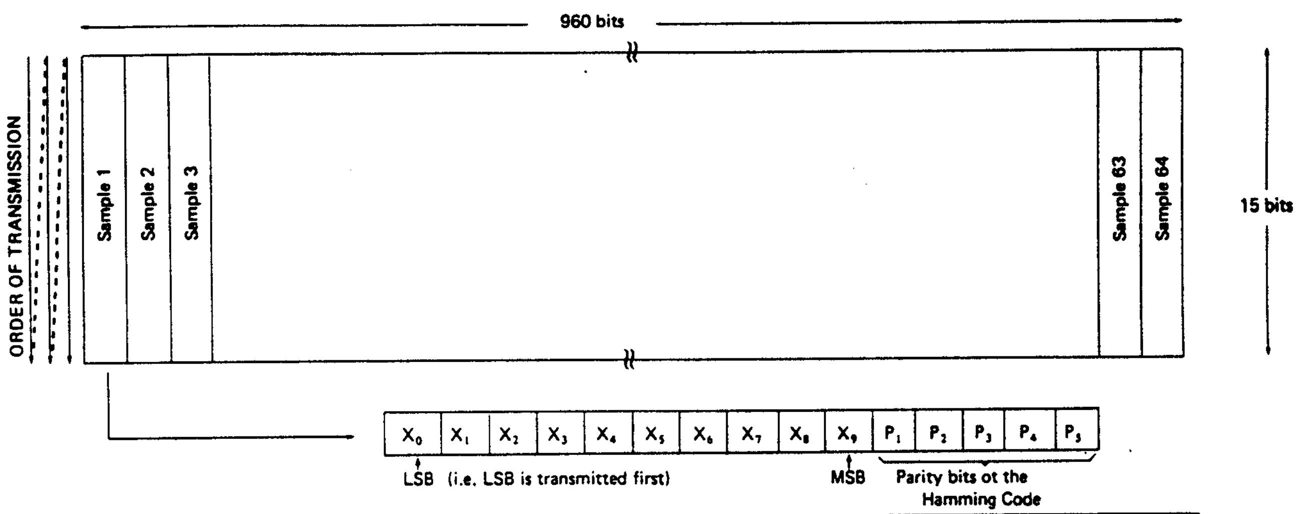
	SAMPLES																		
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ stereophonic companded sound	A_1	B_1	A_2	B_2	....	A_{15}	B_{15}	A_{16}	B_{16}	A_{17}	B_{17}	A_{18}	B_{18}	A_{19}	....	A_{31}	B_{31}	A_{32}	B_{32}
	Block n																		
HQ/MQ Monophonic companded sound	A_1	A_2	A_3	A_4	....	A_{29}	A_{30}	A_{31}	A_{32}	A_1	A_2	A_3	A_4	A_5	....	A_{29}	A_{30}	A_{31}	A_{32}
	Block n									Block n + 1									

Figure 37b: Arrangement of 90-byte companded sound coding blocks
in relation with the first level protection



	SAMPLES													
	1	2	3	....	16	17	18	19	20	21	.....	34	35	36
HQ stereophonic linear sound	A_1	B_1	A_2	....	B_8	A_9	B_9	A_{10}	B_{10}	A_{11}	.....	B_{17}	A_{18}	B_{18}
	Block n													
HQ/MQ Monophonic linear sound	A_1	A_2	A_3	....	A_{16}	A_{17}	A_{18}	A_{19}	A_{20}	A_{21}	.....	A_{34}	A_{35}	A_{36}
	Block n							Block n + 1						

Figure 37c: Arrangement of 90-byte linear sound coding blocks in relation with the second level protection



	SAMPLES																			
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ Stereophonic Companded sound	A_1	B_1	A_2	B_2	....	A_{15}	B_{15}	A_{16}	B_{16}	A_{17}	B_{17}	A_{18}	B_{18}	A_{19}	....	A_{31}	B_{31}	A_{32}	B_{32}
	Block n																		
HQ/MQ Monophonic companded sound	A_1	A_2	A_3	A_4	....	A_{29}	A_{30}	A_{31}	A_{32}	A_1	A_2	A_3	A_4	A_5	....	A_{29}	A_{30}	A_{31}	A_{32}
	Block n									Block n + 1									

Figure 37d: Arrangement of 120-byte companded sound coding blocks in relation with the second level protection

	sample bits											parity bits				
Hamming coding nomenclature	b_1	b_2	b_3	b_4	b_5	b_6	b_7	b_8	b_9	b_{10}	b_{11}	P_i^1	P_i^2	P_i^3	P_i^4	P_i^5
Linear coding bits (see figure 37c)	x_3	x_4	x_5	x_6	x_7	x_8	x_9	x_{10}	x_{11}	x_{12}	x_{13}	P_1	P_2	P_3	P_4	P_5
Companded coding bits (see figure 37d)	0	0	0	0	0	x_4	x_5	x_6	x_7	x_8	x_9	P_1	P_2	P_3	P_4	P_5

Figure 38: Extended Hamming code (16,11) and (11,6) for companded mode
Relation between code table, sample bits and parity bits

Coding relations:

$P_i^1 = b_1 \oplus b_4 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_9 \oplus b_{11};$

$P_i^1 = b_1 \oplus b_2 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{10};$

$P_i^1 = b_1 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_{10};$

$P_i^1 = b_2 \oplus b_3 \oplus b_5 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{11};$

$P_i^1 = b_3 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_8 \oplus b_{10} \oplus b_{11};$

$P'_i5 = P_i5 \oplus R_j.$

Decoding relations:

$S_4 = P_i^1 \oplus b_1 \oplus b_4 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_9 \oplus b_{11};$

$S_3 = P_i^2 \oplus b_1 \oplus b_2 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{10};$

$S_2 = P_i^3 \oplus b_1 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_{10};$

$S_1 = P_i^4 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{11};$

$S_0 = P{*}_i^5 \oplus b_3 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_8 \oplus b_{10} \oplus b_{11};$

where $P{*}_i^5 = P'_i{}^5 \oplus R_j.$

Correction table:

S_4	S_3	S_2	S_1	S_0	S_p	Interpretation	Sample treatment: Linear	Sample treatment: Companded
0	0	0	0	0	0	no error	accept	accept
1	0	0	0	0	1	P_i^1 error	accept	accept
0	1	0	0	0	1	P_i^2 error	accept	accept
0	0	1	0	0	1	P_i^3 error	accept	accept
0	0	0	1	0	1	P_i^4 error	accept	accept
0	0	0	0	1	1	P_i^5 error	accept	accept
1	1	1	0	0	1	b_1 error	correct x_3	conceal
0	1	1	1	0	1	b_2 error	correct x_4	conceal
0	0	1	1	1	1	b_3 error	correct x_5	conceal
1	1	0	0	1	1	b_4 error	correct x_6	conceal
1	0	1	1	0	1	b_5 error	correct x_7	conceal
0	1	0	1	1	1	b_6 error	correct x_8	correct x_4
1	1	1	1	1	1	b_7 error	correct x_9	correct x_5
1	0	1	0	1	1	b_8 error	correct x_{10}	correct x_6
1	1	0	1	0	1	b_9 error	correct x_{11}	correct x_7
0	1	1	0	1	1	b_{10} error	correct x_{12}	correct x_8
1	0	0	1	1	1	b_{11} error	correct x_{13}	.correct x_9
remaining codes					0	multiple errors	conceal	

Where;

$$S_p = \sum_{k=0}^{4} S_k = \sum_{k=1}^{11} b_k \oplus \sum_{n=1}^{5} P_i^n$$

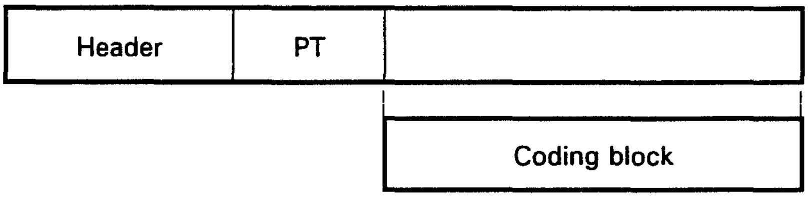
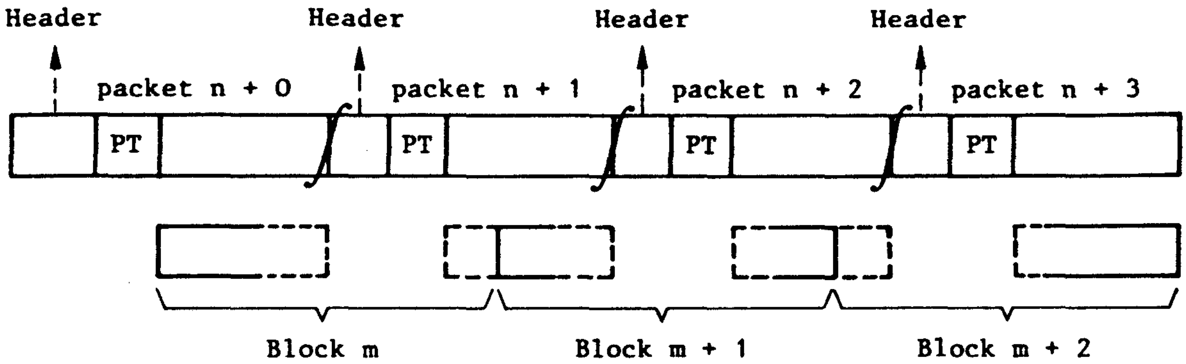


Figure 39: Insertion of a 90-byte coding block

Coding group



Header configurations	packet n + 0	address X	00	parity bits Golay code
	packet n + 1	address X	01	parity bits Golay code
	packet n + 2	address X	10	parity bits Golay code
	packet n + 3	address X	11	parity bits Golay code

X value is the same for the four packet types.

Figure 40: Insertion of a 120-byte coding block

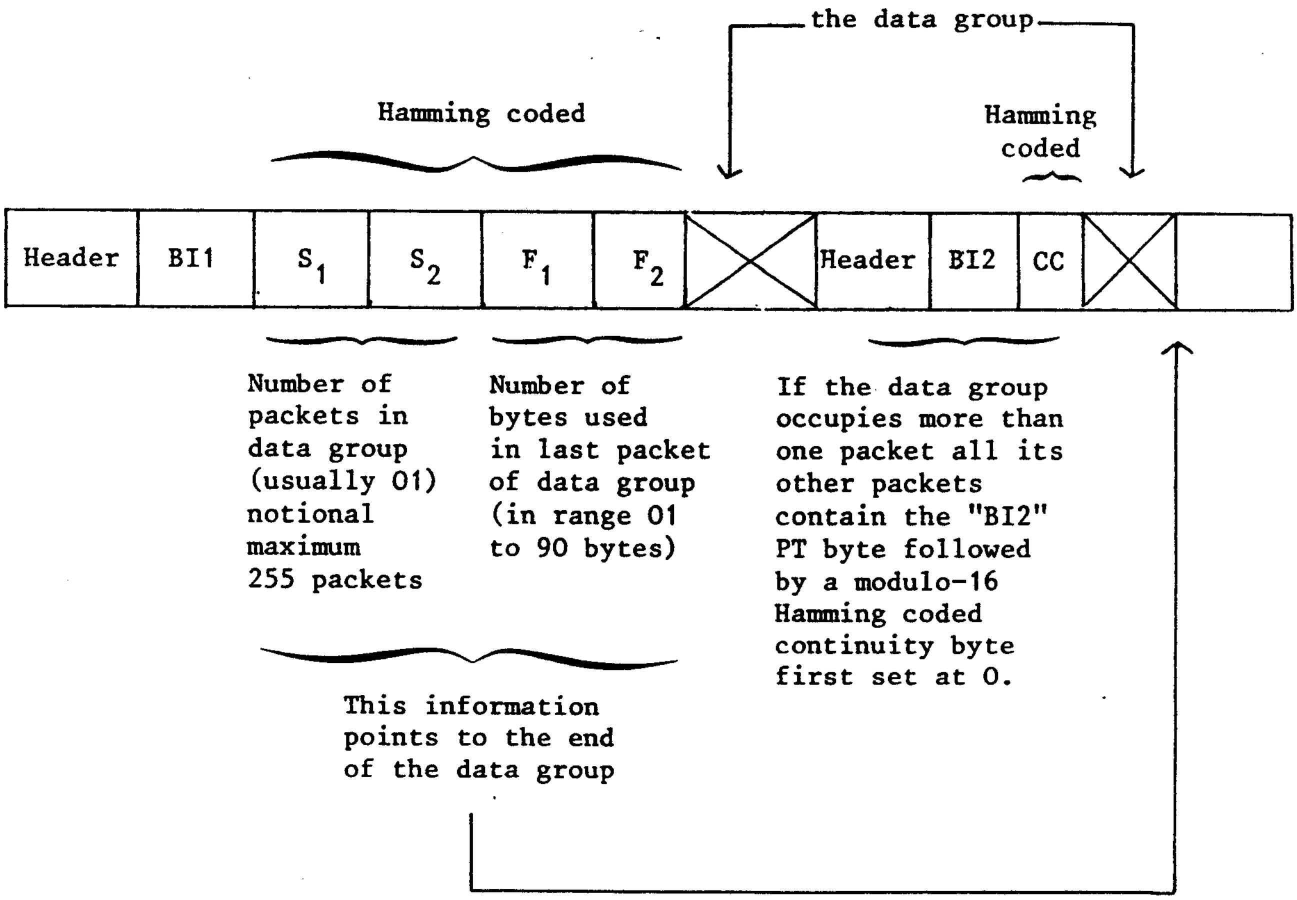
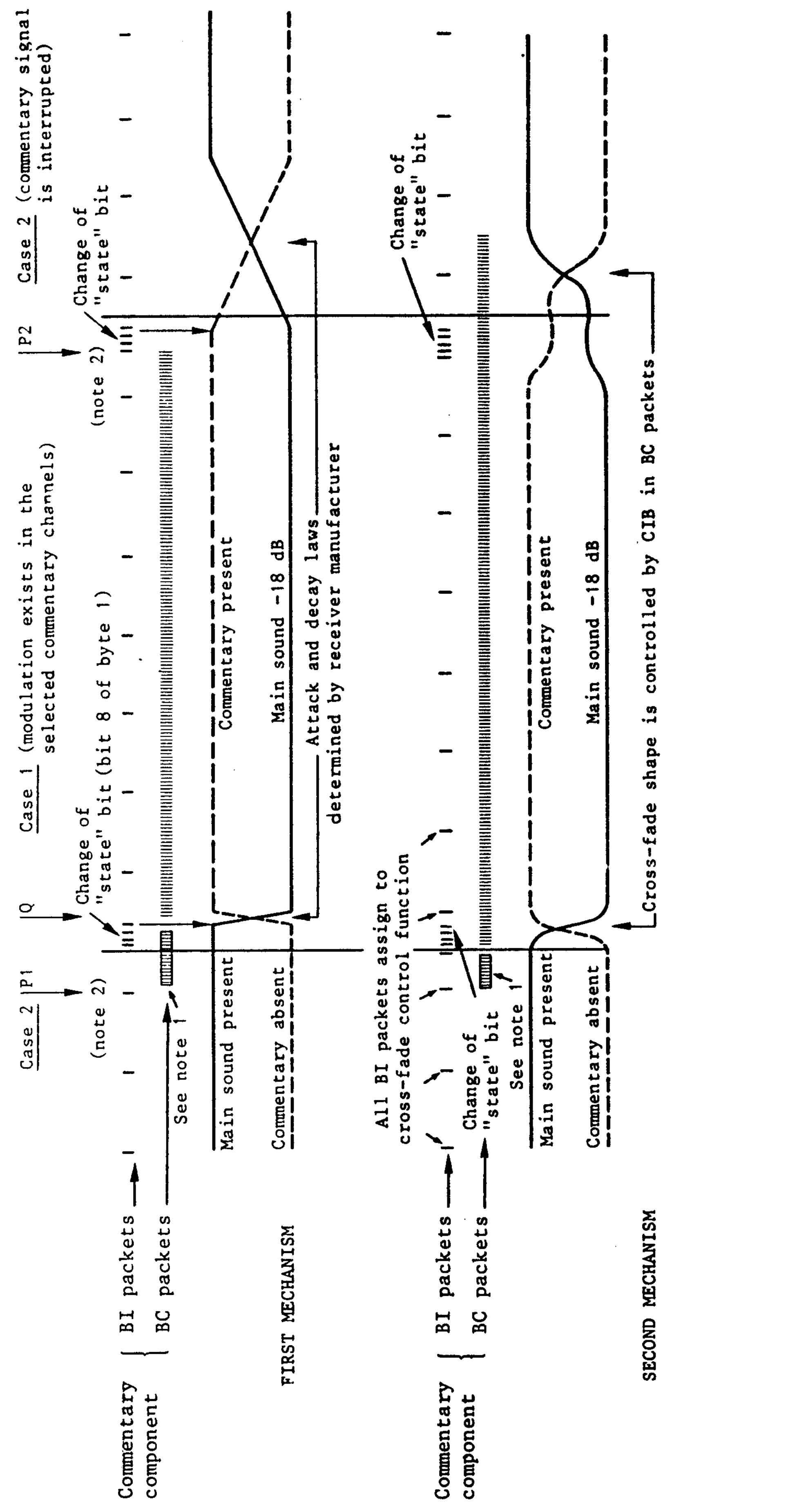


Figure 41: Positioning of the data group in the interpretation packets
(The values of BI1 and BI2 are defined in subclause 6.5.1)



Note 1: If co-timing is required by the broadcaster, these extra BC packets are needed (see the procedure described in Section 7.1).

Note 2: P1 and P2: signals from commentator's box
Q: start of "cue" signal to commentator

Figure 42: Automatic mixing of main sound and commentary
(BC and BI packets of main sound continue normally throughout this operation)

7 Specification of the system for data services

7.1 Introduction

Clause 7 contains the specifications for the transmission of data.

Subclause 7.2 specifies the transmission of teletext systems A and B given in CCIR Recommendation 653-1 [3] and [4] respectively. These systems are known in Europe as the variable-format teletext system (system A) and the fixed-format teletext system (system B). One or both teletext systems may be transmitted in the field-blanking interval to provide a teletext service or to subtitle a television service, for use by existing receivers.

Subclause 7.3 specifies the transport mechanisms using the sound/data multiplex for teletext, subtitling or any other data broadcasting service characterized by data messages originally structured according to CCIR teletext systems A or B.

Subclause 7.4 specifies the protocol of general purpose data services carried via the sound/data multiplex.

The contents of subclauses 7.3 and 7.4 are also applicable to services present in the DATV/data multiplex.

7.2 Teletext service in the field blanking interval

The teletext data shall be transmitted in the order defined in [3] and [4] for systems A and B, whereas the service identification data concerning these signals shall be transmitted in the order defined in Clause 8.

7.2.1 Data transmission in the field-blanking interval

7.2.1.1 Data coding

The data corresponding to the teletext components shall be transmitted in lines of the field-blanking interval, coded in duobinary form (see subclause 4.4.2.4) at a bit-rate of 10,125 Mbit/s. Signal generation starts from a 20,25 MHz clock: even samples shall carry bits from a teletext data line of a teletext service or subtitle component of the television service, whilst odd samples shall correspond to transitions. This bit-stream shall be multiplexed with the data described in Clause 4 (burst of 105 bits + line 625 + bits set to zero during all the periods of time not allocated to duobinary digital components), and constitute a continuous data flow which is duobinary encoded as a whole and in a single operation.

7.2.1.2 Relationship between the bits of data and the sampling structure (figure 43)

The lines of the field-blanking interval can be used for the transmission of blocks of 532 bits. The first bit of the teletext data line transmitted on even sample numbers shall correspond to sample 230, and the last bit to sample 1 292.

7.2.1.3 Clamping transitions

In the lines of the field-blanking interval allocated to teletext transmission, the clamping period shall be obtained by forcing the duobinary signal to zero from sample 208 to 228 inclusive.

7.2.2 Transmission of teletext components

7.2.2.1 Allocated lines

The lines allocated to CCIR systems A and B teletext shall be defined in line 625 according to TDMCTL procedure described in subclause 4.5.4. Any free lines of the intervals from line 1 to 22 inclusive and from line 311 to 334 inclusive may be used. The allocated lines may overlap with the DATV/data multiplex as described in subclause 4.5.5.

7.2.2.2 Transmission of CCIR system A teletext

CCIR system A teletext shall be transmitted as data lines of 320 bits as specified in CCIR Recommendation 653-1 [3]. These data lines shall be inserted in the first part of the 532 bits defined in subclause 7.2.1.2, without scrambling, interleaving or scrambling for spectrum shaping purposes. The non-allocated bits shall be forced to "0".

7.2.2.3 Transmission of CCIR system B teletext

CCIR system B teletext shall be transmitted as data lines of 360 bits as specified in CCIR Recommendation 653-1 [4]. These data lines shall be inserted in the first part of the 532 bits defined in subclause 7.2.1.2, without scrambling, interleaving or scrambling for spectrum shaping purposes. The non-allocated bits shall be forced to "0".

7.2.3 Service identification data

Access coordinates of teletext components transmitted in the field-blanking interval shall be given by the parameter DCINF of the service-identification channel (see subclause 8.2.3) characterized by the parameter identifier 'Fx where x has the following meaning:

x =	'0	CCIR system B cyclic teletext;
	'1	CCIR system B non-cyclic teletext;
	'4	CCIR system A cyclic teletext;
	'5	CCIR system A non-cyclic teletext;
	'8	TV subtitles with CCIR system B teletext;
	'C	TV subtitles with CCIR system A teletext.

The 16-bit sequence of access coordinates shall be as follows:

a) CCIR system A teletext:

- b16 = bit D defined in CCIR Recommendation 653-1 [3] (information on bit-rate);
- b15 = 0;
- b14, b13: two most significant bits of data channel address (A1, A2, A3 of CCIR Recommendation 653-1 [3]);
- b12 = 0;
- b11 = 0;
- b1 to b10: ten least significant bits of data channel address (A1, A2, A3 of CCIR Recommendation 653-1 [3]).

b) CCIR system B teletext:

- b16 = 0;
- b13 to b15: magazine number;
- b12 = 0;
- b11 = 0;
- b9, b10: irrelevant;
- b1 to b8: page number.

The complementary access coordinates shall be composed of 2 bytes coded as a 16-bit sequence as follows:

a) CCIR system A teletext: page number on the 12 LSBs;

b) CCIR system B teletext: page sub-code on the 13 LSBs.

The complementary access coordinates shall be transmitted in the case that CCIR system A teletext is used for subtitling.

7.3 Teletext service in the sound/data and DATV/data multiplex

Subclause 7.3 gives the specification of the transport mechanisms using the sound/data and DATV/data multiplex for teletext, subtitling or any other data broadcasting service characterized by data messages originally structured according to CCIR teletext system A or B.

The organization of the channels dedicated to the services using data broadcasting, or their service components, shall be described in the service identification channel, which is formed by packets in the sound/data multiplex with the packet address "0" (see Clause 8).

7.3.1 Teletext transmission structure

7.3.1.1 Teletext data block description

CCIR teletext system A or B shall be transmitted as teletext data blocks of 360 bits (45 bytes). These shall be drawn from the data bits of the system A or B teletext lines, as specified in CCIR Recommendations 653-1 [3] and [4], respectively, with the 3 bytes for clock run-in and framing code removed.

In the case of CCIR system A, the 45-byte teletext data block (see figure 44a) shall be divided into:

- a 6-byte prefix;
- an information field of 36 useful bytes plus 1 filling byte;
- a 2-byte CRC suffix.

The 6-byte prefix shall contain in sequence:

- one Control Byte (CB);
- three address bytes (P1, P2, P3);
- one Continuity Index (CI);
- one format indicator Packet Suffix (PS).

In the case of CCIR system B, the 45-byte teletext data block (see figure 44b) shall be divided into a:

- 3-byte prefix;
- information field of 40 bytes;
- 2-byte CRC suffix.

The 3-byte prefix shall carry in sequence:

- one CB;
- the 16-bit magazine and teletext data packet address group.

7.3.1.2 Data block prefix description

All prefix bytes shall be (8,4) Hamming encoded. The coding of the message bits b2 b4 b6 b8 of the CB shall be as follows:

Bit number	2	4	6	8	
	1	1	1	1	Transparent transmission of the teletext data packet (i.e. no further knowledge of the contents of the teletext data packets or repetitive nature of the teletext service is implied).
	0	1	1	1	Last or only transmission of the teletext data packet.

The control byte shall be transmitted LSB first.

In the case of CCIR system A, the coding of the message bits of the three address bytes, the continuity index and the format indicator shall be as specified in CCIR Recommendation 653-1 [3]; in the case of CCIR system B, the coding of the message bits of the 16-bit magazine and teletext data packet address group shall be as specified in CCIR Recommendation 653-1 [4].

7.3.1.3 Data block suffix description

The data block suffix shall be a CRC generated by the polynomial:

$$x^{16} + x^{12} + x^5 + 1.$$

In the case of CCIR system A, the 37 bytes of the information field of each teletext data block are composed in sending order of bits m_{295} to m_0 followed by check bits r_{15} to r_0 in the same order. The check bits shall be such that the polynomial:

$$m_{295}x^{311} + m_{294}x^{310} + L + m_0 x^{16} + r_{15}x^{15}15 + L . + r_1 x + r_0$$

is a multiple (modulo 2) of the polynomial:

$$x^{16} + x^{12} + x^5 + 1$$

In the case of CCIR system B, the 40 bytes of the information field of each teletext data block are composed in sending order of bits m_{319} to m_0 followed by check bits r_{15} to r_0 in the same order. The check bits shall be such that the polynomial:

$$m_{319}x^{335} + m_{318}x^{334} + L + m_0 x^{16} + r_{15}x^{15} + L + r_1 x + r_0$$

is a multiple (modulo 2) of the polynomial:

$$x^{16} + x^{12} + x^5 + 1$$

7.3.1.4 Levels of error protection

Two levels of error protection are defined. In the first level, no forward error correction is introduced. This type of protection, relying upon the use of CRC, is primarily intended for cycled data services where error correction can be provided by using majority logic, or bit variation, at the receiver. In the second level of protection (24, 12) Golay encoding shall be used. Each successive 12-bit segment from the teletext data block is appended by 11 error check-bits. The code is defined by the generator polynomial:

$$G(x) = x^{11} + x^{10} + x^6 + x^5 + x^4 + x^2 + 1$$

The resulting 23-bit code shall be appended by a single parity bit giving overall odd parity for the 24-bit block. This type of protection provides a high level of error correction on a single transmission of the teletext data.

7.3.2 Insertion of teletext data blocks into the MAC/packet multiplex

The packet structure is shown in figure 45a in the case of first-level protection and in figure 45b in the case of second-level protection. The packet shall commence with a 23-bit packet header followed by a PT-byte which shall be used to signal, on a packet basis, the access mode. Scrambling as specified in subclause 9.3 may be applied to the content of the packet.

The PT-byte values shall be assigned as follows:

PT-byte	access mode;
'C7	free access, scrambled;
'F8	controlled access, scrambled;
'00	unscrambled;
'3F	unallocated.

7.3.3 Service identification data

Access coordinates of the service or service component transmitted in the packet multiplex shall be given by the parameter DCINF of the service identification channel (see subclause 8.2.3) characterized by the parameter identifier 'Bx, where x has the following meaning:

x =	'0	teletext;
	'1	teletext subtitles;
	'2	replacement teletext;
	'3	programme delivery control.

The four most significant bits of the DCINF access coordinates shall indicate the level of error protection and the type of teletext system as follows:

'0	first level, CCIR system A;
'1	first level, CCIR system B;
'2	second level, CCIR system A;
'3	second level, CCIR system B;
'4 to 'F	reserved for future use.

The remaining 12 bits of the DCINF access coordinates are as specified in subclause 8.2.3.

In the case of system A teletext only, the first two bytes of the complementary access coordinates shall be transmitted and are specified as:

- the most significant half-byte of the first byte is reserved for future use;

- the least significant half-byte of this byte and the second byte represent the teletext data channel address.

For service components provided in teletext format, it may be useful to identify the teletext page containing either the complete component or an index page for the component. This page number can be indicated in the complementary access coordinates, by extending the field of the complementary access coordinates from 2 to 4 bytes in the case of system A teletext, or by providing a 2-byte field of complementary access coordinates in the case of system B teletext.

For system A teletext, the two further bytes of the complementary access coordinates (if provided) shall be coded as follows:

- Byte 3:

    - 4 MSBs reserved for future use;

    - 4 LSBs most significant digit or page number.

- Byte 4:

  - 4 MSBs middle digit of page number;

  - 4 LSBs least significant digit of page number.

For system B teletext, the complementary access coordinates (if provided) shall be coded as follows:

- Byte 1:

  - 5 MSBs reserved for future use;

  - 3 LSBs magazine number.

- Byte 2:

  - 4 MSBs most significant digit of page number;

  - 4 LSBs least significant digit of page number.

When CCIR teletext systems A or B are used for subtitling the complete complementary access coordinates shall be transmitted.

7.4 General Purpose Data service

Subclause 7.4 specifies the protocol of General Purpose Data (GPD) services carried via the sound/data and DATV/data multiplex. GPD services are used for the one-way distribution of data, possibly combined with conditional access.

GPD service components are distinguished from teletext components carried in the sound/data and DATV/data multiplex by having a different format. The one-way distribution of general data may be carried either by teletext components or by GPD components.

7.4.1 Description of GPD service protocol

7.4.1.1 General

In terms of the OSI model, a channel carrying GPD services is a network. To facilitate inter-working through other types of network, the transport and higher OSI layer protocols for GPD services is external to the D2-HDMAC/packet specification.

GPD services may be carried as one or more HDMAC/packet digital components (i.e. each having a unique packet address). The protocol is described in general terms with reference to the OSI model, as follows:

- Layer III (network):

  messages, data messages (see figure 46a) consist of:

  - an optional data message header;

  - a data segment containing user-data;

  - an optional data segment CRC.

- Layer II (link):

  - Data messages are carried either by individual MAC packets, or packets linked by packet header CI. The 90 byte area following the PT byte in packets carrying GPD service components may optionally be protected by Golay (24,12) coding as defined in subclause 8.3.2. The performance improvement using Golay coding is illustrated in subclause 7.4.5.

7.4.1.2 OSI link and network layer features

Where the greatest efficiency in use of packet data capacity is required, a data message structure which has no header may be used. Such data messages are carried in the entire useful data area of individual packets. In this case, user-data is carried using the following OSI link and network layer features:

- Layer III (network):

  - packet multiplexing;

  - error detection by optional data segment CRC;

  - service management via the SI channel.

- Layer II (link):

  - framing by packet structure;

  - link addressing by packet address;

  - error detection by packet header CI;

  - Forward Error Correction (FEC) by optional Golay coding.

For some purposes it may be desirable to enhance the above features for GPD services. The use of the data message header, with optional packet linking for improved transport efficiency, signals the optional addition of the following features, each of which may be individually selected in a given component:

- Address extension:

  - this allows several independent data streams to be carried under one packet address. This could be advantageous, for example, in efficiently utilising a given data capacity to carry several variable-rate sub-services.

- Data segment counter:

  - this provides error detection at Layer III (data message loss).

- Segment length indicator:

  - this enables a specific indication of segment length when a data message does not fill exactly the available packet space.

7.4.1.3 Identification of GPD services

The SI description of a GPD service component shall include the following information:

- identification of service and component;

- error protection methods;

- whether the data message header is used;

- general class of transmission;

- whether data messages are carried by single or linked packets;

- whether conditional access scrambling may be in use.

The following information shall be signalled within a GPD service component:

- whether packets are scrambled, as indicated by the packet PT byte;

- extended address, if in use;

- a segment counter, if in use;

- data segment length for messages which do not fill a single packet, or group of linked packets.

7.4.1.4 Data message structure (see figure 46)

A data message shall consist of:

- an optional variable length data message header;

- a segment of user-data;

- an optional segment protection CRC.

The data message header shall consist of:

- a Format Descriptor (FD) byte;

- a header extension, of variable length according to FD coding.

7.4.1.4.1 Data message header

The FD byte shall be Hamming (8,4) coded according to table 21. The four information bits shall be coded as follows:

b2 = 0	The data message fills a single packet for a non-linked component, or a complete group of four linked packets for a linked component. There is no segment length indicator in the message header extension field.
b2 = 1	The data message does not fill a single packet for a non-linked component, or a complete group of four linked packets for a linked component. The message header extension includes a segment length indicator.
b4 = 0	There is no segment counter field in the message header extension.
b4 = 1	The segment header extension includes a segment counter field.
b6 = 0	The component is a single stream of user-data, and no address extension is in use. There is no address extension field in the message header extension.
b6 = 1	The component includes more than one stream of user-data, and address extension is in use. The message header extension includes an address extension field.
b8 = 0	The FD information bits b2, b4 and b6 are defined as above.
b8 = 1	Reserved.

The remainder of the message header, after the FD byte, shall consist of a variable length message header extension, containing any of the following fields, in the following order:

Address extension field:	present if FD b6 = 1 absent if FD b6 = 0
segment counter field:	present if FD b4 = 1 absent if FD b4 = 0
Segment length indicator:	present if FD b2 = 1 absent if FD b2 = 0

Coded as Hamming (8,4) according to table 21 giving the number of Hamming (8,4) bytes in the following variable length extended address field (valid values are from 1 to 15). The LSB of the extended address is b2 of the first byte; the MSB of the extended address is b8 of the last byte. A data channel using address extension is uniquely defined by a combination of packet address and the complete address extension field, including address extension length indicator. Two extended addresses, of different lengths but the same numerical value when leading zeroes are ignored, shall define different data channels.

The segment counter field, if present, shall consist of a 16-bit binary number coded into four Hamming (8,4) bytes. The LSB of this 16-bit number is b2 of the 1st Hamming (8,4) byte, and the MSB is b8 of the 4th Hamming (8,4) byte. The segment counter shall increment modulo-65536 for successive data messages.

The segment length indicator, if present, shall consist of three Hamming (8,4) bytes, representing a binary number giving the number of bytes in the remainder of the data message, counting inclusively from the first byte of user-data following the message header, to the last byte of the message, including the segment CRC if present.

7.4.1.4.2 Segment description

The data segment, immediately following the message header, shall consist of a sequence of bytes of user-data, each transmitted LSB first.

7.4.1.4.3 Data segment CRC description

The segment CRC, if present, immediately follows the data segment. It shall be computed from all bytes of the data segment in transmission order. The message header shall not be included in the computation. The generator polynomial and computation method are as specified in subclause 8.3.2.

7.4.2 Assembly of messages into packets

When a message header is not used, packets shall not be linked. When a message header is used, packets may be individual or linked in groups of up to four packets. The use of linked packets improves the efficiency of the transport mechanism when a message header is required.

7.4.2.1 Non-linked component

Each data message, with Golay coding if used, shall fit within the 90 bytes following the PT byte of a single packet. The packet header CI value shall increment modulo-4 for successive packets having the same packet address, irrespective of address extension if in use.

If the message header is not used, or if it is used and the FD byte bit b2 = 0, the data message shall fill the packet with no bytes left unused. Thus the last byte of the packet shall either be the last byte of the data segment if there is no segment CRC, or the second byte of the CRC if present.

If the message header is used and the FD byte bit b2 = 1, the data message may occupy less than the space available following the PT byte, as indicated by the segment length indicator. Any unused bytes shall come after the end of the data segment, and after the end of the segment CRC if present.

For a non-linked component without Golay protection, the data message shall be contained within a maximum of 90 bytes. If necessary, arbitrary padding bytes shall be added after the message to complete 90 bytes. The resultant 90 byte block is then loaded into the 90 byte space following the PT byte of a single packet.

For a non-linked component with Golay protection, the data message before Golay encoding shall be contained within a maximum of 45 bytes. If necessary, arbitrary padding bytes shall be added after the message to complete 45 bytes. The block of 45 bytes shall then be coded into thirty Golay (24,12) words as specified in subclause 8.3.2. The resultant 90 byte block is then loaded into the 90 byte space following the PT byte of a single packet.

7.4.2.2 Linked component

Each data message, with Golay coding if used, shall fit within the 360 bytes following the PT bytes of four linked packets, which are linked by packet header CI. CI = 0 shall be used for the synchronising packet (i.e. the first packet in the linked group), and shall increment for successive packets in the linked group.

If the message header FD byte bit b2 = 0, the data message shall fill the space available in the complete group of four linked packets with no bytes left unused.

If the message header FD byte bit b2 = 1, the data message may occupy less than the space available in a complete group of four linked packets. Any unused bytes shall come after the end of the data segment, and after the end of the segment CRC if present. If the message requires less than four linked packets to carry it, the additional packets to complete the group of four may or may not be transmitted.

For a linked component without Golay protection, the data message shall be divided, from the start of the message header, into a maximum of four 90 byte blocks with arbitrary padding bytes added to complete the final 90 byte block if necessary. These 90 byte blocks are then loaded into the 90 byte spaces following the PT bytes of the linked packets.

For a linked component with Golay protection, the data message shall be divided, from the start of the message header, into a maximum of four 45 byte blocks with arbitrary padding bytes added to complete the final 45 byte block if necessary. Each block shall then be coded into thirty Golay (24,12) words as specified in subclause 8.3.2. The resultant 90 byte blocks are then loaded into the 90 byte spaces following the PT bytes of the linked packets.

7.4.2.3 Use of PT byte

GPD service components packets shall be identified by PT byte as follows:

PT = 'F8 Data message packet with conditional access scrambling;
PT = '00 Data message packet without scrambling;
PT = '3F Reserved for potential use with control or interpretation packets (see NOTE below);
PT = 'C7 Reserved.

NOTE: Control or interpretation packets may be found useful to provide extra service management within each GPD component.

7.4.3 Multiplexing rules

Packets of one linked group of a GPD service component shall not be interleaved with the packets of any other linked group having the same packet address.

When address extension is in use with linked components, the packets of a linked group shall not be interleaved with the packets of any other linked group having the same packet address but different address extension field.

7.4.4 Service identification

A GPD component is defined as a digital channel identified by a particular packet address and conforming to the protocol of the preceding subclauses 7.4.1 to 7.4.3. A GPD service shall contain at least one GPD component, and may consist of a group of associated components, including other types of component.

A special method of Service Identification (SI) is required for GPD services. This is due mainly to the fact that a HDMAC transmission may carry large numbers of GPD services. It is possible for over a thousand different GPD services to be transmitted at the same time within a given subframe or subframe pair.

To avoid overloading the dedicated packet 0 channel used for the service identification of the traditional broadcasting services of TV, Radio and Teletext, GPD service identification shall be carried under a separate packet address, the value of which may be from 1 to 1 022 (decimal) inclusive. The value of packet address used for GPD SI in any particular case shall be included in the information conveyed by the DATX parameter in the Network Command in DG0 of packet 0 SI, as described in subclause 8.2.1.

A GPD service may include non-GPD components such as sound, teletext, or video, where such components are, as far as the user is concerned, part of a GPD service and not part of a traditional broadcasting service. In such cases, the non-GPD components shall be included within the GPD service description, and shall not appear in packet 0 SI.

A GPD service shall be identified to its users in at least one of the following three ways:

Network Specific Identity (NSI):

This is a number in the range 0-65535 (decimal) inclusive. A given value of NSI shall remain constant for as long as the associated GPD service remains in existence in the commercial or other sense, and shall be uniquely associated with one, and only one, GPD service transmitted on a given MAC network. The same GPD service carried by a different network may have a different value of NSI, and the same value of NSI on a different network may be used for a different GPD service. The NSI is intended to support automatic acquisition of a GPD service on a particular network, so that unattended equipment can be used as a data terminal for a non-continuous data service. (It should be noted that a given GPD service may be carried under different packet addresses when transmitted at different times.)

Universal identity (UI):

This is a number in the range 0 - (256 - 1) which shall be unique to a particular GPD service on a world-wide basis. It may be represented in GPD SI by a coding space of from 4 to 7 bytes (32 to 56 bits). A GPD service shall be uniquely defined by the numerical value of UI, irrespective of the number of leading zeroes. A given GPD service, as defined by its purpose, data provider, intended user-group, etc.., shall have one and only one UI value, irrespective of the network or time of transmission. The UI is intended to support automatic acquisition of a GPD service, irrespective of network or time of transmission. It is particularly relevant to data services which may need to use different satellite channels, either concurrently or consecutively.

Name server:

A GPD service may be uniquely identified by a plain text name, transmitted in a name-server within the GPD SI protocol. This is intended for presentation as a menu of available services, with provision for automatic acquisition of a given service after selection by the user.

7.4.4.1 General description of GPD service identification

All GPD SI information shall be carried under one value of packet address as signalled by DATX in packet 0 SI. This MAC channel will be called the GPD SI channel.

In order to provide the flexibility required, the GPD SI channel is subdivided into different types of lists used to carry different classes of information. Each list, with the exception of list type '0, may be further subdivided into separate messages. Each message may be carried by a single packet, or a group of linked packets. The various messages making up a list shall be transmitted cyclically so that all information in the list is transmitted repetitively.

When a list is subdivided into more than one message, any single message can be selected by use of a Message Index (MI) value.

In the following description of the GPD SI lists, a distinction is drawn between an active GPD service and an inactive one. The point arises from the fact that data may be transmitted via a GPD component in an intermittent manner, possibly with long gaps between data messages. This may happen even though the data provider intends that receivers keep the relevant packet acquisition channel (or channels) open. In this case, however long the gap between data messages, the GPD service shall continue to be described in GPD SI messages, and the service is considered active. When a GPD service becomes inactive, the GPD SI description shall be removed, and receivers are not expected to keep packet acquisition channels open to receive it. The distinction largely concerns the arrangements between a data service provider and the operator of a satellite channel, since a GPD service shall have one or more packet addresses allocated to it while active, whereas these packet addresses can be used for other purposes when it is inactive.

The different types of GPD SI lists shall be identified by Type of List (TL), and the following values are defined:

Type '0: Network summary:

This list may be used as an option and consists of one message. It may be used to duplicate the parameters NWO, NWNAME, and DATX from the packet 0 Network Command.

Type '1: Service descriptions list:

This list shall be used and it carries the service descriptions of all active GPD services. To select a particular message, the receiver acquires the packet, or group of linked packets, having TL = '1, and a MI value equal to a Primary Index (PX) obtained from one of the following lists. To select a particular service description within a message, the receiver uses a Secondary Index (SX), also obtained from one of the following lists. Each entry in the list consists of a GPD service description. These are in the form of parameters similar in principle to packet zero SI, but with a more specialised and compact coding. PX and SX are allocated by the network operator for each transmission of a GPD service, but may not change while the service remains listed as a current service in GPD SI. The purpose of PX and SX is to provide an efficient location mechanism for the required service description. All services having a given value of PX shall be described in the same message having MI = PX, and within this message, no two service descriptions shall have the same value of SX.

Type '2: NSI cross-reference list:

This list may be used as an option, but any active GPD services which have an NSI shall appear in an NSI list. To select a particular message in the list, the receiver acquires the packet, or group of linked packets, having TL = '2, and a message index value which may be calculated from the NSI value. Within each message, NSI values and the corresponding values of PX and SX are listed in fixed format.

Type '3: UI cross-reference list:

This list may be used as an option. All active GPD services which have a UI and which do not appear in either the NSI cross-reference or the name server list (see below) shall appear in the UI cross-reference list. To select a particular message in the list, the receiver acquires the packet, or group of linked packets, having TL = '3, and a message index value which may be calculated from the UI value. Within each message, UI values and the corresponding values of PX and SX are listed in fixed format. The length of each entry is the same throughout any given UI list, but is not a fixed value. The UI coding size shall be signalled in DATX.

Type '4: Name server list:

This list is may be used as an option. All active GPD services which do not appear in either the NSI or UI cross reference lists shall appear in the name server list. Where this list consists of more than one message, the messages shall be transmitted cyclically with message index values which shall increment modulo-N, where N is the number of messages which comprise the complete list. The value of N shall be signalled in DATX. The receiver uses the index to ensure that all messages in the list have been acquired. Where the list consists of only one message, the index shall have the value zero, and N is signalled with the value 1. Each name server message carries entries for services having the same value of PX, with PX signalled in the message. Each entry shall consist of the SX value plus a plain text service name.

All other values of TL are unallocated and reserved.

7.4.4.2 Description of GPD SI list transport protocol

The following details, illustrated in figure 47a, are common to all types of GPD SI list.

A GPD SI list may consist of one or more messages. Each message shall be coded as follows:

- a message header consisting of 5 or 7 bytes;
- the message contents, of variable length;
- a message CRC, computed for all header and message content bytes. The same generator polynomial is used as for packet and SI data group suffices, as described in subclause 8.3.2.

The message header shall be coded as follows:

- Byte 1: Type of List (TL) byte:
    - this byte identifies which type of GPD SI list each message belongs to. Values '00 to '04 are defined, as described in subclause 7.4.4.1, with values '05 to 'FF reserved. All messages in a given GPD SI list shall have the same value of TL byte.
- Byte 2: Message Index (MI):
    - MI uniquely identifies each message in a list, as specified for each type of list in subclause 7.4.4.3.
- Byte 3: Coding Sequence Number (CSN) byte:
    - this byte indicates updating of the GPD SI lists. Whenever the coding of a message content is changed, CSN shall be simultaneously incremented by 1 (modulo-256). CSN operates independently for each message in each type of GPD SI list.

- Byte 4: Reserved byte:

  - this byte is reserved for future use and shall be set to '00.

- Byte 5 or bytes 5-7: Message Length Indicator (LI):

  - this is a 1-byte or 3-byte length indicator, coded as described in subclause 8.3.1. It gives the number of bytes which follow in the message, including the 2 CRC bytes.

Bytes 1 to 3 permit messages to be selected according to requirements. Filtering on byte 1 selects any message of a given Type of List (TL). Filtering on both bytes 1 and 2 allow a particular message to be selected from the list by MI. When a message has been acquired and decoded, a decoder may be set up to acquire the next message only when there has been a coding change, by filtering on all three bytes.

7.4.4.3 Use of Message Index (MI) in each type of list

7.4.4.3.1 Network summary list

If present, this shall consist of one message, for which MI is set to 0.

7.4.4.3.2 Service description list

In each message of the list, MI shall have the value of PX, as given in the name server or cross-reference lists. All services listed as having a given value of PX, and no other services, shall be described in the message identified by MI = PX. At any time, there shall be only one message in the service description list, transmitted repetitively, having a given value of MI. There may be from 1 to 256 messages in the service description list, according to the number of PX values in use.

7.4.4.3.3 NSI cross-reference list

In each message of the list, MI shall have the value of a pointer computed from NSI. All services having values of NSI yielding the same pointer value, and no other services, shall appear in the message identified by MI equal to this pointer. At any time, there shall be only one message in the NSI cross reference list, transmitted repetitively, having a given value of MI. The NSI message pointer shall be computed as a variable number of bits from zero to 7, signalled in DATX. There may be zero messages in the list, or any power of two up to 128 messages.

7.4.4.3.4 UI cross-reference list

In each message of the list, MI shall have the value of a pointer computed from UI. All services having values of UI yielding the same pointer value, and no other services, shall appear in the message identified by MI equal to this pointer. At any time, there shall be only one message in the NSI cross reference list, transmitted repetitively, having a given value of MI. The UI message pointer shall be computed as a variable number of bits from zero to 7, signalled in DATX. There may be zero messages in the list, or any power of two up to 128 messages.

7.4.4.3.5 Name server list

There may be any number of messages in this list from zero to 255, as signalled in DATX. When there is only one message in the list, MI shall be set to zero. When there are N messages in this list, they shall be transmitted with MI incrementing modulo-N, i.e., in a repeating cycle from 0 to (N-1).

7.4.4.4 Computation of NSI and UI message pointers

The pointers used to locate messages in the NSI and UI cross reference lists, have variable coding widths. The pointer sizes may be chosen independently for the two lists, and shall be signalled in the DATX parameter (see subclause 8.2.1). The same algorithm shall be used to compute a pointer from either NSI or UI, according to the number of bits n in the pointer, as follows:

If,

n = 0, the pointer = 0, there is only one message in the list, and this has MI = 0.

Otherwise,

Starting from the least significant bit of NSI or UI, split the binary representation of the identity into 8-bit words. Perform modulo-2 addition (i.e. bit-by-bit exclusive OR) on these 8-bit binary values.

If n is less than 5, split the above 8-bit result into a least-significant 4 bits and a most-significant 4 bits and perform modulo-2 addition (i.e. bit-by-bit exclusive OR) on these two 4-bit words.

The least-significant n bits of the result represents the binary value of the pointer.

7.4.4.5 Assembling messages into packets

All GPD SI messages shall be carried in packets having Golay error protection. This is independent of the use of Golay protection for packet 0 SI, and of the setting of the SIFT bit in line 625.

Each message shall be divided, from Byte 1 of its header, into blocks of 45 bytes, adding arbitrary padding bytes if necessary after the CRC to complete the last (or only) block. Each block shall be coded into 90 bytes each consisting of 30 Golay (24,12) words as described in subclause 8.3.2.

When a message, so coded, consists of only one 90-byte block, it shall be loaded into the useful data area of one packet, and transmitted with packet header continuity index = '0, and Packet Type PT = '00.

When a message, so coded, consists of more than one 90-byte block, each block shall be loaded into the useful data area of a set of linked packets, as follows:

- first or synchronising packet: PT = '00;
- other or continuation packets: PT = '3F;
- the packet header continuity index shall be set to zero for the first (or synchronising) packet, and shall increment modulo-4 for the remaining packets.

If an NSI or UI message pointer calculation is such that no entries are required in a particular message, the message shall be transmitted in empty form.

7.4.4.6 Rules of multiplexing GPD SI messages

7.4.4.6.1 Transmission rates

If present, the Network Summary message shall be transmitted on average every 12th television frame, with never more than 25 frames between transmissions. All other GPD SI messages may be transmitted at a repetition rate determined for the service requirements.

If present, the Name Server messages shall be transmitted in a cycle during which the Message Index (MI) of consecutive messages increments modulo-N, where N is the number of messages in use. There is no requirement on the transmission order of the Service Description or NSI and UI cross reference messages.

7.4.4.7 The coding of GPD SI lists

In the following Sections, for all bytes, whatever they are representing, the least significant bits are transmitted first, and for all sequences of numbered code bits transmission is in increasing numerical order. This rule for bit transmission order does not apply to bit sequences used for CRC purposes.

7.4.4.7.1 Network summary message (see figure 47b)

The network summary message shall consist of a duplicate of any or all of the packet 0 Network Command parameters NWO, NWNAME and DATX. The message content does not include the associated Command Identifier (CI) and Length Indicator (LI). Thus the first byte of message content shall be the Parameter Identifier (PI) of whichever parameter comes first. The parameters shall be ordered in increasing value of PI.

A change in the coding of any parameter in the Network Summary message shall be accompanied by an increment in the value of the message CSN byte, and shall occur at nominally the same time as the associated coding change in the packet 0 channel.

7.4.4.7.2 Service description list (see figure 47c)

The contents of each message in this list shall consist of one or more GPD service descriptions. Each service description shall be coded as follows.

First byte:

- Secondary Index (SX) value, as given in the name server or cross reference lists.

Second byte:

- A one-byte length indicator giving the number of bytes which follow in the GPD service description.

Service description:

- One or more parameters, each coded in general as follows:
    - First byte.

      A combined parameter identifier and length indicator:

        - least-significant 4 bits: Length Indicator (LI);
        - most-significant 4 bits: Parameter Identifier (PI).

      Remaining bytes
      Coded in a manner specific to each parameter.

The following parameters are defined, with reference to existing packet 0 service identification parameters:

DACMM PI = '0

This parameter may be used as an option, with data field as ACMM: this parameter is included so that different GPD services may conveniently use different over-air addressing (OAA) services: alternatively, the OAA service for GPD services may be described in packet 0 SI and associated with DACMM (see below) via the CA system byte and system-specific data of the ACMM and DACCM parameters.

DACCM PI = '1

This parameter shall be used when conditional access scrambling is in use: data field as ACCM.

DGPD PI = '2

This parameter shall be used describing each GPD component: the data field shall be coded as follows:

- Bytes 1 and 2 are coded as a 16-bit field. The 12 LSBs shall be coded as for DCINF, and the 4 MSBs shall be coded as follows:
    - bit 13, indicates the presence (bit = 1), or absence (bit = 0) of a segment suffix in the GPD component;
    - bit 14, indicates the presence (bit = 1), or absence (bit = 0) of Golay protection in the GPD component;
    - bit 15, indicates the presence (bit = 1), or absence (bit = 0) of message headers in the GPD component;
    - bit 16, the most-significant bit, indicates the presence (bit = 1) or absence (bit = 0) of packet linking in the GPD component.
- Byte 3 may be present as an option. If provided it shall be coded as follows:
    - the least significant 4 bits describe a general class of transmission based upon the maximum average throughput of user data which a GPD component may carry. Where address extension is in use in the component data message header, the maximum throughput applies to all sub-addressed channels in aggregate. The following values are defined:
        - '0 - no limits on throughput;
        - '4 - maximum throughput 64 kbit/s;
        - '7 - maximum throughput 7 680 bit/s;
        - All other values are unallocated and reserved.
    - the most significant 4 bits, bit 5 to bit 8, indicate when certain functions of the component segment header are in use. When bit 8 is set to 0, the functions of bits 5, 6 and 7 are undefined and reserved. When bit 8 is set to 1, bits 5, 6 and 7 have the following meanings:
        - bit 5 indicates whether the segment length indicator may be in use (bit = 1) or is not in use (bit = 0);
        - bit 6 indicates whether the segment counter may be in use (bit = 1) or is not in use (bit = 0);
        - bit 7 indicates whether address extension may be in use (bit = 1) or is not in use (bit = 0).
- Byte 4 may be used as an option for a language byte, coded as for packet 0 sound DCINF. If this byte is provided when byte 3 is not required, byte 3 shall still be supplied, set to '00, in order to permit a correct interpretation of the optional bytes.

DSOUND **PI = '3**

A mandatory parameter for each digital sound component associated with a GPD service and not part of a TV, radio or teletext service: data field as sound DCINF, including optional complementary access coordinates giving BI information.

DTEXT **PI = '4**

This parameter shall be present for each digital teletext component associated with a GPD service and not part of a TV, radio or teletext service: data field is coded as the corresponding DCINF (Parameter Identifier 'B0) as given in Part 4B.

DVIDEO **PI = '8**

This parameter shall be used if video carried in the sub-frames identified by TDMCID '10 and '11 is part of a GPD service: the data field shall consist of one byte coded as for VCONF.

Other values of PI are unallocated and reserved.

The association between conditional access parameters (DACMM and DACCM) and service component parameters (DGPD, DSOUND, DTEXT, DVIDEO) shall be effected by parameter ordering. There is no requirement to order parameters in ascending order of parameter identifier. Component parameters shall be associated with the closest preceding conditional access parameter, if present. The parameters for all unscrambled components shall appear before any DACCM parameters.

GPD service descriptions, coded as above, shall be assembled into service description messages in ascending order of SX value. All service descriptions in a given message shall have the same value of PX, which is the value of the message index in the message header. No two service descriptions in the same message shall have the same value of SX.

7.4.4.7.3 NSI cross-reference list (see figure 47d)

An NSI cross reference list message shall consist of binary information in fixed-length entries, each coded as follows:

- bytes 1 and 2: a 16-bit group, transmitted LSB first, giving the value of NSI;
- byte 3: one byte giving the value of PX;
- byte 4: one byte giving the value of SX.

NSI cross reference entries, coded as above, shall be assembled into messages in ascending order of NSI value. Each NSI cross reference message shall reference all, and only, those GPD services appearing in the list which have the same value of NSI message pointer, according to the pointer coding width signalled in DATX.

7.4.4.7.4 UI cross-reference list (see figure 47e)

A UI cross reference list message shall consist of binary information in fixed-length entries, coded as follows. The length of the entry shall depend upon the size of UI coding space which is signalled in DATX. For UI values represented by n bytes:

- first n bytes: a single multi-bit group, transmitted LSB first, giving the value of UI;
- next byte: one byte giving the value of PX;
- last byte: one byte giving the value of SX.

UI cross reference entries, coded as above, shall be assembled into messages in ascending order of UI value. Each UI cross reference message shall reference all, and only, those GPD services appearing in the list having the same value of UI message pointer, according to the pointer coding width signalled in DATX.

7.4.4.7.5 Name server list (see figure 47f)

Entries in the name server list shall be grouped under PX values in a similar manner as for the service description list. All services described in the name server and having a given value of PX shall be entered into the same name server message. However, the message index of name server lists is not equal to PX, but is an incrementing index for successive messages as described in subclause 7.4.4.3.5.

The first byte of each name service message shall give the value of PX for all entries in the message. The rest of the message shall consist of entries coded as follows:

- byte 1: this gives the value of SX for the entry;
- byte 2: this is a length indicator giving the number of bytes in the rest of the entry;
- remainder: this consists of a variable length plain text service name coded as defined in subclauses A.3.1 and A.3.2.

7.4.4.8 Coding changes

Whenever any content of a GPD SI message is changed, the Coding Sequence Number (CSN) in the message header shall simultaneously be incremented modulo-256.

If the change is associated with a change of packet 0 SI, such as a change in the data field of DATX, the GPD SI messages shall change coding at nominally the same time that packet 0 changes and introduces UPDAT.

When the change in a GPD service description affects the decoding of a component which is neither starting nor stopping, such as a change between error protection or presence of header in a GPD component, the data messages shall stop before the first SI message containing the change, and shall start with the new coding after a sufficient interval of time to allow all decoders to respond to the change. Similarly, when a new GPD service or service component is introduced, the GPD SI change shall be made at a sufficient time interval before the new service or component starts in order to allow all decoders to acquire it.

7.4.5 Performance of GPD protocol in the presence of channel bit errors

Figure 48 compares the channel bit-error ratio with the effective bit-error ratio for the useful data (i.e. after error correction and protocol decoding) for GPD services. The various graphs show the performance for a "simple" reference data message (without a data message header) and a "complex" reference data message (where a 48 bit message header and packet linking is used), with and without the use of Golay forward error correction.

At high bit-error ratios, the effective bit error-ratio of the complex service degrades at a faster rate than the bit-error ratio of the channel. This effect is due to the presence of the data message header which introduces the dominant error mechanism at these high channel bit-error ratios.

A further improvement in the error performance of a GPD service may be achieved by the use of repetition. This is particularly effective in the presence of packet loss (due to precipitation attenuation etc..). This feature is not included in the GPD protocol of OSI layers II and III as specified in subclause 7.4.1.2, but should be implemented at OSI layer IV (transport).

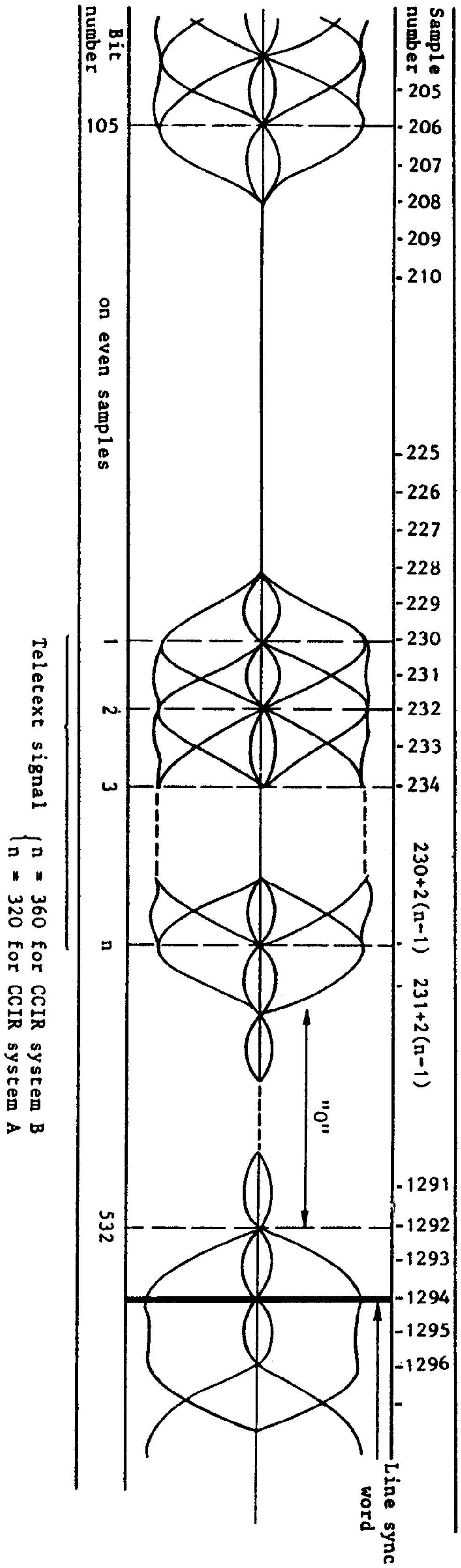


Figure 43: Relationship between the bits of data and the sampling structure

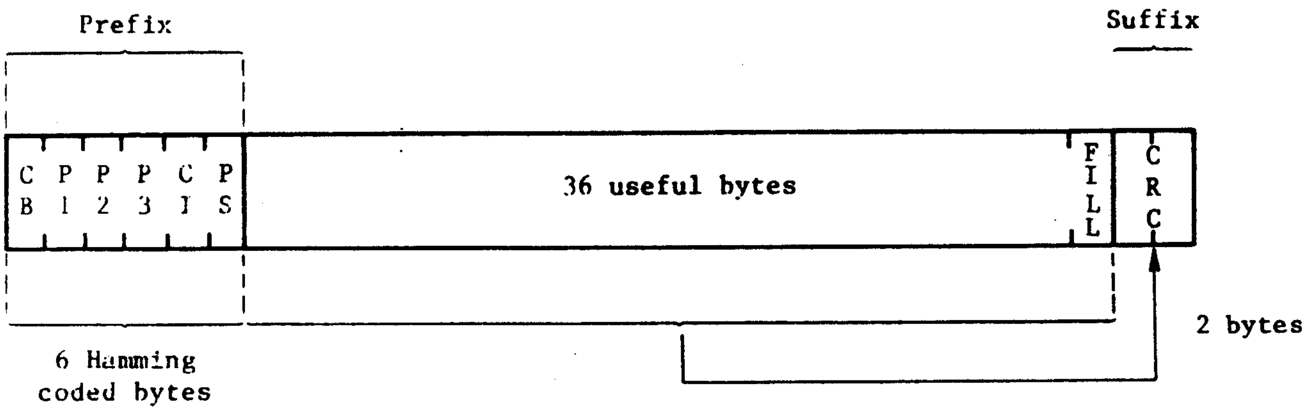


Figure 44a: CCIR system A teletext

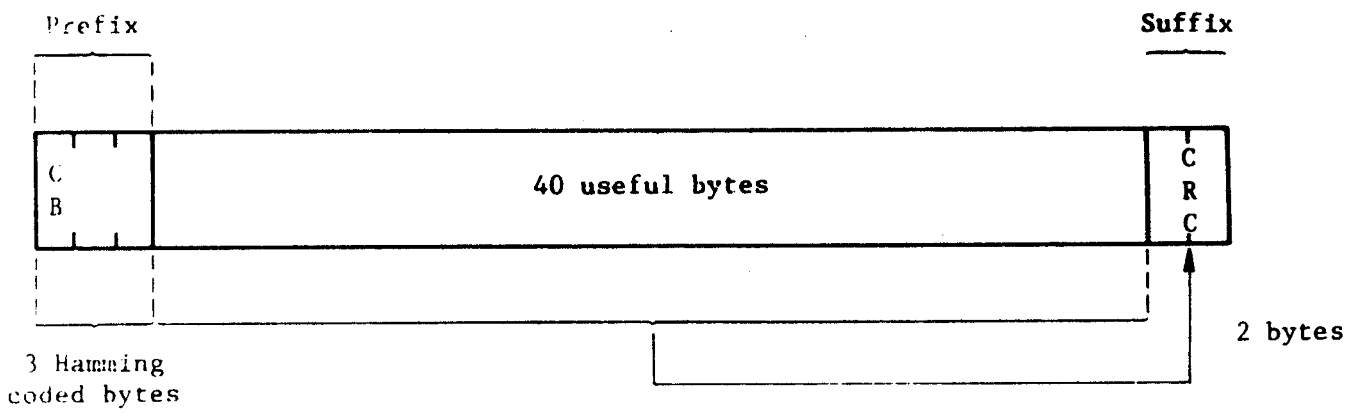


Figure 44b: CCIR system B teletext

Figure 44: Construction of a 45-byte teletext data block from a teletext data packet with CRC suffix

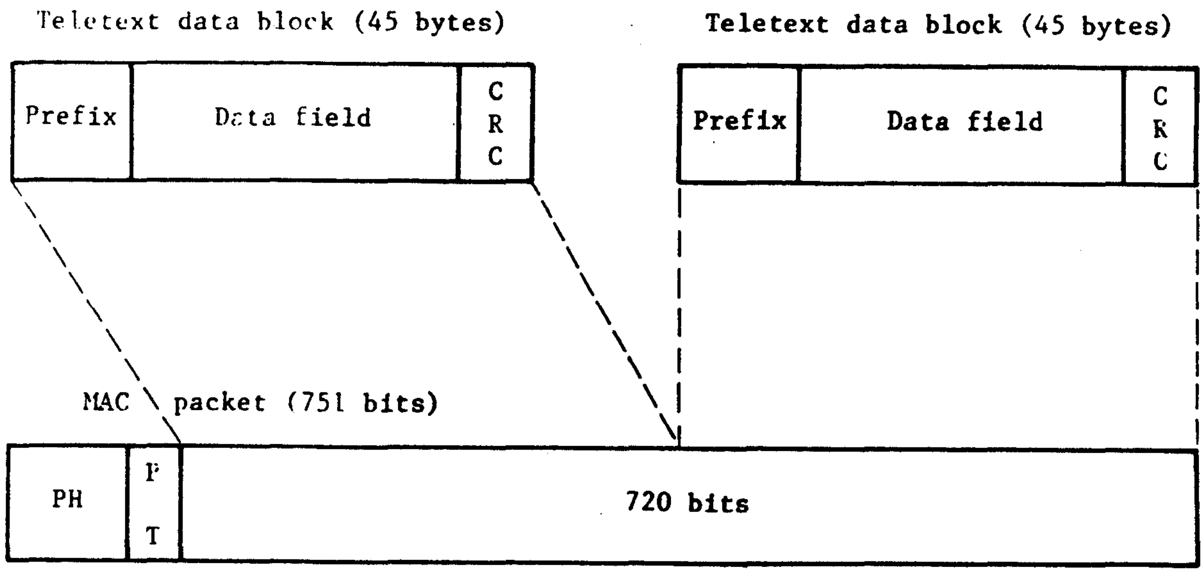


Figure 45a: First level protection

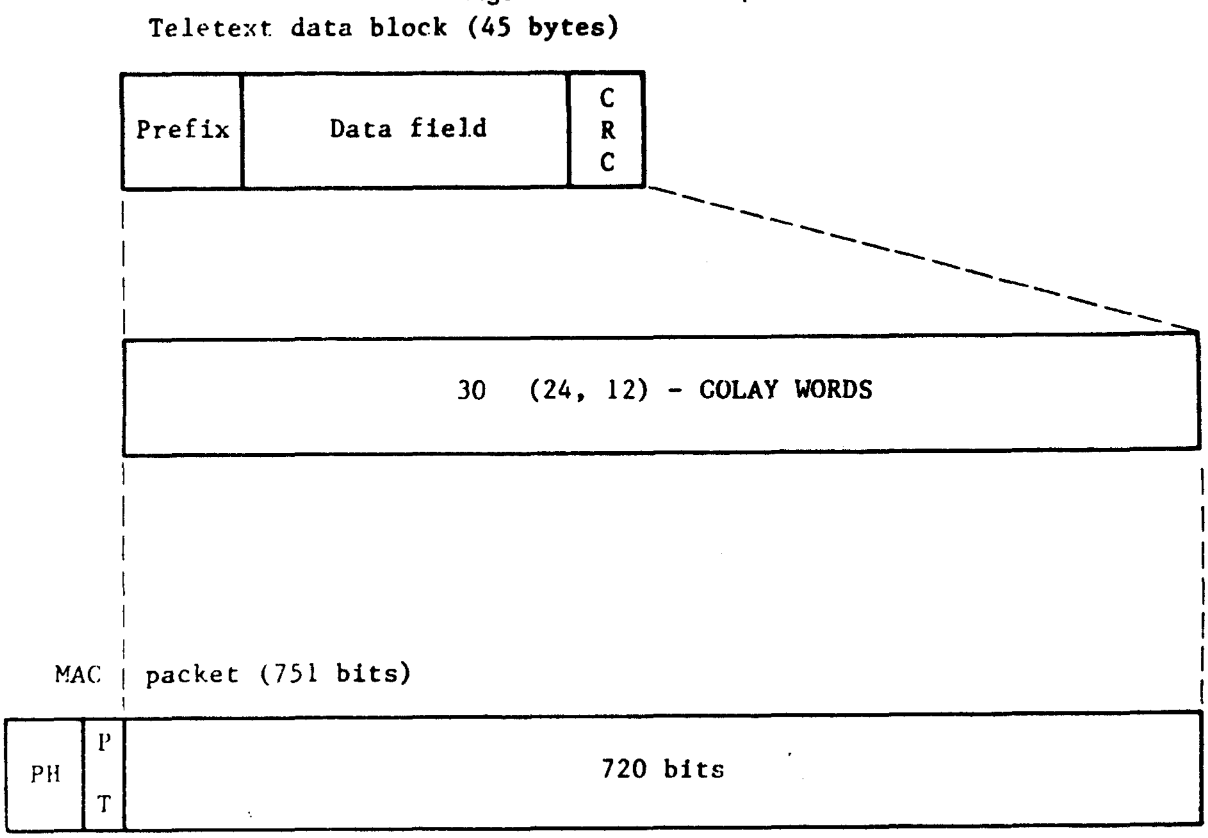


Figure 45b: Second-level protection

Figure 45: The insertion of teletext data blocks in a HDMAC packet

a) Data message format

Message header (variable, optional)	Data Segment (variable)	Segment CRC (2 bytes, optional)

b) Message header format

Format Descriptor (1 byte)	Message Header Extension (variable)

c) Message header extension format

Address Extension field (variable, optional)	Segment counter field (4 bytes, optional)	Segment length field (3 bytes, optional)

d) Address extension field format

Address extension length indicator (1 byte)	Extended address (variable)

Figure 46: Data message format

a) GPD SI list message format

Overall message structure

Message header 5 - 7 bytes	Message contents variable	CRC 2 bytes

Header structure

Type of List TL 1 byte	Message Index MI 1 byte	Coding Sequence Number CSN 1 byte	Reserved 1 byte	Length Indicator LI 1 or 3 bytes

b) Format of network summary message

Overall message structure

Message header 5 - 7 bytes	Entry 1 variable	Entry 2 variable optional	Entry 3 variable optional	CRC 2 bytes

Structure of each entry (NWO, NWNAME, or DATX parameter as packet 0 SI)

Parameter Identifier PI 1 byte	Length indicator LI 1 byte	Parameter Data Field variable

c) Format of service description messages

Overall message structure

Message Header 5 - 7 bytes	Entry 1 variable	Entry 2 variable optional	. . . remaining entries . . . optional	CRC 2 bytes

Structure of each entry

Secondary Index SX 1 byte	Length Indicator LI for rest of entry 1 byte	Parameter variable	Parameter variable optional	. . . Remaining Parameters . . . optional

Structure of each parameter

Combined Parameter/Length Indicator 1 byte		Parameter Data field coded as specified for each type of parameter
PI MS 4 bits	LI LS 4 bits	

Figure 47: Format of GPD SI messages

d) Format of NSI cross reference messages

Overall message structure

Message Header 5 - 7 bytes	Entry 1 variable	Entry 2 variable optional	. . . remaining entries . . . optional	CRC 2 bytes

Structure of each entry

NSI 2 bytes	Primary Index PX 1 byte	Secondary Index SX 1 byte

e) Format of UI cross reference messages

Overall message structure

Message Header 5 - 7 bytes	Entry 1 variable	Entry 2 variable optional	. . . remaining entries . . . optional	CRC 2 bytes

Structure of each entry

UI 4 - 7 bytes	Primary Index PX 1 byte	Secondary Index SX 1 byte

f) Format of name server messages

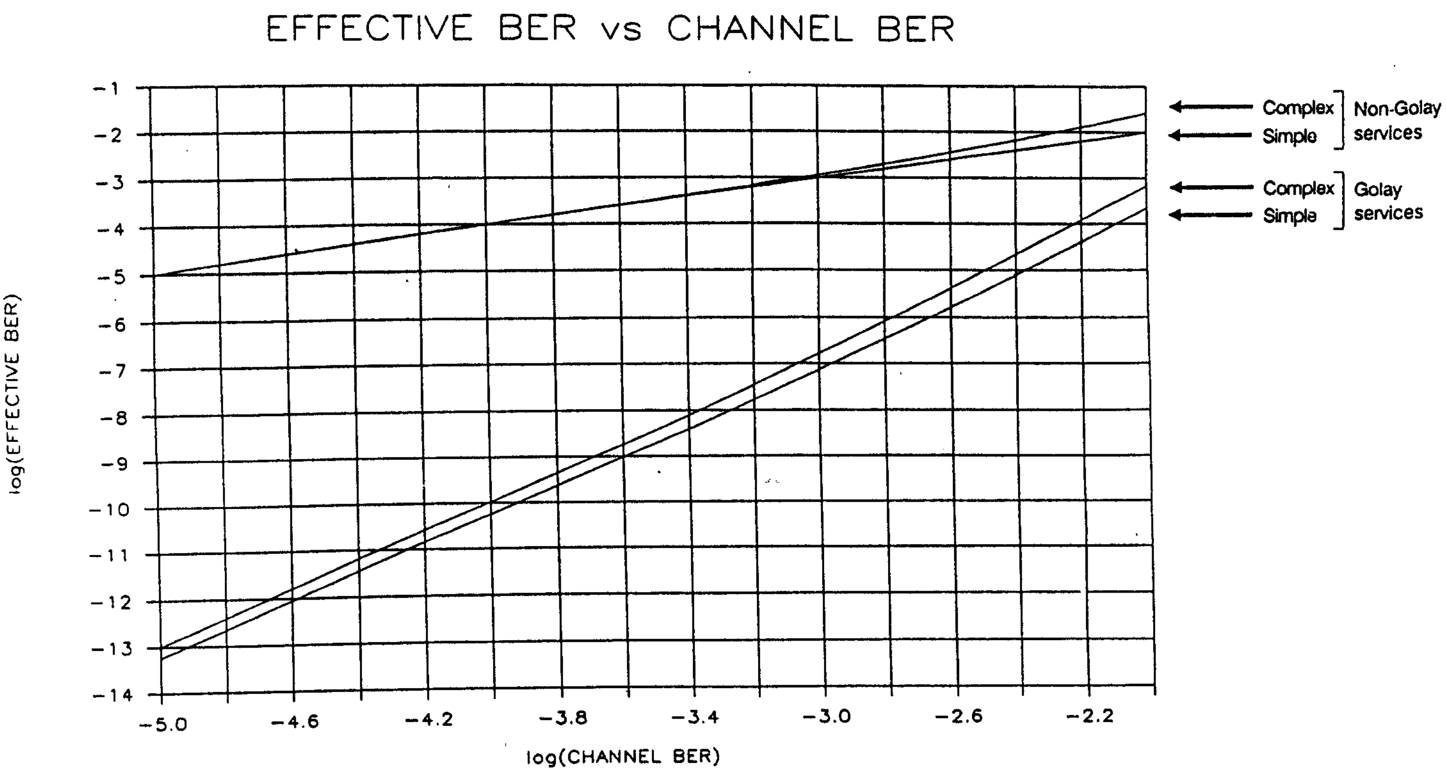
Overall message structure

Message Header 5 - 7 bytes	PX 1 byte	Entry 1 variable	Entry 2 variable optional	. . . remaining entries . . . optional	CRC 2 bytes

Structure of each entry

Secondary Index SX 1 byte	Length Indicator LI 1 byte	Plain text service name variable

Figure 47: Format of GPD SI messages (continued)



Simple reference message : 1 packet with no message header

Complex reference message : 4 linked packets with 48-bit message header consisting of FD byte, address extension length indicator byte, and a 4-byte extended address.

Figure 48: Effective BER vs. channel BER

8 Specification of the service identification channel

8.1 Introduction

Clause 8 gives the specification of the data broadcast in the service identification channel. This channel is formed by packets in the sound/data multiplex with the packet address "0" (see subclause 4.3.6.1.1). Service identification provides a comprehensive description of the elements contained in the D2-HDMAC/packet signal. This information gives access to the various television, sound and data services. In the case that service components other than DATV are present in the DATV/data multiplex, this multiplex may carry its own SI channel, providing information needed by the user to access these service components.

A service may comprise several components; for example, a television service comprises a vision component, a DATV component and one sound component, possibly more sound components (possibly in different languages), subtitles (again possibly in different languages), and "commentaries" giving supplementary information, for example in the form of teletext pages. The information in the service identification channel identifies the network and lists the services available. It describes all the components of each service and carries information that allows the receiver to be configured to receive the particular components of any service selected by the user.

The network and its services are described by a series of data items called parameters; these parameters are defined in subclause 8.2. For transmission, the parameters may be grouped into parameter groups and are coded into commands (see subclause 8.3.1). The commands are assembled into data groups each of which consists of one or more whole packets (see subclause 8.3.2). There are sixteen numbered data groups; data group 0 carries parameters that relate to the whole network or that give access to parameters that may be carried in other data groups. Subclause 8.4 gives rules for multiplexing the packets carrying the data groups in the whole sound/data multiplex.

Annex A specifies coding tables for certain information transmitted by the service identification system. Annex G gives an example of the coding of the service identification data describing a complete network; this example is intended to help the reader to understand the specification.

In the following subclauses, the least significant bits shall be transmitted first for all sequences of bits representing magnitudes, for all bytes, whatever they are representing (hexadecimal pairs, BCD pairs or characters of a text message), and for all sequences of numbered code bits, whereby transmission is in increasing numerical order. This rule for bit transmission order does not apply for bit sequences used for CRC purposes. The transmission order of a sequence of bytes will be specified case by case.

8.2 Parameters

The following parameters are used to describe the network and the services available.

8.2.1 Network identification and description in terms of its services

(NWO) Network origin (PI = '10)

This parameter shall be used, its information consists of three bytes followed by a variable-length text. The three bytes give explicitly the satellite channel number in the range 0-99, the satellite orbital position in the range -179° (west) to 180° (east) and the polarisation. This information shall be conveyed in the following way:

- first byte: two BCD digits giving in sequence the "tens" and "units" of the satellite channel number;
- second byte: in sequence four bits defining the sign and "hundreds" of the satellite orbital position (-1, -0, +0, +1 correspond to 1001, 1000, 0000 and 0001, respectively) and a BCD digit defining the "tens" of the satellite orbital position.

- third byte: in sequence a BCD digit defining the "units" of the satellite orbital position and four bits giving the polarisation defined as:

  0000: Left hand circular;
  0001: Right hand circular;
  0010: Linear polarisation X, tilting from the horizontal plane (see NOTE);
  0011: Linear polarisation Y, tilting from the vertical plane (see NOTE).

- the text gives the name of the country of origin coded according to the standard given in subclause A.3.

NOTE: The "X" and "Y" polarisation's are orthogonal. The vertical reference is the meridian plane through the satellite pitch (north/south) axis, the horizontal plane being orthogonal to that plane and parallel to the satellite roll-yaw plane.

(NWNAME) Network name (PI = '14)

This parameter shall be used and it contains the name of the network as it is known by the public, requiring a variable length of text, coded according to the standard given in Clause A.3.

A short form of the information in these two parameters is sent in the CHID group in special data line 625 in order to facilitate automatic receiver operation.

(MODPRM) Modulation parameters (PI = '16)

This is parameter shall be used. This information shall consist of 8 or 9 bytes. The first 8 bytes give the centre frequency of the carrier, its deviation sensitivity, energy dispersal and pre-emphasis characteristics. The information shall be conveyed in the following way:

- first three bytes: six BCD digits giving in sequence the "ten thousands", "thousands", "hundreds", "tens", "units" and "tenths" of the centre frequency in MHz;

- next three bytes: three BCD digits giving in sequence the "tens", "units" and "tenths" for the deviation of the vision signal in MHz/V. The remaining bits are encoded as three BCD digits providing the same information for the data signal on the D2-HDMAC channel;

- seventh byte: two BCD digits giving in sequence the "thousands" and "hundreds" of the peak-to-peak energy dispersal in kHz;

- eighth byte: the most significant four bits indicate pre-emphasis characteristics:

  - xx11: both linear and E7E non-linear pre-emphasis (see subclause 10.2.1). xx are two bits reserved for future use and shall be set to 0. HDMAC decoders should not interpret these two bit;

  The least-significant four bits are unallocated.

- ninth byte (optional): this byte is defined only in the case of linear polarisation. The byte is coded with two BCD digits giving in sequence the "tens" and "units" of the tilt angle of the polarisation plane, as seen from the satellite with clockwise reference and with respect to the equatorial plane.

(LISTX) List of the index values (PI = '18)

This parameter shall be used. It gives for each type of service, at the moment of broadcasting, a list of the index values of the services of that type is given. The index value is used in the precise description of the service (see "SREF" in subclause 8.2.2). The list provides the means for verifying the complete reception of the descriptions of all available services of a particular type. The data field shall have a variable length and be formatted as follows:

- One byte is needed to specify the type of service. At present, three broadcasting services have been identified (television, radio sound, teletext) as well as a fourth service (over-air addressing) used as a possible means to distribute the entitlement management messages (see subclause 9.5.1) for controlled access to the previous services. A fifth service type (additional service identification) may be used to describe services carried outside the packet multiplex in which it is found. These five services shall be coded as follows (hexadecimal notation):

    - television service: code '01;
    - radio sound service: code '02;
    - teletext service: code '03;
    - over-air addressing service: code '04;
    - Additional Service Identification (ASI): code '05.

- For each service of the particular type:

    - one byte is needed for the index value;
    - followed by a 16-bit sequence;

        - for the data group type for the remaining service identification (four most significant bits);
        - and the packet address of the main digital component of the associated service (ten least-significant bits).

          However, when the main digital component of a service is not included in the sound/data multiplex, the value of the 10-bit field containing the packet address shall be set to zero. Access to this component requires the use of other information present in the SI channel. This is the case for teletext services using teletext component in the field-blanking interval (see subclause 7.2.3).

        - The two remaining bits (bits 11 and 12) are defined at present only in subframes characterized by TDMCID '01-'02, where they are used for digital component location in the following way:

- bit 11 = 1 bit 12 = 0 - bit 11 = 0 bit 12 = 1 - bit 11 = 1 bit 12 = 1	The packets of the digital component are inserted in the subframe characterized by an even or odd TDMCID code in the range '01 - '0E (see subclause 4.5.4)
- bit 11 = 0 bit 12 = 0	The packets of the digital component are not inserted in the subframe characterized by an even or odd TDMCID code in the range '01-'02. This digital component is not available.

The first eight entries in the list are eight ordered default choices for that type of service.

The parameters NWO, NWNAME, LISTX, DATX, COMD (for network commentary description) and TIMD (for local time description) together with UPDAT (see subclause 8.2.4) if applicable shall be sent in data group type "0" of packet address "0".

(DATX) List of Data Index services (PI = '19)

This parameter shall be used if any services of type General Purpose Data (GPD) are being transmitted. Such services do not have a LISTX type of service allocation, and the service descriptions and other information required to acquire the services are carried in a data channel with packet address not equal to zero. DATX indicates which packet address is used for this purpose, and provides information on the way in which GPD services are referenced. See subclause 7.4.4 for details of GPD SI description and acquisition.

DATX shall consist of 9 bytes coded as follows:

Byte 1 and 2: a 16-bit group (transmitted LSB first):

- bits 1-10: the packet address used for GPD service information;
- bits 11-16: unallocated but reserved for future use.

Byte 3: set to zero, reserved for future use.

Byte 4:

- bit 8: presence (bit 8 = 1) or absence (bit 8 = 0) of a Network Summary message. When bit 8 = 0, bits 1-7 are undefined;
- bits 1 and 2: the MPX location of the Network Summary message;
- bits 3-7: unallocated.

Byte 5:

- bits 1 and 2: the MPX location of the service description messages;
- bits 3-8: unallocated.

Byte 6:

- bit 8: presence (bit 8 = 1) or absence (bit 8 = 0) of NSI messages. When bit 8 = 0, bits 1-7 are undefined;
- bits 1 and 2: the MPX location of the NSI messages;
- bits 3-5: coding size of NSI message pointer in bits. A value of zero indicates that there is one NSI cross-reference message having MI = 0;
- bits 6 and 7: unallocated.

Byte 7:

- bit 8: presence (bit 8 = 1) or absence (bit 8 = 0) of UI messages. When bit 8 = 0, bits 1-7 are undefined;
- bits 1 and 2: the MPX location of the UI messages;

- bits 3-5: coding size of UI message pointer in bits. A value of zero indicates that there is one UI cross-reference message having MI = 0.

- bits 6 and 7: coding size of UI, indicated as follows:

  '0 4 bytes;
  '1 5 bytes;
  '2 6 bytes;
  '3 7 bytes.

Byte 8:

- bit 8: presence (bit 8 = 1) or absence (bit 8 = 0) of Name server messages. When bit 8 = 0, bits 1-7 are undefined;

- bits 1 and 2: the MPX location of the Name server messages;

- bits 3-7: unallocated.

Byte 9:

- bits 1-8: the number of Name server messages currently being transmitted. If this value is n, the Name service messages shall be transmitted in rotation with MI incrementing modulo-n.

All unallocated bits are reserved for future use, and shall be set to 0.

(TIME) Local time (PI = '20)

This parameter gives a local time, of medium precision, and may be used as an option intended for display and related to a relevant time-zone. The information shall be coded, in accordance with the standard given in Clause A.3, as text of limited length, for example in the form: day of week/year/month number/day of month / hour / minute. The length shall be limited to 32 characters. Changes to the value of TIME are simply broadcast; they are not announced through the UPDAT mechanism. However, UPDAT shall be used in association with TIME if the TIME parameter is being added to or deleted from a data group.

8.2.2 Service and programme-item description

For each available service, all or some of the items of information given in the following list are provided:

(SREF) Service reference (PI = '40)

This parameter shall be composed of two items of information and coded in the given order as follows:

- index value, to distinguish among several services of the same type; for a given type of service, the list of index values is given in the network information (see LISTX in subclause 8.2.1). The coding uses one byte, without other specific significance;

- name of the service, a limited-length clear text allowing suitably-equipped receivers to select a service by its name as known by the public. It consists of a variable-length text, coded according to the standard given in Clause A.3. The length shall be limited to 32 characters.

The parameter SREF shall be used for all non-TV service descriptions which are introduced with a LISTX parameter, but it should be used also for the TV service description.

(PREF) Programme item reference (PI = '48)

This parameter may be used as an option. When used it shall be composed of three items of information and coded in the given order as follows:

- programme-type:
  - in terms of nature of content or intended audience such as light entertainment, serious music, public affairs, drama or immigrant groups, young people, etc.. coded as one byte. The coding table is given in Clause A.4. Value "0" indicates that no programme-type information is provided. As only one code can be transmitted for each programme, the broadcasting organization decides which aspect (content or intended audience) is predominant when choosing the code to be transmitted.

- programme item number:
  - this code in association with SREF should enable the receiver to respond to a particular programme item that the user has preselected; the programme item number consists of three bytes representing a 24-bit code based upon CCIR Recommendations for the expression of date and time. It comprises six binary coded decimal digits representing the scheduled broadcast start day and time. The first byte carries the two least significant decimal digits of the Modified Julian Day (MJD). It is followed by the "tens" and "units" of hours (second byte) and the "tens" and "units" of minutes (third byte), on the Coordinated Universal Time scale (UTC). This implies that programmes anywhere in the world scheduled at the same absolute time have the same programme item number.

- name of the programme item:
  - within the service, a limited length clear text allowing suitably-equipped receivers to select a programme item by its name. It shall consist of a variable-length text, coded according to the standard given in Clause A.3. The length shall be limited to 32 characters.

(NXPREF) Next programme item reference (PI = '4A)

This parameter may be used as an option, which when used provides information about the next programme item to be transmitted. It shall be coded in the same manner as PREF.

(VPS) Video Programme System (PI = '49)

This parameter may be used as an option and shall be composed of five bytes of information coded according to the following specification: "Technische Richtlinie ARD/ZDF/ZVEI No. 8R2 Video Programme System (VPS)" (bytes 5 and 11 to 14 of the above indicated specification).

In addition, line 16 of the field-blanking interval may carry a biphase data signal in accordance with the above specification for terrestrial broadcasting. This signal is processed as a luminance signal. This signal may be added in line 16 in order to provide compatibility with video tape recorders for which D2-MAC/packet signals are converted into a composite TV signal. This line, when used, shall be signalled within the time division multiplex control of line 625.

8.2.3 Service components description

Depending on the service type and the complexity of the programme, one or several components shall be present. A service shall have at least one component. For each of them, a component description is given. Depending on the nature (analogue or digital) of the component, different information is provided:

(ACMM) Access management related message (PI = '78)

This parameter gives the information concerning entitlement management messages. It shall be used when a network contains an over-air addressing service for conditional access. It may be used within an over-air addressing command, or within a service description. When used within an over-air addressing command it shall apply to one type of conditional-access system. When used within another type of service description it shall apply to one CA system, itself applicable to the service components following it in the command or parameter group.

The parameter field shall be encoded as follows:

- bits 1-10: these give the packet address for the over-air addressing component;

- bits 11-12: these two bits indicate the subframe related location of this digital component in the same manner as bits 11 and 12 in LISTX;

- bits 13-16: unallocated, reserved for future use;

- CA system byte: this byte gives the type of conditional-access system to which ACMM applies (see subclause 9.1);

- variable length field: a further field may be added, coded in a manner specific to the type of conditional-access system.

(ACCM) Access checking related message (PI = '88)

This parameter gives the information concerning entitlement checking messages (ECMs) described in subclause 9.5.1. This parameter shall be used for all service components which are subject to conditional-access control. The ACCM content is associated with one or more service components.

The coding shall be as follows:

- bits 1-10: these give the packet address for the ECMs. If entitlement checking is provided for, without the use of a dedicated packet address (i.e. MSB = 0), the packet address value is set to "0".

- bits 11-12: these two bits indicate the subframe related location of this digital component in the same manner as bits 11 and 12 in LISTX.

- bits 13-14: unallocated.

- bit 15: this bit indicates the way the control word is derived. If bit 15 is set to 1, the control word shall be derived from the 20 bit CAFCNT parameter carried in line 625. If bit 15 is set to 0, the control word shall be derived by other methods.

- bit 16 (MSB): this bit indicates the presence (bit 16 = 1) or absence (bit 16 = 0) of an associated entitlement checking message in the digital multiplex.

- CA system byte: this byte gives the type of conditional-access system required for access to the service components to which ACCM applies (for coding see parameter ACMM).

- variable length field: a further field may be added, coded in a manner specific to the type of conditional-access system.

The entitlement checking procedure defined by an ACCM parameter shall apply to all components following it within a parameter group or command which are specified (e.g. in the VSAM field in line

625 for vision, in BI packets for sound, and with the PT byte for packet teletext) as being subject to conditional access.

In the Eurocrypt conditional-access system, the entitlement checking procedure defined by an ACCM parameter shall apply to all the components which follow it in a parameter group or a command; the ACCM parameter indicates that these components are subject to identical access conditions or are in a transient phase toward this configuration.

A service component or group of service components may be subject to control by several different types of conditional-access systems.

(VCONF) Video configuration (PI = '90)

This parameter shall be used and it allows the correct "interpretation" of the video signal. It shall be coded as one byte according to the following rules:

- The three least significant bits duplicate the VSAM sub-group data of MVSCG (see subclause 4.5.3), i.e.:

  xxxxx000: free access, double-cut component rotation scrambling.
  xxxxx100: controlled access, double-cut component rotation scrambling.
  xxxxx010: free access, single-cut line rotation scrambling.
  xxxxx110: controlled access, single-cut line rotation scrambling.
  xxxxx001: free access, unscrambled.

- Two bits describe the source picture aspect ratio (AR) and compression ratios for the luminance and colour-difference signals (Cy and Cu respectively). The following bit combinations are at present allocated:

	aspect ratio	compression ratio	
00110xxx	AR = 16:9	Cy = 3:2	Cu = 3:1

(ASCONF) Analogue sound configuration (PI undefined)

This parameter allows the correct "interpretation" of an analogue sound signal. It can be introduced for terrestrial applications, when an analogue sound component is present.

(DCINF) Digital component information (several PI values)

This parameter shall be used for any service with one or more digital components. A separate DCINF parameter shall be supplied for each digital sound, DATV, data and teletext component. DCINF shall consist of three items (the last being optional), and shall be coded in the given order as follows:

- language used within this component;

  - coded in one byte. The value 0 indicates that no language information is provided or that this kind of information does not apply to the component. Value 40 refers to background sound (or "clean feed"). Values up to 127 are common to all components. Other values, above 127, are specific to each kind of component and do not necessarily refer to human languages (for example programming languages for telesoftware). For any given service no more than one component of each type (as identified by the DCINF PI value) shall have a particular language byte value. The coding table for codes up to 127 is given in Clause A.5.

- access coordinates for the digital channel carrying this component;

  - coded as a 16-bit sequence. The ten least significant bits shall represent the packet address (PA) of the digital component (transmitted in the same order). As in LISTX the two bits adjacent to the packet address bits (bits 11 and 12) shall give an indication on the subframe related location of this digital component. The four most significant bits shall be coded according to the type of component. For TV and radio sounds, the four most significant bits shall be coded as described in subclauses 11.3.2 and 11.4.2. For teletext within the packet multiplex, the four most-significant bits of the data field shall be coded as specified in subclause 7.3.3. For teletext in the field-blanking interval, this data field shall be coded as described in subclause 7.2.3. For DATV the data field shall be coded as described in subclause 5.8.4.

- complementary access coordinates;

  - a variable length optional field of hexadecimal digits (two per byte), allowing the possibility of defining components as a part of the content of a digital channel. The use of this field depends on the component and has to be specified for each component. For teletext, reference is made to subclause 7.2.3 for components in the field-blanking interval and to subclause 7.3.3 for components in the packet multiplex.

  - For digital audio components with DCINF values 'A4, 'A5, 'A8, 'A9, 'AA, 'AB and 'AC the first two bytes of the complementary access coordinates reflect, when included, information contained in the BI packets within the component. If the digital audio component description contained within the complementary access coordinates is being updated, then the complementary access coordinates correspond to the BI data associated with CI = 2 in the sound channel itself. Otherwise, the complementary access coordinates correspond to CI = 1. The coding of the complementary access coordinates bytes shall be identical to the corresponding BI data specified in Clause 6 with the following exceptions:

    - byte 1 bit 1: this bit is chosen such that the modulo-2 addition of all the bits in bytes 1 and 2 of the complementary access coordinates is "0";

    - byte 1 bit 8: this bit is reserved and shall be set to zero. (Bit 8 in the BI data specifies whether the sound signal is present or interrupted);

    - All other bits shall have the same meaning as the corresponding BI data. If at any time the description at a sound component provided in the SI channel differs from that provided in the BI packets in the component, then the description in the BI packets shall prevail.

8.2.4 Miscellaneous information

Some types of information are common to the network on the one hand, and the service component and programme description on the other. These are:

(UPDAT) Updating messages (PI = '00)

The UPDAT message shall be used. It signals the changing of the content of a previous network or service description. The coding of those messages will be given in subclause 8.3.1.

(ULIST) Update list parameter (PI = '01)

This parameter may be used as an option, which is provided only in association with an UPDAT parameter to indicate changes of SI content with greater precision. The coding of a ULIST parameter is a list of bytes corresponding on a one-for-one basis with the data field of the associated UPDAT parameter. ULIST is not used in association with an UPDAT parameter having a zero length data field. ULIST data bytes shall be specified in terms of the corresponding data bytes of the associated UPDAT parameter:

- LISTX index byte;
- DCINF language byte;
- EDCINF language byte;
- ACMM CA system type byte;
- ACMEM CA system type byte;
- ACCM CA system type byte;
- ACCEM CA system type byte.

- All other values of UPDAT data bytes shall have a corresponding dummy ULIST data byte with all bits set to zero. When ULIST is used to identify change in LISTX parameters, the index values listed for services in LISTX shall be allocated so that no index values are repeated throughout all the LISTX fields in the network command. PI values '02, '03 and '04 are reserved for future use in connection with the UPDAT mechanism.

- PI value '05 is also associated with UPDAT.

(COMD) Commentary description (PI = '61)

The COMD parameter may be used as an option, giving information on an existing direct or indirect commentary. Commentary messages can be of two types: direct commentaries, which are broadcast within the service identification channel and indirect commentaries, which are carried by other data channels of the sound/data multiplex. The presentation features for indirect commentaries will not necessarily be those specified in Clause A.3. The field coding of COMD shall be identical to the field coding of DCINF. Furthermore, for a commentary description of the direct commentary, the four most significant bits of the access coordinates shall give the value of the data group type which carries this commentary.

(TIMD) Description of local time (PI = '62)

The TIMD parameter may be used as an option. It shall have the same coding as that specified for COMD, in order to provide the access coordinates of the local time information (TIME) (see subclause 8.2.1).

(DCOM) Direct commentary (PI = '60)

The DCOM parameter may be used as an option, giving information on the network or on the services broadcast in variable-format clear text. It shall be coded as follows: one byte representing the language used (according to the same table as used for DCINF), followed by a variable-length clear text coded according to the standard given in Clause A.3.

(LKPNT) Link pointer (PI = '70)

When used within extension and continuation commands, gives information necessary to locate the data group in which extension is to be found or the data group which is extended by the continuation. It shall be coded in one byte as follows:

- the two most significant bits are unallocated;

- the next two most significant bits give an indication of the subframe related location of the associated data group (as in LISTX except that the value 00 is excluded);

- the final four bits indicate the data group type for the associated data group.

8.2.5 Further parameters required for additional service identification (ASI)

(MID) Multiplex identity (PI = '17)

This parameter identifies in the ASI service a subframe other than the one which carries the ASI. It shall be coded as one byte representing the value of the TDMCID code of a subframe (see subclause 4.5.4). Services and service components described by parameters following a MID parameter shall be carried in the subframe identified by MID (if applicable), unless a different TDMCID is explicitly indicated with an ACMEM, ACCEM, or EDCINF parameter.

(EDCINF) Extended digital component information

This parameter consists of four items (the last being optional), which shall be coded in the given order as follows:

- language used: definition as for DCINF.

- access coordinates:

  - for the digital channel carrying this component, coded as a 16-bit sequence. The ten least significant bits represent the packet address (PA) of the digital component (transmitted in the same order).

  - For a digital component carried in a subframe (characterized by TDMCID values '42 to '4F), bits 11 and 12 are unallocated. The four most significant bits are reserved for future use (for example to give information on the bit rate within that channel).

- multiplex location:

  - one byte representing the value of the TDMCID code of the unpaired subframe which carries the digital component (see subclause 4.5.4).

  - Complementary access coordinates: definition as for DCINF.

(ACCEM) Access checking-related extended messages (PI = '89)

The parameter field shall be as defined for ACCM except that the first 16-bit field is extended to 24 bits. The additional 8 bits (MSBs) shall provide the multiplex location as defined in EDCINF.

(ACMEM) Access management-related extended messages (PI = '79)

The parameter field shall be as defined for ACMM, except that the first 16-bit field is extended to 24 bits. The additional 8 bits (MSBs) shall provide the multiplex location as defined in EDCINF.

8.3 Coding structure

Most of the parameters listed in subclause 8.2 have variable length; some of them are optional and do not need to be transmitted at each cycle of the service-identification data broadcast. So a principle of coding has been adopted, which makes it possible to identify numerous parameter fields, and to encode the length of each independently.

Since the information of the service identification channel is cyclically repeated and with a repetition rate which will ensure rapid acquisition of services by a newly tuned-in receiver, error protection by use of Cyclical Redundancy Check (CRC) code is provided. In addition, Golay forward error correction coding may be used to protect the information and allow correct reception of the first transmission of the service identification data even when the received bit-error ratio is poor. If the SI data carried in packet address zero is transmitted in Golay encoded form, an appropriate SI description shall be also transmitted in the uncoded form.

8.3.1 Commands and parameters

Such a coding structure is currently used for the control procedure for the teletext service (CCITT Recommendation S.62), and is now under study at the ISO and the CCITT as a general coding method for session protocol for teletext, telecopy and videotext. The coding principles are illustrated in figure 49.

Several classes of information (referring to network or types of service information) and two categories of information (in relation to the access time or priority) have been identified. The low-priority information consists of TIME and DCOM (direct commentary). All other information is considered to be of medium priority. (Signalling data in line 625 is considered to be of high priority).

To make the coding as flexible as possible, a particular command for each class and each category (priority) of information has been defined.

The currently defined CI (command identifiers) are:

Network information:	medium priority (M);
Network information:	low priority (L);
Television service description and composition:	medium priority (M);
Television service description and composition:	low priority (L);
Sound services description and composition:	medium priority (M);
Sound services description and composition:	low priority (L);
Teletext services description and composition:	medium priority (M);
Teletext services description and composition:	low priority (L);
Additional service identification description and composition:	medium priority (M);
Additional service identification description and composition:	low priority (L);
Over-air addressing description and composition:	medium priority (M);
Over-air addressing description and composition:	low priority (L).

This list should be extended if new types of service are defined.

Inside each command field, the relevant information shall be included within one or several parameter field(s), each of them being introduced by a PI (parameter identifier). Inside a given type of service command, all the parameter fields describing the same service shall be included within a PG (parameter group), introduced by a PGI (parameter group identifier), which is optional if there is only one service described within the command. The number of bytes of a command field or a parameter field or a parameter group is defined by an LI (length indicator) code.

The LI shall indicate the number of bytes of useful data in the field following the LI. When the value of LI is 0 to 254 it is coded as one byte. If LI is in the range 255 to 65535 then LI is coded as 3 bytes. The first byte is set to 255, indicating that the next 2 bytes give the value of the length, most significant byte first.

Each command is independent and can have a different cycle time.

Each different digital component information shall be introduced by a different parameter identifier. The components listed hereafter have been identified:

- for the television service:
  - analogue television picture;
  - (analogue television sound);
  - digital television original sound;
  - digital television additional sound;
  - digital subtitles;
  - digital news flash sound (see NOTE);
  - digital replacement teletext;
  - digital replacement sound.

- for the radio sound services:

  - digital radio sound;
  - digital radio additional sound;
  - digital news flash sound (see NOTE);
  - digital replacement sound.

- for the teletext services:

  - digital sound for teletext;
  - cyclic teletext;
  - non-cyclic teletext;
  - telesoftware.

- for the data services:

  - digital data (synchronous or asynchronous, coded or uncoded);
  - digital sound for data.

- for all types of services:

  - access related messages (in the case of controlled-access) if they are not included in one component (for example in the BI blocks) or in a digital channel other than the packet address "0" (see Clause 9).

NOTE: A given service cannot have more than one component of this type.

Table 22 gives a coding scheme for CI, PI and PGI showing the significant combinations of PI inside each CI.

Digital replacement sound and teletext components may be provided for a service. These components are intended to be selected automatically by a receiver to replace any components which are subject to conditional access, when access is denied by the receivers CA sub-system or sub-systems. These replacement components should not be available for normal selection by the user.

The only PGs defined at the present time are used to regroup all the data for one particular service or to associate access related parameters with a set of service components. The parameters which may be so associated are ACMM, ACCM, ACMEM and ACCEM. The corresponding PGI is coded '80 and has no specific parameter field. The ordering rules for parameters in commands containing parameter groups are indicated in table 23.

Updating messages introduced by the UPDAT parameter identifier shall be sent at the beginning of a command or parameter group when changes in the information conveyed by the command or parameter group occur, in order to facilitate the recognition and subsequent processing of the changes by the receiver. The UPDAT parameter field consists of a variable-length list of bytes, corresponding to the values of the parameter identifiers (or parameter group identifiers) whose contents have been modified by the change. The absence of a parameter field in an updating message shall indicate a major redefinition of the command or parameter group contents.

The ULIST parameter may be used in association with UPDAT to identify the particular updated parameter or parameters when several identical parameter identifiers exist. If ULIST is not provided and a parameter updating is signalled in UPDAT, the SI processor in the receiver should examine all the parameter fields introduced by the given value of parameter identifier (if several exist). If an UPDAT parameter field indicates the modification of a parameter group, then the particular parameter group will be identifiable by a further UPDAT within the modified parameter group.

Updating messages shall be broadcast for about two seconds to enable them to be acquired by all the decoders concerned. It is not necessary to have a precise time relationship between the programme-identification data and the effective change in the service itself, because it is intended that this event should be signalled within the individual digital components. However, the service change should nominally coincide with the end of the period over which the updating messages are broadcast. The SI channel may use parameters known only to a set of decoders, for instance DATV description. If a parameter has an unknown PI value, the PI and its data field should be ignored and the next parameter is processed.

8.3.2 SI Data channel

Packet address "0" shall be reserved for the service-identification data channel. This data channel uses the same basic transport mechanism as the digital sound with supplements in order to provide delimitation for messages and an error detection/correction capability.

The delimitation for messages is provided by the definition of a transport entity called a data group. Each group shall contain a data group header (DGH), a sequence of commands and a Data Group Suffix (DGS). The incorporation of several commands in the same data group, called "blocking", contributes to improve the efficiency of the transport layer, especially in the case of short command messages.

The DGH shall contain five parameters carrying information about the content or the transport mechanism of the data group; data group type (TG), data group continuity (C), data group repetition (R), number of packets in the data group (S1, S2), and number of bytes in the last packet (F1, F2). In addition there is a four bit field (N). The DGH shall be Hamming coded (see tables 23 and 24).

Up to 16 data groups with packet address '0' can coexist independently, distinguished by the TG parameter in the data group header.

A data group may be carried by a number of packets, the first and succeeding packets shall be indicated by the packet type byte (PT) in the corresponding packet header (see tables 23 and 24).

The information contained within a data group may be protected by a Cyclic Redundancy Check (CRC) or a CRC combined with forward error correction using Golay (24,12) encoding. Where Golay encoding is used it shall apply to all the data within the data group, with each Golay word comprising 12 bits of SI data with 11 Golay protection bits and one parity bit.

The data group size shall be expressed as the number of packets (S1, S2) and number of bytes in the last packet (F1, F2), excluding the packet suffix (where used) as defined below, but including the data group suffix. Where Golay encoding is used the byte count shall refer to the SI data before Golay encoding. In the case where the data group is wholly contained within a single packet, then the (F1, F2) byte count also includes the data group header in addition to the data itself and any data group suffix. S1 and F1 are the most significant bytes.

When only data group error detection (using the CRC of the DGS) is used, a restriction which may reduce receiver storage requirements shall be applied to the lengths of data groups. In the subframes characterized by TDMCID values '01 and '02 and when in line 625, bit 2 (bm) of the multiplex and video scrambling control group (MVSCG) is set to "1", a data group size limitation to five packets in length is introduced, and data group "0" is further restricted to two packets. When the capacity required exceeds the available length of one data group, a method for forward and reverse linking between data groups is provided by special commands, in conjunction with the parameter link pointer (see subclause 8.2.4, LKPNT - PI value '70). These commands are defined as the extension command (CI value 'FF) and the continuation command (CI value '00).

Error detection is performed at the packet layer by means of the use of a continuity index (I) in the packet header to detect the loss of data packets due to transmission errors, and either a Packet Suffix (PS) to detect errors in the packet (see table 23) or Golay encoding of the data.

The value of the continuity index I shall be incremented by one (modulo 4) each time that a packet is transmitted, unless that packet is a synchronising packet. If the packet is a synchronising packet then its continuity index shall be reset to an initial value, i.e. 00.

The packet suffix has a length of 2 bytes and shall be a CRC generated by the polynomial:

$$x^{16} + x^{12} + x^5 + 1$$

The message to be sent shall be composed in the sending order of bits.

m_{8n-1} to m_0, followed by check bits r_{15} to r_0 in the same order. The check bits are such as the polynomial:

$$m_{8n-1}\, x^{8n+15} + m_{8n-2}x^{8n+14} + \ldots + m_0x^{16} + r_{15}x^{15} + \ldots + r_1x + r_0$$

is a multiple (modulo 2) of the polynomial:

$$x^{16} + x^{12} + x^5 + 1$$

The message checked by the packet suffix consists of the useful part of the data block, i.e. the 88 bytes following the PT byte.

Error correction can be achieved either by decoding the Golay protected data or by processing several repetitions of data groups of the same data group type, the number of which remaining is signalled by a data group repetition indicator (R) contained in the data group header (see table 23).

When the identical information is repeated several times in consecutive data groups of the same type, the data group repetition byte shall be set and decremented at each transmission until it is zero at the last. When the repetition continues indefinitely, the repetition byte shall be set to carry the message bits 1111 (implying 15 repetitions), after which a change (possibly involving a discontinuity) to a sequence decrementing to 0000 indicates the end of this period of repetition.

In addition, the loss of a data group (due to the loss of its first packet) can be detected by means of a data group continuity index (C). The value of this byte shall be incremented by one (modulo 16), each time a data group of the same type (TG) is transmitted.

When used, (24,12) Golay encoding shall be applied to each successive 12-bit segment of data by appending 11 error check-bits. The code is defined by the generator polynomial:

$$G(x) = x^{11} + x^{10} + x^6 + x^5 + x^4 + x^2 + 1$$

The resulting 23-bit code shall be appended by a single parity bit giving overall odd parity for the 24 bit block.

A Data Group Suffix (DGS) is added to enable any residual errors to be detected (see table 23 and table 24). The data group suffix shall be a CRC generated by the same polynomial as used for the packet suffix.

The message checked by the data group suffix consists of the useful part of the data group following the data group header.

Bit b2 in the last four bit field (N) of the data group header shall be set to "1" in order to provide compatibility with other applications of the same transport mechanism. Bits b4, b6 and b8 are reserved for future use.

The transmission coding described above is summarized in table 23 and table 24, which are arranged in accordance with the layer architecture. All the bytes of the data group header shall be Hamming protected according to the Hamming code table given in table 21.

All the information of medium priority (M) concerning any particular service shall be contained in LISTX in data group type '0' together with only one other data group type.

8.4 Rules for multiplexing

Each data group carrying information of medium priority (M) (concerning the network and any particular service) shall be transmitted (on average) every sixth television frame and with never more than twelve television frames between such data groups.

Each data group carrying information of low priority (L) (commentaries and time) shall be transmitted (on average) every 25th television frame and with never more than thirty-seven television frames between such data groups.

Data groups may be transmitted in any order but all the packets of one occurrence of one data group shall be transmitted in sequence before the next data group is transmitted. However, during transmission of a data group, packets with different addresses can be interleaved freely with packets having address "0".

Two occurrences of the same data group type shall always be separated by at least 50 ms (measured between the end of the last packet of the first occurrence and the beginning of the first packet of the second occurrence). However, the last packet of a data group of one type may be followed immediately by the first packet of another type.

When unused capacity is left by digital components of the services, it is desirable to increase the repetition rate for the data groups of channel 0, while complying with the foregoing rules.

8.5 Use of error protection levels

Two levels of error protection are specified: one with a packet suffix providing error detection and one with a forward error correction code (Golay). The following rules shall apply for the use of one or two levels of error protection simultaneously:

- any information may optionally be duplicated and transmitted using the two levels of error protection;

- mandatory information allowing the network identification shall be present in the non-Golay protected form, even if all the services within the channel are subject to conditional access;

- mandatory information which is necessary to identify and locate the services shall be present in the non-Golay protected form;

- the use of the Golay form is not in any case mandatory.

NOTE: The table given on the next pages shows the validity of parameters within commands in the SI service and ASI service (indicated by table entry) and in the ASI service alone (indicated by table entry A). The other indices note the following:

1) shortened form of SREF with index value only;
2) high-priority limited-length network commentary;
3) must be confirmed by further studies.

As further commands and parameters are added in future extensions and enhancements of the features of the MAC/packet family this table will of necessity become incomplete. Some coding space may be allocated for particular applications without full definition being given in the specification. All coordination of code allocations is undertaken by the EBU Technical Department who will provide an updated table on request.

Table 22: Coding of commands and parameter identifiers (CI, GPI, PI)

	CI	Continuation Command	Network Command		TV Command		Radio sound Command		Teletext Command		Over-air Addressing Comm'd		Additional SI Command		Extension Command
			Priority Medium	Priority Low	Priority Medium	Priority Low	Priority Medium	Priority Low	Priority Medium	Priority Low	Priority Medium	Priority Low	Priority Medium	Priority Low	
PARAMETER NAME and (MNEMONIC)	PI	'00	'10	'11	'90	'91	'A0	'A1	'B0	'B1	'C0	'C1	'D0	'D1	'FF
Update (UPDAT)	'00		*	*	*	*	*	*	*	*	*	*	*	*	
Update list (ULIST)	'01		*	*	*	*	*	*	*	*	*	*	*	*	
(reserved code values)	'02 to '05														
Network origin (NWO)	'10		*												
Other network origin (ONO)	'11		A												
Network name (NWNAME)	'14		*												
Modulation parameters (MODPRM)	'16		*												
Multiplex identity (MID)	'17		A	A	A	A	A	A	A	A	A	A	A	A	
List of index values (LISTX)	'18		*												
General Purpose Data Index (DATX)	'19		*												
Local time (TIME)	'20			*											
Service reference (SREF)	'40				*	* 1	*	* 1	*	* 1	*	* 1	*	* 1	
Programme item reference (PREF)	'48				*		*		*						
Video programming system (VPS)	'49				*										
Next programme item ref; (NXPREF)	'4A				*		*		*						
(reserved code values)	'4B to '4D				*		*		*						
Direct commentary (DCOM)	'60		* 2	*		*		*		*		*		*	
Commentary description (COMD)	'61		*		*		*		*		*		*		
Local time description (TIMD)	'62		*												
Link pointer (LKPNT)	'70	*													*
Access management related messages (ACMM)	'78				*		*		*		*				
Access management related extended messages (ACMEM)	'79				*		*		*		*				

Table 22: Coding of commands and parameter identifiers (CI, GPI, PI) (continued)

CI → / PI ↓		Contin-uation Command	Network Command Priority		TV Command Priority		Radio sound Command Priority		Teletext Command Priority		Over-air Addressing Comm'd Priority		Additional SI Command Priority		Exten-sion Command
			Medium	Low	Medium	Low	Medium	Low	Medium	Low	Medium	Low	Medium	Low	
PARAMETER NAME and (MNEMONIC)	PI	'00	'10	'11	'90	'91	'A0	'A1	'B0	'B1	'C0	'C1	'D0	'D1	'FF
Parameter group identifier (PGI)	'80				*	*	*	*	*	*	*	*			
Access checking related messages (ACCM)	'88				*		*		*						
Access checking related extended messages (ACCEM)	'89				A		A		A						
Analogue TV picture (VCONF)	'90				*										
DIGITAL COMPONENT INFORMATION (DCINFs)															
TV original sound	'A4				*										
TV additional sound	'A5				*										
Radio sound	'A8						*								
Radio additional sound	'A9						*								
Replacement sound	'AA				*		*		*						
News flash sound	'AB				*		*								
Sound for teletext	'AC								*						
Teletext in the packet multiplex	'B0				*				*						
Teletext Subtitles (packets)	'B1				*										
Replacement teletext (packets)	'B2				*				*						
Programme delivery control (packets)	'B3				*										
Telesoftware (packets)	'C0								* 3						
(reserved code value)	'C1				*		*		*						
(reserved code value)	'98				*										

Table 22: Coding of commands and parameter identifiers (CI, GPI, PI) (concluded)

PARAMETER NAME and (MNEMONIC)	CI / PI	Continuation Command	Network Command Priority Medium	Network Command Priority Low	TV Command Priority Medium	TV Command Priority Low	Radio sound Command Priority Medium	Radio sound Command Priority Low	Teletext Command Priority Medium	Teletext Command Priority Low	Over-air Addressing Comm'd Priority Medium	Over-air Addressing Comm'd Priority Low	Additional SI Command Priority Medium	Additional SI Command Priority Low	Extension Command
		'00	'10	'11	'90	'91	'A0	'A1	'B0	'B1	'C0	'C1	'D0	'D1	'FF
EXTENDED DIGITAL COMPONENT INFORMATION (DCINFs)															
Additional Service Identification	'C4														
Radio sound	'C8						A						*	*	
Radio additional sound	'C9						A								
Replacement sound	'CA				A		A								
News flash sound	'CB				A		A								
Sound for teletext	'CC								A						
Teletext in the packet multiplex	'D0								A						
Teletext Subtitles (packets)	'D1				A										
Replacement teletext (packets)	'D2				A				A						
Programme delivery control (packets)	'D3				A										
Telesoftware (packets)	'E0								A 3						
DIGITAL COMPONENT INFORMATION (DCINFs) for TELETEXT IN THE VERTICAL BLANKING INTERVAL															
CCIR System B - cyclic	'F0				*[4]		*[4]		*		*[4]				
CCIR System B - non-cyclic	'F1								*						
CCIR System A - cyclic	'F4								*						
CCIR System A - non-cyclic	'F5								*						
CCIR System B - Subtitles	'F8				*										
CCIR System A - Subtitles	'FC				*										
DIGITAL COMPONENT INFORMATION (DCINFs) for DATV in the DATV/data multiplex															
DATVD	'FE				*										

Table 23: Transmission coding for service-identification within the sound/data packet multiplex using CRC error detection and data correction by repetition

LAYER V: SERVICE CONTROL (service-identification data sequence of commands)

CI 1	LI 1	service-identification data 1	/	CI 2	LI 2	service-identification data 2	/	CI 3	LI 3	service-identification data 3

Within a command sequence, CI codes are transmitted in an increasing order, i.e. each CI value is equal to or greater than the preceding one.
The extremities of a command are shown by "/".

LAYER IV: DATA TRANSMISSION CONTROL (data group)

data group header	+	sequence of commands	+ DGS

Data group header:

- TG: data group type
- C : data group continuity
- R : data group repetition
- S1, S2: number of packets carrying the data group
- F1, F2: number of data group bytes in the last packet
- N : data group suffix indicator configuration $b_2\ b_4\ b_6\ b_8$ = 1XXX

(8,4) A total of 8 bytes, each in Hamming code (see Table 4 of Part 3)

DGS: data group suffix

LAYER III: DATA MULTIPLEXING (packets)

packet 1 header	: data group header	+ sequence	: PS
packet 2 header	: of	/	: PS
packet 3 header	: commands	/	: PS
packet 4 header	: + DGS :		: PS

PS: packet suffix

Packet header:

PA	Address field	10 bit
I	Continuity index	2 bit
	Protection	11 bit
		23 bit
PT	Packet type	Code (8,2)*

The extremities of a command are shown by ",".
The extremities of a sequence of command data (data group) are shown by "+".
The extremities of a data group (useful part of data blocks) are shown by ":".

---

* Packet type coding (hexadecimal, LSB first):
'F8 = presence of synch. packet (i.e. Packet 1 of a sequence of packets)
'C7 : absence of sync. packet (subsequent packet in sequence)

Table 24: Transmission coding for service-identification within the sound/data packet multiplex using Golay (24,12) forward error correction coding

LAYER V: SERVICE CONTROL (service-identification data sequence of commands)

CI	LI	service-identification data	/ CI	LI	service-identification data	/ CI	LI	service-identification data
1		1	/ 2		2	/ 3		3

Within a command sequence, CI codes are transmitted in an increasing order, i.e. each CI value is equal to or greater than the preceding one.
The extremities of a command are shown by "/".

LAYER IV: DATA TRANSMISSION CONTROL (data group)

data group header	+	sequence of commands	+ DGS

Data group header:

- TG: data group type
- C : data group continuity
- R : data group repetition
- S1, S2: number of packets carrying the data group
- F1, F2: number of data group bytes in the last packet
- N : data group suffix indicator configuration $b_2\ b_4\ b_6\ b_8$ = 1XXX

(8,4) A total of 8 bytes, each in Hamming code (see Table 4 of Part 3) before Golay encoding.

DGS: data group suffix

LAYER III: DATA MULTIPLEXING (packets)

packet 1 header	:	data group header	+ sequence	:	PS
packet 2 header	:	of	/	:	PS
packet 3 header	:	commands	/	:	PS
packet 4 header	:	+ DGS :		:	PS

PS: packet suffix

Packet header:

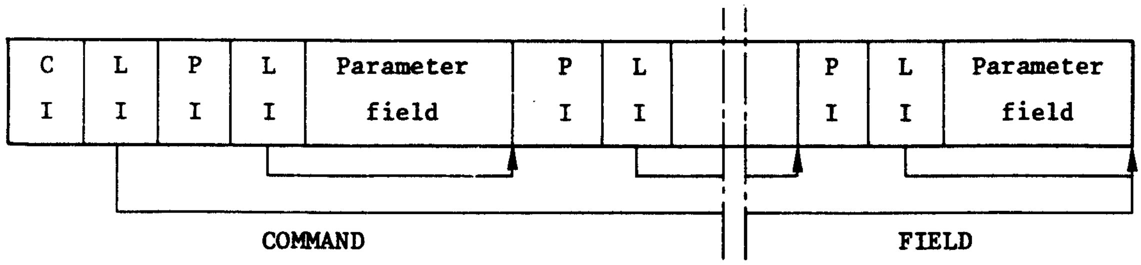
PA	Address field		10 bit
I	Continuity index		2 bit
	Protection		11 bit
			23 bit
PT	Packet type	Code	(8,2)*

The extremities of a command are shown by "/".
The extremities of a sequence of command data (data group) are shown by "+".
The extremities of a data group (useful part of data blocks) are shown by ":".

---

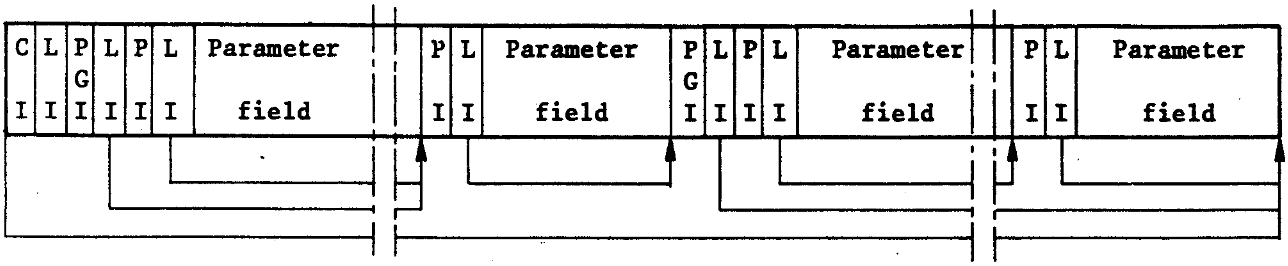
* Packet type coding (hexadecimal, LSB first):
'00 : presence of synch. packet (i.e. Packet 1 of a sequence of packets)
'3F : absence of sync. packet (subsequent packet in sequence)

a) without PG (Parameter Group):



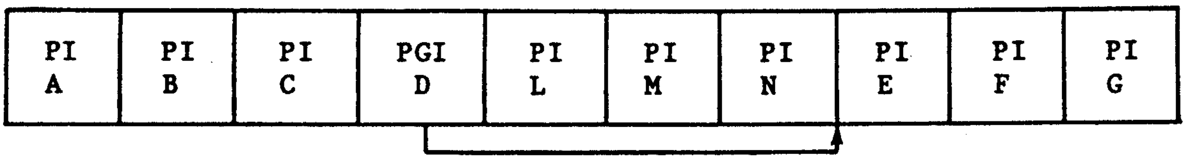
- within the command field, the PI codes are transmitted in increasing order.

b) with PG (Parameter Group):



- within the command field, parameter groups can be mixed with isolated parameters;
- within the command field, PGI codes and PI codes of isolated parameters are transmitted in an increasing order;
- within a parameter group, PI codes are transmitted in an increasing order.

c) showing PI/PGI value order value in general:



1) A to G and L, M and N denote PI values;
2) LI's and parameter fields are omitted for clarity;
3) The numerical values of the Pi's ascend from A to G and independently from L to N i.e.:
   $A < B < C < D < E < F < G$ and $L < M < N$
4) L, M and N may have lower values than A, B, and C or higher values than E, F and G.

Legend:

CI = Command Identifier;
LI = Length Indicator;
PI = Parameter Identifier;
PGI = Parameter Group Identifier.

Figure 49: Coding structure of a command

9 Specification of the conditional access system

9.1 Introduction

This subclause contains information on the conditional-access system. That is, the system which may be used to ensure that services or service components are available only to those who have fulfilled certain conditions, such as payment. Subclause 9.2 contains a general description of the conditional-access system, and describes the basic principles of the system. The conditional-access system can, for convenience, be considered in two parts: the scrambling system and the encryption system. The scrambling system is broadly the part which processes the signals carrying the service or the service components, and the encryption system is broadly the part which processes the key signals needed to unlock the service or the service components. Subclause 9.3 contains a description of the scrambling system. Subclause 9.4 contains a description of a conceptual internal interface across which key and control signals pass in the receiver. Subclause 9.5 contains a description of the transport mechanisms etc., for appropriate key and control signals ('conditional-access data').

The internal format of the conditional-access data signals is not part of this specification. The allocation of the CA-byte as specified in subclause 8.2.3 is given in EN 50 094.

As far as the terminology used in Clause 9 is concerned, the reader should note the following equivalencies:

- ECM (entitlement checking message) = SMM (service management message);

- EMM (entitlement managing message) = CMM (customer management message).

9.2 General description of the conditional-access system

9.2.1 The scrambling and encryption systems

A block diagram of the overall conditional-access system is given in figure 50. Certain functions (such as demultiplexing) are omitted for clarity, and figure 50 is intended to give only an overall picture of the system and does not describe the details of all access-control systems. In particular, the key hierarchy given in figure 50 is an example. The source component in figure 50 may be a picture, sound, or data signal which the service operator wishes to make conditionally available. This is rendered unintelligible by processing in the scrambler, making use of a scrambling sequence which is a continuously changing pseudo-random binary number. In the receiver, the reciprocal process can occur via the descrambling sequence generator, descrambling sequence (which is the scrambling sequence regenerated) and the descrambler.

The scrambling sequence generator is a pseudo-random binary sequence generator with a very long cycle time. Its output is made more unpredictable, by using a control word and a cyclic 8-bit frame count, FCNT, which has a cycle of 256 TV frames, or approximately 10 seconds. The descrambling system must be synchronized at the source and receiver. The scrambled source component, the frame count and the synchronization are obtained from signals transmitted in the same channel.

The service operator is able to choose whether to send the signal unscrambled (in clear form) or scrambled. If it is to be scrambled, he has two choices. He can either use a local control word, which is invariable and also stored in the receiver, or a regenerated control word derived from the encryption system. Whether or not the users receiver is able to generate internally the correct control word in this case depends on whether the user has met required conditions for access to the service. The control word in the receiver is supplied (when the local control word is not used) by the control word generator or by the ECM decrypter, corresponding to two methods for provision of control words (see figure 50). The choice of method for a particular service will be made by the service operator, as each offers different advantages which may be more or less important in particular cases.

The first method implies that the Control Word (CW) is generated at the transmit side and sent to the receiver as an addressed cryptogram. The cryptogram, associated with programme data (P), shall be encrypted by the Service Key (SK) (see figure 50). The control word shall be changed every 256 TV frames (approximately 10 seconds), but transmitted at approximately 0,5 second intervals to allow rapid descrambling for receivers which tune to the channel. The cryptogram SK(P,CW) is sent in an entitlement Checking Message (ECM), which shall be sent in the same channel as the service itself.

The second method implies that the control word is generated in the same way at both transmit and receive sides. This shall be obtained by encrypting a known number with the service key. The result is the control word, and will only be available to those who have got the service key. To ensure that the control word changes frequently, the known number shall be derived from the 20 bit CAFCNT parameter which is sent in the same channel as the service. This 20-bit CAFCNT increments every 256 TV frames in a cycle which lasts about four months.

Both methods ensure that new control words are fed to the descrambler at approximately 10 second intervals, provided that the service key is available at the receiver, and that it is entitled to generate the control word.

The service key shall be common to all customers for a particular service (or group of services) but changes less frequently than the control word. The intervals between the changes of the service key allow time for the next service key to be transmitted to each customer by means of an over-air addressing process and/or by other means. The service key together with customer entitlements (E) shall be encrypted with each customers Distribution Key (DK) or Unique Key (UK). These cryptograms shall be sent in Entitlement Management Messages (EMMs).

The time required to cycle around the audience base can be substantially reduced by incorporating a number of different customers entitlements (E) within the same shared or collective message, using a shared or collective distribution key and a shared or collective customer address. The mechanism allows a number of customers to be simultaneously updated with entitlements for subscriptions or pay-per-view for several programmes.

Such messages are sent over air in shared or collective EMMs to shared or collective customer addresses. The EMMs may be sent in the same channel as the service itself, or in some cases also via other channels.

Addresses and distribution keys for each shared or collective group shall be sent in unique entitlement management messages, which shall be encrypted with the unique key of the customer and addressed to his unique customer address. These messages may be sent in channel, via other channels, or via telecommunications networks, smart card or other means. For clarity the customer addresses are not shown in figure 50.

Customer addresses, distribution keys and entitlements are associated with a particular service group, and must be stored in the access-control system. The access-control system has to store one set of parameters for each service group it is authorized to receive. A service group is associated with a network (as defined in Clause 8) on a full or part-time basis, or several networks at a time.

The entitlement management messages can also be used to distribute individual or group messages to a unique customer or a group of customers. Such messages can be used to prepare replacement states for a series of separate groups or to present special management messages. Such replacement states may for example be a special teletext page which can replace the vision signal in case of 'black out' of the corresponding user group. Replacement of other services (e.g. teletext, sound ..) is also possible. These messages may also provide fingerprinting configurations, i.e. insertion of the customers 'signature' on selected frames of the vision signal.

9.2.2 Example of functional operation

The D2-HDMAC channel may contain one or more services (e.g. television, radio, or teletext services) with service components (e.g. TV = video + DATV + main sound + additional sounds in several languages + subtitles in several languages).

A service component or a group of service components may be scrambled by individual control word(s). Alternatively, all components within a service or a group of services may be scrambled by the same control word. The way the control words may be made available to the customers determines the access-control modes of the service or service components.

As a function of time, the programme which is broadcast within a service can be split in several programme elements, each programme element having its own access-control modes.

For each programme element, the allocation of control words to the various components of a service, and the selection of the access-control modes are made by the service operator (programme provider), at the broadcasting side. The customer can get access to a programme element by one of several access-control modes, provided he has got the appropriate entitlements, sent by the entitlement provider (management centre) of the service operator.

Several access-control methods are based on the broadcasting of control words within ECMs. Other access-control methods may be based on the local generation of control words, without need for ECMs. Such methods are mainly suited for static access control, without rapid changes.

The different access-control methods for a programme element may be:

- subscription, in which customers purchase a programme for a given period (e.g. a month). The service operator may use the various subscription methods that are specified and implemented in the different access-control systems;

- pay-per-view, in which customers purchase the programme elements when viewed. In this case, there are three possible types of purchase:

    - pre-booked pay-per-view per programme: customers purchase one programme element, or a series of programme elements in advance from the entitlement provider;

    - impulse pay-per-view per programme: each programme element has a price which is broadcast together with the programme element itself. Customers may buy the programme when broadcast, provided that they have a sufficient credit in the access-control system. They may obtain this credit via the entitlement provider in advance. When the programme is bought, the credit is reduced by the price of the programme;

    - impulse pay-per-view per time: customers are also assigned a credit as explained above. When the programme is bought, the credit is debited on a time basis (price per time unit).

Additional functions can be included in the conditional-access system, such as:

- Group control: the entitlement provider may allocate customers to group addresses, which allows addressing them in accordance with geographical and subject groups. This group control may be used for:

    - black-out of groups (black-out is active for the addressed groups or for all others but the addressed groups), and replacement of the service for blacked-out groups. Black-out implies that the access to the service is denied, replacement implies that the original service components are replaced by other components, as described in the separate conditional-access systems specifications;

    - fingerprinting control, which allows indication of the customer unique address or a specific text on the screen for a specific number of frames;

- Maturity rating: each programme element may be associated with a maturity level, that indicates which age group it is intended for:

    - homing/parallel transfer of EMMs via several networks: each network may broadcast EMMs for other networks, in addition to those for its own services.

Communication with the customer may include:

- transmission of text messages to addressed customers or to addressed groups of customers;
- local control of the access control system, using a "parental code" (PIN code);
- use of a man-machine dialogue.

9.3 Sound, data, DATV and picture scrambling

9.3.1 Introduction

All services not subject to subscription or other special payment are considered to be in the free-access mode. Scrambling of sound, data and picture components shall be used for the controlled-access mode, but may be used as an option for the free-access mode. For those free-access services which are subject to scrambling prior to transmission, the relevant control word information (the "local control word") for descrambling the various components is available in the receiver (see subclause 9.3.3).

In the free-access mode of the conditional-access system, the vision signal, when present, may either be non-scrambled or scrambled at the option of the service originator. The scrambling process of a picture component is described in subclause 5.7.

The digital sound and data signals are subjected first to scrambling for both modes of the conditional-access system and then to bit interleaving and transmission scrambling (for spectrum shaping purposes). The conditional access scrambling process is described in subclause 9.3.2.

9.3.2 Description of the sound and data scrambling process

To scramble sound and data, a Pseudo-Random Binary Sequence (PRBS) shall be added modulo 2 to the useful data of the sound or data service packets.

The sound encoding blocks shall be scrambled after being subjected to error protection; this means that, in the receiver, error detection or correction takes place after descrambling;

The packet headers, packet type bytes (PT), interpretation blocks (BI) and service identification data (SI) shall not be scrambled;

The derivation of the scrambling sequence for the DATV/data burst shall use the same method as that for the subframe in the LBI. This means that the scrambling sequence is restarted at the beginning of each DATV/data multiplex (FBI1 and FBI2) with an identical sequence. Scrambling for the DATV/data multiplex is so arranged that a given service component in the n-th packet position of each TDM component would be scrambled with the same scrambling sequence as if it occupied the n-th packet position of the LBI.

When scrambling is used, the digital sound and data signals shall be subjected first to scrambling for the conditional access system and then to bit interleaving and transmission scrambling (for spectrum shaping purposes) as described in Clause 4.

9.3.3 Derivation of the descrambling sequences

The descrambling sequences are provided by a pseudo-random binary sequence (PRBS) system. A block diagram for this is given in figure 51. This would be the necessary configuration for the basic television service (vision and associated sound components). This configuration is related to one set of PRBS generator(s) with combinational logic for video and one set for the sound/data packets. One additional set with an independent control word is required for each additional sound or data service with independent scrambling.

All services may be subject to scrambling but, in the case of the free-access scrambled mode, the descrambled control word is provided locally in the receiver.

For descrambling, a 60-bit control word is required. For the free-access scrambled mode this is a word in which all 60 bits are set to logic 'one' and which is stored in the receiver. In the controlled-access mode, two 60-bit control words are provided from the access-control system.

The block diagram given in figure 51 shows three PRBS generators. These are described in figures 52a, 53a and 54a. A more detailed diagram of each of these PRBS generators is given in figures 55b, 56b and 57b.

The PRBS system requires an 8-bit frame count, and the location of this information within line 625, and its specification, are given in subclause 4.5.4 (code FCNT). As shown in figure 51, the initialization words of the PRBS generators are obtained by combining the control words with the 8-bit frame count FCNT. The PRBS generators are synchronized with the frame count as described in the following subclause.

9.3.4 Synchronization of the descrambling sequences

Consider two consecutive frames, F_n and F_{n+1}. During line 625 at the end of F_n a new value of frame count (FCNT equal to n) is received. This value n is combined with CW1 and CW2 (figure 51) to generate new values of IW1 and IW2, which are loaded into PRBS generator 1 and PRBS generator 2, respectively. It is important to note that the first 61 bits generated by PRBS generator 1, after it is thus loaded, are used by PRBS generator 3 to descramble the first packet in each subframe received during frame F_{n+1}. Likewise, the first 16 bits generated by PRBS generator 2 after it is loaded are used to descramble line 1 of frame F_{n+1}.

A new control word is received by the PRBS system every 256 frames. The new control word becomes the valid control word when FCNT = 0, i.e. at the beginning of the frame F1.

The following diagram summarises the relationships between the frame number, FCNT, CAFCNT, and the derivation of the Initialization Word (IW).

	Frame 255	Frame 256	Frame 257
FCNT &	255	000	001
CAFCNT	0	1	1
in line 625			
IW derivation	(CW_x & FCNT 254)	(CW_x & FCNT 255)	(CW_{x+1} & FCNT 0)

The first bit of the sequence generated by each PRBS generator is the value present at the output after it has been loaded and before any shift operations have taken place.

9.4 The conditional-access interface

9.4.1 Introduction

Reception of access controlled transmissions requires a receiver with an Access-Control System (ACS). The ACS may consist of one or more conditional-access (CA) sub-systems, each prepared for control of one or more services. The CA sub-systems may be of one or more types, and each controlled by one or more operating/controlling organization(s). The CA sub-system(s) may be buried in the receiver, detachable, or contain both a buried and a detachable part.

The CA interface between the ACS and the basic receiver (figure 58) allows that:

a) different operating/controlling organizations may control services within the same channel;

b) different revenue collection systems may be used;

c) three identified methods for delivery of the entitlement management message may be used; over-air addressing, post and user action, telecommunications networks.

9.4.2 Functions within the access-control system

The precise implementation of the access-control system depends on the operating organizations' choice for the various CA sub-systems. However, the following transactions take place with the CA sub-system(s) that is(are) contained with the ACS:

a) initialization (What are you? Who are you?);

b) entitlement checking functions;

c) entitlement management functions;

d) control functions.

Each function is carried out by the execution of a transaction with the sub-system (see subclause 9.4.3), which may be detachable and which is always the slave. As a result of a transaction, the sub-system delivers information (computation results, stored data) and/or modifies its content (data storage, event memorization).

A transaction is composed of successive phases:

- activation of the sub-system;

- reset, where appropriate;

- processing of one or more instructions contained in the sub-system;

- de-activation of the sub-system.

These transactions take place, through the CA interface, between the basic receiver and the access-control system and subsequently between the common part of the access-control system and the sub-system.

9.4.3 Basic functions of the CA interface

9.4.3.1 General diagram

$\boxed{A} \Rightarrow$ CA interface $\Rightarrow \boxed{B}$

A: main receiver circuits;

B: access-control system (possibly including a detachable sub-system).

9.4.3.2 Role of the CA interface

Through the interface bus, the receiver first initializes the access-control system, and is then instructed to collect data (usually through the network but sometimes from, say, a user key-pad). The access-control system passes control words for the receiver descramblers and messages for display or interaction with the user. Displayable text messages arriving via the sound/data multiplex will use the SI system, the teletext system or ECMs/EMMs.

9.4.3.3 Practical interface specification

The following is an outline specification for a universal interface between a receiver and an access-control system. The interface is located inside the receiving unit as a whole, as shown in figure 58, so as to allow the use of any form of access-control system.

This ETS does not attempt to specify the physical form of the interface nor to propose any standard. However, one of several forms of available bus system may be chosen, and the manner of its operation determined by receiver manufacturers. Manufacturers are encouraged to define and use a unique standard.

Similar remarks apply to the form and physical connection of detachable sub-systems themselves, which are matters for agreement between service operating organizations and the receiver manufacturing industry. No preference for any particular form of sub-system is stated or implied in this ETS.

9.4.3.4 Data crossing the CA interface

Table 25 shows the data which can be handled by the interface. A specific conditional-access system may use a sub-set of the messages listed.

9.4.3.5 Structure of information exchanged across the interface

The main traffic (illustrated in table 25) is devoted to entitlement checking. Indeed, the main role of the Access-Control System (ACS) is to check entitlements presented by the user. The ACS is also able to manage entitlements and to provide information on their status. A fuller description of the message types is given in subclause 9.5.

a) Entitlement checking:

- Rx⇒ACS initialization (on service change), SI information (e.g. ACCM, LISTX);
- ACS⇒Rx ready, CA-byte plus optional parameters, plus request for CAFCNT if needed;
- Rx⇒ACS contents of selected packets and/or CAFCNT;
- ACS⇒Rx control words automatically repeated;
- In special cases (e.g. pay-per-view):
    - ACS⇒Rx display data, configuration data;
    - Rx⇒ACS key-pad entry.

b) Entitlement management:

1) Rx⇒ACS initialization (e.g. ACMM parameter);
2) ACS⇒Rx addresses for the demultiplexer (see NOTE 1);
3) Rx⇒ACS contents of selected packets (see NOTE 2);
4) ACS⇒Rx display data, configuration data, external data (e.g. to modem) (see NOTE 3);
5) Rx⇒ACS key-pad entry (see NOTE 3).

NOTE 1: May also occur after 3).

NOTE 2: Occurs whenever appropriate packets arrive in multiplex.

NOTE 3: Depends on the content of the EMM packets.

c) Control (e.g. entitlement status, man/machine dialogue):

- Rx⇒ACS initialization;
- Rx⇒ACS key-pad entry;
- ACS⇒Rx requests for selected packets and teletext pages;
- Rx⇒ACS contents of selected packets and teletext pages;
- ACS⇒Rx messages for display;
- Message dialogue repeated as required.

Table 25: Main traffic exchanged across the CA interface

General list of messages crossing the CA interface	Data signals crossing the CA interface
From receiver to ACS:	From receiver to ACS:
Conditional-access frame count	CAFCNT (from line 625)
Contents of selected packets	EMM-U data packets EMM-S data packets EMM-C data packets EMM-G data packets ECM data packets
SI information	Part of LISTX (from packet "0") Part of ACCM (from packet "0") Part of ACMM (from packet "0")
Man/machine dialogue information	Key-pad data
Initialization	Initialization
From ACS to receiver:	From ACS to receiver:
Control words	Odd and even control words
Customer addresses	Unique customer address Shared customer address Collective customer address Group customer addresses
Configuration data	Configuration data
Man/machine dialogue information	Display data
Data to the receiver, extracted from the EMMs	Display data, configuration or external data (e.g. modem)
Initialization	CA-byte, plus optional parameters

9.5 Conditional-access data

9.5.1 Introduction

The CA interface can be considered to define, in the receiver, the frontier between the basic receiver and the access control system (see figure 58).

The basic receiver includes the functions associated with data demultiplexing, packet selection and the user interface.

The access-control system receives messages (ECMs and EMMs), data from the SI system, and data from the user interface. It delivers control words and other data to the receiver, which are used for correct descrambling and operation of the receiver, and to generate the necessary indications to the user.

Messages are sent via the packet data multiplex for use by the access-control system. This implies that the access-control system has to present its customer addresses to the basic receiver in order to allow selection of the messages addressed to it.

The Entitlement Checking Messages (ECMs) (which provide the function of service management) apply to all customers, but may be specific to a service, service components or group of services. The ECMs can only be sent over air. Their presence and addresses in the data multiplex are indicated by the ACCM parameter of the SI (see subclause 8.2.3).

For selection of keys and entitlements in the access-control system, a set of parameters is required from the SI (see subclause 8.2) and the ECM packet, or line 625 when no ECM is used. For operation without ECMs, both the LISTX and ACCM parameters and the CAFCNT parameter in line 625 are required. The LISTX and DCINF parameters are also required to identify special conditional-access service components, such as replacement teletext.

The entitlement management messages, EMMs (which provide the function of customer management), are used to provide to each customer the means to obtain the services which he is authorized to receive. EMMs can be sent as an over-air addressing service, as indicated by the LISTX parameter in the SI (see subclause 8.2.1). Four different types of EMMs (EMM-U (Unique), EMM-S (Shared), EMM-C (Collective) and EMM-G (General), see subclause 9.5.2) which may contain a customer address, are specified. These addresses together with the ACMM parameter in the SI (see subclause 8.2.2) indicate the sub-system in the access control system for which the EMMs are intended.

The EMMs can also, optionally, be sent via telecommunications networks or other means.

Local control messages (requests) may be generated at the User Interface, which is part of the basic receiver.

The access control system may provide configuration data to the receiver, for use in connection with replacement or fingerprinting functions, or user messages.

9.5.2 Message formats

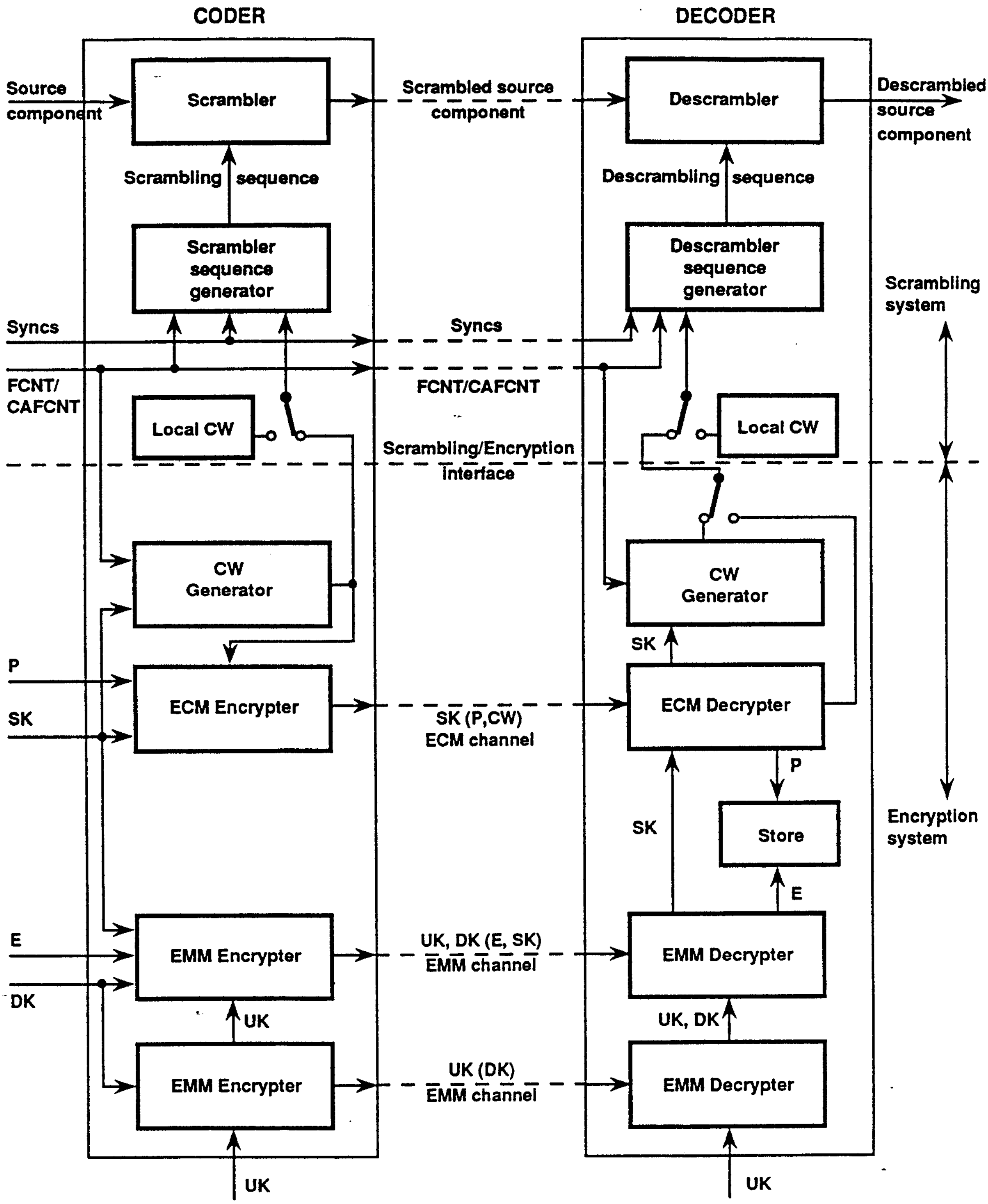
The five general packet formats which may be used for the entitlement management messages and the entitlement checking messages are shown in figure 59.

The first two fields, PH and PT, are common to all packets (as defined in subclause 4.3.6). The Packet Header (PH) contains a 10-bit address field identifying the packet in the data multiplex and a 2-bit continuity index (I). Packets of the same type can be linked using the continuity index (I) of the packet header. The first synchronising packet of a message shall have a zero continuity index (I=0), while subsequent packets have continuity index values in the range, the value of I being incremented by 1 in modulo 3 sequence each time a packet of the same type is transmitted.

The type of message shall be indicated by the PT byte which is Hamming (8,2) protected. The values of the PT byte defined for EMMs shall be as shown in figure 59.

After the PT parameter, all packets shall contain an error protected data field. Protection shall be provided by a (24,12) Golay code; the data block, then, consists of 30 (24,12) Golay words which corresponds to a useful data field length of 360 bits with the protection removed. The 24-bit code words shall be formed by the (23,12) code as specified for the packet headers, the 24th bit specifying an overall odd parity. The Golay protection shall be applied as the last stage in the construction of the message.

Three of the possible types of EMMs are specified with a customer address field at the start of the useful data field.



EMM	= Entitlement management message
ECM	= Entitlement checking message
CW	= Control Word
DK	= Distribution Key
SK	= Service Key
FCNT	= Frame Count (from line 625)
CAFCNT	= Conditional Access Frame Count (from line 625)
E	= Customer Entitlements
UK	= Unique Key

P	= Programme Data (e.g. Access Conditions)
SK(P, CW)	= P and CW encrypted with SK
UK(E,SK) or DK(E,SK)	= E and SK encrypted with UK or DK
UK(DK)	= DK encrypted with UK

Figure 50: General structure of the conditional-access system with an example of key hierarchy

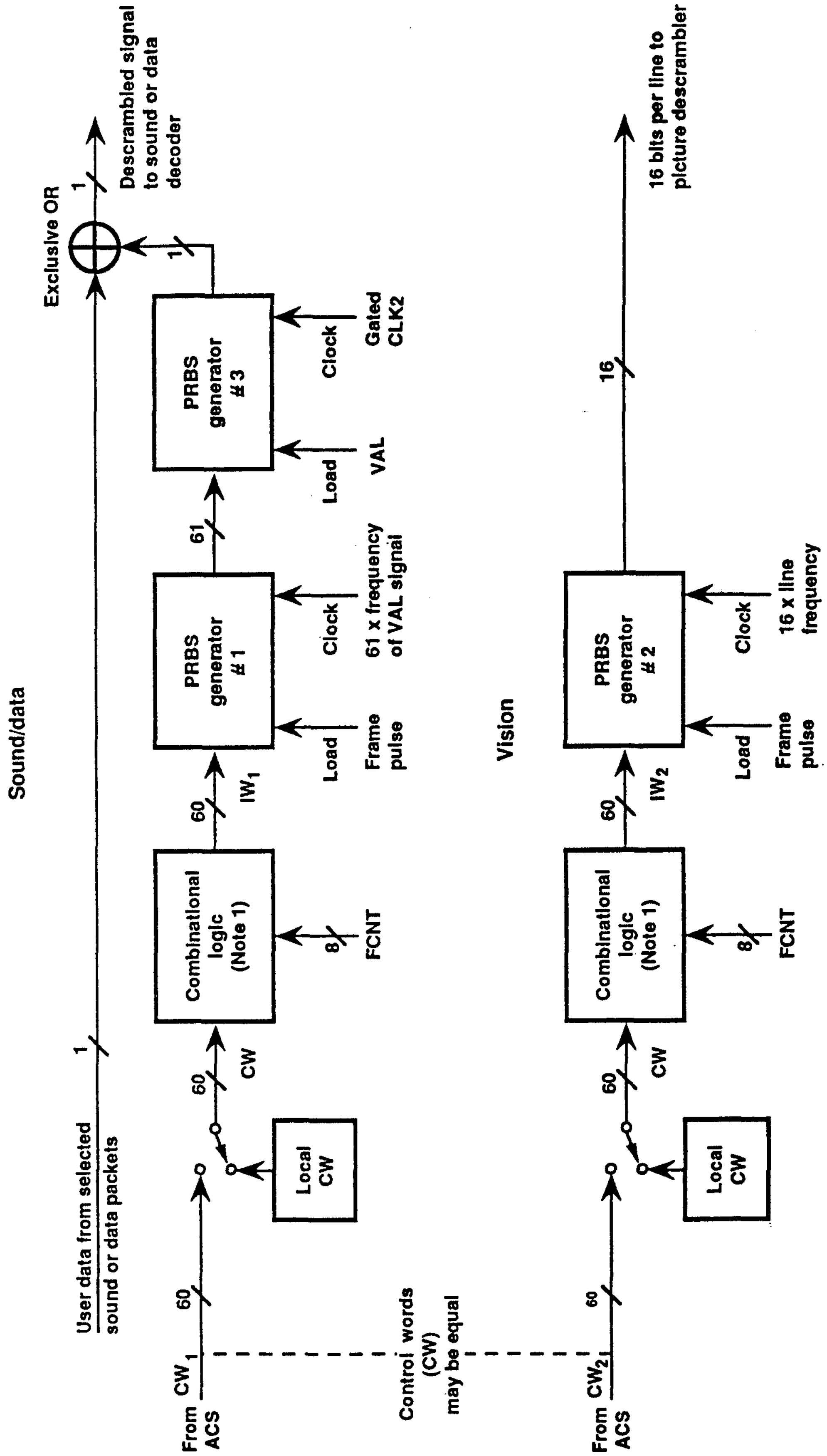


Figure 51:
PRBS system for conditional-access system (ACS) (shown for vision and one sound/data channel)

Description of the combinatorial logic of figure 51.

- The combinational logic functions, used to combine the two 60-bit control words (CW1 and CW2) with the 8-bit frame count (FCNT), to give the two 60-bit initialization words (IW1 and IW2), are the same in both cases and shall be defined as follows:
  - the 8-bit frame count (FCNT) is written as $F_0, F_1, \ldots F_7$;
  - the 60-bit control word (CW) is written as follows:

$$\underbrace{C_0C_1C_2C_3C_4C_5C_6C_7}_{CW0}\quad \underbrace{C_8C_9C_{10}C_{11}C_{12}C_{13}C_{14}C_1}_{CW1}\quad \underbrace{C_{48}C_{49}C_{50}C_{51}C_{52}C_{53}C_{54}C_5}_{CW6}\quad \underbrace{C_{56}C_{57}C_{58}C_{59}}_{CW7}$$

  - Then the 60-bit Initialization Word (IW) may be written as follows:

$$\underbrace{I_0\ I_1\ I_2\ I_3\ I_4\ I_5\ I_6\ I_7}_{IW0}\ \underbrace{I_8\ I_9\ I_{10}\ I_{11}\ I_{12}\ I_{13}\ I_{14}\ I_{15}}_{IW1}\ \underbrace{I_{48}\ I_{49}\ I_{50}\ I_{51}\ I_{52}\ I_{53}\ I_{54}\ I_{55}}_{IW6}\ \underbrace{I_{56}\ I_{57}\ I_{58}\ I_{59}}_{IW7}$$

  - IW shall be obtained from CW and FCNT as follows:

$$IW0 = CW0 \oplus FCNT \quad IW1 = CW1 \oplus \overline{FCNT} \quad IW2 = CW2 \oplus FCNT \quad IW3 = CW3 \oplus \overline{FCNT}$$
$$IW4 = CW4 \oplus FCNT \quad IW5 = CW5 \oplus \overline{FCNT} \quad IW6 = CW6 \oplus FCNT \quad IW7 = CW7 \oplus \overline{FCNT}$$

  - In addition, it should be noted that:

a) FCNT is the complement of FCNT (i.e. $FCNT \oplus \overline{FCNT} = 11111111$);
b) in the transformation from CW7 to IW7 only the least significant half byte of $\overline{FCNT}$ is used.

Load pulse for PRBS 1 and PRBS 2:

- the frame pulse occurs once per frame such that its leading edge (rising edge) marks the frame boundary.

Clocking of PRBS 1:

- PRBS generator 1 shall be clocked 61 times between each VAL signal. The validation signal (VAL) is a clock signal which occurs at the packet rate.

Loading and clocking of PRBS 3:

- the clock shall be gated such that PRBS 3 is clocked for 720 bits per packet and in synchronism with the burst of useful data arriving at the descrambling modulo-2 adder.

Fixed local control word:

- in the scrambled free-access mode, all the bits of the control word shall be set at logic "1".

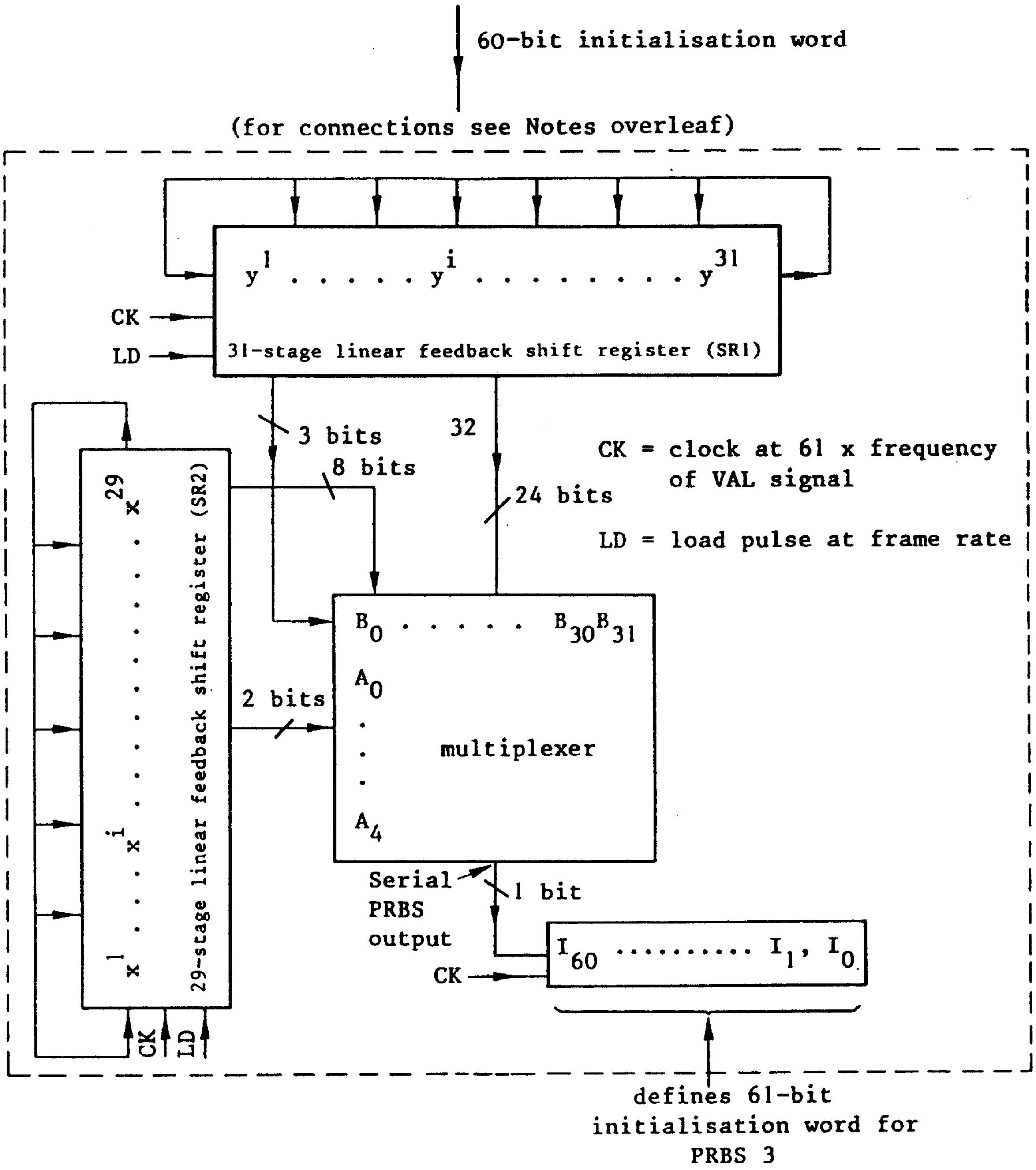


Figure 55a: PRBS generator 1 in figure 51

Description of PRBS generator 1.

The interconnections for the above block diagram are as follows.

The PRBS generator has two multi-stage linear-feedback shift-registers SR1 and SR2. The polynomials defining the feedback configurations for these are as follows:

SR1: $1 + y + y^2 + y^3 + y^7 + y^{14} + y^{19} + y^{25} + y^{31}$

SR2: $1 + x^2 + x^3 + x^4 + x^8 + x^{11} + x^{16} + x^{20} + x^{29}$

The initialization word loading for SR1 and SR2 shall be defined as follows. The 60-bit initialization word is defined by the nomenclature I_{59} to I_0, i.e. I_i for each bit, where i has the values 0 to 59.

SR1: I_i is loaded to $y^{(31-i)}$ for i = 0 to 30

SR2: I_i is loaded to $x^{(60-i)}$ for i = 31 to 59

The shift register outputs are input to a multiplexer. The inputs to the multiplexer are given by the notation A_0 to A_4 and B_0 to B_{31}. The connections are defined by the following:

- x^1 is loaded to A_0;
- x^2 is loaded to A_1;
- x^i is loaded to $B_{(i-3)}$ for i = 3 to 10;
- y^1 is loaded to A_2;
- y^2 is loaded to A_3;
- y^3 is loaded to A_4;
- y^n is loaded to $B_{(n+4)}$ for n = 4 to 27.

The A signal forms a five-bit word A_4 ... A_0 which defines the address of the index of the B signal (a binary number corresponding to 0 to 31 for B_{31} to B_0). Sixty-one consecutive values of the B signal, as defined by the A address, are output to form the initialization word for PRBS 3 at a rate of once per packet (i.e. VAL signal frequency) such that the least significant bit of the initialization word (I_0) is output first.

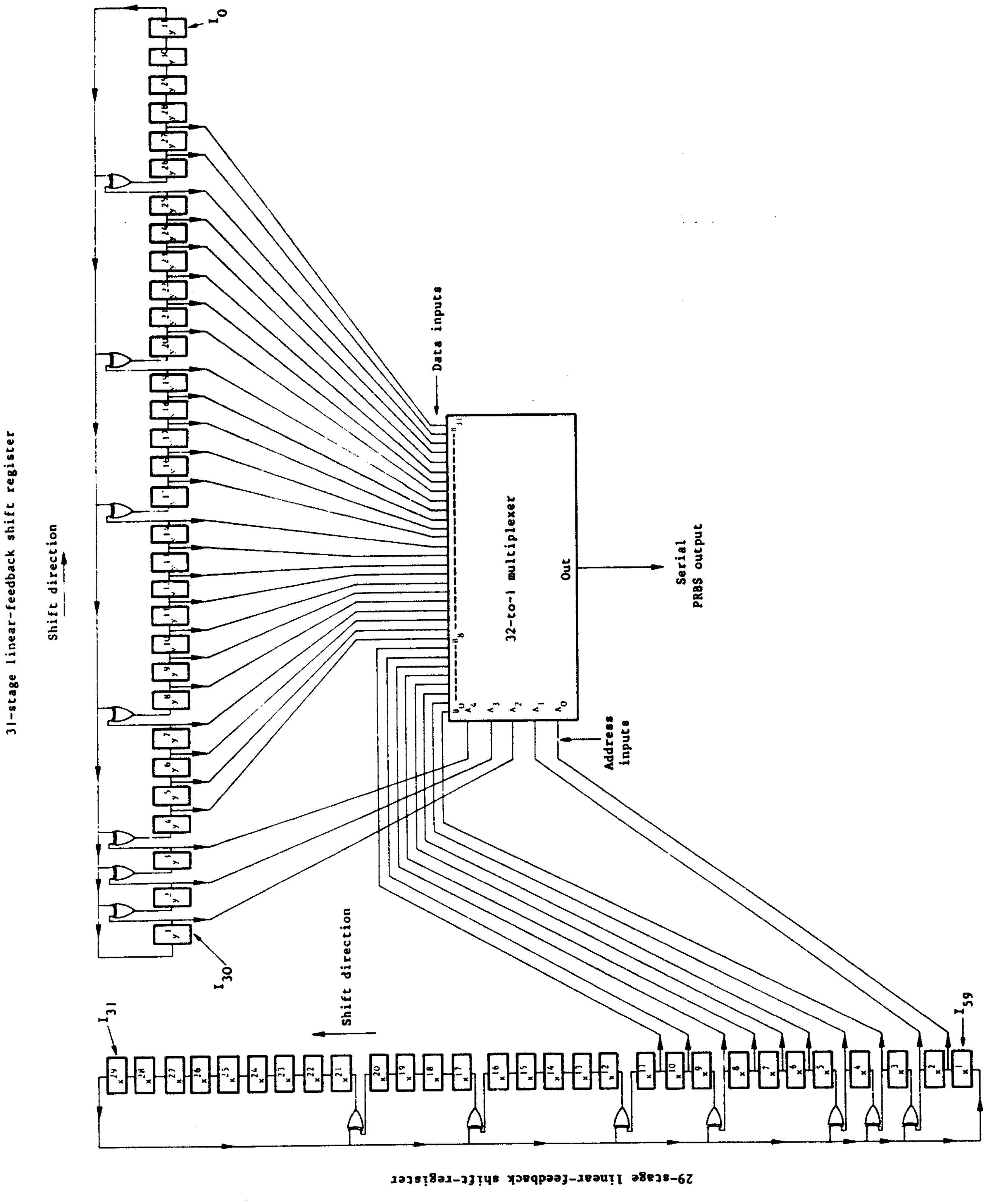


Figure 55b: more detailed diagram of PRBS generator 1 in figure 51

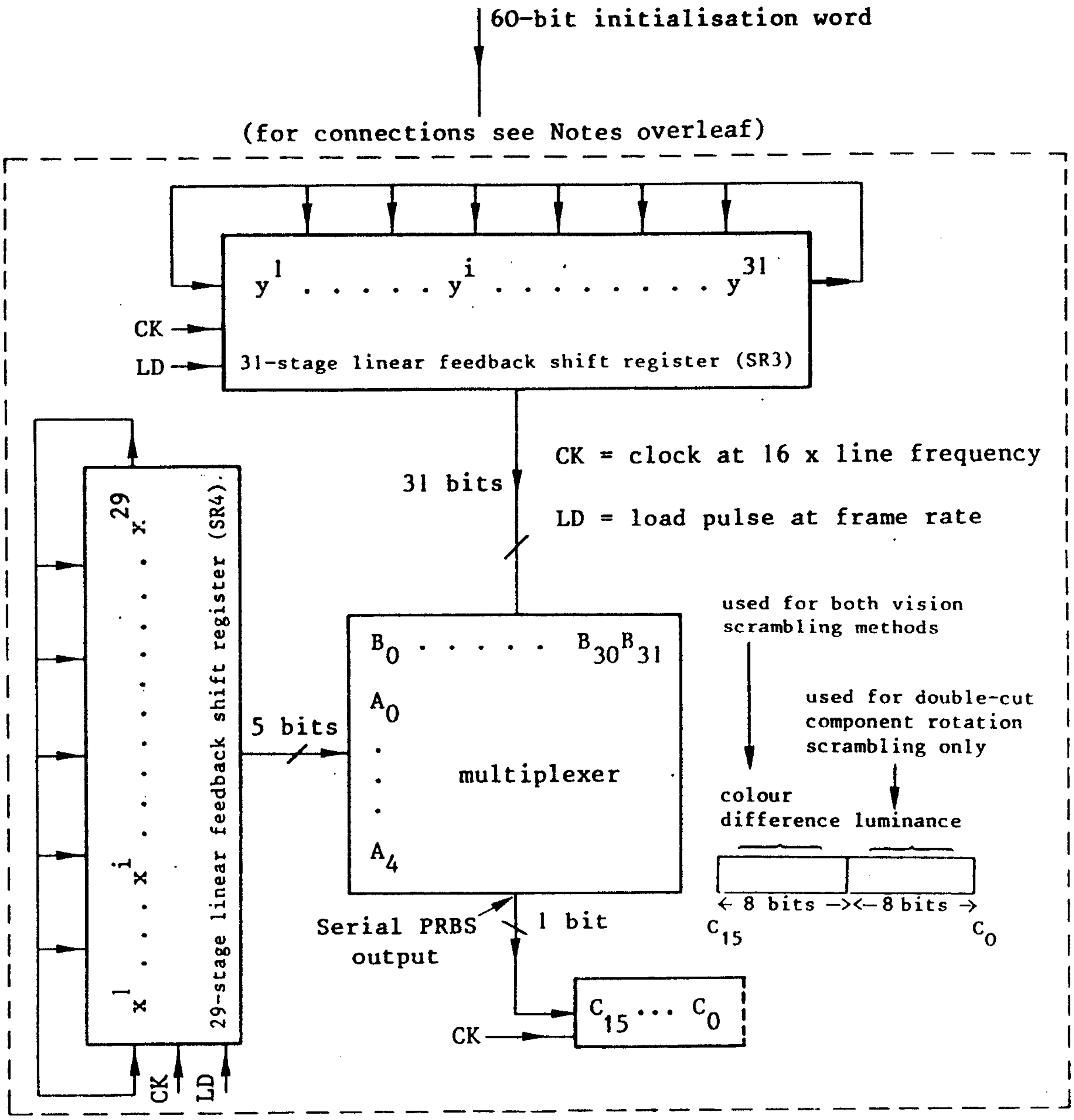


Figure 56a: PRBS generator 2 in figure 51

Description of PRBS generator 2.

The interconnections for the above block diagram are as follows.

The PRBS generator has two multi-stage linear feedback shift registers SR3 and SR4. The polynomials defining the feedback configurations for these are as follows:

SR3: $1 + y + y^2 + y^3 + y^5 + y^6 + y^7 + y^9 + y^{10} + y^{11} + y^{15} + y^{19} + y^{23} + y^{27} + y^{31}$

SR4: $1 + x^2 + x^3 + x^4 + x^5 + x^7 + x^{11} + x^{13} + x^{14} + x^{20} + x^{29}$

The initialization word loading for SR3 and SR4 are defined as follows. The 60-bit initialization word is defined by the nomenclature I_{59} to I_0, i.e. I_i for each bit, where i has the values 0 to 59.

SR3: I_i is loaded to $y^{(31-i)}$ for $i = 0$ to 30

SR4: I_i is loaded to $x^{(60-i)}$ for $i = 31$ to 59

The shift register outputs are input to a multiplexer. The inputs to the multiplexer are given by the notation A_0 to A_4 and B_0 to B_{31}.

The connections are defined by the following:

- x^i is loaded to $A_{(i-1)}$ for $i = 1$ to 5;
- y^i is loaded to $B_{(i-1)}$ for $i = 1$ to 31;
- in addition B_{31} is strapped to B_{30}.

The A signal forms a five-bit word A_4 ... A_0 which defines the address of the index of the B signal (a binary number corresponding to 0 to 31 for B_{31} to B_0). Sixteen consecutive values of the relevant B signal are output to form the 16-bit image descrambling word at line-rate.

For the double-cut component rotation system, the luminance component cut-point number (in the range 0 to 255, see figure 26d.) is the binary number C_7 ... C_0 given by the 8 least significant bits of the 16-bit shift register at the output of PRBS generator 2.

The colour-difference component cut-point number (in the range 0 to 255, see figure 26c) is the binary number C_{15} ... C_8 given by the 8 most significant bits of the 16-bit shift register at the output of PRBS generator 2.

For the single-cut line rotation system only, the colour-difference cut-point number described above is used.

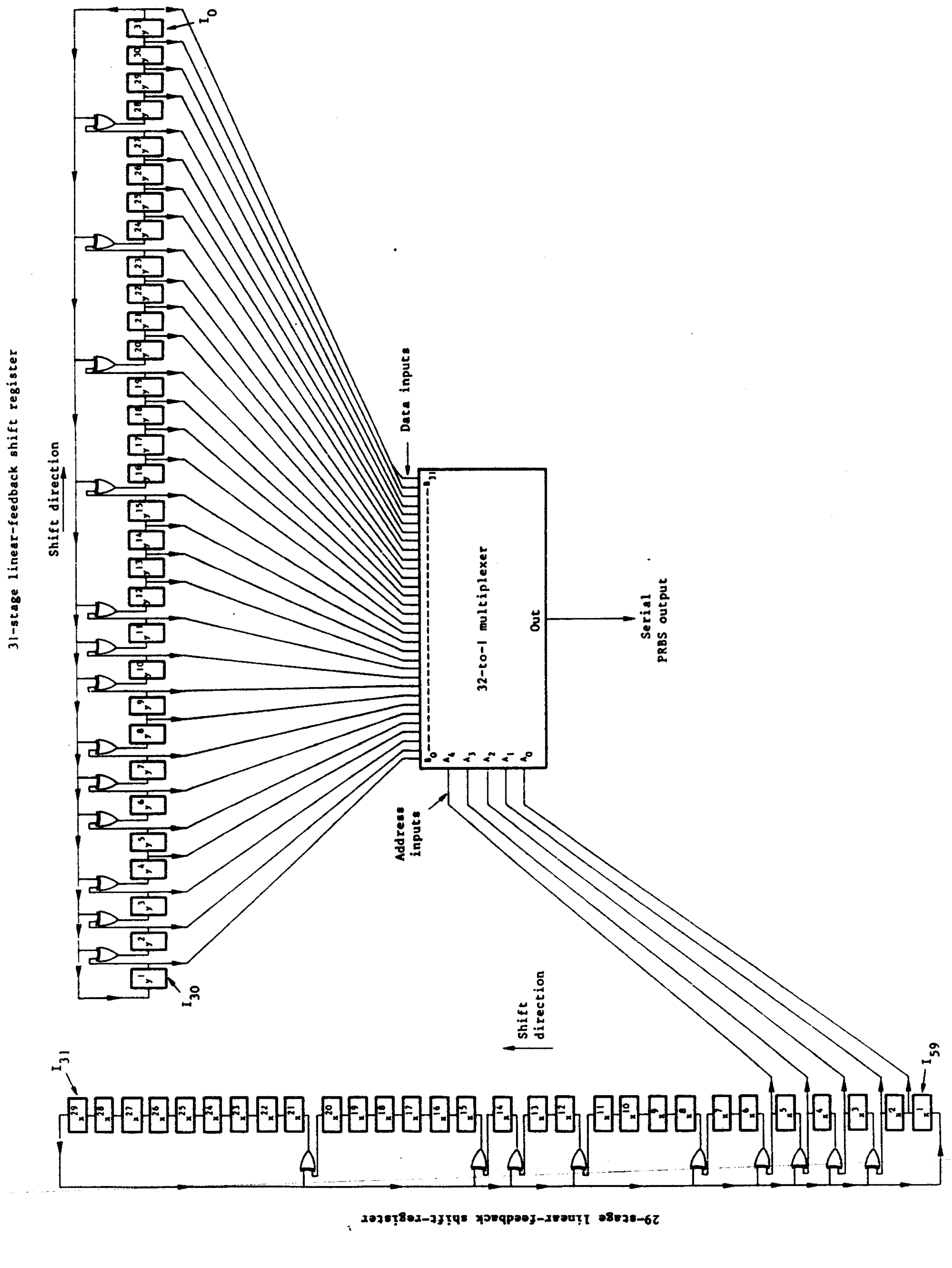


Figure 56b: a more detailed diagram of PRBS generator 2 in figure 51

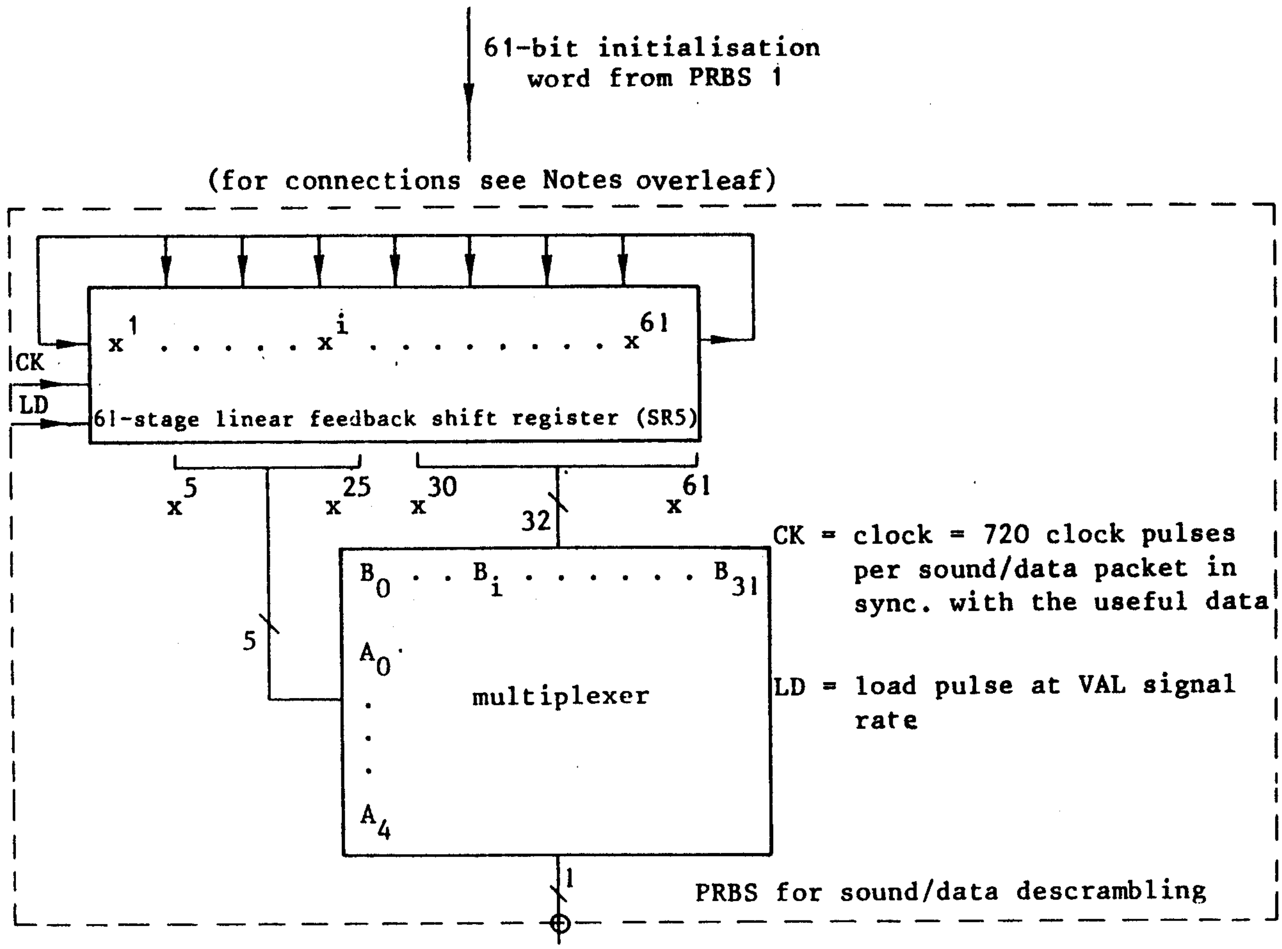


Figure 57a: PRBS generator 3 in figure 51

Description of PRBS generator 3.

The complete interconnections for the above block diagram are as follows.

The PRBS generator has one 61-stage linear feedback shift register SR5. The polynomial defining the feedback configuration is as follows:

SR5: $1 + x^2 + x^3 + x^7 + x^8 + x^9 + x^{10} + x^{12} + x^{15} + x^{19} + x^{20} + x^{22} + x^{24} + x^{25} + x^{28} + x^{30} + x^{33} + x^{34} + x^{37} + x^{40} + x^{43} + x^{44} + x^{46} + x^{54} + x^{56} + x^{60} + x^{61}$

The initialization word loading for SR5 is as follows.

The 61-bit initialization word is defined by the nomenclature I_{60} to I_0, i.e. I_i for each bit, where i has values 0 to 60.

SR5: I_i is loaded to $x^{(60-i)}$ for $i = 0$ to 60.

The shift register outputs are input to the multiplexer. The inputs to the multiplexer are given by the notation A_0 to A_4 and B_0 to B_{31}. The connections are defined by the following:

- $x^{5(i+1)}$ is loaded to A_i for $i = 0$ to 4;
- $x^{(30+i)}$ is loaded to B_i for $i = 0$ to 31.

The A signal forms a five-bit word $A_4 \ldots A_0$ which defines the address of the index of the B signal (a binary number corresponding to 0 to 31 for B_{31} to B_0). The value of the B signal, at the address defined by A is switched to the output at clock instants (CLK 2) to form a serial output data stream at the data rate of the useful data.

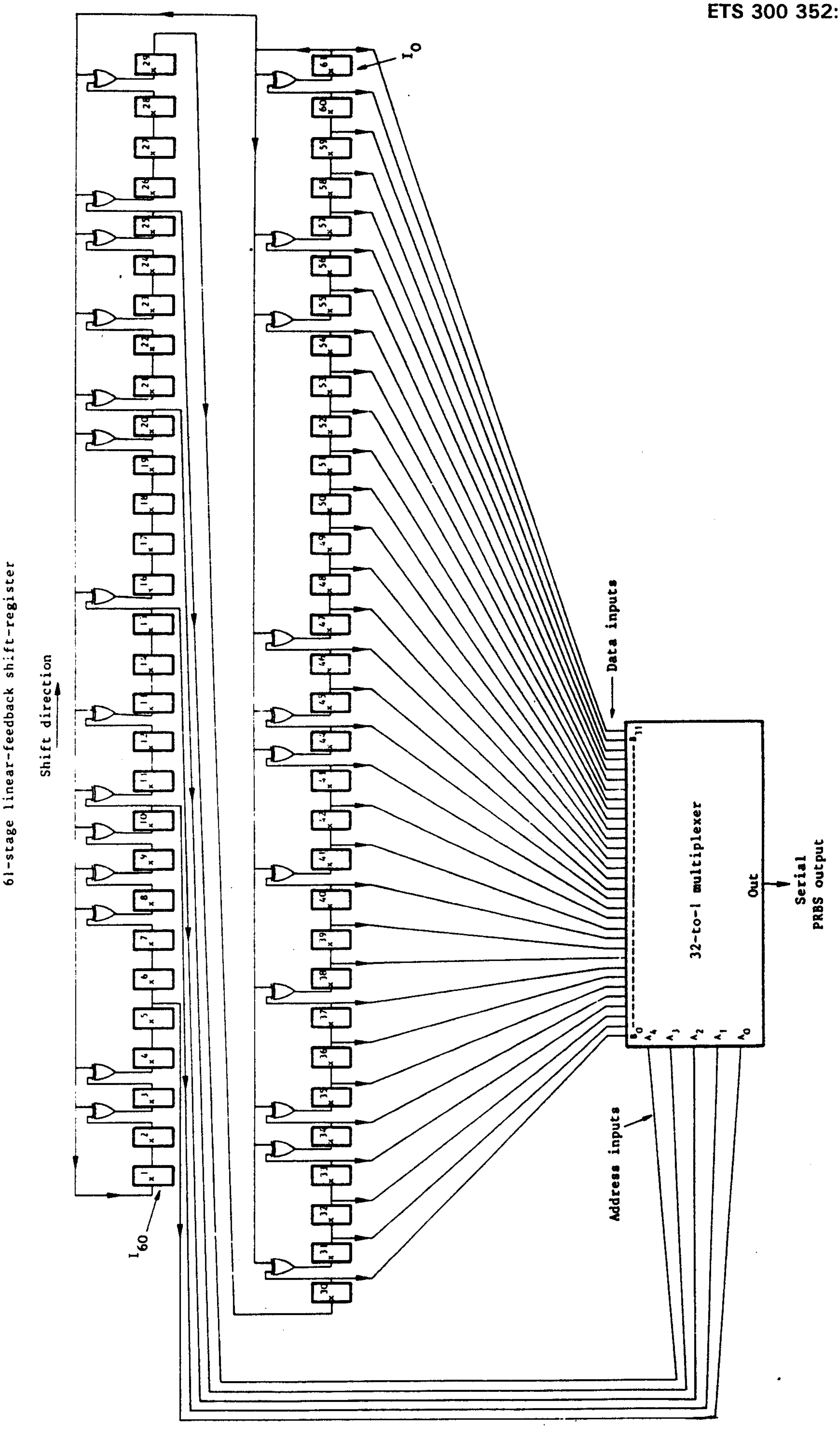


Figure 57b: a more detailed diagram of PRBS generator 3 in figure 51

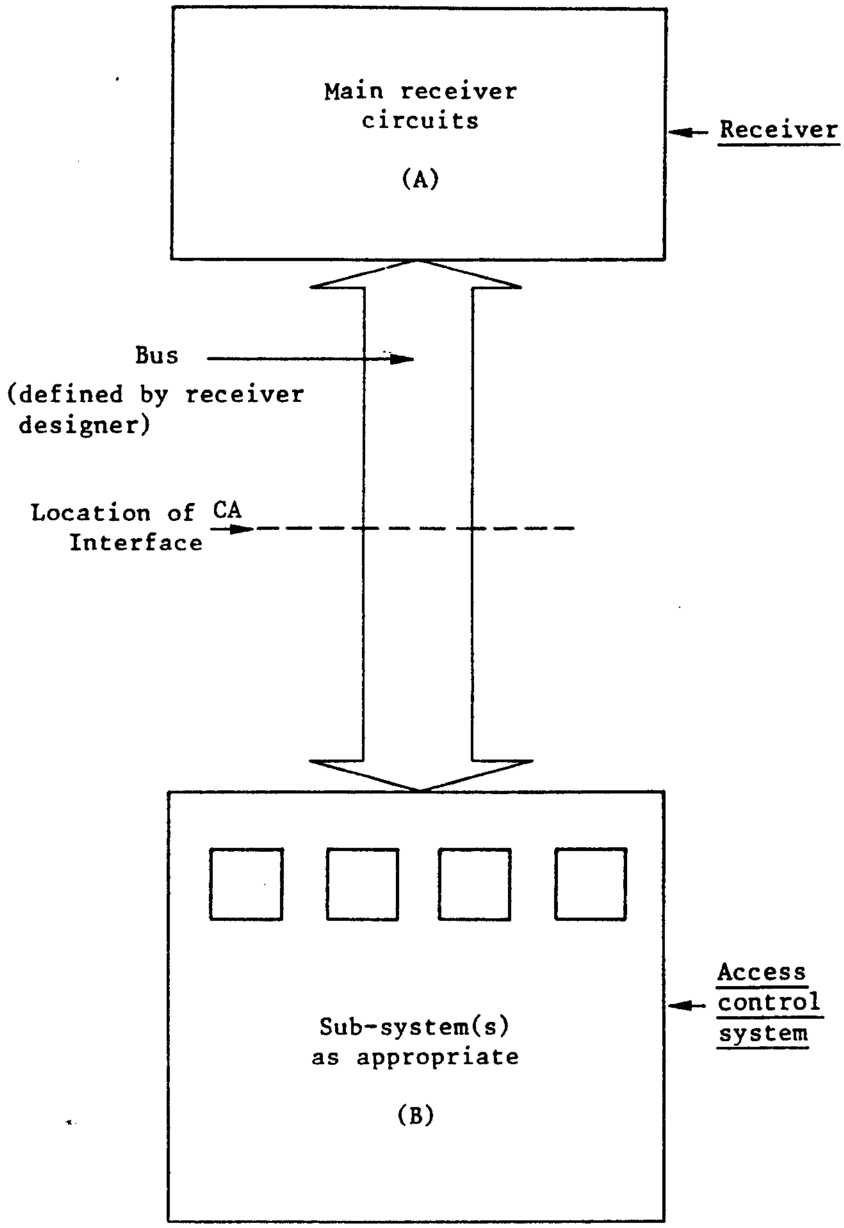


Figure 58: Conditional-access interface

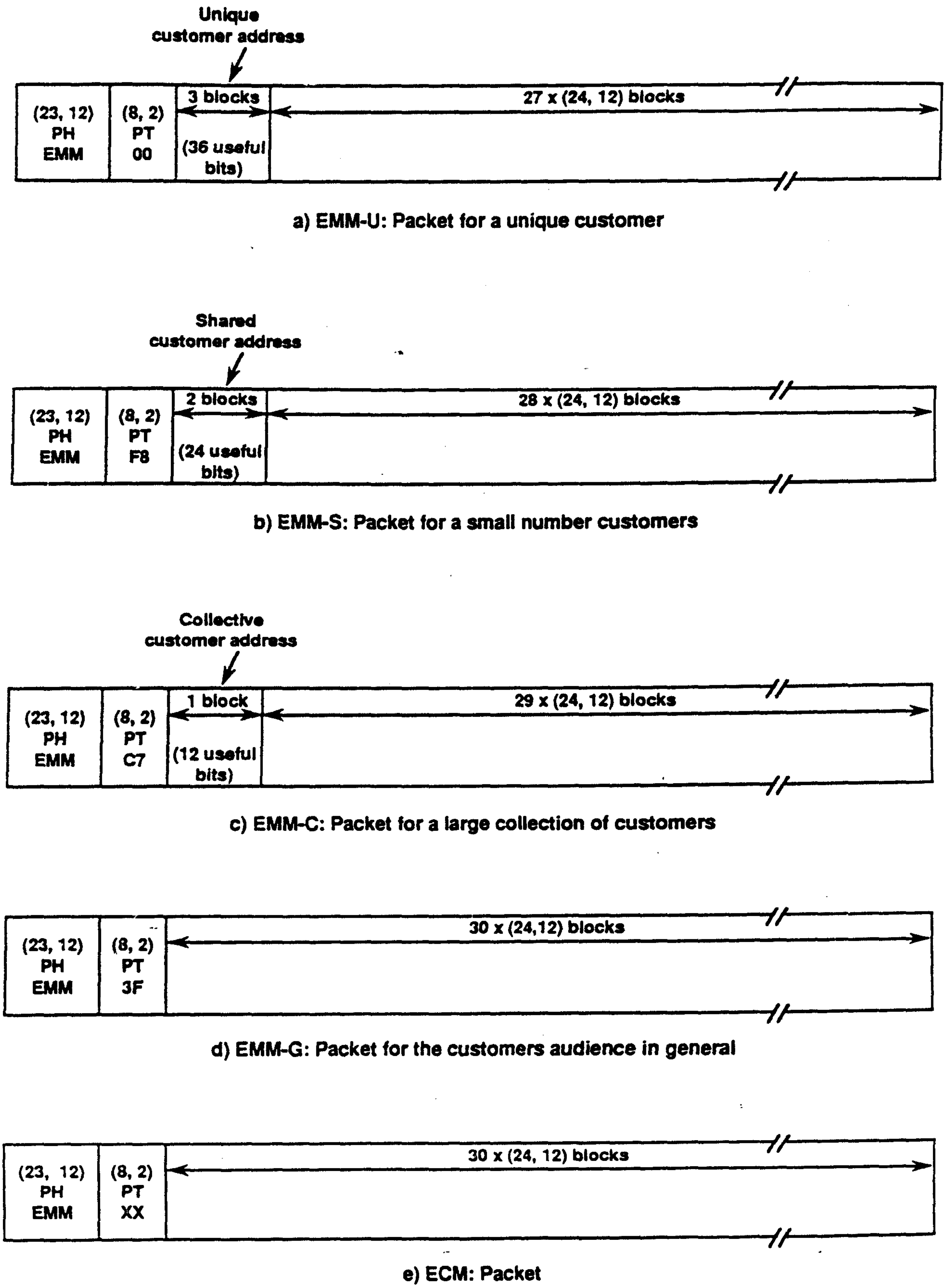


Figure 59: EMM and ECM packet formats

10 Specification of the modulation parameters

10.1 Introduction

Clause 10 contains the specification of the modulation parameters of the D2-HDMAC/packet system for satellite broadcasting receivable by domestic satellite receiving equipment and for cabled distribution networks.

Subclause 10.2 is concerned with the baseband processing for transmission of the D2-HDMAC/packet signal. Subclause 10.3 is concerned with the modulation parameters suited to satellite channels in the Broadcasting Satellite Service (BSS) and in the Fixed-Satellite Service (FSS). Subclause 10.4 is concerned with the modulation parameters suited for cabled distribution networks which use AM/VSB in the hyperband.

10.2 Baseband signal processing

10.2.1 Pre-emphasis

The pre-emphasis consists of two parts, a linear pre-emphasis and a non-linear pre-emphasis. The non-linear pre-emphasis will be further referred to as the Enhanced E7 (E7E) pre-emphasis.

The characteristics of the E7E network are such that the subjective picture quality of the signal received by a normal D2-MAC receiver equipped with an E7 de-emphasis (see ETS 300 250 [1] for description of E7 pre- and de-emphasis) does not degrade. In case the normal D2-MAC receiver is not equipped with an E7 de-emphasis a reduced subjective picture quality may occur. The presence of the E7E pre-emphasis is conveyed in the Service-Identification (SI) channel, parameter MODPRM (see subclause 8.2.1).

10.2.1.1 Linear pre-emphasis

The baseband signal of the D2-HDMAC system shall, before transmission, be pre-emphasised in the case of satellite broadcasting by means of a network with transfer characteristic (as shown in figure 60) defined by the expression:

$$H(f) = A * \frac{1 + j(f/f_1)}{1 + j(f/f_2)}$$

The characteristics of the pre-emphasis network are given below and in figure 60

- $A = 1 / \sqrt{2}$.
- $f_1 = 0{,}84$ MHz.
- $f_2 = 1{,}5$ MHz.

10.2.1.2 Non-linear pre-emphasis (E7E)

The E7E pre-emphasis is shown in figure 61a. The E7E de-emphasis network is shown in figure 61b.

The E7E pre-emphasis network consists of a feed-forward error correction structure. The general structure of the subblocks BP and BD are depicted in figures 62a and 62b respectively.

Within these subblocks F_{1a}, F_{1b} and F_2 represent filters, with frequency responses as shown in figure 63. The coefficients of these filters shall comply to the values defined in table 26.

The subblocks N_{1p}, N_{2p}, N_{1d} and N_{2d} are non-linear functions as shown in figures 64a and 64b. The non-linear functions are defined by the functions listed below, depending on parameters A, B, C and G (see also figure 65). The parameters shall comply to the values defined in table 27.

The non-linear functions N_{1p} and N_{2p} are defined by:

$$V_o = \frac{V_i}{C} + \frac{1}{B}\log_e\left(\frac{V_i + \sqrt{V_i^2 + (2AC)^2}}{2AC}\right),$$

where V_o is the output from the network and V_i is the input to the network. Note that the input and output levels V_i and V_o are expressed as normalised values relative to the input range of the non-linear pre-emphasis network, in which the range -0,5 ; 0,5 corresponds to the difference between the black and white levels (as referenced in line 624).

The non-linear functions N_{1d} and N_{2d} shall be the inverse of N_{1p} and N_{2p} respectively and can be obtained by solving V_i from the below equation:

$$V_i = \frac{V_o}{C} + \frac{1}{B}\log_e\left(\frac{V_o + \sqrt{V_o^2 + (2AC)^2}}{2AC}\right)$$

Table 26: filter coefficients

coefficient	F_{1a}	F_{1b}	F_2
C_{-3}	0	- 16	0
C_{-2}	0	- 64	19
C_{-1}	0	16	- 64
C_0	64	64	90
C_1	6	0	- 64
C_2	- 64	0	19
C_3	- 16	0	0

Table 27: parameters for non-linear functions

	A	B	C	G
N_{1p}	0,00215	49,5	2,125	0,95
N_{1d}	0,00215	49,5	2,125	0,95
N_{2p}	0,0017	24,75	2,3125	1
N_{2d}	0,0017	24,75	2,3125	1

The pre-emphasis network is chosen complex in order to allow a simpler implementation of the de-emphasis network.

The E7E pre-emphasis shall be applied only to active vision samples. The transitions between E7E mode and non-E7E mode are controlled transitions. The weighting network is included as shown in figure 61a.

For active vision lines 23 to 310 and 335 to 622 inclusive, transition weights shall be as follows:

Transition weight W	Sample numbers		
	Scrambled video (double cut)	Scrambled video (single cut)	Unscrambled video
0	1 to 225 inclusive	1 to 225 inclusive	1 to 231 inclusive
1/8	226	226	232
1/2	227	227	233
7/8	228	228	234
1	229 to 1 287 inclusive	229 to 1 288 inclusive	235 to 1 287 inclusive
7/8	1 288	1 289	1 288
1/2	1 289	1 290	1 289
1/8	1 290	1 291	1 290
0	1 291 to 1 296 inclusive	1 292 to 1 296 inclusive	1 291 to 1 296 inclusive

10.2.2 Nyquist filtering

The coding of the HDMAC vision signal, based on subsampling, implies that the overall transmission channel preserves the independence between consecutive samples, at 20,25 MHz. This condition is fulfilled if the equivalent baseband channel meets the first Nyquist criterion in the vicinity of 10,125 MHz.

The Nyquist filter shall be applied to the HDMAC vision signal after the E7E pre-emphasis and to the test signals of lines 312, 623 and 624.

The Nyquist filtering shall be performed at baseband level by a Nyquist filter with a roll-of factor of 10%, equally-split between the transmitter (HDMAC encoder) and the receiver (HDMAC receiver front-end). The Nyquist filtering shall be identical for satellite broadcasting and cabled distribution.

The half-Nyquist filter shall have a theoretical transfer function defined by the expression (see figure 66a):

$$H(f) = \begin{cases} 1 & \text{if } |f| < \frac{1}{2T}(1-\alpha) \\ \sqrt{\frac{1}{2} + \frac{1}{2}\sin \pi T\left(\frac{\frac{1}{2T} - |f|}{\alpha}\right)} & \text{if } \frac{1}{2T}(1-\alpha) < |f| < \frac{1}{2T}(1+\alpha) \\ 0 & \text{if } |f| > \frac{1}{2T}(1+\alpha) \end{cases}$$

where: $1/T = 20{,}25$ MHz

$\alpha = 0{,}1$

The template given in figure 66b shall be complied to for a digital hardware implementation on the transmitter side.

10.3 Modulation parameters for satellite broadcasting

This subclause contains the specification of the modulation parameters for use in satellite broadcasting both in the Broadcasting Satellite Service (BSS) and in the Fixed-Satellite Service (FSS) frequency ranges.

Frequency modulation shall be used for the whole baseband signal. Linear pre-emphasis shall be applied to the whole baseband of the D2-HDMAC signal, energy dispersal shall be added, and the signal shall be used to frequency modulate a carrier. However, the additional non-linear pre-emphasis shall be applied to the vision samples only (see subclause 10.2.1.2).

Both data and vision are processed together in the modulating equipment.

Two sets of parameters are defined, depending on the bandwidth available with respect to the frequency planning arrangements and the interference situation. These sets may be used independently of the frequency range according to the radio frequency regulations. In the following, they are referred to as:

- Option a: narrow bandwidth channels (< 30 MHz);
- Option b: wide bandwidth channels (≥ 30 MHz).

10.3.1 Service continuity

For systems with frequency deviation of 13,5 MHz/V and 27 MHz bandwidth, it is assumed that the service continuity requirement is fulfilled if a bit-error ratio of 10^{-3} is achieved at a carrier-to-noise ratio of 8 dB, measured in a 27 MHz noise bandwidth.

10.3.2 Modulation method

Frequency modulation shall be used for the baseband signal.

A transition from black to white of the luminance signal corresponds to an increase in frequency at the receiver input.

10.3.3 Deviation sensitivity

The frequency deviation sensitivity in MHz/V at the transition frequency of the pre-emphasis network shall be as follows (the actual value of the deviation sensitivity is conveyed in the SI channel, parameter MODPRM (see Clause 8)):

Channel bandwidth	Frequency bands	
	BSS	FSS
Option a	13,5-16 MHz/V	13,5-16 MHz/V
Option b	18-22 MHz/V	22 MHz/V

10.3.4 Energy dispersal

An energy dispersal signal shall be added (see NOTE 1) to the whole baseband signal of the D2-HDMAC system (see figure 67). The dispersal signal shall consist of a frame synchronous triangular wave form of frequency 25 Hz with a deviation after modulation in the radio frequency channel as given in the following table:

Channel bandwidth	Frequency bands	
	BSS	FSS
Option a	600 kHz p-p	1,2 MHz p-p
Option b	600 kHz p-p	2 MHz p-p (see NOTES 2 and 3)

NOTE 1: For full channel digital mode of operation (option C) energy dispersal is not always required.

NOTE 2: A lower value can be used if it complies with the Radio Regulations requirement not to cause interference (this depends on the satellite EIRP and the angle of elevation).

NOTE 3: The use of this value could result in some degradation of the picture quality (depending on the deviation sensitivity) with receivers using 1992 technology.

The dispersal deviation at the start of line 1 shall correspond to a reduction of the carrier frequency of one half of the peak-to-peak deviation. The actual value of the energy dispersal is conveyed in the SI channel, parameter MODPRM (see subclause 8.2.1).

10.3.5 DC restoration

DC restoration shall be applied to the analogue components at the input of the frequency modulator. The carrier frequency corresponding to zero colour-difference is given by:

$$f_o + f_d$$

where:

f_o is the channel centre frequency; and
f_d is the instantaneous frequency deviation produced by the energy dispersal signal.

10.3.6 RF bandwidth

The RF bandwidth of the transmission channel may vary depending on the satellite employed:

option a: 26 and 27 MHz;

option b: 33 and 36 MHz.

In order to ensure reception of all services, the -3 dB bandwidth of the receiver filters of domestic receiving equipment should be:

option a: 27 MHz;

option b: 33 MHz.

NOTE: The satellite transponder bandwidth can be greater than the transmission channel bandwidth.

10.3.7 Out of band radiation for the overall signal

Measured in a 4 kHz bandwidth, the radiated signal shall comply to the mask given in figure 68.

10.4 Modulation parameters for cabled distribution networks

This subclause contains the specification of the D2-HDMAC/packet system for use in cabled distribution networks using AM/VSB in the hyperband.

Only the 12 MHz channel spacing allows the transmission of HD-MAC signals.

This subclause conforms to the European Standard EN 50080 "RF characteristics of MAC AM-VSB cable receivers", which is applicable to 12 MHz channels in the hyperband.

10.4.1 Frequency allocation and channel spacing

In the hyperband, between 300 and 470 MHz, a 12 MHz channel spacing shall be used. This range can be extended downwards if needed.

The use of a PAL or SECAM channel adjacent to a D2-HDMAC channel without special precautions shall be avoided.

10.4.2 Modulation method

VSB amplitude modulation shall be used for the composite multiplex signal. There shall be no energy dispersal or linear pre-emphasis.

10.4.3 Polarity of modulation

The polarity of modulation shall be such that a transition from white to black of the luminance signal corresponds to an amplitude increase at the receiver input (negative modulation).

10.4.4 Depth of modulation

The peak amplitude of the carrier is taken as 100%. The minimum level of the carrier shall be 10%. Under these conditions, level "1" in the data signal corresponds to 91% and 19%, while level "0" corresponds to 55% (see figure 69).

10.4.5 Transmitter filtering

10.4.5.1 Transmitter Nyquist filtering around the carrier

A + 500 kHz full Nyquist filtering shall be used at the transmitter, according to the definition of the vestigial sideband which considers the slope at the -6 dB point. An example of a possible Nyquist slope is given in the following formula:

$$H(f) = \frac{1}{2}\left[1 + \sin\left(\frac{\pi}{2}\,\frac{f_0 - f}{0{,}75}\right)\right]$$

where f_0 is the carrier frequency, f is the current frequency, f and f_0 are expressed in MHz, and $f_0 - f < 0{,}75$ MHz.

10.4.5.2 IF filter specification

The complete Nyquist-filtering at the vision carrier (Carrier Nyquist) shall be done in the HDMAC VSB/AM modulator.

The Nyquist filtering at the second Nyquist point at 10,125 MHz (HD Nyquist) shall be done at baseband level, equally-split between transmitter (HDMAC Encoder) and receiver (HDMAC Front-end) side.

a) Modulator filtering:

- Nyquist filtering in the region of the vision carrier:

  - the VSB shall be filtered with a response, which is skew-symmetric to the vision carrier frequency in a linear scale.

- Amplitude characteristic:

  - the recommended template of the amplitude characteristic of the transmitter IF filter is given in figure 70a. The nominal value of the steepness of the Nyquist characteristic is ± 500kHz (linear). For details see template.

b) Receiver filtering:

- Amplitude characteristic:

  - the passband amplitude characteristic should be essentially flat up to 11,14 MHz i.e. (1 + α/100)*10,125 MHz, in relation to the vision carrier frequency (α = 10%, steepness of the Nyquist-filter at the 10,125 MHz point).

  - the recommended template of the amplitude characteristic for an HDMAC receiver IF filter is given in figure 71a.

10.4.5.3 Transmitter phase-frequency response

a) Modulator filtering:

- no pre-correction for the receiver group delay characteristic shall be applied;

- the transmitter filter shall be substantially phase linear;

- the recommended template of the phase characteristic is given in figure 70b.

b) Receiver filtering:

- the filter should be substantially phase-linear;

- the recommended template of the phase characteristic is given in figure 71b.

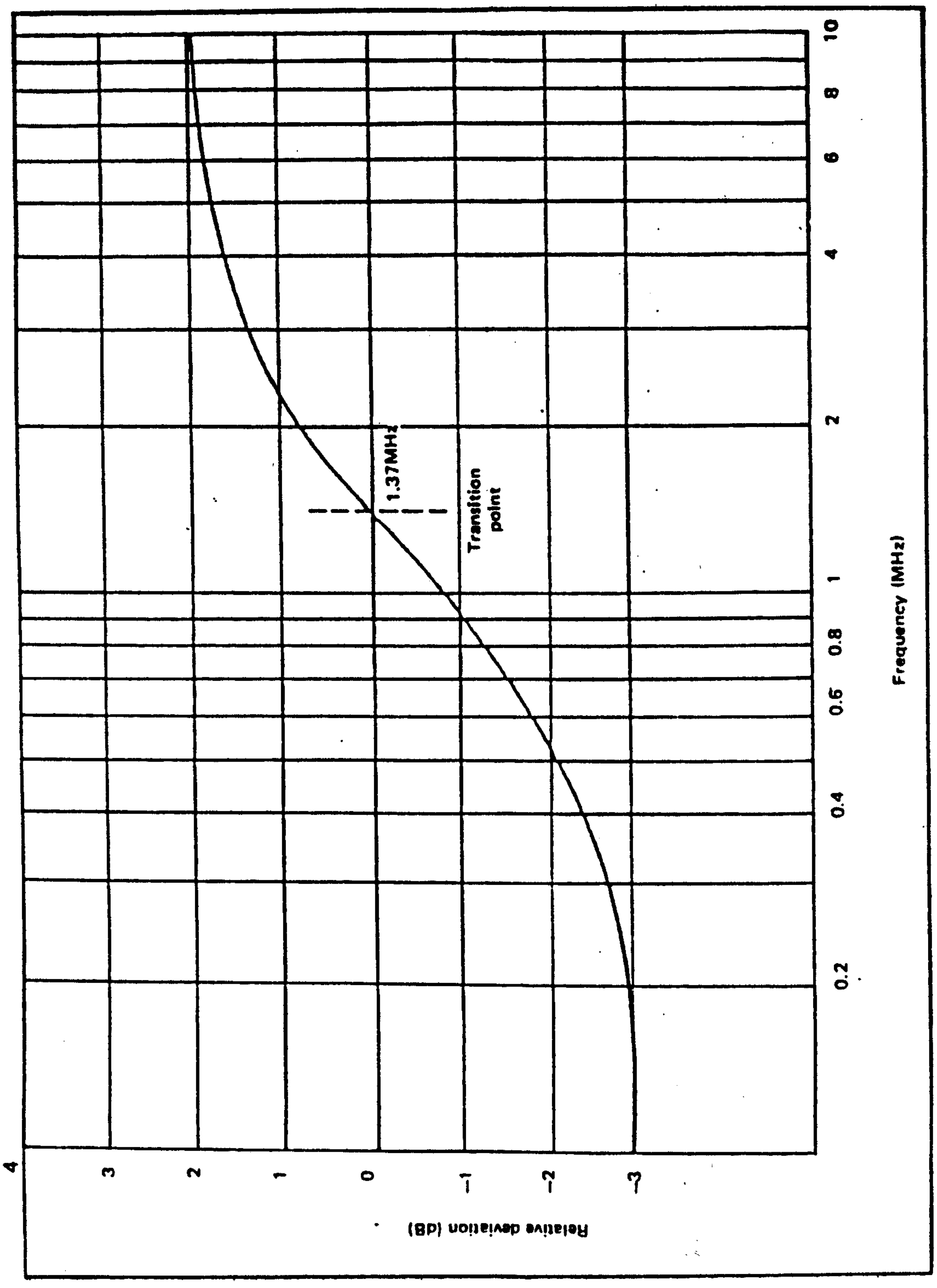


Figure 60: (HD)MAC linear pre-emphasis network

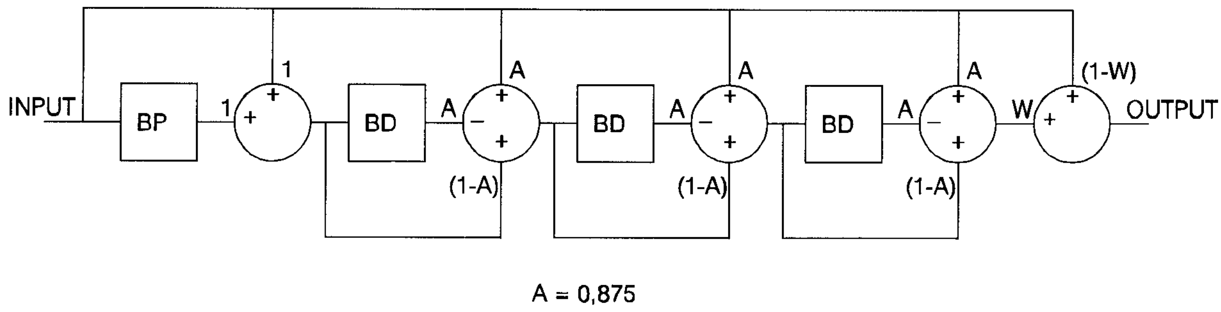


Figure 61a: E7E pre-emphasis network

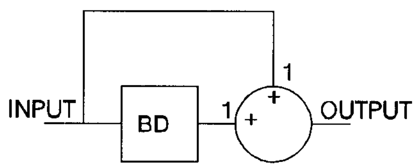


Figure 61b: E7E de-emphasis network

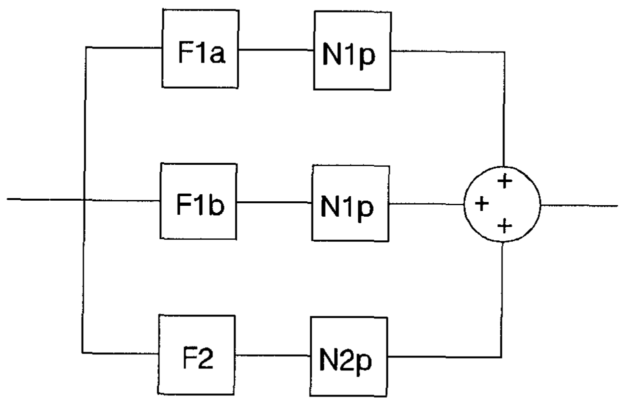


Figure 62a: Subblock BP

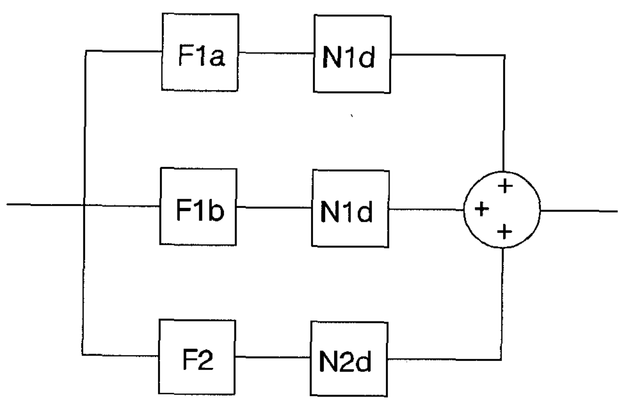


Figure 62b: Subblock BD

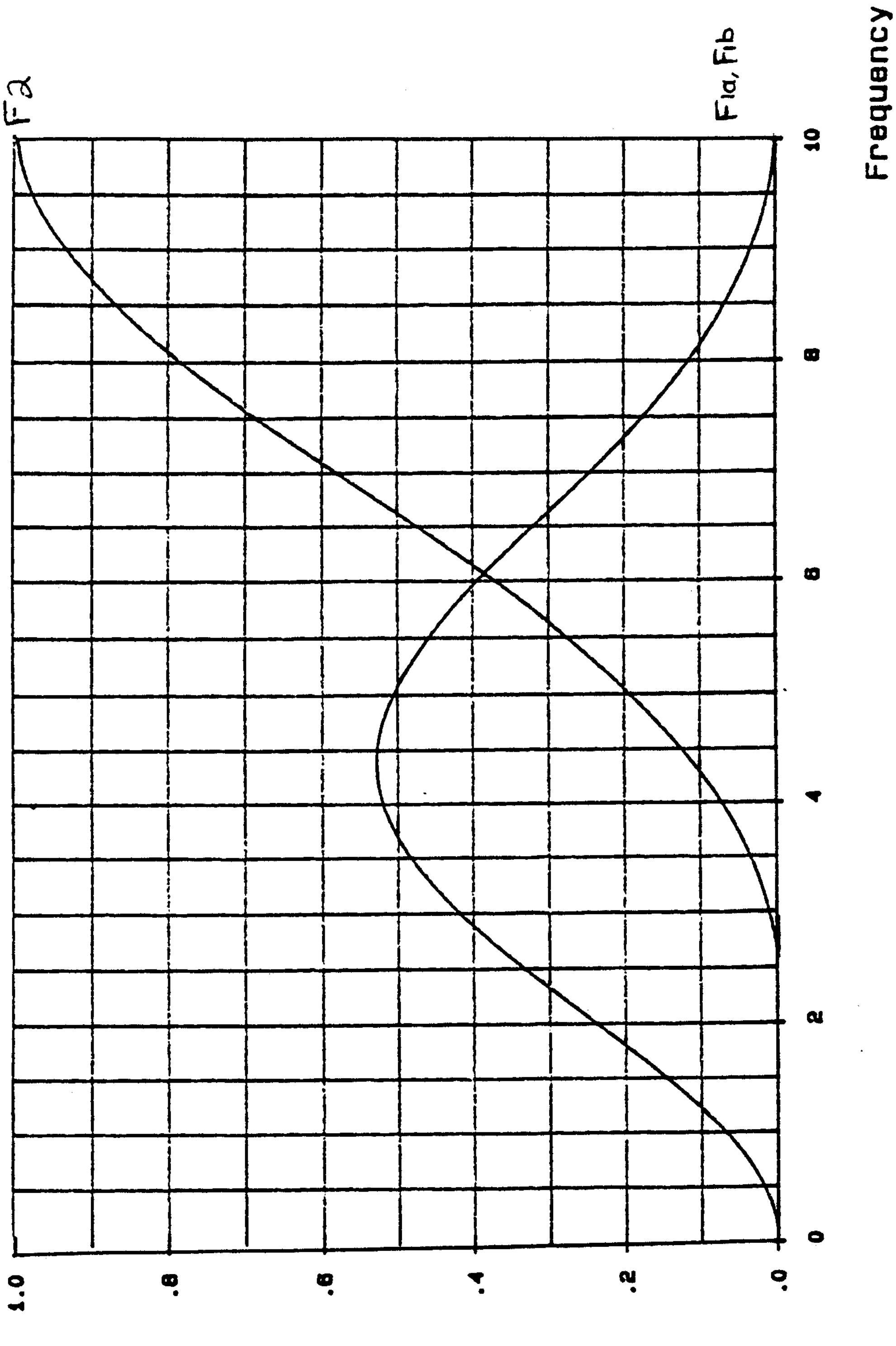


Figure 63: Magnitude response of filters F_{1a}, F_{1b} and F_2

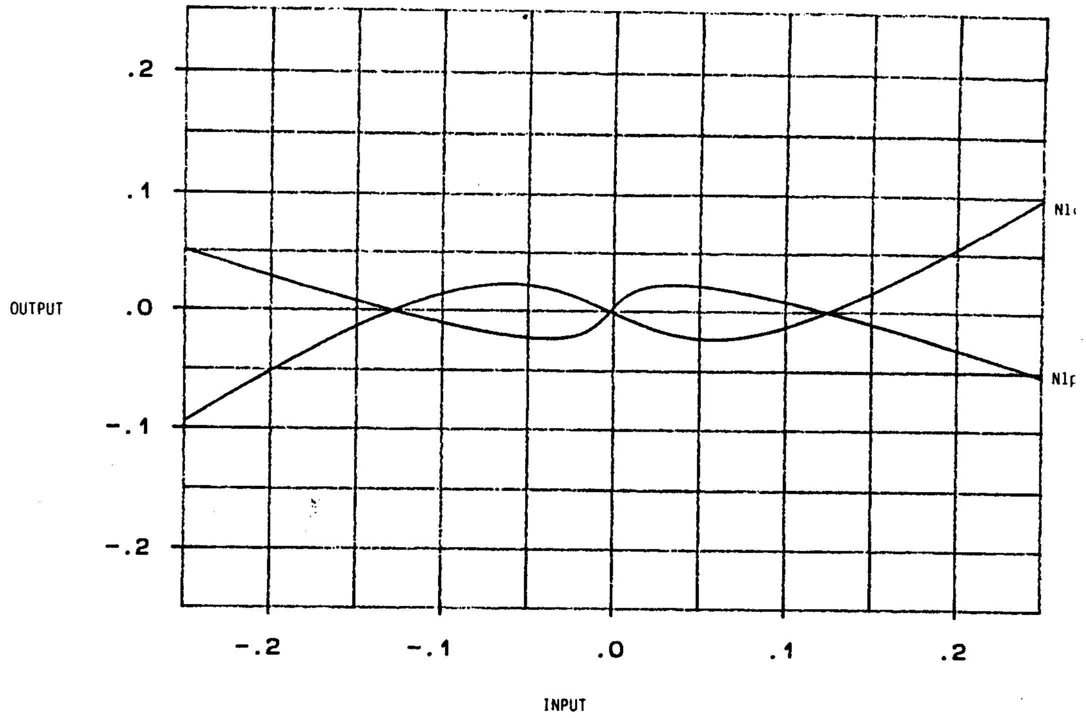


Figure 64a: Transfer function of N_{1p} and N_{1d}

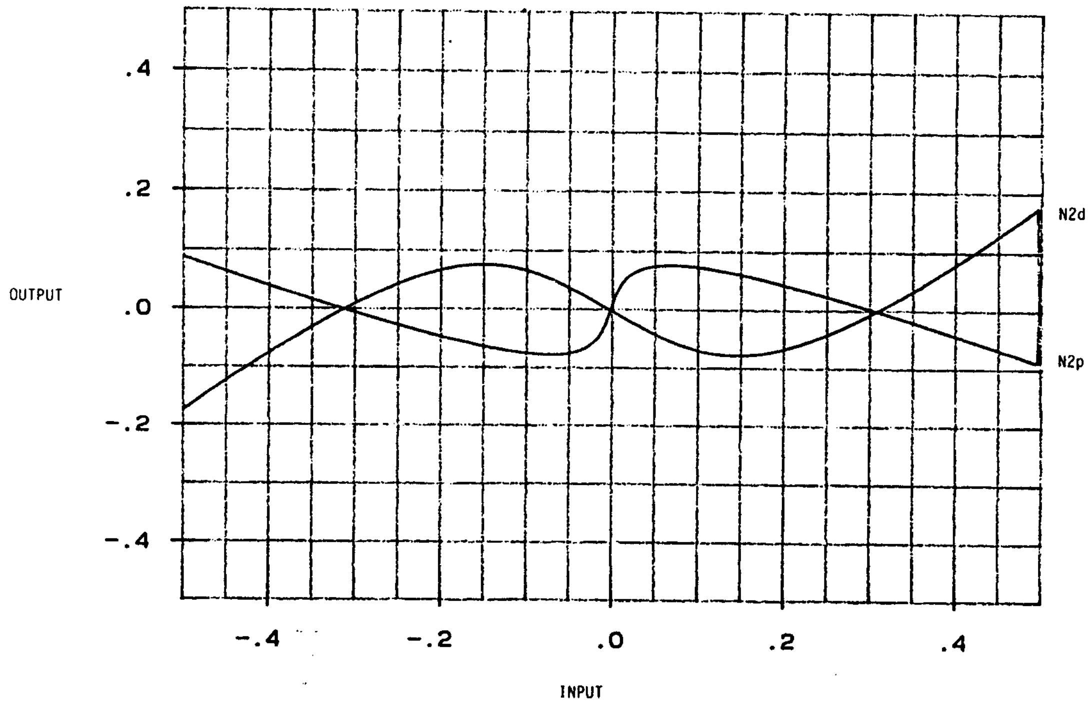


Figure 64b: Transfer function of N_{2p} and N_{2d}

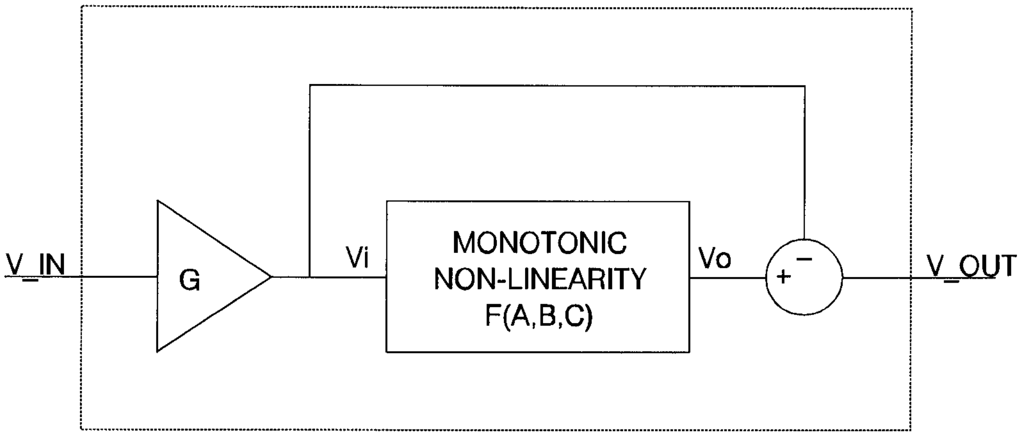


Figure 65: General structure of the non-linear functions

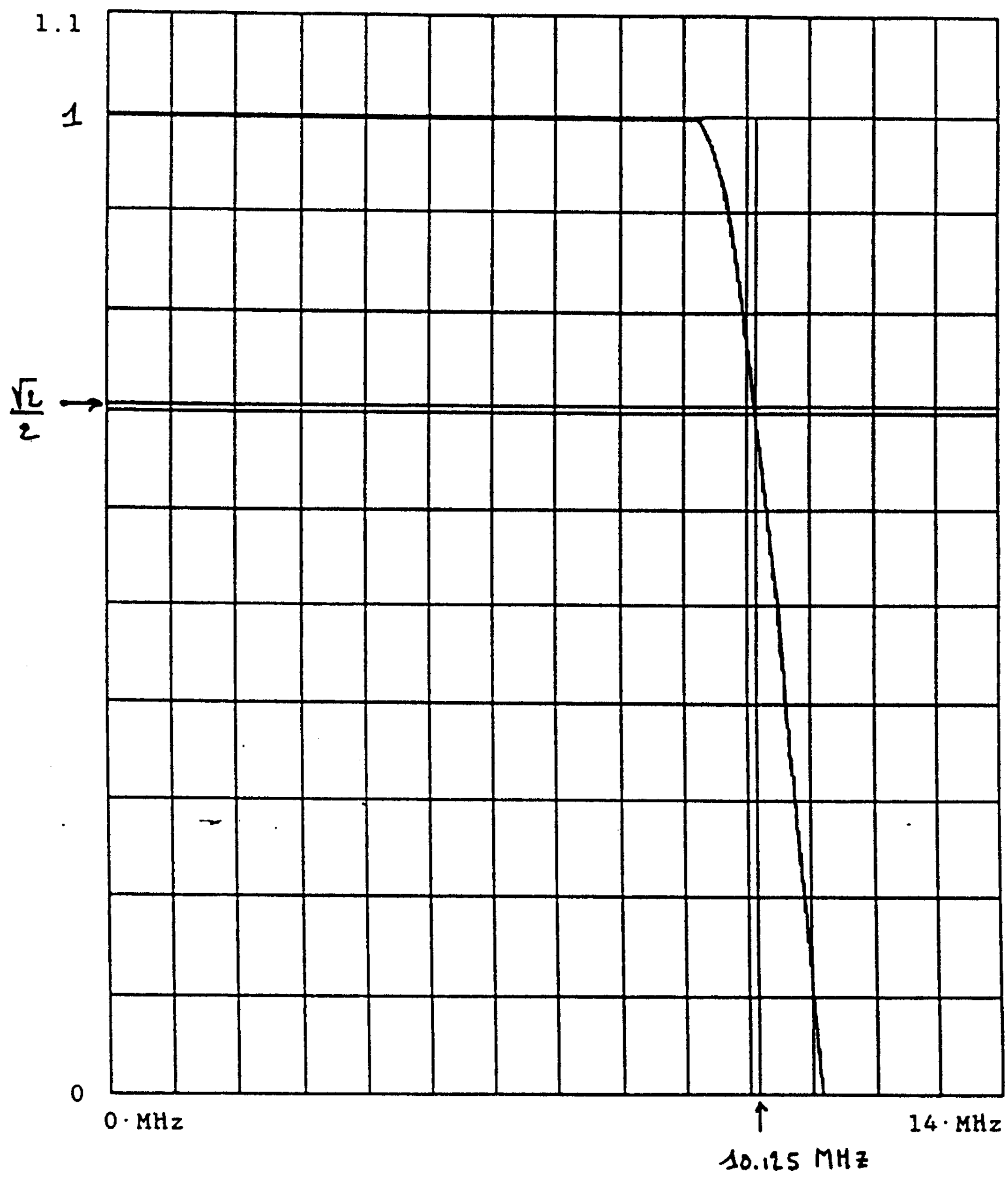


Figure 66a: Theoretical response of the 10% roll-off half-Nyquist filter

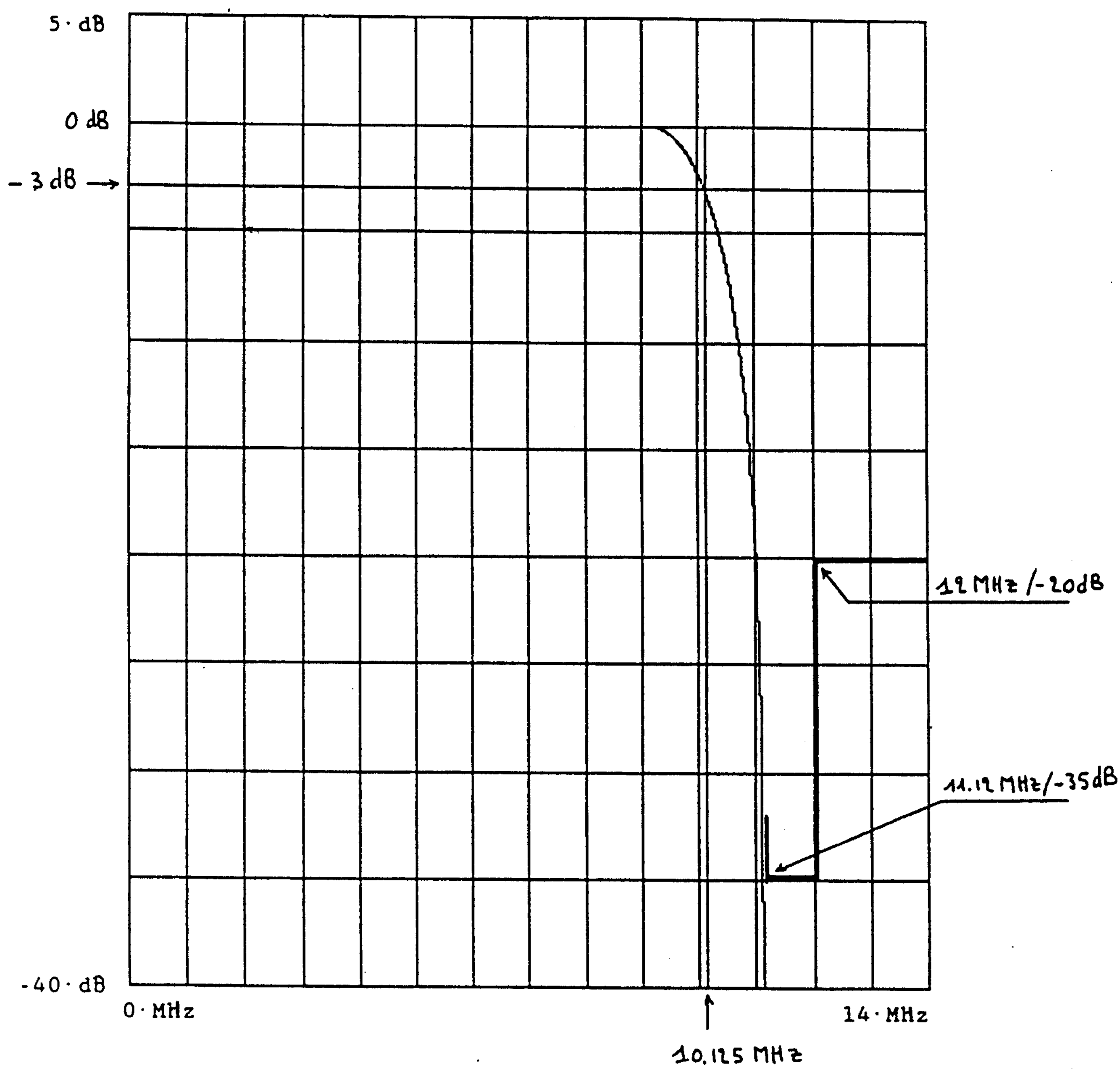


Figure 66b: Template for the out-band rejection of the transmitter half-Nyquist digital filter

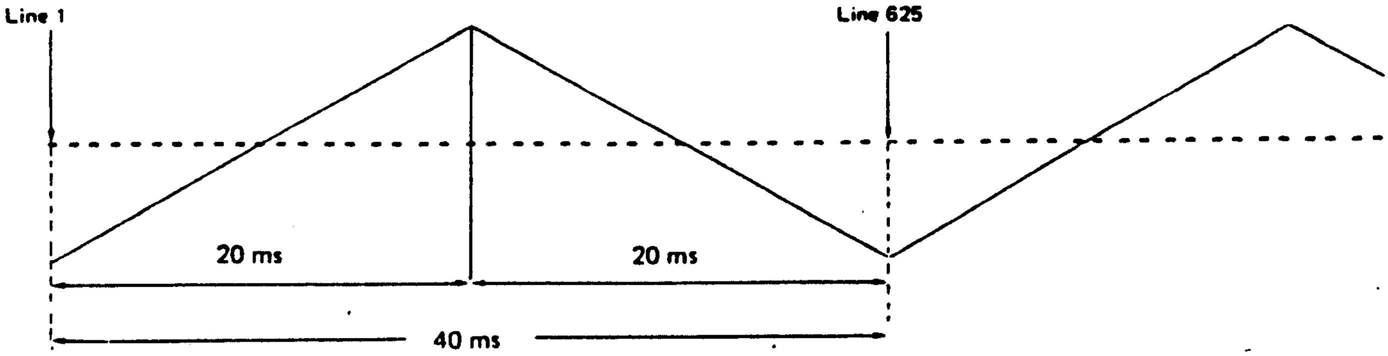


Figure 67: Energy dispersal wave form added to the baseband signal

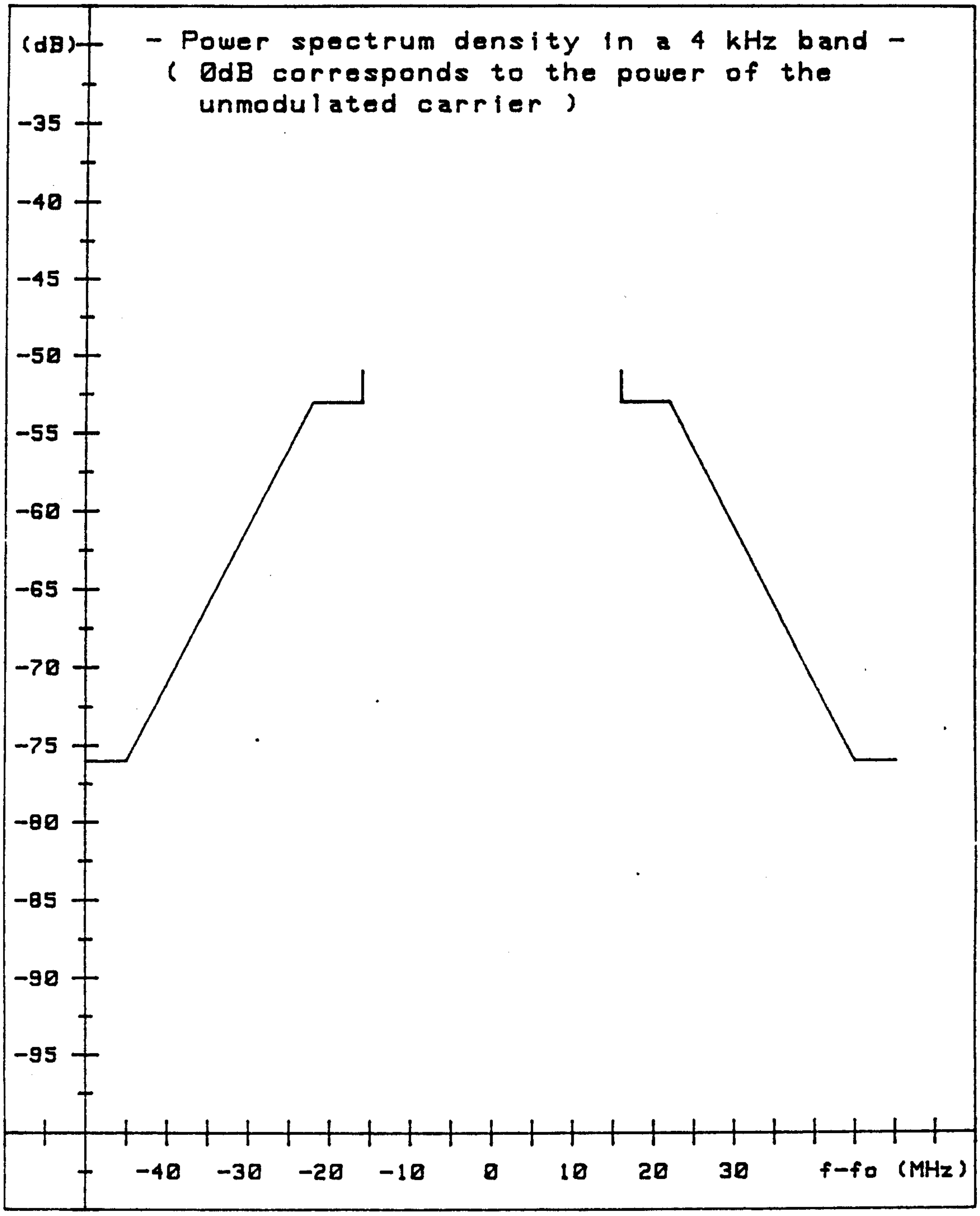


Figure 68: Mask for the overall signal at the output of the satellite

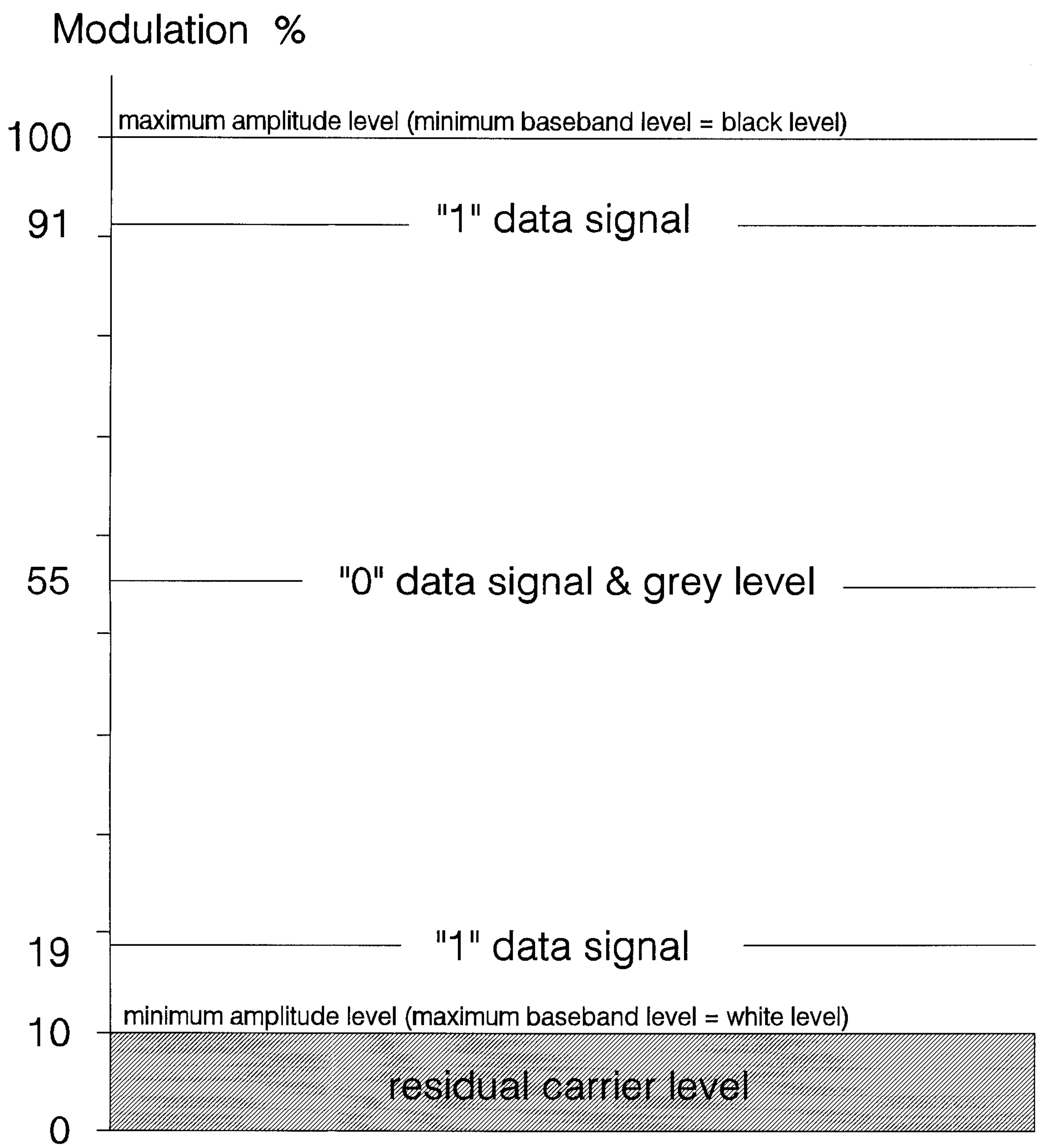


Figure 69: Modulation levels (before transmission filtering)

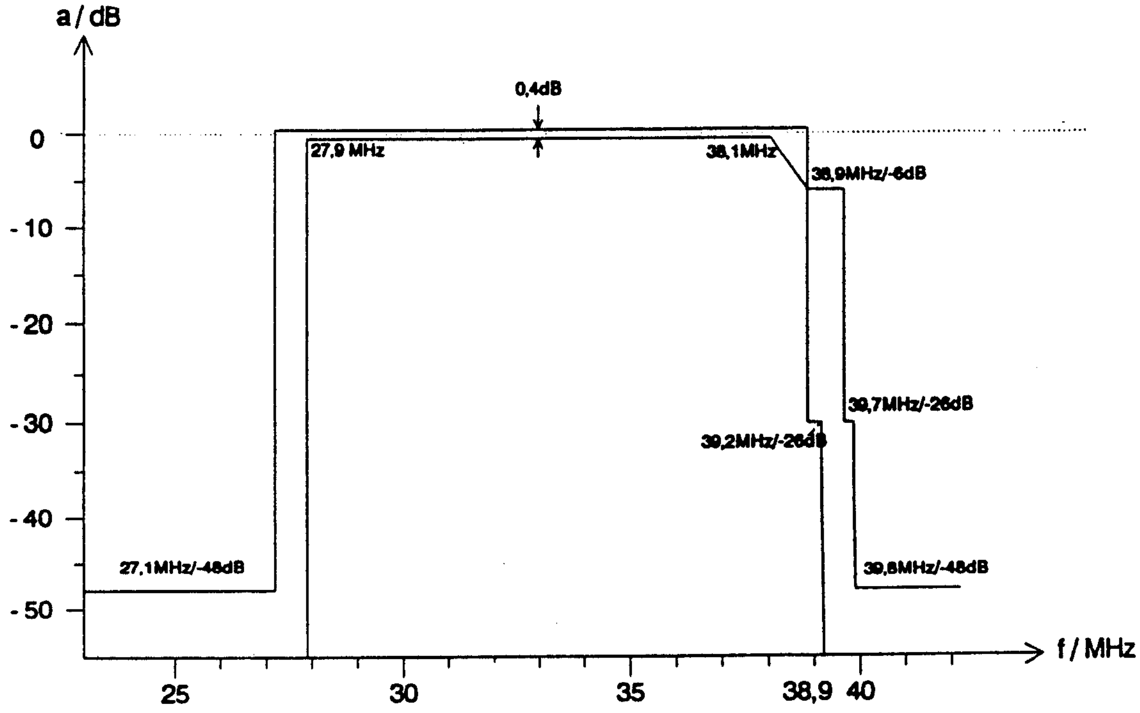


Figure 70a: Modulator IF filter, Amplitude response template

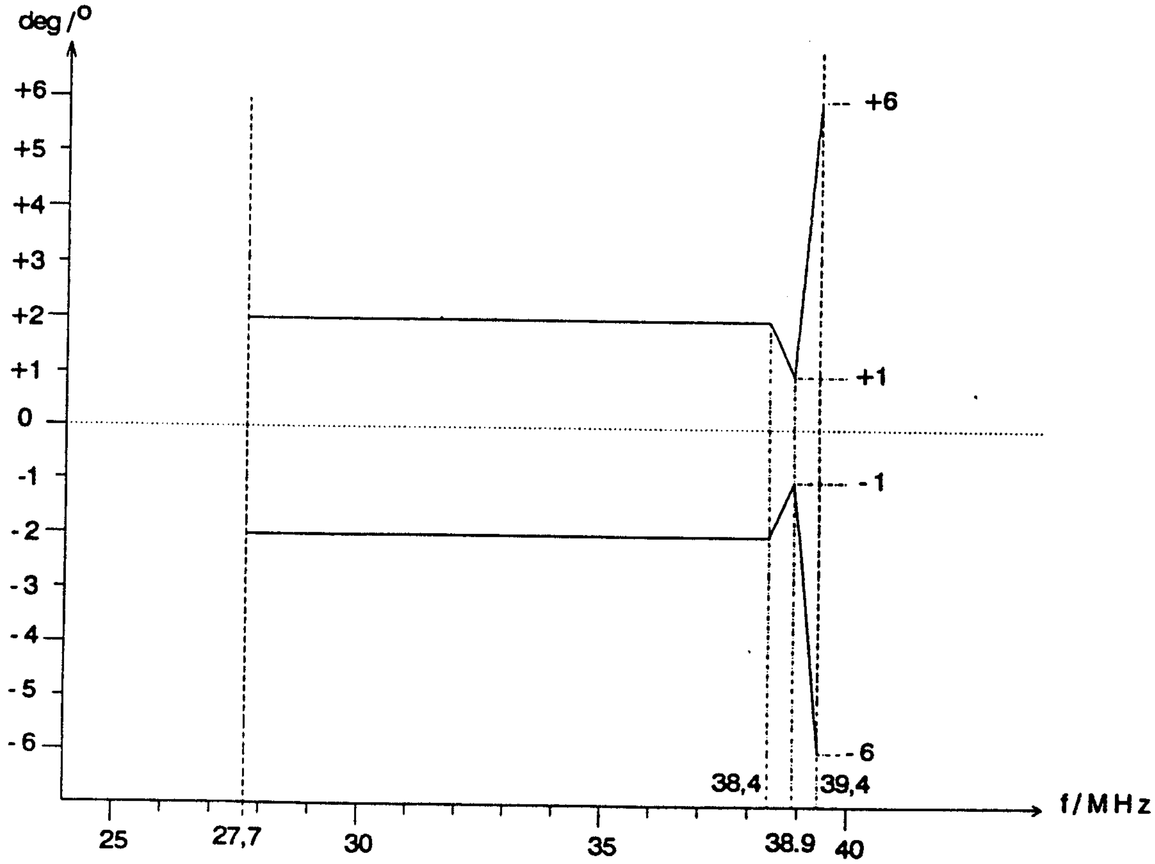


Figure 70b: Modulator IF filter, Phase response template

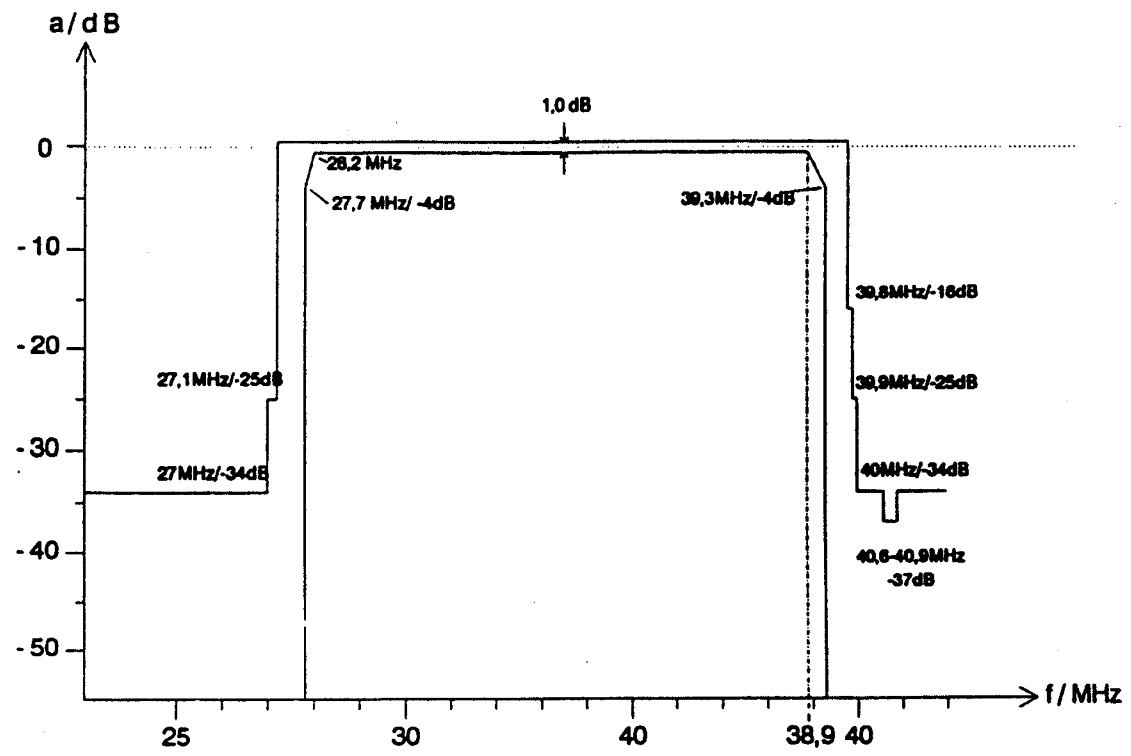


Figure 71a: Receiver IF filter, Amplitude response template

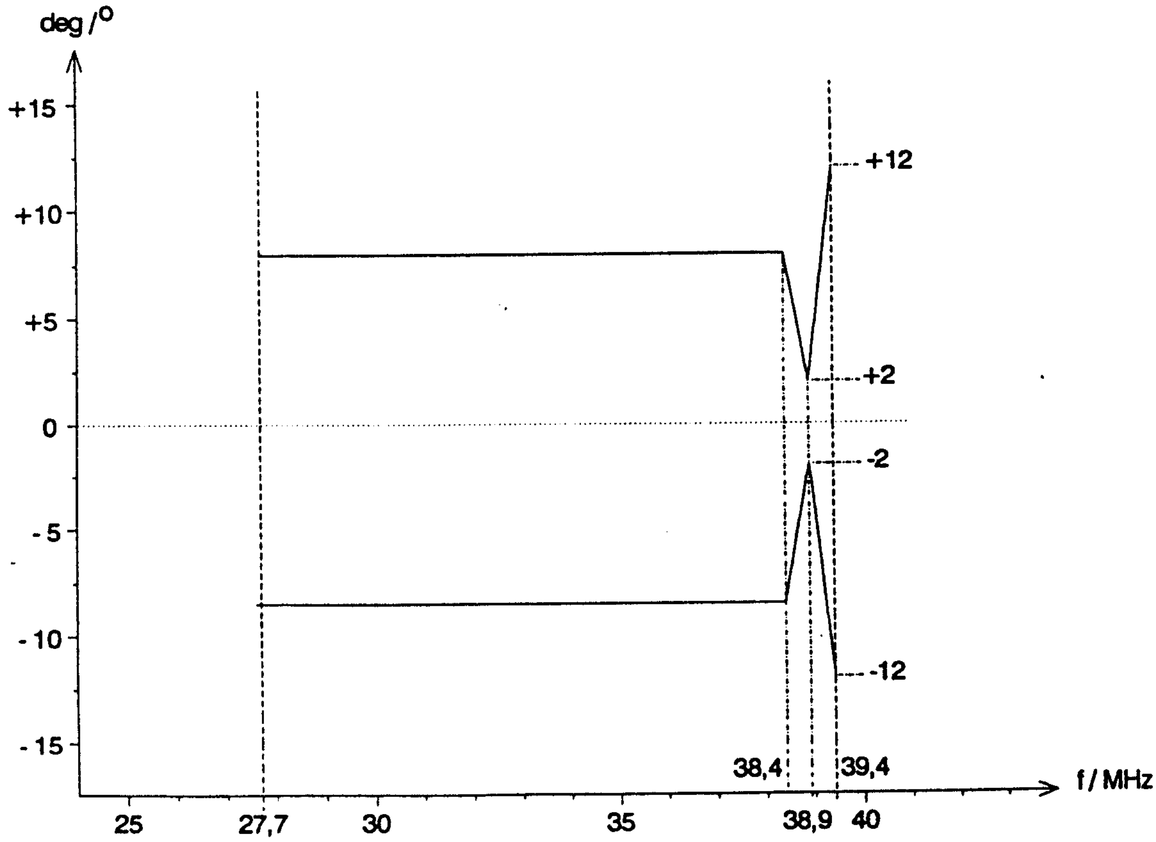


Figure 71b: Receiver IF filter, Phase response template

11 Code of Practice

11.1 Introduction

Clause 11 contains the "Code of Practice" for the broadcasting of services using the D2-HDMAC/packet specification for compatibility with basic HDMAC receivers and for backward compatibility with D2-MAC receivers according to the technical facilities of the HDMAC system.

The rules of operation which apply to services that use a particular conditional-access system are described in the specification of this conditional-access system (see EN 50 094).

The sound components may be included into the "radio services" independent from the "TV services", and from each other.

11.2 Rules applying to all services

The rules applying to all services shall be as follows:

- basic format for data burst (105 bits);
- teletext components in the field blanking interval;
- DATV/data in the field blanking interval (see subclause 11.6).

All the service components (picture, DATV, sounds, subtitles, teletext, entitlement control messages and entitlement management messages) shall be described in the broadcast signal as indicated in this ETS (in line 625, in the SI channel and in the sound interpretation blocks).

Packet number 82 in the LBI (i.e. the last packet of the LBI) shall not contain an EMM, ECM or SI packet. It should be noted that some decoders are capable of decoding only BC sound packets carried on this packet position.

11.3 Rules applying only to television services

11.3.1 General characteristics

The general characteristics shall be as follows:

- 16:9 picture format with optional transmission of panning information in accordance with the coding structure described in subclause 4.5.6;
- main sound shall be transmitted;
- DATV shall be transmitted.

11.3.2 Special arrangements for the broadcasting of TV sounds

In order to provide backward compatibility with basic D2-MAC receivers, a restriction concerning the use of the specification is defined for the broadcasting of sounds. A subset of packet addresses is defined. This subset (128 addresses) shall be reserved for TV and radio sound components (see also subclause 11.4.2) and shall be identified by the binary values 001 of the three most significant bits in the packet address (b10, b9, b8).

Inside this subset, the 7 bits of packet address shall be used as follows:

- 4 bits (b7 to b4) correspond to the coding characteristics of the sound signal;

- 3 bits (b3 to b1) give a sound channel address. If the coding characteristics of the sound component change during a transmission, bits b3 to b1 shall remain the same. The contents of interpretation blocks BI and the synchronization of the change of this content (BC1/BC2) are in accordance with this ETS (see subclause 6.7).

For the changes allowed by the BI other than coding characteristics, bits b3 to b1 may stay the same or change depending on operation choices.

The precise allocation of the bits shall be as follows:

b10	b9	b8	b7	b6	b5	b4	b3	b2	b1
0	0	1	1: stereo 0: mono	1: HQ 0: MQ	Protection 1: level 2 0: level 1	Coding 1: linear 0: companded	000 for main TV sound 7 other values for other sounds (free use)		

These rules imply a modification of the 10-bit packet address which is synchronized with any change in the coding characteristics of the sound signal. This modification consists of the modification of the value of the 4 bits (b7 to b4) described above. In accordance with the specification of the BI (procedure of updating the parameters of a sound configuration, see subclause 6.7.3), these 4 bits change their value on the third packet after the change of BC1/BC2 (as described in subclause 6.7.4).

Again, in order to provide backward compatibility with basic D2-MAC receivers, three complementary arrangements are defined:

a) In packet multiplexing, a series of four consecutive packets (continuity index from 0 to 3) containing three 120-byte sound coding blocks shall not be interleaved with packets of another sound component using 120-byte coding blocks.

b) Some information available in the BI shall be duplicated in the SI channel. This information shall be situated in the four most significant bits (see subclause 8.2.3) of the access coordinates of the parameter DCINF characterized by the parameter identifiers 'A4 or 'A5 (see subclause 8.3.1) as follows:

Access coordinates of DCINF	BI information		Significance
b13	b5 of byte 1		0: continuous 1: intermittent
b14	b4 of byte 2	additional sound channel	bit4 = 0 : mixing not intended bit4 = 1 : mixing intended
		main sound channel	bit4 = 0 : equal attenuation bit4 = 1 : significant attenuation of the main sound
b15	b5 of byte 2		0 : no scrambling 1 : scrambling
b16	b6 of byte 2		0 : free-access mode 1 : controlled-access mode

c) Procedure for changing the state from "present" to "interrupted" of an additional sound (e.g. a commentary channel):

- in the SWITCH-OFF procedure ("present" to "interrupted") (see figure 73), the broadcaster can possibly send silences (scale factor = 000 : sample value below - 36 dB) after the time P2; in any case the eight last BC packets shall contain silence. There shall not be more than eight BC packets after the BC1/BC2 transition.

- to activate the BC1/BC2 transition at SWITCH-ON time ("interrupted" to "present") and to enable the activity detection within the receiver, the broadcaster shall transmit around 100 milliseconds of BC packets before the BC1/BC2 transition.

- the operation is illustrated by figures 42, 72 and 73.

- the timing between sound component switching and UPDAT in the SI channel is described in subclause 11.6.3.

11.4 Rules applying to sound broadcasting services multiplexed with a television service

11.4.1 General characteristics of radio services

The concept of "radio services" and the technical regulations relating to their transmission are defined and specified in this ETS (in particular, in Clauses 6 and 8, concerning the coding of sound information and the SI channel). The arrangements described below comply with these basic elements.

Each radio service can comprise one or more sound channels. The radio services consist of services which shall be independent of the TV service and of each other.

This independence concerns service access in particular, whether unscrambled or conditional access.

A given D2-HDMAC/packet broadcasting channel may support 0, 1 or several radio services up to a theoretical maximum of 7, multiplexed to a TV service.

The radio services can use all the procedures offered for sound channels management by the D2-HDMAC/packet standard, in particular multiple channels and mixing channels. As far as reception is concerned, the channel selection and processing algorithms used are the same as those planned for the TV sound channels.

11.4.2 Special arrangements for the broadcasting of radio sounds

The address prefix value shall be the same as for TV sound i.e. binary value 001 for the three most significant bits (b10, b9, b8).

The seven remaining packet address bits shall be assigned as follows:

- 4 bits (b7 to b4) corresponding to the sound signal coding specifications;

- 3 bits (b3 to b1) specifying the sound channel address;

- the address 000 shall not be accessible to radio services.

The set of packet addresses for a given radio service need not be constant throughout an operating period. There may be variations in the main sound and, where appropriate, in additional sounds (e.g. service commentaries). The value of the addresses effectively used is specified in the SI channel.

The channel operator is responsible for sharing the available addresses (eight in all, including the main TV sound address) among the different services, depending on the requirements of the programmes to be transmitted. No addressing "sub-field" is pre-defined other than address 000.

The other arrangements described in subclause 11.3 also apply to the radio services. They include address management and the duplication of certain items of interpretation block data in the SI channel. Consequently, parameter fields 'A8 and 'A9 of DCINF shall be encoded using the same regulations as those applied to parameters 'A4 and 'A5.

The timing between sound component switching and UPDAT in the SI channel is described in subclause 11.6.3.

11.5 Rules applying to teletext components

Transmission of teletext components complies with Clause 7 of this ETS.

Programme providers should keep in mind that some receivers only process level 1 as specified in CCIR teletext system B defined in the addendum to CCIR Recommendation 653-1 [4].

11.6 Rules applying to overlapping windows of DATV and teletext in the FBI

The available FBI teletext component capacity may be maximized by an overlapping mechanism in the FBI which allocates a certain number of lines both to the DATV/data multiplex as well as to the FBI teletext multiplex (see subclause 4.5.5).

The implementation of the overlapping mechanism requires a horizontal allocation of the DATV/data multiplex on the FBI lines which avoids any interference to teletext decoders. The relevant values shall be assigned as follows:

- first active sample: 294;

- last active sample: 1 290.

Any sample in the FBI multiplex not allocated to be active for DATV or teletext shall be set to clamp level (logic zero).

In a TV service without FBI teletext transmission, the full size horizontal allocation of the DATV/data multiplex (FCP = 229; LCP = 1289) may be used. D2-HDMAC receivers should have the flexibility to cope with any signalled boundaries.

If so desired, line 16 can remain available for the special VPS signal.

11.7 SI channel

11.7.1 SI channel coding rules applying to all services

Mandatory information, which is necessary to identify and locate the services, shall at least be present in the non-Golay protected form (see subclause 8.5).

11.7.2 Supplementary SI coding rules applying to television and sound broadcasting services without scrambled components

Some of the data conveyed in the SI channel are duplicated and certain restrictions shall be taken into account to allow for the limited capacity of some decoders which can only receive the first packet of each data group of the SI channel.

SI data shall be encoded according to Clause 8. In addition, the following coding rules shall apply:

- all data concerning the television and sound broadcasting services (UPDATE, NWO, NWNAME, LISTX) in data group 0 (DG0) shall be transmitted in the first packet of DG0. Consequently, the permissible length of NWO and NWNAME (at the beginning of the network command) may be restricted below the maximum permissible length of 64 bytes each.

- if the size of the data group containing the command of medium priority describing the TV or Radio service (data group type DGa) is longer than one packet (excluding the data group suffix), a condensed description of the TV or Radio service shall be provided in one or two additional data groups (data group type = DGb and DGc). The size of each additional data group shall be limited to one packet.

These additional data groups shall contain a TV or Radio medium priority command regrouping one or several complete parameters describing the TV or Radio service.

The data groups DGa, DGb and DGc shall be signalled in the parameter LISTX of the TV or Radio service in data group 0. After the byte indicating the type of service, a 3-byte sequence follows for each data group:

- an eight bit field, containing the index value;

- a sixteen bit data field, containing the access coordinates (i.e. data group type and packet address of the main digital component), where the data group takes the values DGa, DGb and DGc as necessary, and the index value and the packet address are kept constant.

There is no UPDAT parameter in the commands contained in DGb and DGc because UPDAT in the command contained in DGa also concerns the data in DGb and DGc. In addition to the general principles, PI codes shall be transmitted in increasing order via the first packet of DGa and the single packets of DGb and DGc. Parameters contained completely in the first packet of DGa shall not be duplicated in order to avoid double reception of the same parameter field by the first generation receivers. Consequently, disappearance of the UPDAT parameter in the command contained in DGa can modify the content of the data groups DGb and DGc.

Figure 74 gives a diagram of the relation between various data groups describing the television service. A coding example is given in Annex 2 of Part 8 of ETS 300 250 [1].

11.7.3 Guideline to broadcasters for the choice of transmitted parameters

All the parameters that are indicated as mandatory in Clause 8 shall be transmitted when relevant.

Transmission of the parameters of the following list is highly desirable (when they are available) in order to be used by the most complete decoders: VPS, TIME, TIMD, DCOM, COMD, SREF (for television service) and PREF.

The processing of low priority information in the SI channel and the information CHID, SDFSCR and UDT in line 625 is an option of the decoder manufacturers.

In addition, it is recalled that MAC/packet family decoders should ignore all not yet defined or unknown commands or parameters in order not to be disturbed.

11.7.4 Timing between sound component switching and UPDATE in the SI channel

11.7.4.1 Definition of the reference frame

The reference frame shall contain the data group describing the component with:

- the repetition byte of data group header equal to zero;

- the last broadcasting of the UPDATE parameter on the component. The UPDATE period should last approximately 2 seconds.

11.7.4.2 Switching window

The BC1/BC2 or BC2/BC1 transition shall occur during a period of 12 frames, in which the reference frame is the sixth one.

11.8 Transitions between D2-HDMAC and D2-MAC

The following two subclauses describe the process of switching (transition) in both directions between D2-HDMAC and D2-MAC during transmission.

11.8.1 D2-HDMAC to D2-MAC transition

The high to normal definition transition shall be signalled by the MVSCG field in line 625 (bit 5 goes from 0 to 1) during 16 frames in advance of the transition and effectively occurs when the FCNT modulo 16 is equal to 1 (see subclause 4.5.3).

After transition, the DATV description in SI channel shall not exist. DATV disappearance shall be signalled by the SI parameter updating mechanism (see subclause 8.3.1) and the end of this updating period should effectively occur at the same time as in MVSCG with an accuracy of $\pm$ 8 frames.

DATV/data multiplex subframe disappearance shall be signalled by the RDF updating mechanism (see subclause 4.5.4) and the end of this updating period shall effectively occur at the first time after the transition that the FCNT modulo 128 is equal to 1. After this period, neither the DATV/data multiplex in TDM nor its description in RDF (TDMCID '40 and '41) shall exist.

The D2-HDMAC to D2-MAC transition is illustrated in figure 75.

11.8.2 D2-MAC to D2-HDMAC transition

The normal to high definition transition shall be signalled by the MVSCG field in line 625 (bit 5 goes from 1 to 0) during 16 frames in advance of the transition and effectively occurs when the FCNT modulo 16 is equal to 1 (see subclause 4.5.3).

After transition, the DATV description in the SI channel shall exist. DATV appearance shall be signalled by the SI parameter updating mechanism (see subclause 8.3.1) and the end of this updating period should effectively occur at the same time as in MVSCG with an accuracy of $\pm$ 8 frames.

After transition, the DATV/data multiplex in TDM and its description in RDF (TDMCID '40 and '41) shall exist. DATV/data multiplex subframe appearance shall be signalled by the RDF updating mechanism (see subclause 4.5.4) and the end of this updating period shall effectively occur at the last time before the transition that the FCNT modulo 128 is equal to 1.

DATV packets should be broadcast at least three frames before the transition. If not, an HDMAC decoder is able to completely process the high definition picture by the second odd frame after transition.

The D2-MAC to D2-HDMAC transition is illustrated in figure 76.

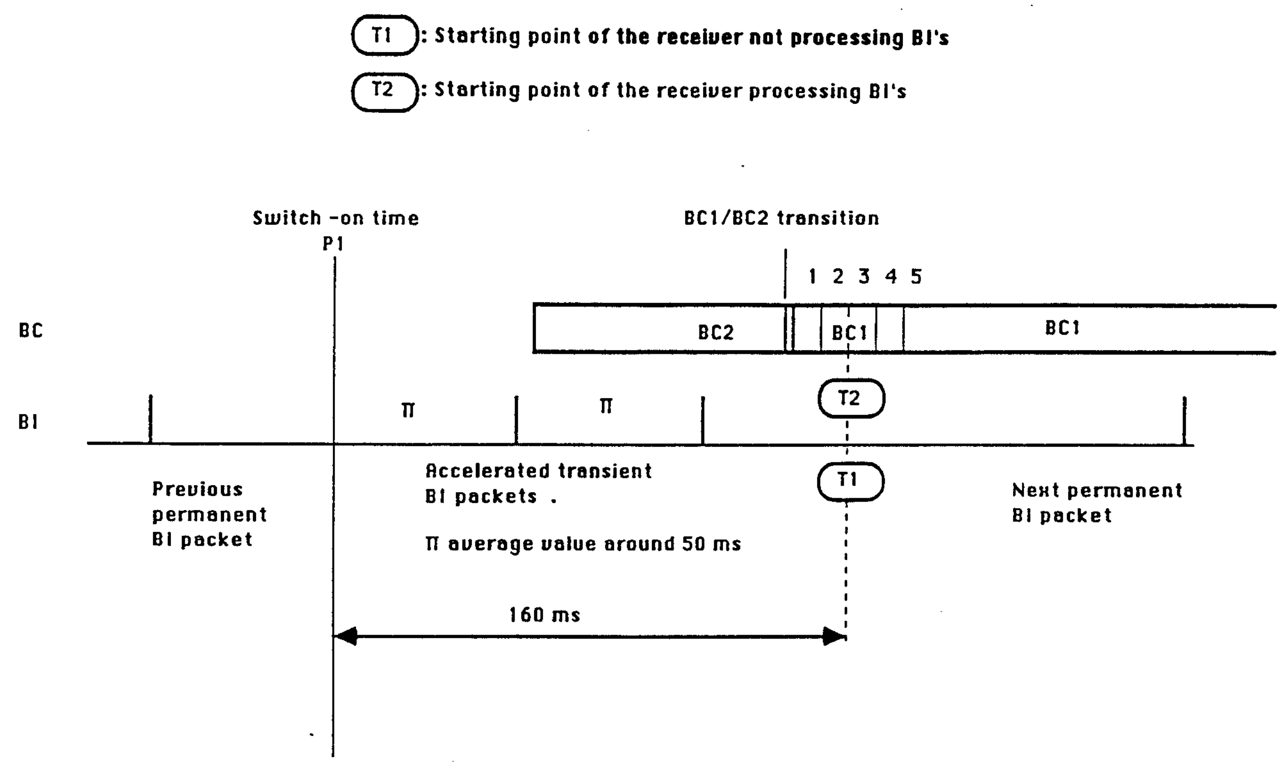
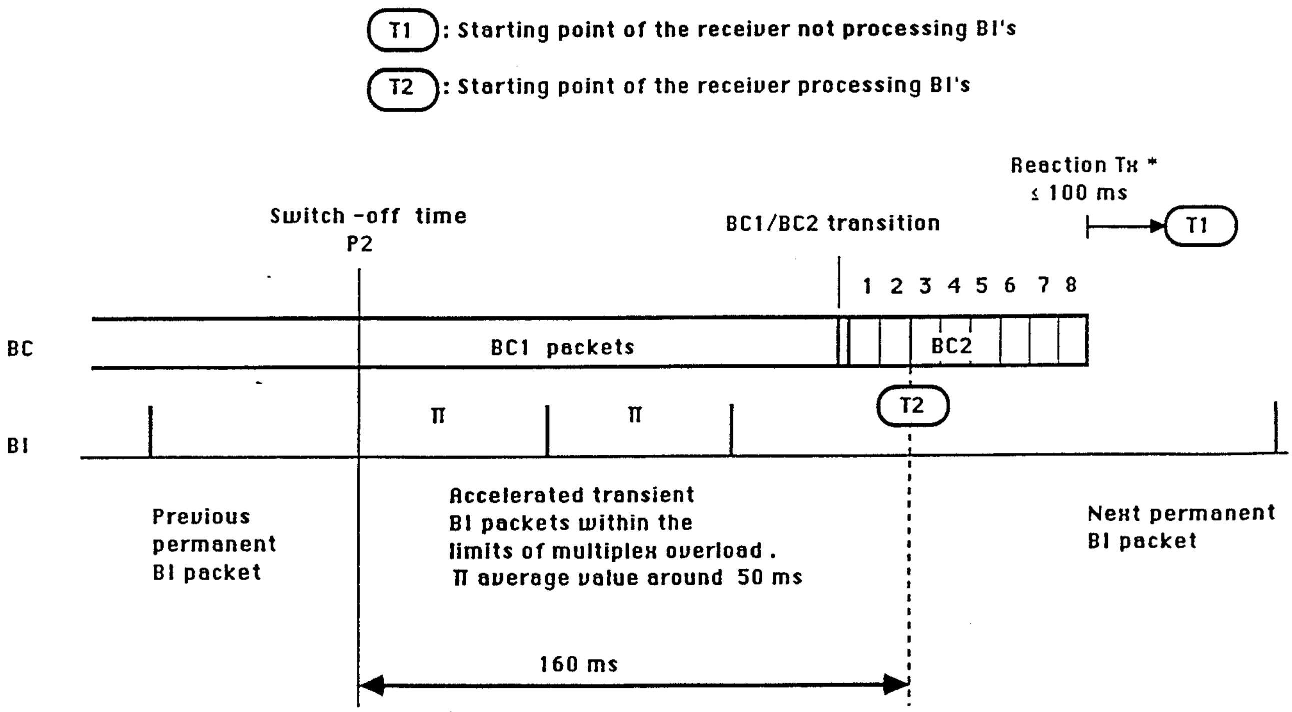
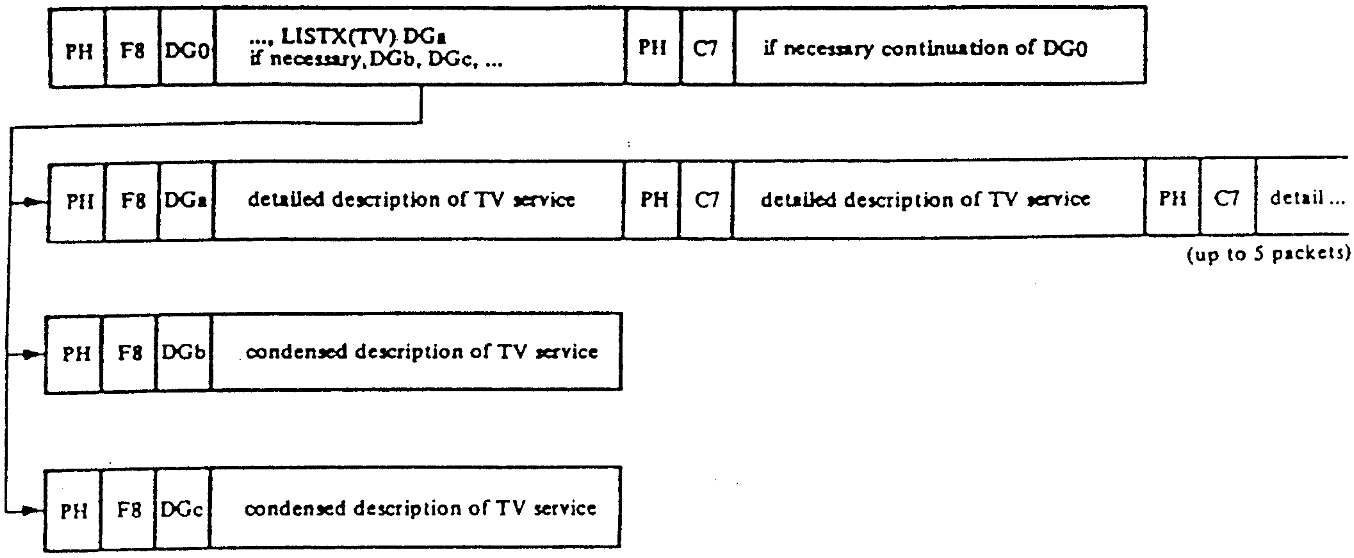


Figure 72: Switch-on procedure for the commentary channel



* Tx is the time necessary for a receiver not proccessing BI's to detect the inactivity of the channel

Figure 73: Switch-off procedure for the commentary channel



PH:	packet header;
'F8:	packet type (synchronizing packet, non-Golay coded);
'C7:	packet type (following packets, non-Golay coded);
DG0:	data group type 0;
DGa, DGb, DGc:	data group type a, b, c.

Figure 74: Relation between data groups describing the television service

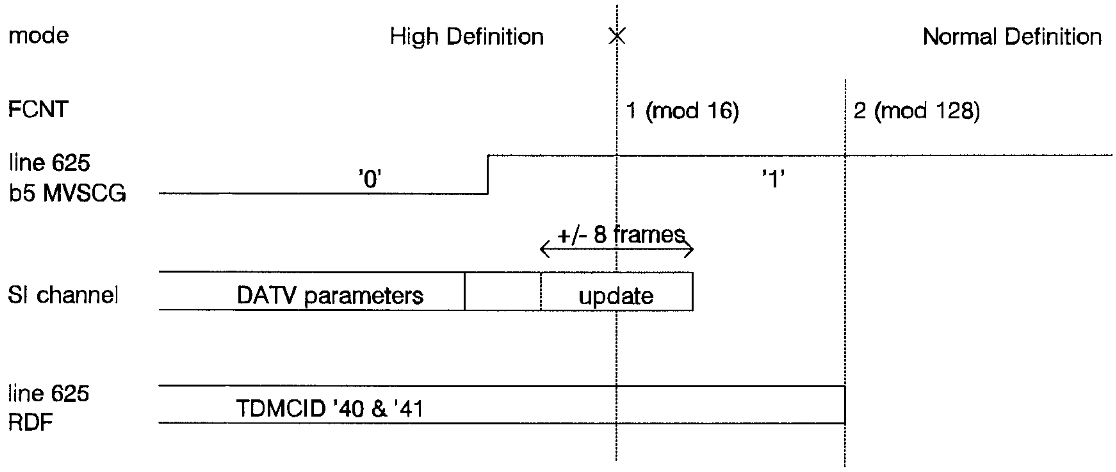


Figure 75: D2-HDMAC to D2-MAC transition

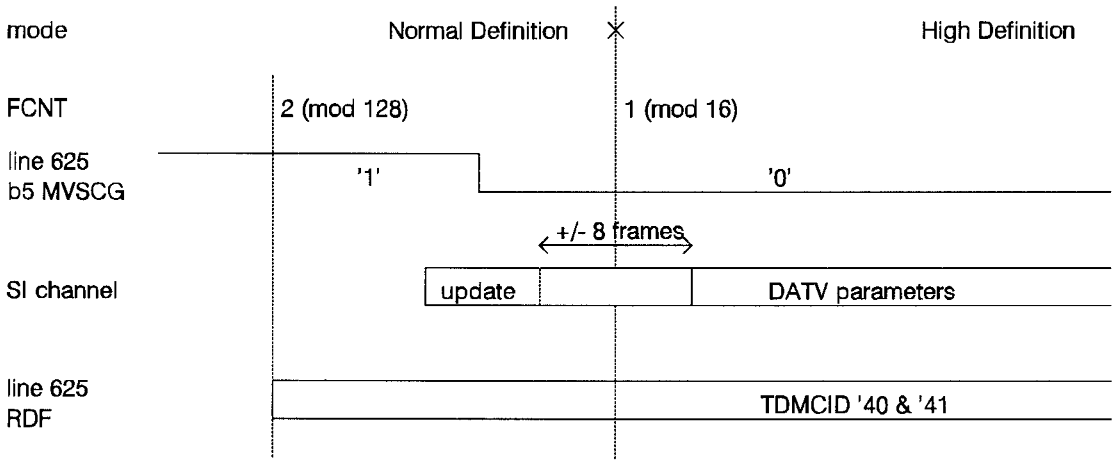


Figure 76: D2-MAC to D2-HDMAC transition

Annex A (normative): Code tables

A.1 Introduction

A number of special codes are defined in Clause 8, with reference to code tables or coding methods. The corresponding details are included in the present normative annex.

A.2 CHID codes

To be allocated (see subclause 4.5.3).

A.3 Coding of characters used for various text messages transmitted in the dedicated channel (see subclause 8.2)

A.3.1 Repertoires

The basic character repertoire to be used for the following displayable texts: (NWO) Network origin, (NWNAME) Network name, TIME, Name of service, Name of programme item, and (DCOM) Direct commentary, transmitted for service-identification purposes within the dedicated channel shall be that given in Appendix 2 to EBU Tech. doc 3232 [7]. This set of 221 characters is "the complete repertoire of Latin-based characters required (for teletext broadcasting) by EBU Member-organizations", and contains the characters needed for composing texts in twenty-five European languages. For certain non Latin-based alphabets, suitable repertoires are also given in the above mentioned document.

A.3.2 Coding tables

Each of the three repertoires contains about 220 characters; this exceeds the capacity of a single seven-bit coding table, but is within that of an eight-bit coding table. The transmission channel of the service-identification system is organized as eight-bit bytes, and it is therefore appropriate to employ an eight-bit coding table directly. The set of coding tables already devised for use in the EBU Radio Data System RDS (Appendix 5 of EBU Tech doc 3244 [8]) shall be adopted directly for use in the service identification system. They are reproduced here as figures A.1, A.2 and A.3. In this context, it is of interest to note that the seven-bit ISO 646 coding table, which was originally proposed for this application, is a sub-set of each of the eight-bit tables, except for the characters in a few of the positions for which provision for alternative assignments is made in ISO 646. Decoders should not react to any bit combinations not defined in Clause A.3.

A.3.3 Control characters

In order to simplify the receiver, the number of display attributes within the SI system has been kept to a minimum; in particular, no provision is made for colour attributes. In accordance with EBU Tech. doc 3244 [8], the applicable coding table shall be determined by the non-spacing control characters as follows:

- 0/15, 0/15: selection of code table of figure A.1;
- 0/14, 0/14: selection of code table of figure A.2;
- 1/11, 6/14: selection of code table of figure A.3.

In default of a control character, the code table shown in figure A.1 shall be applicable. Receivers should not respond to any other characters in columns 0 and 1 of the code tables, which are reserved for possible future extensions in accordance with EBU Tech. doc 3244 [8].

A.3.4 Display of clear text messages

The maximum length of any clear text message shall be 64 characters. If a display device which has fewer than 64 character positions is used, then memory should be provided in the receiver/decoder so that the elements of the message can be displayed sequentially. This may, for example, be done by displaying elements of text (of length to fit the available display) one at a time in sequence or, alternatively by scrolling the displayed characters of the message from right to left.

If a teletext decoder is used to display these clear-text messages, it may be necessary to provide suitable transcoding and reformatting facilities in the receivers.

A.4 Coding of programme type (see subclause 8.2.2)

A.4.1 Introduction

In subclause 8.2.2, provision is made for the transmission of one up to 256 codes to identify the nature of the content of each programme broadcast in a service. It is therefore necessary to establish a code table to define the meaning of the codes available for this purpose. This code table is given below.

A.4.2 Inventory of programme types

In order to establish an inventory of the programme types that it is likely to be necessary to identify in this way, reference was made to the classification developed by the EBU Statistics Group, and to the programme-type code table for sound broadcasting given in Appendix 6 to EBU Tech. doc 3244 [8]. The EBU Statistics Group has produced an elaborate multi-dimensional classification system known as "ESCORT" (EBU doc. SPG 2353), which is intended to resolve the problems arising from the coexistence of different principles of classification. As the number of codes required to implement ESCORT in full is many times greater than that available, codes have been defined for only the more important subdivisions of the two categories that are considered to be most useful in this context, "INTENDED AUDIENCE" and "CONTENT".

A.4.3 Compilation of the code table

For maximum compatibility, the higher-level categories of the "ESCORT" system have in general been adopted directly, but in a few cases two or more categories have been grouped together, so as to obtain a coding system suitable for a binary transmission system. Where a separate category of "other" programme types is not provided, the next higher level in the hierarchy should be adopted. The reference numbers used in the ESCORT system are shown for convenience.

In accordance with the "ESCORT" system, the codes for the programme type have been classified on the basis of various different principles, in accordance with the table below. Hexadecimal code 00 has been reserved for use when information on the programme type is not available, as required by subclause 8.2.2 and hexadecimal code 3F has been assigned to identify alarm/emergency messages.

Tables A.1, A.2 and A.3 are coded according to the following principles of classifications:

Code (Hexadecimal)	Principle of classification
00	Information not available;
01-3E	Intended audience;
3F	Alarm/emergency identification;
40-7F	Content;
80-BF	Codes specific to each service (to be defined);
C0-FF	Codes specific to each service.

Table A.1: Code table for programme type

Code (Hexadecimal)	Programme type	ESCORT ref. number
	INTENDED AUDIENCE	
08	general audience	2.0.0
	special groups:	
10	ethnic and immigrant groups	1.1.0
11	ethnic groups	1.1.1
12	immigrant groups	1.1.2
18	age groups	1.2.0
19	children (0 - 13 years)	1.2.1
1A	young people (14 years or more)	1.2.2
1F	retired people	1.3.0
20	disabled people	1.4.0
21	blind people	1.4.1
22	deaf people	1.4.2
28	householders	1.5.0
30	occupational status groups	1.6.0
31	unemployed people	1.6.1
32	students	1.6.2
33	farmers	1.6.3
34	fishermen and sailors	1.6.4
38	travellers	1.7.0
39	motorists	1.7.1
3A	tourists	1.7.2
	CONTENT	
40	public affairs	
41	general domestic affairs	1.1.0
42	legal and social affairs	1.2.0
43	economic, industrial and financial affairs	1.3.0
44	housing, environment and health affairs	1.4.0
45	communication affairs	1.5.0
46	educational and cultural affairs	1.6.0
47	international relations and defence affairs	1.7.0
48	science and the humanities	2.0.0
49	natural sciences	2.1.0
4A	social sciences	2.2.0
4B	humanities	2.3.0
4C	other sciences or humanities	2.9.0
50	music	3.1.0
51	serious music	3.1.1
52	light classical music	3.1.2
53	light music	3.1.3
54	jazz	3.1.4
55	folk music	3.1.5
56	rock music	

(continued)

Table A.1: Code table for programme type (concluded)

Code (Hexadecimal)	Programme type	ESCORT ref. number
57	other music	3.1.9
58	drama, arts	3.0.0
5A	ballet and dance	3.2.0
5B	drama	3.3.0
5C	literature/poetry	3.4.0
5D	media affairs	3.5.0
5E	painting, sculpture, architecture	3.6.0
5F	other drama, arts	3.9.0
60	philosophies of life	
61	Christian religion	4.1.0
62	non-Christian religion	4.2.0
63	non-religious philosophy of life	4.3.0
67	other philosophies of life	4.9.0
68	sports	5.0.0
69	non-instrumental ball games	5.1.0
6A	instrumental ball games	5.2.0
6B	winter sports	5.3.0
6C	water sports	5.4.0
6D	racing and equestrian sports	5.5.0
6E	athletics	5.6.0
6F	martial arts	5.7.0
70	leisure and hobbies	6.0.0
71	do-it-yourself	6.1.0
72	gardening	6.2.0
73	tourism	6.3.0
74	keep fit	6.4.0
77	other leisure or hobbies	6.9.0
78	light entertainment, folklore and human interest	7.0.0
7A	light entertainment	7.1.0
7B	folklore/festivities	7.2.0
7C	human interest	7.3.0
7F	other light entertainment, etc.	7.9.0

75 out of 128 available codes between 00 and 7F have been assigned. 53 spare codes applicable to all services, and 128 codes applicable to individual services, remain to be assigned.

A.5 Coding of languages (see subclause 8.2.3)

A.5.1 Introduction

In subclause 8.2.3, provision is made for the transmission of up to 256 codes to identify the language used for each component of a broadcast service. Half of these are common to all components; the other half are specific to each type of component. It is therefore necessary to establish a code table to define the meaning of the codes available for this purpose. The code table common to all components is given below.

A.5.2 Inventory of languages

In order to establish an inventory of the languages that it is likely to be necessary to identify in this way, two sources were consulted: EBU Tech. Doc 3232 [7] and the European section of the 1983 edition of the annual "World Radio-TV Handbook". On this basis, a list of some 42 European languages written in Latin-based characters, and some 60 other languages, that are currently used or envisaged to be used for broadcasting in Europe was compiled.

A.5.3 Compilation of the code table

In order to provide an impartial basis for the allocation of codes, the codes for European languages written in Latin-based alphabets have been assigned consecutively, starting at "01", in the alphabetical order of the names of the languages in the languages themselves. The codes for the other languages are assigned in reverse alphabetical order of the English names of the languages, starting at hexadecimal "7F". In this way, codes for additional languages can be assigned consecutively until all spare codes have been used, while maintaining the subdivision into "European Latin-based" and "Other" sections. In accordance with subclause 8.2.3, code '00 shall indicate that no language information is provided or that such information is not meaningful in the case of the component concerned. The sixteen codes '30 to '3F are reserved for assignment, if required, to languages that are not already included in the code table, on a national basis.

Table A.2: Code table for the languages common to all service components
European languages written in Latin-based alphabets

Code (Hexadecimal)	Language	Code (Hexadecimal)	Language
00	Unknown / not applicable	20	Polish
01	Albanian	21	Portuguese
02	Breton	22	Romanian
03	Catalan	23	Romansh
04	Croatian	24	Serbian
05	Welsh	25	Slovak
06	Czech	26	Slovene
07	Danish	27	Finnish
08	German	28	Swedish
09	English	29	Turkish
0A	Spanish	2A	Flemish
0B	Esperanto	2B	Walloon
0C	Estonian	2C	
0D	Basque	2D	
0E	Faroese	2E	
0F	French	2F	
10	Frisian	30	)
11	Irish	31	)
12	Gaelic	32	)
13	Galician	33	)
14	Icelandic	34	)
15	Italian	35	)
16	Lappish	36	Reserved for
17	Latin		national
18	Latvian	38	assignment
19	Luxembourgian	39	)
1A	Lithuanian	3A	)
1B	Hungarian	3B	)
1C	Maltese	3C	)
1D	Dutch	3D	)
1E	Norwegian	3E	)
1F	Occitan	3F	)

Table A.3: Other languages

Code (Hexadecimal)	Language	Code (Hexadecimal)	Language
7F	Amharic	5F	Marathi
7E	Arabic	5E	Ndebele
7D	Armenian	5D	Nepali
7C	Assamese	5C	Oriya
7B	Azerbijani	5B	Papamiento
7A	Bambora	5A	Persian
79	Belorussian	59	Punjabi
78	Bengali	58	Pushtu
77	Bulgarian	57	Quechua
76	Burmese	56	Russian
75	Chinese	55	Ruthenian
74	Churash	54	Serbo-Croat
73	Dari	53	Shona
72	Fulani	52	Sinhalese
71	Georgian	51	Somali
70	Greek	50	Sranan Tongo
6F	Gujurati	4F	Swahili
6E	Gurani	4E	Tadzhik
6D	Hausa	4D	Tamil
6C	Hebrew	4C	Tatar
6B	Hindi	4B	Telugu
6A	Indonesian	4A	Thai
69	Japanese	49	Ukrainian
68	Kannada	48	Urdu
67	Kazakh	47	Uzbek
66	Khmer	46	Vietnamese
65	Korean	45	Zulu
64	Laotian	44	
63	Macedonian	43	
62	Malagasay	42	
61	Malaysian	41	
60	Moldavian	40	Background sound / clean feed

8-BIT CODING TABLES OF DISPLAYABLE CHARACTERS FOR THE SERVICE-IDENTIFICATION SYSTEM

The complete EBU Latin-based repertoire

					Displayable characters from the code table of ISO Norm 646:						Additional displayable characters for: EBU common-core (7 languages)		Additional displayable characters for: Complete Latin-based repertoire (25 languages)					
				b8	0	0	0	0	0	0	1	1	1	1	1	1	1	1
				b7	0	0	1	1	1	1	0	0	0	0	1	1	1	1
				b6	1	1	0	0	1	1	0	0	1	1	0	0	1	1
				b5	0	1	0	1	0	1	0	1	0	1	0	1	0	1
b4	b3	b2	b1		2	3	4	5	6	7	8	9	10	11	12	13	14	15
0	0	0	0	0		0	@	P	‖	p	á	â	ª	º	Á	Â	Ã	ã
0	0	0	1	1	!	1	A	Q	a	q	à	ä	α	¹	À	Ä	Å	å
0	0	1	0	2	"	2	B	R	b	r	é	ê	©	²	É	Ê	Æ	æ
0	0	1	1	3	#	3	C	S	c	s	è	ë	‰	³	È	Ë	Œ	œ
0	1	0	0	4	¤	4	D	T	d	t	í	î	Ğ	±	Í	Î	Ŷ	ŵ
0	1	0	1	5	%	5	E	U	e	u	ì	ï	ě	İ	Ì	Ï	Ý	ý
0	1	1	0	6	&	6	F	V	f	v	ó	ô	ň	ń	Ó	Ô	Õ	õ
0	1	1	1	7	'	7	G	W	g	w	ò	ö	ő	ű	Ò	Ö	Ø	ø
1	0	0	0	8	(	8	H	X	h	x	ú	û	π	µ	Ú	Û	Þ	þ
1	0	0	1	9	)	9	I	Y	i	y	ù	ü	₠	¿	Ù	Ü	Ŋ	ŋ
1	0	1	0	10	*	:	J	Z	j	z	Ñ	ñ	£	÷	Ř	ř	Ŕ	ŕ
1	0	1	1	11	+	;	K	[(I)	k	{ (I)	Ç	ç	$	°	Č	č	Ć	ć
1	1	0	0	12	,	<	L	\	l	|	Ş	ş	←	¼	Š	š	Ś	ś
1	1	0	1	13	−	=	M	] (I)	m	} (I)	ß	ğ	↑	½	Ž	ž	Ź	ź
1	1	1	0	14	.	>	N		n		¡	ı	→	¾	Đ	đ	Ŧ	ŧ
1	1	1	1	15	/	?	O		o		Ĳ	ĳ	↓	§	Ľ	ŀ	ð	

Figure A.1:
Code table for 218 displayable characters forming the complete EBU Latin-based repertoire. The characters shown in positions marked (I) in the table are those of the "international reference version" of ISO-646 that do not appear in the "complete Latin-based repertoire" given in Appendix 2 of EBU Tech. doc 3244 [8]

Combined repertoire: Latin-based common-core, Cyrillic and Greek

					Latin (ISO Norm 646)						EBU common-core		Part of the EBU complete Latin-based repertoire		Cyrillic etc.		Greek	
				b8	0	0	0	0	0	0	1	1	1	1	1	1	1	1
				b7	0	0	1	1	1	1	0	0	0	0	1	1	1	1
				b6	1	1	0	0	1	1	0	0	1	1	0	0	1	1
				b5	0	1	0	1	0	1	0	1	0	1	0	1	0	1
b4	b3	b2	b1		2	3	4	5	6	7	8	9	10	11	12	13	14	15
0	0	0	0	0		0	@	P	‖	p	á	â	ª	º	Є	ý	Π	π
0	0	0	1	1	!	1	A	Q	a	q	à	ä	ľ	¹	Я	Љ	α	Ω
0	0	1	0	2	"	2	B	R	b	r	é	ê	©	²	Б	ď	β	ρ
0	0	1	1	3	#	3	C	S	c	s	è	ë	‰	³	Ч	Ш	ψ	σ
0	1	0	0	4	¤	4	D	T	d	t	í	î	Ă	±	Д	Ц	δ	τ
0	1	0	1	5	%	5	E	U	e	u	ì	ï	ě	İ	Э	Ю	ε	ξ
0	1	1	0	6	&	6	F	V	f	v	ó	ô	ň	ń	Ф	Щ	φ	Θ
0	1	1	1	7	'	7	G	W	g	w	ò	ö	ő	ű	Ѓ	Њ	γ	Γ
1	0	0	0	8	(	8	H	X	h	x	ú	û	ť	ţ	Ъ	Џ	η	Ξ
1	0	0	1	9	)	9	I	Y	i	y	ù	ü	ę	¿	И	Й	ι	υ
1	0	1	0	10	*	:	J	Z	j	z	Ñ	ñ	£	÷	Ж	З	Σ	ζ
1	0	1	1	11	+	;	K	[(1)	k	{ (1)	Ç	ç	$	°	Ќ	č	κ	ς
1	1	0	0	12	,	<	L	\	l	|	Ş	ş	←	¼	Л	š	λ	Λ
1	1	0	1	13	-	=	M	] (1)	m	} (1)	β	ğ	↑	½	Ћ	ž	μ	Ψ
1	1	1	0	14	.	>	N	—	n	—	İ	ı	→	¾	Ђ	đ	ν	Δ
1	1	1	1	15	/	?	O	—	o		IJ	ij	↓	§	Ы	ć	ω	

Figure A.2:
Code table for a combined repertoire consisting of the EBU common-core Greek and upper-case Cyrillic alphabets (together with certain characters from the EBU complete Latin-based repertoire, and the lower-case characters required for texts in Serbo-Croat, Slovenian, Slovakian, Hungarian and Romanian). The characters shown in positions marked (I) in the table are those of the "international reference version" of ISO-646 that do not appear in the "complete Latin-based repertoire" given in Appendix 2 of EBU Tech. doc 3244 [8]

Combined repertoire for Latin, Arabic, Hebrew, Cyrillic and Greek

					Latin (ISO Norm 646)						Arabic		Hebrew		Cyrillic etc.		Greek	
				b8	0	0	0	0	0	0	1	1	1	1	1	1	1	1
				b7	0	0	1	1	1	1	0	0	0	0	1	1	1	1
				b6	1	1	0	0	1	1	0	0	1	1	0	0	1	1
				b5	0	1	0	1	0	1	0	1	0	1	0	1	0	1
b4	b3	b2	b1		2	3	4	5	6	7	8	9	10	11	12	13	14	15
0	0	0	0	0		0	@	P	‖	p	ب	ظ	א	ן	Є	ý	Π	π
0	0	0	1	1	!	1	A	Q	a	q	ت	ع	ב	ס	Я	Љ	α	Ω
0	0	1	0	2	"	2	B	R	b	r	ة	غ	ג	ע	Б	ď	β	ρ
0	0	1	1	3	#	3	C	S	c	s	ث	ف	ד	פ	Ч	Ш	ψ	σ
0	1	0	0	4	¤	4	D	T	d	t	ج	ق	ה	ף	Д	Ц	δ	τ
0	1	0	1	5	%	5	E	U	e	u	ح	ك	ו	צ	Э	Ю	ε	ξ
0	1	1	0	6	&	6	F	V	f	v	خ	ل	ז	ץ	Ф	Щ	φ	Θ
0	1	1	1	7	'	7	G	W	g	w	د	م	ח	ק	Ѓ	Њ	γ	Γ
1	0	0	0	8	(	8	H	X	h	x	ذ	ن	ט	ר	Ъ	Џ	η	Ξ
1	0	0	1	9	)	9	I	Y	i	y	ر	ه	י	ש	И	Й	ι	υ
1	0	1	0	10	*	:	J	Z	j	z	ز	و	כ	ת	Ж	З	Σ	ζ
1	0	1	1	11	+	;	K	[(1)	k	{ (1)	س	ي	ך	°	Ќ	č	χ	ς
1	1	0	0	12	,	<	L	\	l	|	ش	←	ל	¼	Л	š	λ	Λ
1	1	0	1	13	-	=	M	] (1)	m	} (1)	ص	↑	מ	½	Ћ	ž	μ	Ψ
1	1	1	0	14	.	>	N		n		ض	→	ם	¾	Ђ	đ	ν	Δ
1	1	1	1	15	/	?	O		o	\	ط	↓	נ	§	Ы	ć	ω	

Figure A.3:
Code table for a combined repertoire consisting of the ISO-646 Latin-based alphabet, Greek, upper-case Cyrillic, Hebrew and Arabic. The characters shown in positions marked (I) in the table are those of the "international reference version" of ISO-646 that do not appear in the "complete Latin-based repertoire" given in Appendix 2 of EBU Tech doc. 3244 [8]

Annex B (normative): Specification of the 656+ interface

B.1 Introduction

This Clause specifies a digital interface format based on CCIR Recommendation 656 [5], further referred to as "656+" (see also figure B.1). This interface facilitates interconnection between equipment such as a BRE, a digital VTR, a panning vector inserter and a D2-HDMAC multiplexer.

B.2 General description of the 656+ interface

The 656+ interface shall make use of the format specified in CCIR Recommendation 656 [5] for carrying synchronization, video samples and ancillary (ANC) data. For this annex 8-bit coding is assumed.

The sync part shall conform to the CCIR Recommendation 656 [5] for horizontal and vertical syncs and field parity. However, since the frame parity of the D2-HDMAC multiplexer is locked to the BRE frame parity, D2-HDMAC requires an additional image parity (IP). The IP information shall be included in the ancillary data (see subclause B.5.1).

In the video part, Y, CR and CB matrixing shall comply to CCIR Recommendation 601-2 [6].

The data part carries the following information:

a) HDMAC Identification;

b) Image Parity;

c) DATV data structured in 91-byte blocks for insertion in the packet multiplex in the FBI of the D2-MAC/Packet signal;

d) in addition, panning vectors, if they are generated by the production center for insertion within the D2-HDMAC/packet frame.

B.3 Compatibility of 656+ with CCIR Recommendation 656 and digital video recording

Signalling conforming to the 656+ interface may be recorded on digital component VTR's without modification, since the headroom for FBI recording allows the recording of ancillary data such as DATV and panning vectors.

B.4 Structure of the line

Two types of lines are considered:

a) video lines (lines 23 to 310 inclusive and lines 336 to 623 inclusive) shall have the following arrangement: " CB, Y, CR, Y ";

- the arrangement of CR and CB at the 656+ output interface of the BRE takes account of the vertical subsampling of colour signals in D2-HDMAC. The colour difference samples which are not transmitted in D2-HDMAC are the result of vertical interpolation between those of the adjacent lines:

    - an odd line shall carry a true CR, while CB is an average of the two CB vertical adjacent samples;

    - an even line shall carry a true CB, while CR is an average of the two CR vertical adjacent samples;

- this allows the signal to be normally displayable and recordable on digital 625/50/2 component recorders;

- for test purposes, the average may be replaced by decimal level 128.

b) field blanking lines (lines 11 to 22 inclusive and lines 324 to 335 inclusive). For these lines:

  - two types of data can be present:

    1) DATV data blocks, which shall have a rate of three blocks per line, preceded by a byte indicating the image parity IP. See subclause B.5.1. The DATV data shall be formatted as follows:

       - a header which shall consist of '00, 'FF, 'FF (hexadecimal notations), followed by codes TT, MM, LL where:

         - TT shall identify, according to its value, the type of data which follows;

         - MM and LL form a length indicator whose value corresponds to the number of ancillary data bytes;

       - three DATV data blocks at the rate of three blocks per line, preceded by a byte indicating the image parity.

    2) an optional panning vector data block at the rate of one per image (or two if some redundancy is introduced for recording on D1 tapes). See subclause B.5.2. The panning data shall be formatted as follows:

       - a header which shall consist of '00, 'FF, 'FF (hexadecimal notations), followed by codes TT1, TT2, TT3, LL1, LL2, where:

         - TT1, TT2, TT3 shall identify, according to its value, the type of data which follows;

         - LL1 and LL2 form a length indicator whose value corresponds to the number of ancillary data bytes;

       - a panning vector data block inserted at the rate of one per image (or two if some redundancy is introduced for recording on D1 tapes).

  - the completion of the 1440 bytes shall be achieved by the repeated sequence: '10,'80,'10,'80,... when there is no panning vector or '80,'10,'80,'10,...in the opposite case.

The format of a field blanking line having only DATV data blocks is given in figure B.2. The format of a field blanking line having both the DATV data blocks and a panning vector block is given in figure B.3.

B.5 Content of DATV and panning data

B.5.1 DATV data

For DATV, the header shall be coded as follows:

- TT shall have the hexadecimal value 'DA;
- MM, LL shall have the values corresponding to the number of DATV bytes per line, which for the here specified case of 553 bytes per recordable line leads to:
  - MM = '10;
  - LL = '52.

The header shall be followed by:

- the image parity IP takes the value 0 (even image parity) for even frames and 1 (odd image parity) for odd frames (as defined in table 1) and shall be Hamming coded according to table 21 ; b1 to b8 of this table refer to D0 to D7 of CCIR Recommendation 656 [5] respectively;
- each of the 3 following data blocks shall be composed of 184 bytes.

The contents of DATV data blocks, as illustrated in figure B.2, shall be coded as follows:

- PA (present/absent):
  - one Hamming [8,4] coded byte (according to table 21), where:
    - bit b2 = 1 indicates that the following data is DATV data;
    - bit b2 = 0 indicates that the following data is not DATV data;
    - bits b4, b6 and b8 are reserved for future use and shall be ignored. The bits b1, b3, b5 and b7 contain the Hamming redundancy bits.
  - when PA value means absent, the 183 following bytes are replaced by the repeated sequence '80, '10, ...;
  - when PA value means present it is followed by:
    - one byte, set to '15 (until further specified) and to be ignored by the D2-HDMAC chain;
    - 182 bytes (B1 to B182) carrying 91 bytes Hamming encoded (according to table 21) relevant to the DATV data.

All the blocks with significant DATV data (PA = present) shall be adjacent, starting from the ANC code respectively of line 11 and line 324. The irrelevant blocks (PA = absent) shall be inserted after the relevant blocks, in each blanking interval, in order to have 36 blocks per FBI.

This protocol assumes that the BRE ensures the completion of all the DATV packet data block to 91 bytes, and that all the 36 data blocks in each field blanking are signalled by the PA byte.

The DATV data of the 656+ interface are formatted in order to be inserted within the useful data part of a DATV packet of the D2-HDMAC signal.

Consequently, the first 2 Hamming bytes of the DATV block correspond to the position of PT byte in the DATV packet and are not relevant (set to any value) in the 656+ interface. The PT byte shall be set by the D2-HDMAC multiplexer on its own. The length of 91 useful bytes, for each block of the 656+ interface, is identical to the useful data part of a packet of the HDMAC/packet signal.

The LSB information bit of each Hamming encoded byte shall be transmitted first in the DATV packets.

B.5.2 Panning data

The first panning vector data block may be set on any line in the range of line 11 to line 22, either after the three DATV blocks or at any still vacant location modulo 16 after the time reference (XY). Some additional redundancy can be introduced by duplicating the panning vector data block, in such a way that the two identical blocks can be recorded on two different tracks of the D1 tape.

If a panning vector data block is inserted after the three DATV data blocks, a stuffing byte ('10) shall be added after the 559 byte DATV data in order that the panning vector data block starts at the beginning of a slot of 16 bytes.

The completion of the 1440 bytes shall be achieved by the repeated sequence: '80,'10,'80,'10,... .

The header shall be formatted as follows:

- TT1, TT2 and TT3 identifies the panning vector data and consists of 12 information bits, which are Hamming (8,4) coded:
    - TT1, TT2 and TT3 are set to '8, '1 and '0 respectively;
    - after Hamming (8,4) coding the TT1, TT2 and TT3 are coded as follows:
        - TT1 = 'D0;
        - TT2 = '02;
        - TT3 = '15.
- LL1 and LL2 consists of 8 bits which are Hamming (8,4) coded, and which contains the length indicator:
    - the 8 bits have the value '0E, corresponding to the number of panning bytes (14 bytes);
    - after Hamming (8,4) coding LL1 and LL2 are coded as follows:
        - LL1 = '15 ;
        - LL2 = 'FD.

The header is followed by a block of 14 Hamming (8,4) coded bytes, which represent the seven panning vectors aimed at being inserted within line 625. Their content is as specified in subclause 4.5.6.

B.6 Timing relationship between DATV data and picture

On the 656+ interface all DATV data blocks present in a given FBI of a sequence of four fields beginning by an odd image refer to the next group of four video fields which follows. This timing relationship is preserved when the video signal goes through the D2-HDMAC chain.

The D2-HDMAC multiplexer shall transmit all DATV data packets as supplied via the 656+ interface. This means that the BRE governs the allocation of the DATV packets for each of the FBI multiplexes. As a consequence the BRE should not transmit more DATV/packets per FBI than the D2-HDMAC multiplex can handle and should take care of compliance to the multiplexing rules (as described in subclause 5.8.2). The D2-HDMAC multiplexer shall place all the DATV packets in the same FBI as provided on the 656+ interface. In the case that spare capacity is available (i.e. the number DATV data blocks signalled present is smaller than the maximum packet capacity of that FBI multiplex), packets of other services may be added into the packet stream on any packet position. In the case of overlapping TDMs as defined in subclause 4.5.5 however the packet with the special marker shall be the last packet of the active part of each DATV/data FBI TDM.

B.7 Relationship between the video fields numbering and the sampling phase of Y, CR, CB and MAC signals

The analogue D2-HDMAC samples shall be transmitted with the correct sampling phase according to the video field parity (as defined in subclause 5.3.6.1).

In figure B.4 the signals are shown for the HD picture, the 656+ interface and the analogue D2-HDMAC baseband, where:

- after the subsampling and shuffling of the HD picture, the luminance and colour signals are sampled in a field quincunx structure (see figure B.4a);
- at the 656+ encoder interface, the samples are arranged according to figure B.4b;
- in the D2-HDMAC signal, analogue samples are transmitted according to figure B.4c.

B.8 Electrical characteristics

The frequency accuracy of the clock should comply with that of the D2-HDMAC/packet specification, being 2,5 parts in 10 million.

The duty cycle of the clock and the clock-to-data timing relationship shall be in accordance with CCIR Recommendation 656 [5].

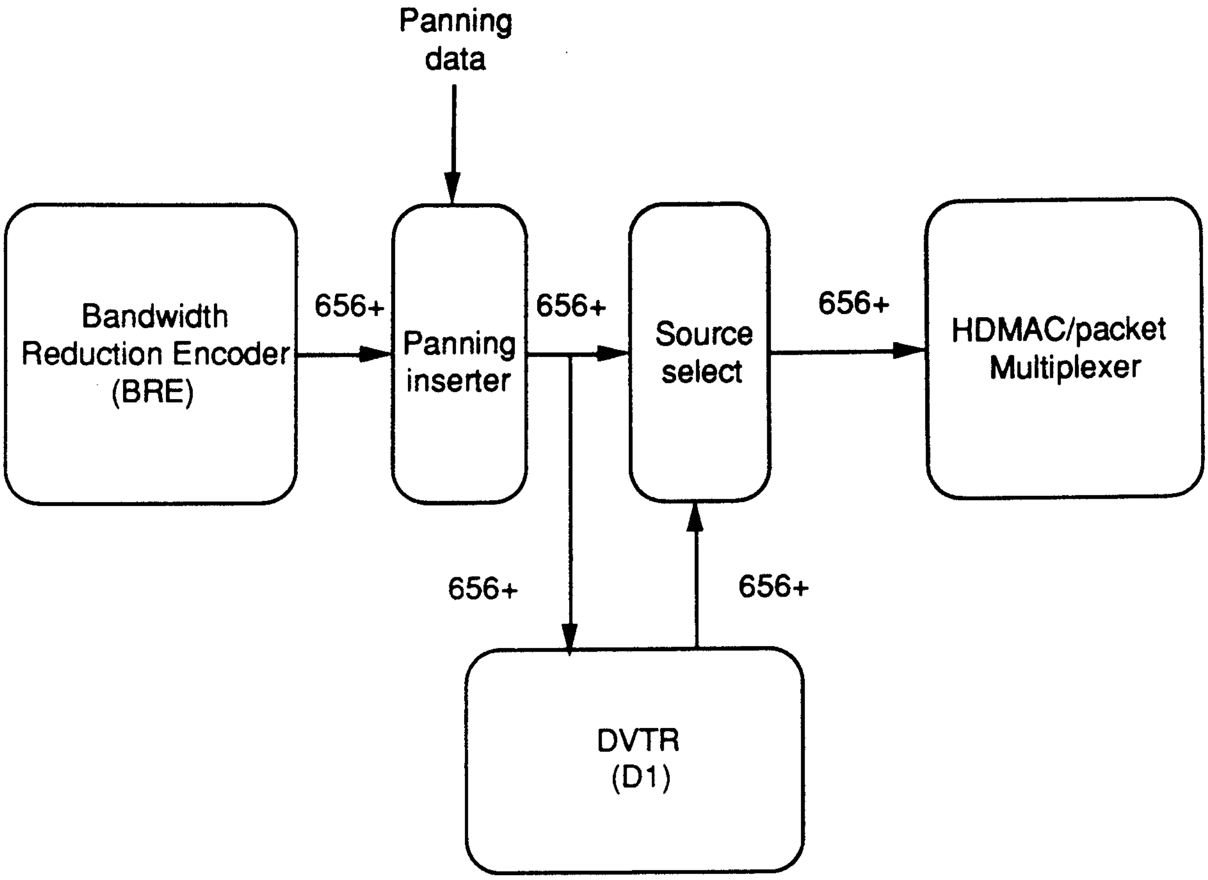


Figure B.1: Typical application of the 656+ interface

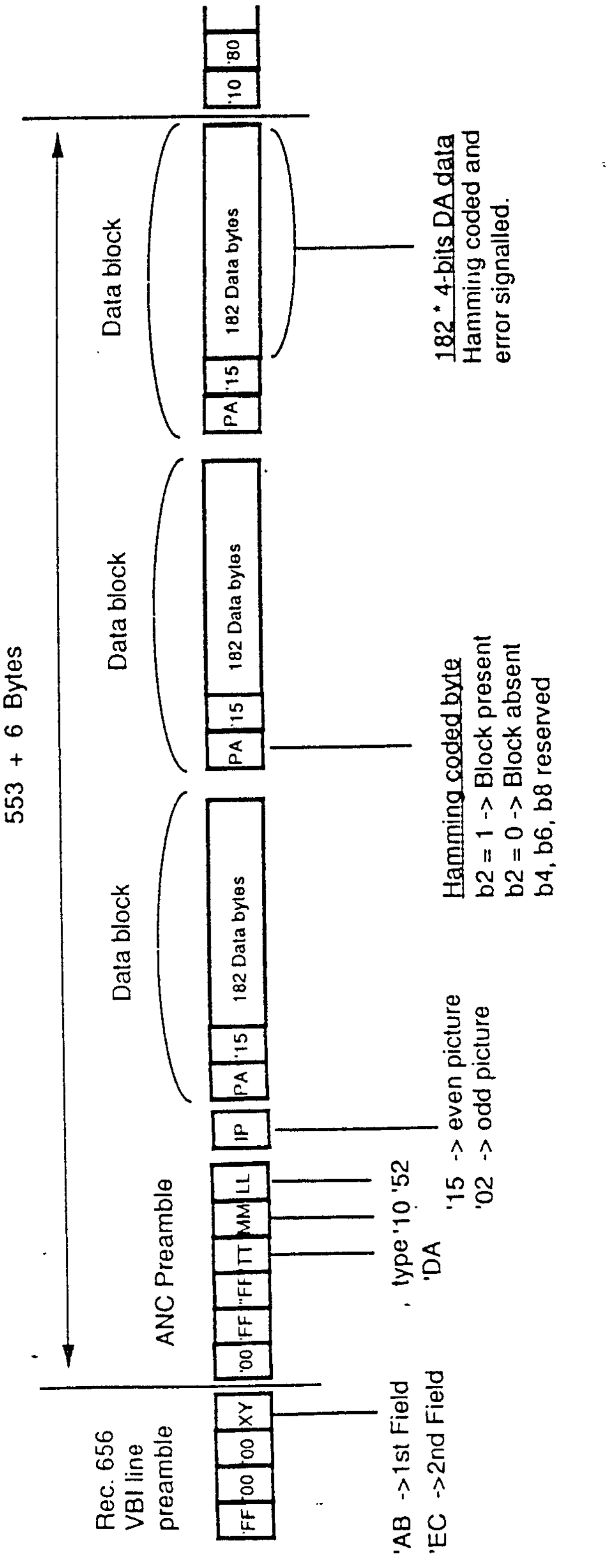


Figure B.2: Structure of a line with DATV data blocks only

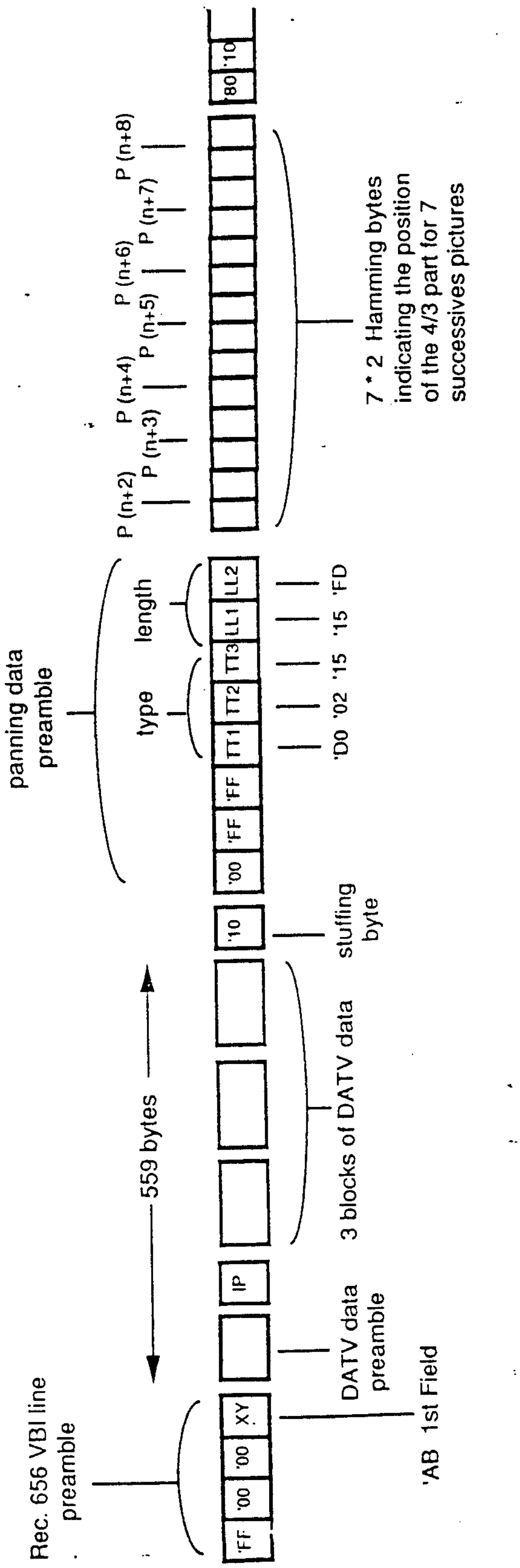


Figure B.3: Structure of a line with DATV data blocks and a panning vector block

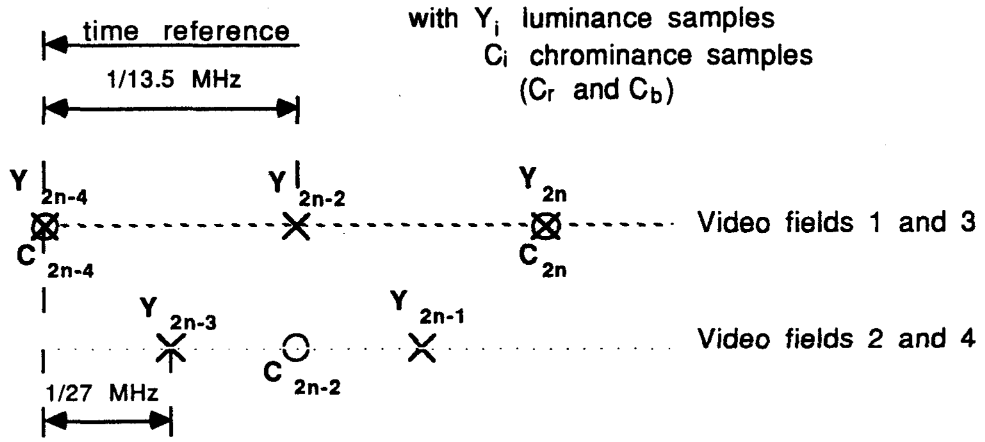


Figure B.4a: Spatial repartition of the video samples after subsampling and shuffling of the HD picture

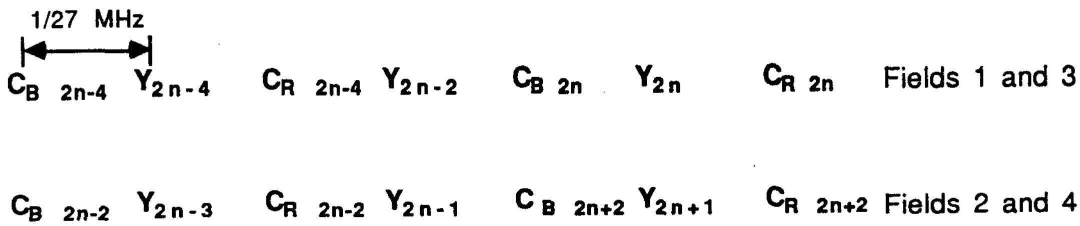


Figure B.4b: Video samples arrangement at 656+ interface

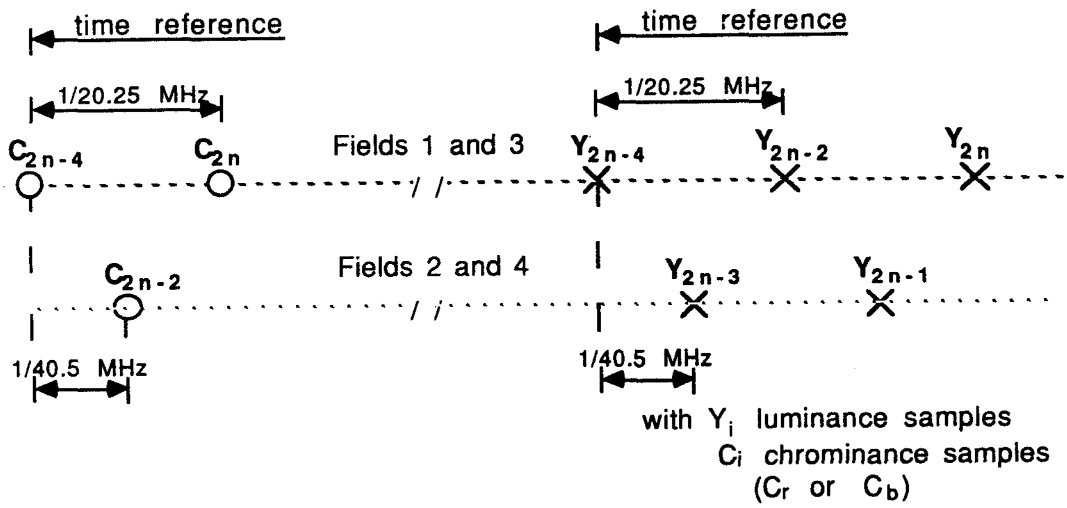


Figure B.4c: Analogue D2-HDMAC samples transmission phase

Annex C (informative) Coding example for the special data in line 625

C.1 Introduction

In order to illustrate the specifications given in subclause 4.5, the following examples show how information is sent in the static data frame, how the configuration of the TDM multiplex is described in repeated data frame and how a modification of the configuration is signalled.

The first example, given in figure C.1, concerns the coding of the Static Data Frame (SDF). In this example, the initial condition of the signal is an unscrambled D2-HDMAC vision signal.

After transition the new condition of the signal is a D2-HDMAC vision signal under controlled access, scrambled according to the double-cut component rotation method.

The incrementation of CAFCNT is also shown.

The second example, given in figure C.2, concerns the coding of the Repeated Data Frame (RDF). In this example, the initial TDM configuration comprises the following components:

- one sound/data subframe identified by TDMCID code '01;
- a D2-HDMAC vision signal identified by TDMCID codes '10 and '11 (colour-difference and luminance respectively);
- two DATV/data subframes identified by TDMCID codes '40 and '41;
- insertion test signal on lines 312 and 623 identified by TDMCID code '30;
- two FBI variable format teletext subframes, which do not overlap the DATV/data subframes, identified by TDMCID code '21.

After transition, the new TDM configuration comprises the following components:

- one sound/data subframe identified by TDMCID code '01;
- a D2-HDMAC vision signal identified by TDMCID codes '10 and '11 (colour-difference and luminance respectively);
- two DATV/data subframes identified by TDMCID codes '40 and '41;
- insertion test signal on lines 312 and 623 identified by TDMCID code '30;
- two FBI fixed format teletext subframes overlapping the DATV/data subframes, identified by TDMCID code '20.

This second example is illustrated in figure C.3 which shows the TDM organization in the frame before and after transition.

C.2 Legend in figure C.1 and figure C.2

The legend of codes in figure C.1 and figure C.2 are the following:

A CHID value;

B SDFSCR code value corresponding to the sound services carried by sound/data subframe before transition;

C SDFSCR code value corresponding to the sound services carried by sound/data subframe after transition;

D initial CAFCNT value;

D+1 incremented CAFCNT value;

E_{SDFn} bit combinations for error control of the static data frame;

E_{253} bit combinations for error control of the repeated data frame in which the FCNT is equal to 253 (static over the fivefold RDF);

" indicates a code value that remains unchanged from the previous transmission of the line 625, or from the previous transmission having the same TDMCID code.

In figure C.1 the FCNT codes are given in decimal notation; TDMC, VSAM, Rp, Rp and SIFT codes are given in binary notation.

In RDF, note that line number 1 is coded as binary 0, clock period 1 is coded as binary 0 and that higher numbers are coded correspondingly (see subclause 4.5.4). In figure C.2 these codes are given in decimal notation, as well as FCNT codes. TDMCID are however given in hexadecimal notation that conforms with the codes that are given in subclause 4.5.4.

Any unused bits in the static data frame are set to "1".

C.3 Remarks on figure C.1

1) FCNT is continuously running. The origin of the code is arbitrary.

2) The future state of video signal is announced 16 frames in advance when FCNT = 1 to 16 (modulo 16). During this period, the vision signal is still in the old state.

3) In the frame following the one in which FCNT is equal to 0 (modulo 16), the video signal has the new state and the future state of video becomes the current state.

4) In this example, it is assumed that the state of sounds changes with video state ; so, the SDFSCR code may change.

5) CAFCNT is incremented when FCNT goes from 255 to 0.

C.4 General comments on figure C.2

The sound/data burst consists of only one subframe, and consequently FLN2 and LLN2 are coded as 1 023 (all "1"s). It occupies clock periods 10-206 in lines 1-623 inclusive (coded as 009-205 and 000-622, respectively).

The MAC colour difference and luminance components are described separately and occupy two subframes each. They comprise lines 23-310 in field 1 and lines 335-622 in field 2 (coded as 022-309 and 334-621). The colour difference comprises clock periods 232-584 in both fields (coded as 231-583), and the luminance comprises clock periods 586-1 286 in both fields (coded as 585-1 285). Refer also to figure 28.

The DATV/data burst in field 1 consists of only one subframe of 20 lines, and consequently FLN2 and LLN2 are coded as 1 023 (all "1"s). It occupies clock periods 230-1 290 in lines 1-20 inclusive (coded as 229-1 289 and 000-019, respectively).

The other DATV/data burst, in field 2, consists of a subframe of 1 line and a subframe of 19 lines. Both subframes occupy clock periods 230-1 290 (coded as 229-1 289). The first subframe occupies the line 311 only (coded as 310), the second the lines 313-331 (coded as 312-330).

The clock periods given for the teletext components are 230-1 292 (coded as 229-1 291). See subclauses 7.2.1 and 7.2.2. Before transition, the variable format teletext occupies 2 lines in each field 1 and 2 (lines 21-22, coded as 20-21, and lines 333-334, coded as 332-333). After transition, the variable format teletext does not exist and the fixed format teletext occupies 11 lines in field 1 (lines 12-22, coded as 11-21) and 9 lines in field 2 (lines 326-334, coded as 325-333). Both subframes overlap the DATV/data subframes, as indicated in figure C.3.

C.5 Remarks on figure C.2

1) FCNT is continuously running. The origin of the code is arbitrary.

2) LINKS alternate between 0 and 1 on each repetition of a TDMS group that completely defines one TDMS component (which normally is defined in one TDMS field). The origin of this sequence is arbitrary and independent among different TDM components.

3) TDMCTL codes are retransmitted cyclically. Cycles may be different for individual components.

4) TDM configuration is static. Typical mean acquisition time (in the example shown) for one component is 60-120 ms (low BER). Maximum acquisition time for all (7) components is 360 ms (processing time is not included).

5) CAFCNT is incremented in SDF. See figure C.1.

6) Start of the new TDM structure definition. Old configuration is still transmitted.

7) TDM components flagged for deletion cannot be acquire by receivers switched on during this period.

8) New TDM component. It can be acquired after the change of TDM configuration.

9) The period during which the new TDMS data is transmitted should be kept at a minimum. In the example shown, new TDMS data are repeated two times prior to the change of structure (corresponding to 520 ms). New TDMS data can be repeated at higher rate if desired.

10) FCNT modulo 128 resets to 0, signalling the change of configuration.

11) TDMS data is now describing new static TDM configuration.

12) New configuration is transmitted, starting from line 1 of this television frame.

13) Turns off decoders that may have failed to recognize the deletion earlier. These TDMS data will only be transmitted temporarily.

Remark 3) indicates that the acquisition time can be minimised for those TDM components that convey information that is essential for setting up the receiver configuration, for example the sound/data burst.

Remarks 9) and 13) indicate practical rules aimed at minimizing the delay and maximizing the security of TDM reconfigurations.

<--STATIC DATA FRAME-------------------------------------->

CHID (sample 212)	SDFSCR	MVSCG <--------------------> TDMC	VSAM	CAFCNT	Rp + Fp + unall. + SIFT (sample 352)	error control	FCNT in RDF	Comments	Remarks
16b	8b	5b	3b	20b	5b	14b			
A	B	11100	100	D	11111	E_{SDF1}	...	static conditions	1
A	B	11100	100	D	11111	E_{SDF1}	47	"	
A	B	11100	100	D	11111	E_{SDF1}	48	"	
A	B	11100	001	D	11111	E_{SDF2}	49	transient conditions	2
A	B	11100	001	D	11111	E_{SDF2}	50	"	
A	B	11100	001	D	11111	E_{SDF2}	51	"	
...	...	...	...	...	...	...	...		
A	B	11100	001	D	11111	E_{SDF2}	63	"	
A	B	11100	001	D	11111	E_{SDF2}	64	"	
A	C	11100	001	D	11111	E_{SDF3}	65	static conditions	3,4
A	C	11100	001	D	11111	E_{SDF3}	66	"	
A	C	11100	001	D	11111	E_{SDF3}	67	"	
...	...	...	...	...	...	...	...		
A	C	11100	001	D	11111	E_{SDF3}	254	"	
A	C	11100	001	D	11111	E_{SDF3}	255	"	
A	C	11100	001	D+1	11111	E_{SDF4}	0	"	5
A	C	11100	001	D+1	11111	E_{SDF4}	1	"	
A	C	11100	001	D+1	11111	E_{SDF4}	2	"	
...	...	...	...	...	...	...	...		

Figure C.1: Example of Static Data Frame coding in line 625

<--Repeated Data Frame--- . .

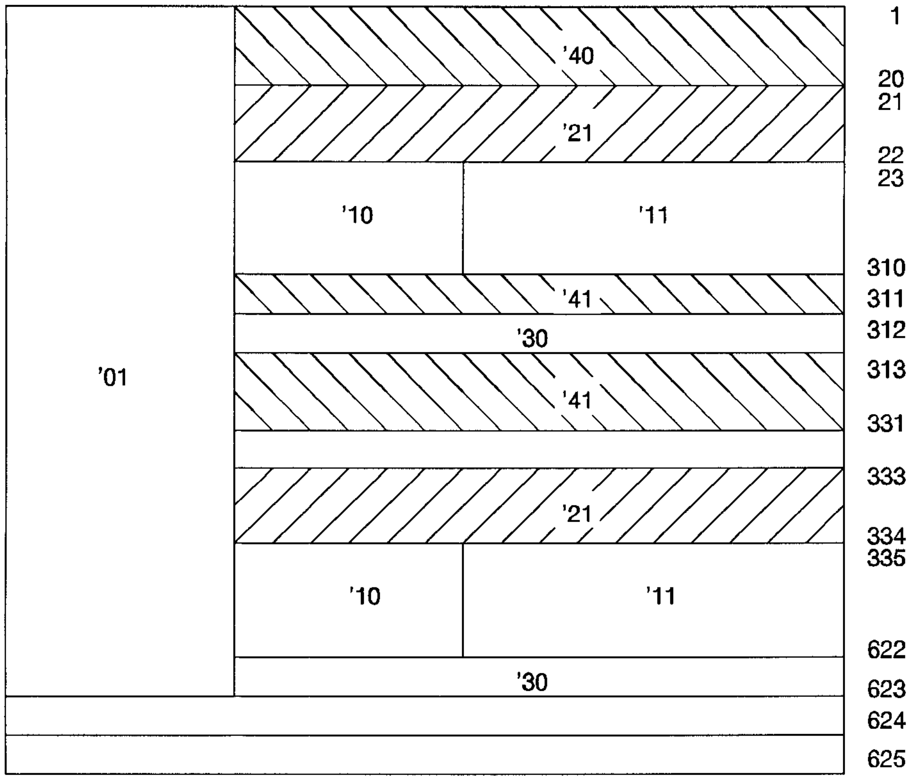
<----------------------------TDMCTL [1] (TDMCTL codes are repeated 5 times)---> <----- . .

sample 354 ... sample 540, sample 542

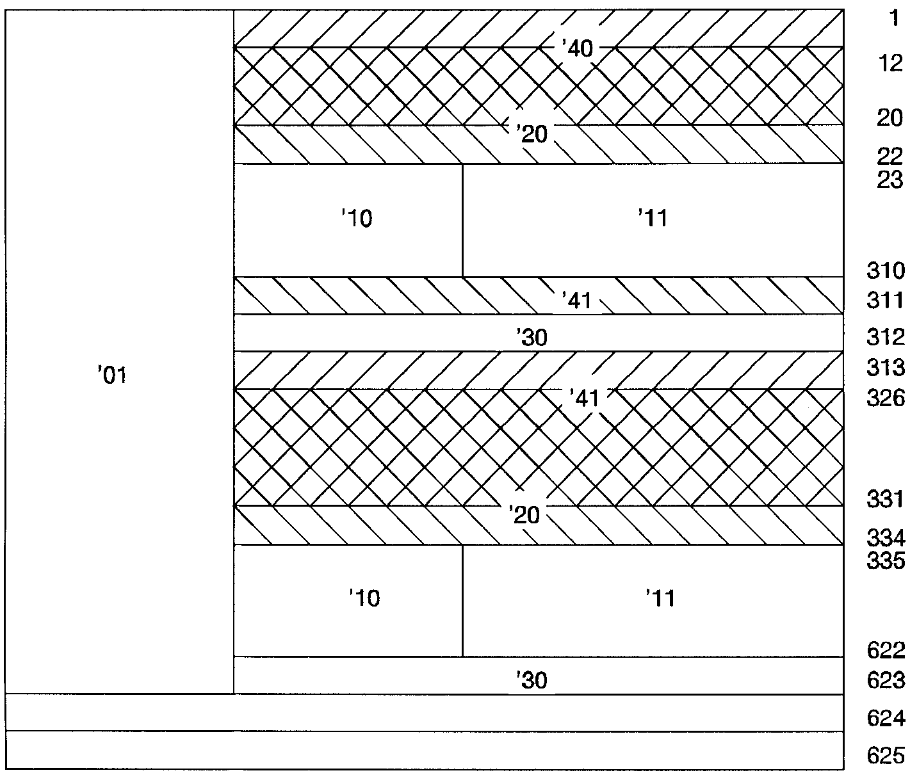
<--------------------------------TDMS-------------------------------->

FCNT	UDF	TDMCID	FLN1	LLN1	FLN2	LLN2	FCP	LCP	LINKS	error control	TDM component description	Re-marks
8b	1b	8b	10b	10b	10b	10b	11b	11b	1b	14b		
...	...	...	...	...	...	...	...	...	...	...		
253	0	'41	"	"	"	"	"	"	0	E_{253}	DA2	1
254	0	'01	0	622	1023	1023	9	205	1	E_{254}	MPX = data burst	2,3,4
255	0	'10	22	309	334	621	231	593	1	E_{255}	CHR = colour differ.	4
0	0	'11	22	309	334	621	585	1285	1	E_0	LUM = luminance	4,5
1	0	'30	311	311	622	622	231	1285	1	E_1	IST = test signal	4
2	0	'40	0	19	1023	1023	229	1289	1	E_2	DA1 = field 1 DATV	4
3	0	'41	310	310	312	330	229	1289	1	E_3	DA2 = field 2 DATV	4
4	0	'21	20	21	332	333	229	1291	1	E_4	VTXT = VF teletext	4
5	0	'01	0	622	1023	1023	9	205	0	E_5	MPX	
...	...	...	...	...	...	...	...	...	...	...		
116	0	'41	310	310	312	330	229	1289	0	E_{116}	DA2	
117	1	'21	1023	1023	1023	1023	1023	1023	0	E_{117}	VTXT	6,7,9
118	1	'20	11	21	325	333	229	1291	0	E_{118}	FTXT = FF teletext	8,9
119	0	'01	0	622	1023	1023	9	205	1	E_{119}	MPX	
120	0	'10	22	309	334	621	231	593	1	E_{120}	CHR	
121	0	'11	22	309	334	621	585	1285	1	E_{121}	LUM	
122	0	'30	311	311	622	622	231	1285	1	E_{122}	IST	
123	0	'40	0	19	1023	1023	229	1289	1	E_{123}	DA1	
124	0	'41	310	310	312	330	229	1289	1	E_{124}	DA2	
125	1	'21	1023	1023	1023	1023	1023	1023	1	E_{125}	VTXT	9
126	1	'20	11	21	325	333	229	1291	1	E_{126}	FTXT	9
127	0	'01	0	622	1023	1023	9	205	0	E_{127}	MPX	
128	0	'10	22	309	334	621	231	593	0	E_{128}	CHR	10
129	0	'11	22	309	334	621	585	1285	0	E_{129}	LUM	11
130	0	'30	311	311	622	622	231	1285	0	E_{130}	IST	12
131	0	'40	0	19	1023	1023	229	1289	0	E_{131}	DA1	
132	0	'41	310	310	312	330	229	1289	0	E_{132}	DA2	
133	0	'20	1023	1023	1023	1023	1023	1023	0	E_{133}	FTXT	
134	0	'21	11	21	325	333	229	1291	0	E_{134}	VTXT	13
135	0	'01	0	622	1023	1023	9	205	1	E_{135}	MPX	
...	...	...	...	...	...	...	...	...	...	...		

Figure C.2: Example of Repeated Data Frame coding in line 625



a) Initial TDM configuration



b) Modified TDM configuration

Figure C.3: The initial and modified TDM configurations with the corresponding TDMCID codes

Annex D (informative): Duobinary coding at 40,5 MHz sample rate

D.1 Introduction

This Annex describes the duobinary coding for data at 20,25 Mbit/s and 10,125 Mbit/s using a 40,5 MHz sample grid. A D2-HDMAC multiplexer may already use the 40,5 MHz sample rate for video processing for the implementation of the half-Nyquist filter (as described in subclause 10.2.2). Extending the duobinary coding rules to the 40,5 MHz grid enables the multiplexing of data and video components (see subclause 4.5.3) in such a D2-HDMAC multiplexer completely in the digital domain.

D.1.1 Relationship between bits of data and the sampling structure

Although the data can be transmitted at a bit rate of 10,125 Mbit/s and 20,25 Mbit/s in D2-HDMAC, the TDM multiplex for this Annex is defined in terms of samples at 40,5 MHz. The relationship between the 20,25 MHz sample numbering and the 40,5 MHz sample numbering is defined to be such that sample "k" of the 20,25 MHz sample grid corresponds to sample "2k" of the 40,5 MHz sample grid.

Using a 40,5 MHz sample grid a line has 2 592 samples. Whenever in the following subclauses the correspondence between a bit number and a sample number results in a number that is negative or zero the sample number can be derived by adding 2 592.

D.2 Digital component coding

D.2.1 20,25 Mbit/s data

The sample number of the first bit of the DATV/data multiplex (see subclause 4.3.2.2) is signalled in the FCP parameter of the TDMS field (see subclause 4.5.5) related to TDMCIDs '40 and '41. The FCP field defines the sample of the first bit number (bit number 1) of each DATV/data burst on a 20,25 MHz sample grid, not including the run-in bit (FCP is coded such that sample 1 is coded 0).

Let FCP be coded as F, then data bit y corresponds to sample number 2y + 2F.

EXAMPLE: Let the FCP parameter equal 229 (corresponding to 20,25 MHz sample number 230). Bit 1 then corresponds to the 40,5 MHz sample number $(2 \times 1 + 2 \times 229) = 460$.

D.2.1.1 Data Precoding of 20,25 Mbit/s data

The binary data stream A_{2k} is precoded to the binary data stream B_{2k} to avoid the propagation of errors using the following relationship:

- $B_{2k} = \overline{A_{2k}} \oplus B_{2k-2}$

During the periods where no digital component is transmitted, the data sequence is $A_{2k} = 0$, resulting in a corresponding B_{2k} sequence of alternately 1 and 0.

Associated with the sequence B_{2k} is the sequence C_{2k} which takes values of -1 and +1 according to:

- $C_{2k} = 2.B_{2k} - 1$

D.2.1.2 Duobinary coding of 20,25 Mbit/s data

The signal coded in duobinary form for data at 20,25 Mbit/s is written as:

- $$D(t) = \sum_j C_j . h(t - jT)$$

T represents the sampling period of the picture signal, which is about 49,4 ns. h(t) represents the pulse response of the filter H corresponding to duobinary coding, of which the frequency response H(f) for 20,25 Mbit/s data is given by:

- $$|f| \leq \frac{1}{2T} \quad : \; H(f) = \left(\frac{MT}{2}\right).\cos(\pi f T)$$
- $$|f| \geq \frac{1}{2T} \quad : \; H(f) = 0$$

where M represents the amplitude of data signal between extreme levels logic "1".

D.2.1.3 Sample coding of 20,25 Mbit/s data

A sampled representation will be taken for D(t).

For data transmitted at 20,25 Mbit/s even numbered duobinary coded samples of the 40,5 MHz sample grid have the following value:

- $$D_{2k} = \frac{M}{4}(C_{2k} + C_{2k-2})$$

For data transmitted at 20,25 Mbit/s odd numbered duobinary coded samples (transitions) of the 40,5 MHz sample grid have the following values:

- $$D_{2k+1} = \frac{M}{\pi}\left[C_{2k} + \frac{1}{3}(C_{2k-2} + C_{2k+2}) - \frac{1}{15}(C_{2k-4} + C_{2k+4}) + \frac{1}{35}(C_{2k-6} + C_{2k+6})\right]$$

This coding constitutes an excellent approximation to the theoretical value of the duobinary odd samples, obtained by truncation of the pulse response of the theoretical filter for 20,25 Mbit/s data as described in subclause D.2.1.2.

The duobinary burst amplitude M corresponds to 80% of that of the nominal black to white amplitude as referenced in line 624, disregarding overshoots.

The value "0" of samples D_{2k} is aligned to the clamp level.

D.2.2 10,125 Mbit/s data

Data transmitted at 10,125 Mbit/s is carried in the 40,5 MHz sample grid for which the sample number modulo 4 results in 0. The other samples correspond to the data transitions.

Data bit y corresponds to sample number 4y-8 (see subclause D.1.1).

EXAMPLE: Bit 1 of the 10,125 Mbit/s data burst corresponds to 40,5 MHz sample number -4. Subclause D.1.1 indicates that in this case 2 592 is to be added, therefore the actual sample number will be 2 588.

D.2.2.1 Data Precoding of 10,125 Mbit/s data

The relationship is written so that the index corresponds to the sample number of the 40,5 MHz sample grid:

- $B_{4k} = \overline{A_{4k}} \oplus B_{4k-4}$

The first transmitted bit in the LBI multiplex of each line corresponds to A_{2588}.

During the periods where no digital component is transmitted, the data sequence is $A_{4k} = 0$, resulting in a corresponding B_{4k} sequence of alternately 1 and 0.

Associated with the sequence B_{4k} is the sequence C_{4k} which takes values of -1 and +1 according to:

- $C_{4k} = 2.B_{4k} - 1$

D.2.2.2 Duobinary coding of 10,125 Mbit/s data

The signal coded in duobinary form for data 10,125 Mbit/s shall be written as:

- $D(t) = \sum_j C_{2j}.h(t-(2j+1)T)$

T represents the sampling period of the picture signal, which is about 49,4 ns. h(t) represents the pulse response of the filter H corresponding to duobinary coding, of which the frequency response H(f) for 10,125 Mbit/s data shall be given by:

- $|f| \le \frac{1}{4T} \quad : \; H(f) = MT.\cos(2\pi fT)$

- $|f| \ge \frac{1}{4T} \quad : \; H(f) = 0$

where M represents the amplitude of data signal between extreme levels logic "1".

D.2.2.3 Sample coding of 10,125 Mbit/s data

A sampled representation will be taken for D(t).

For data transmitted at 10,125 Mbit/s duobinary coded samples of the 40,5 MHz sample grid containing data have the following values:

- $D_{4k} = \frac{M}{4}(C_{4k} + C_{4k-4})$

For data transmitted at 10,125 Mbit/s the duobinary coded samples containing data transitions have the following values:

- $$D_{4k+1} = M\frac{2\sqrt{2}}{\pi}\left[\frac{1}{3}C_{4k} + \frac{1}{5}C_{4k-4} - \frac{1}{45}C_{4k-8} + \frac{1}{117}C_{4k-12} + \frac{1}{21}C_{4k+4} - \frac{1}{77}C_{4k+8} + \frac{1}{165}C_{4k+12}\right]$$

- $$D_{4k+2} = \frac{M}{\pi}\left[C_{4k} + \frac{1}{3}(C_{4k-4} + C_{4k+4}) - \frac{1}{15}(C_{4k-8} + C_{4k+8}) + \frac{1}{35}(C_{4k-12} + C_{4k+12})\right]$$

- $$D_{4k+3} = M\frac{2\sqrt{2}}{\pi}\left[\frac{1}{3}C_{4k} + \frac{1}{21}C_{4k-4} - \frac{1}{77}C_{4k-8} + \frac{1}{165}C_{4k-12} + \frac{1}{5}C_{4k+4} - \frac{1}{45}C_{4k+8} + \frac{1}{117}C_{4k+12}\right]$$

This coding constitutes an excellent approximation to the theoretical value of the duobinary odd samples, obtained by truncation of the pulse response of the theoretical filter for 10,125 Mbit/s data as described in subclause D.2.2.2.

The duobinary burst amplitude M corresponds to 80% of that of the nominal black to white amplitude as referenced in line 624, disregarding overshoots.

The value "0" of samples D_{4k} is aligned to the clamp level.

Annex E (informative): Coding examples for the interpretation blocks for a sound channel in various configurations.

E.1 General

In order to illustrate the specifications set out in subclause 6.7.3, the following examples show how content of the interpretation blocks of a sound channel may vary according to the different configurations of that channel and the up-dating procedures.

E.1.1 Transmission without up-dating

EXAMPLE 1: A high quality sound channel carries a monophonic sound component with linear coding and second level error protection. The timing characteristic is "continuous". The command information (CIB) is allocated to music/speech switching. There is no scrambling and therefore the service is in the free-access mode. Automatic mixing is possible. The PT byte of the coding blocks has the BC1 configuration. There is no news flash indication and the SDFSCR flag bit is such that the information is stored.

EXAMPLE 2: A medium quality channel carries a monophonic sound component with near instantaneous companding and first-level error protection. The timing characteristic is "intermittent". The command information (CIB) is not allocated. There is scrambling and the service is in the free-access mode. Automatic mixing is not possible. The PT byte of the coding blocks has the configuration BC2. There is no news flash indication and the SDFSCR flag bit is such that the information is stored.

BI block		Function	Value	
Byte No.			Example 1	Example 2
1		PT	BI1 = '00	
2		S1	'0	(Hamming code '15)
3		S2	'1	(Hamming code '02)
4		F1	'0	(Hamming code '15)
5		F2	'C	(Hamming code 'A1)
6		CI	'1	(Hamming code '02)
7		LI	'A	(Hamming code '8C)
8	repetition 1	Byte 1	00001001	01110000
9		Byte 2	10001000	01010010
10	repetition 2	Byte 1	00001001	01110000
11		Byte 2	10001000	01010010
12	repetition 3	Byte 1	00001001	01110000
13		Byte 2	10001000	01010010
14	repetition 4	Byte 1	00001001	01110000
15		Byte 2	10001000	01010010
16	repetition 5	Byte 1	00001001	01110000
17		Byte 2	10001000	01010010

E.1.2 Transmission with up-dating

EXAMPLE 3: Present configuration as in Example 1 above.

Future configuration as in Example 2 above except that the SDFSCR flag bit is such that the information is no longer stored. (It will be assumed for this example that at the moment when reconfiguration takes place the signal will not be simultaneously interrupted).

BI block			Function		Value
Byte No.					Example 3
1			PT		BI1 = '00
2			S1	'0	(Hamming code '15)
3			S2	'1	(Hamming code '02)
4			F1	'1	(Hamming code '02)
5			F2	'B	(Hamming code '9B)
6	1st		CI	'0	(Hamming code '15)
7	com		LI	'1	(Hamming code '02)
8	mand	Command No. changed		'2	(Hamming code '49)
9	2		CI	'1	(Hamming code '02)
10	n		LI	'A	(Hamming code '8C)
11	d	repetition 1	Byte 1		00001001
12			Byte 2		10001000
13	c	repetition 2	Byte 1		00001001
14	o		Byte 2		10001000
15	m	repetition 3	Byte 1		00001001
16	m		Byte 2		10001000
17	a	repetition 4	Byte 1		00001001
18	n		Byte 2		10001000
19	d	repetition 5	Byte 1		00001001
20			Byte 2		10001000
21	3		CI	'2	(Hamming code '49)
22	r		LI	'A	(Hamming code '8C)
23	d	repetition 1	Byte 1		01110011
24			Byte 2		01010010
25	c	repetition 2	Byte 1		01110011
26	o		Byte 2		01010010
27	m	repetition 3	Byte 1		01110011
28	m		Byte 2		01010010
29	a	repetition 4	Byte 1		01110011
30	n		Byte 2		01010010
31	d	repetition 5	Byte 1		01110011
32			Byte 2		01010010

Annex F (informative): Mixing operation

When additional sound channels are present, for example separate language commentaries, there is the requirement to mix the selected additional sound channel with the main sound. The table lists possible cases of such mixing. An additional sound channel is "available" if it corresponds to the language preference programmed into the decoder (column 1). The additional sound channel can be transmitted either "continuously" or "intermittently". The intended mixing is signalled in the service identification of the relevant additional sound channel (column 2), while the intended attenuation is signalled in the service identification of the main sound channel (column 3). The expected decoder behaviour is given in column 4.

Table F.1: Possible combinations of sound mixing informations and expected behaviour of the receiver

Additional sound available (see NOTE 1)	Mixing intended bit (in the add. sound channel)	Attenuation bit in the main sound channel	Encoded informations in the multiplex Behaviour of the receiver
Not available	-	-	Main sound with 0 dB attenuation
Continuous or (Intermittent and present)	No Yes Yes	- Equal Attenuated	Additional sound with 0 dB attenuation Additional sound with 3 dB attenuation + main sound with 3 dB attenuation Additional sound with 0 dB attenuation + main sound with 15 dB attenuation (see NOTE 2)
Intermittent and interrupted	No Yes		

NOTE 1: The additional sound is considered available when its description in the Service Identification channel matches the language selection criteria.

NOTE 2: Practical tests have indicated that an attenuation of 15 dB is a maximum.

NOTE 3: This mode of operation is not recommended in the transmitted channel.

NOTE 4: The attenuation is regarded at the output of the receiver. It is left to the manufacturer to define the appropriate gains in the digital and analogue sections of the receiver in order to avoid any unacceptable clipping during mixing operations while preserving a good signal-to-noise ratio.

Annex G (informative): Coding example for information within the dedicated channel of the sound/data multiplex

G.1 General

In order to illustrate the specifications given in Clause 8, the following example shows how the content of a satellite channel can be described by service identification data in the dedicated channel. A functional description of the access procedure is given in figure G.1.

The following services are carried (see also figure G.2):

- one television service with the DATV component in the DATV/data multiplex, one original sound component (companded, first level protected, stereo), one additional sound component (half bandwidth, companded, first level protected, mono), one subtitle component (carried by fixed-format teletext in the field-blanking interval) and an entitlement checking message component;

- one teletext service;

- in addition, a new radio service (linear, high quality, first level protected, mono) is commencing.

Within the dedicated channel, data groups exist (see figure G.3) such as to fully describe these services. (Packets belonging to a radio service also exist but are not sufficient to enable the service to be recovered.)

In the following lists, the mnemonic notation for each parameter and its identifier is followed by a brief description of the parameter and the corresponding data itself. The notation' is used to indicate hexadecimal figures. The text between quotation marks " " is coded for transmission according to the standard defined in subclause A.3. The notation MPX is used to describe the subframe corresponding with TDMCID 01 or TDMCID 02.

G.2 Parameters

The various items in the example below serve to illustrate different features of the coding. In each paragraph, a command is described together with its relevant parameters and the appropriate parameter field coding. Clause G.3 illustrates the coding of commands in their entirety.

G.2.1 Network identification and description in terms of services

G.2.1.1 Medium priority (M) - Data group 0

Parameter name	PI	Details	Parameter field coding
UPDAT	'00	updated, a sound service is commencing, LIST X for the radio sound services is modified	'18
NWO	'10	satellite channel number 17, satellite position and polarisation: 19°, R (1000, 0001, 1001, 0001) country of origin in clear text	 '17 '81 '91 "FRANCE"
NWNAME	'14	clear text	"TDF1"
LIST X	'18	for television service list of indices: index value 2, data group 3, in MPX, packet address 224 (0011, 01, 00 1110 0000)	'01 '02 '34E0
LIST X	'18	for radio services list of indices: index value 2, data group 4, in MPX, packet address 170 (0100, 01, 00 1010 1010)	'02 '02 '44AA
LIST X	'18	for teletext service list of indices: index value 1, data group 6, in MPX, packet address 345 (0110, 11, 01 0101 1001)	'03 '01 '6D59
COMD	'61	pointer to a network commentary (in the dedicated channel in this example) language French data group 9, in MPX, packet address 0 (1001, 11, 00 0000 0000)	 '0F '9C00
TIMD	'62	pointer to local time information (in the dedicated channel in this example) language French data group 9, in MPX, packet address 0 (1001, 11, 00 0000 0000)	 '0F '9C00

G.2.1.2 Low priority (L) - Data group 0

Parameter name	PI	Details	Parameter field coding
TIME	'20	time in clear text	"LUNDI 83 05 30 00:59"
DCOM	'60	direct network commentary language French clear text	 '0F "NOUVEAU: EUROJAZZ COMMENCE BIENTOT"

G.2.2 TV service description

G.2.2.1 Medium priority (M) - Data group 3

Parameter name	PI	Details	Parameter field coding
PREF	'48	programme type: public affairs programme number (from scheduled broadcast start time) (1983 May 30, 00:15 French time) programme name in clear text	'40 '83 '22 '15 "MINUIT 1"
COMD	'61	pointer to commentary (not in dedicated channel) language French in MPX, packet address 347 (0000, 01, 01 0101 1011) page number 123 (0000 0001 0010 0011)	 '0F '055B '0123
PGI	'80	video and original sound component are subject to conditional access	
ACCM	'88	access related message entitlement checking messages exist, CAFCNT not used for key generation, in MPX, packet address 346 (10, 00, 01, 01 0101 1010) Eurocrypt conditional-access system	 '855A '40
VCONF	'90	video configuration controlled access, double-cut component rotation scrambling, AR = 16:9, Cy = 3:2, Cu = 3:1	 '34
DCINF	'A4	(television) original sound component language French in MPX, packet address 224 (1100, 01, 00 1110 0000)	 '0F 'C4E0
DCINF	'A5	(television) additional sound component language English in MPX, packet address 129 (0000, 01, 00 1000 0001)	 '09 '0481
DCINF	'F8	(television) subtitles carried by fixed-format teletext in the field-blanking interval language German magazine number 8, page number 88 (0, 000, 0000, 1000 1000)	 '08 '0088
DCINF	'FE	DATVD (television) component, non scrambled language non-applicable in FBI1 and FBI2, packet address 4 (0, 0, 00, 00, 00 0000 0100)	 '00 '0004

G.2.3 Radio sound service description

G.2.3.1 Medium priority (M)- Data group 4

Parameter name	PI	Details	Parameter field coding
UPDAT	'00	radio service commencing all the parameters transmitted are new	
SREF	'40	service reference index value 2 name in clear text	 '02 "EUROJAZZ"
PREF	'48	programme reference type: jazz music number (from scheduled broadcast start time) (1983 May 30, 01:00 French time)	 '54 '83 '23 '00
DCINF	'A8	(radio) original sound component language: not applicable in MPX, packet address 170 (0000, 01, 00 1010 1010)	 '00 '04AA

G.2.4 Teletext service description

G.2.4.1 Medium priority (M) - Data group 6

Parameter name	PI	Details	Parameter field coding
SREF	'40	service reference index value 1 source of the service in clear text	 '01 "FRANCE SOIR"
PREF	'48	programme reference type: public affairs number: not applicable name: not applicable	 '40 - -
COMD	'61	pointer to indirect commentary (outside the dedicated channel) language French in MPX, packet address 345, page 1, row 1 (0000, 11, 01 0101 1001) (0000, 0000, 0000, 0001) (0000, 0001)	 '0F '0D59 '0001 '01
DCINF	'B0	cyclic teletext component language French in MPX, packet address 345 (0000, 11, 01 0101 1001) teletext data channel 0	 '0F '0D59 '0000

G.3. Network identification and description in terms of services

G.3.1 Medium priority (M) - Data group 0

Total length 50 bytes.

CI	LI	PI	LI	Parameter field			
'10	'30						
		'00	'01	'18			
		'10	'09	'17	'81	'91	"FRANCE"
		'14	'04	"TDF1"			
		'18	'04	'01	'02	'340E	
		'18	'04	'02	'02	'44AA	
		'18	'04	'03	'01	'6D59	
		'61	'03	'0F	'9C00		
		'62	'03	'0F	'9C00		

G.3.1.1 Low priority (L) - Data group 9

Total length 63 bytes.

CI	LI	PI	LI	Parameter field
'11	'3D			
		'20	'14	"LUNDI 83 05 30 00:59"
		'60	'25	'0F "NOUVEAU : EUROJAZZ COMMENCE BIENTOT!"

G.3.2 TV service description

G.3.2.1 Medium priority (M) - Data group 3

Total length 57 bytes.

CI	LI	PI	LI	Parameter field				
'90	'37							
		'40	'04	'02	"TF1"			
		'48	'0C	'40	'83	'22	'15	"MINUIT 1"
		'61	'05	'0F	'055B	'0123		
		'80	'0D					
		'88	'03	'855A	'40			
		'90	'01	'34				
		'A4	'03	'0F	'C4E0			
		'A5	'03	'09	'0481			
		'F8	'03	'08	'0088			
		'FE	'03	'00	'0004			

G.3.3 Radio service description

G.3.3.1 Medium priority (M) - Data group 4

Total length 26 bytes.

CI	LI	PI	LI	Parameter field
'A0	'18			
		'00	'00	
		'40	'09	'02 "EUROJAZZ"
		'48	'04	'54 '83 '23 '00
		'A8	'03	'00 '04AA

G.3.4 Teletext service description

G.3.4.1 Medium priority (M) - Data group 6

Total length 34 bytes.

CI	LI	PI	LI	Parameter field
'B0	'20			
		'40	'0C	'01 "FRANCE SOIR"
		'48	'01	'40
		'61	'06	'0F '0D59 '0001 '01
		'B0	'05	'0F '0D59 '0000

G.4 Packet rate requirements for the SI dedicated channel example

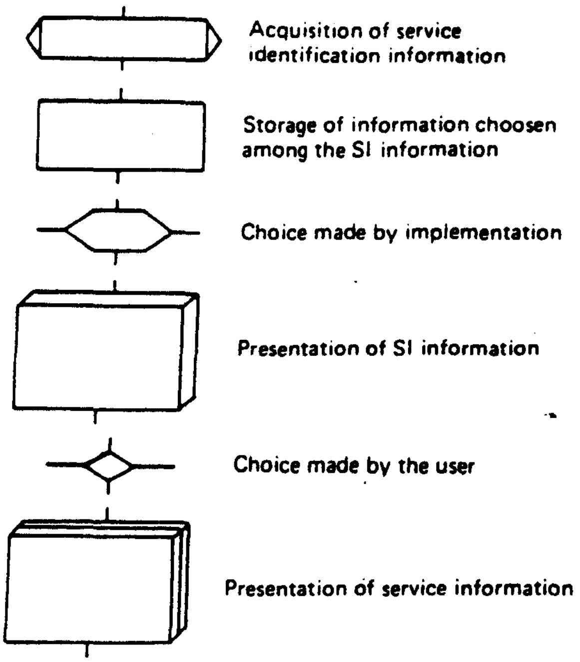
The data group and packet structures required for the transmission of commands and parameters of the foregoing example are illustrated in figure G.4. The SI dedicated channel capacity requirement is as detailed below:

Medium priority:	data group 0	(network)	1 packet
	data group 3	(TV)	1 packet
	data group 4	(radio)	1 packet
	data group 6	(teletext)	1 packet
		Medium priority total:	4 packets.
Low priority:	data group 9	(network)	1 packet
		Low priority total:	1 packet

Assuming that the mean rate of repetition of medium priority information is four times per second and that of low priority information is once per second, the mean capacity utilised by the SI example becomes:

(4 x 4 + 1) = 17 packets/second.

LEGEND



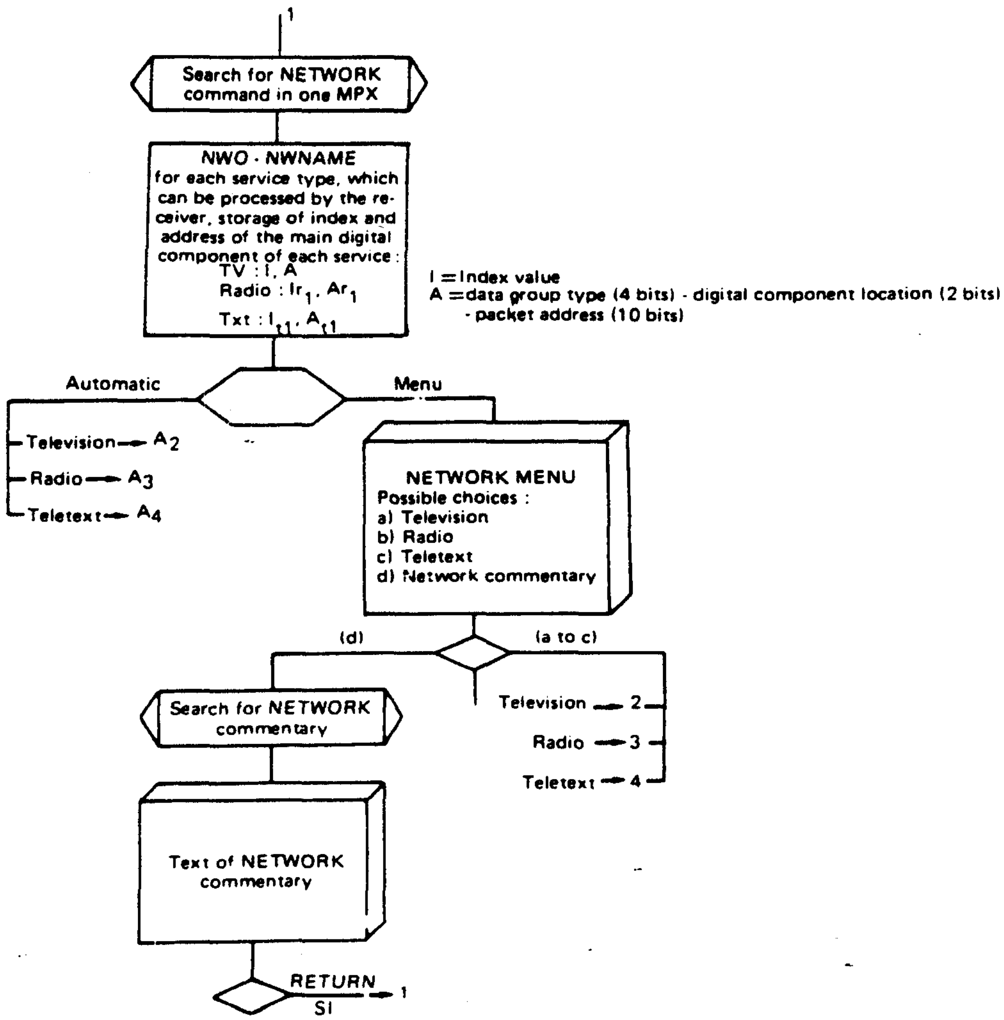


Figure G.1: functional description of the access procedure

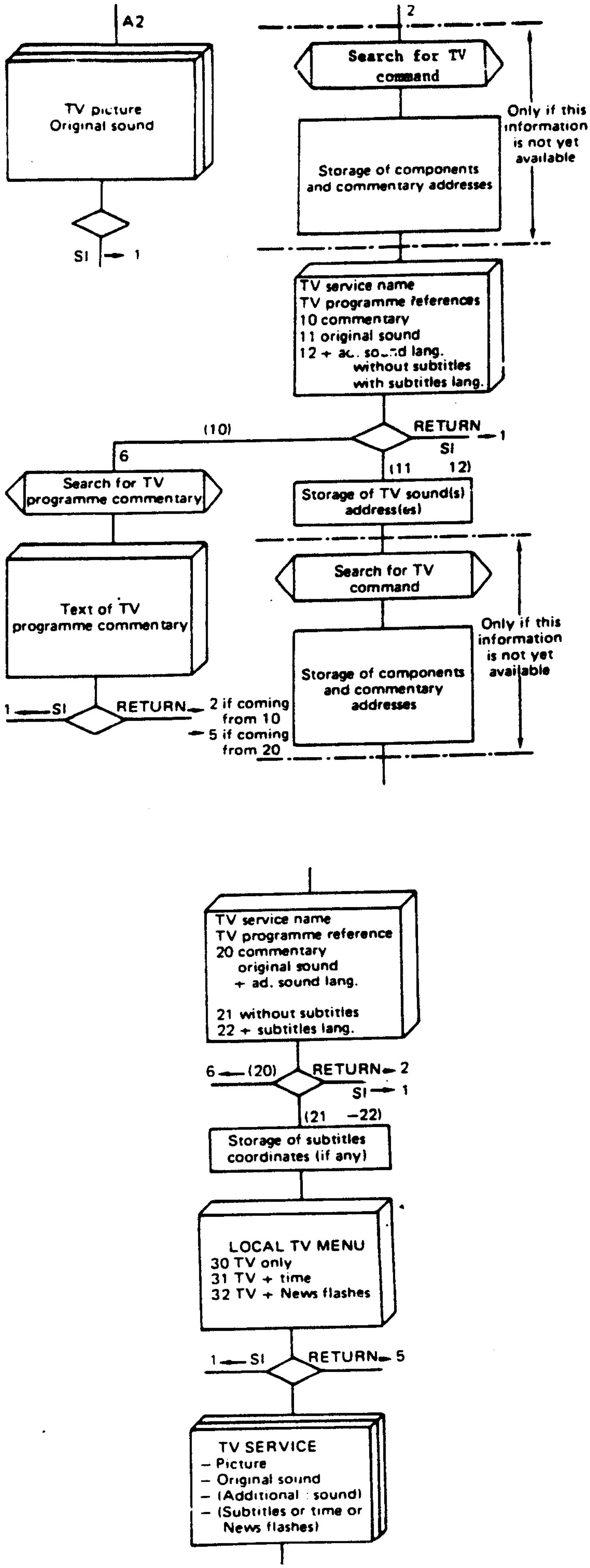


Figure G.1: functional description of the access procedure (continued)

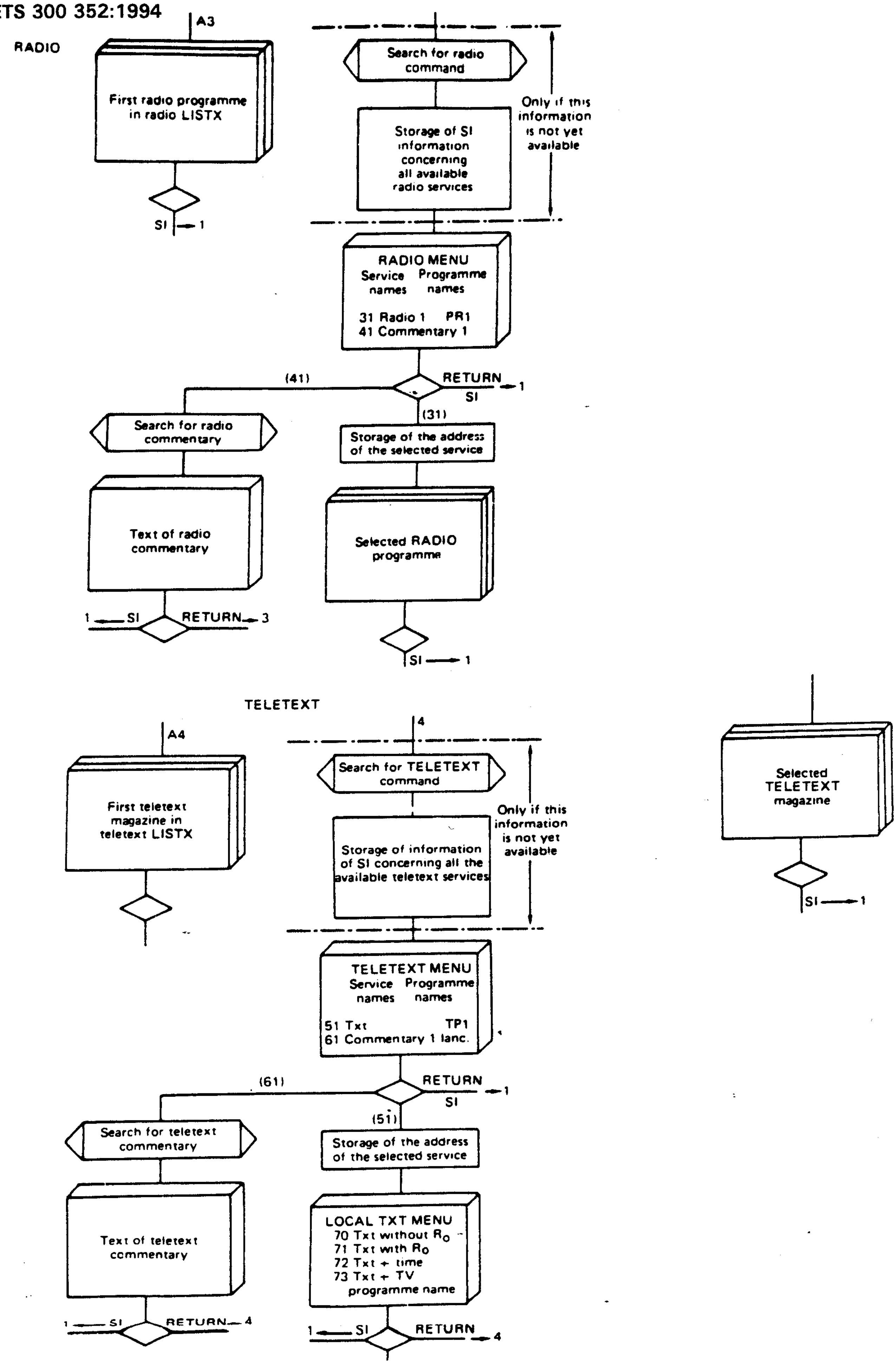


Figure G.1: functional description of the access procedure (concluded)

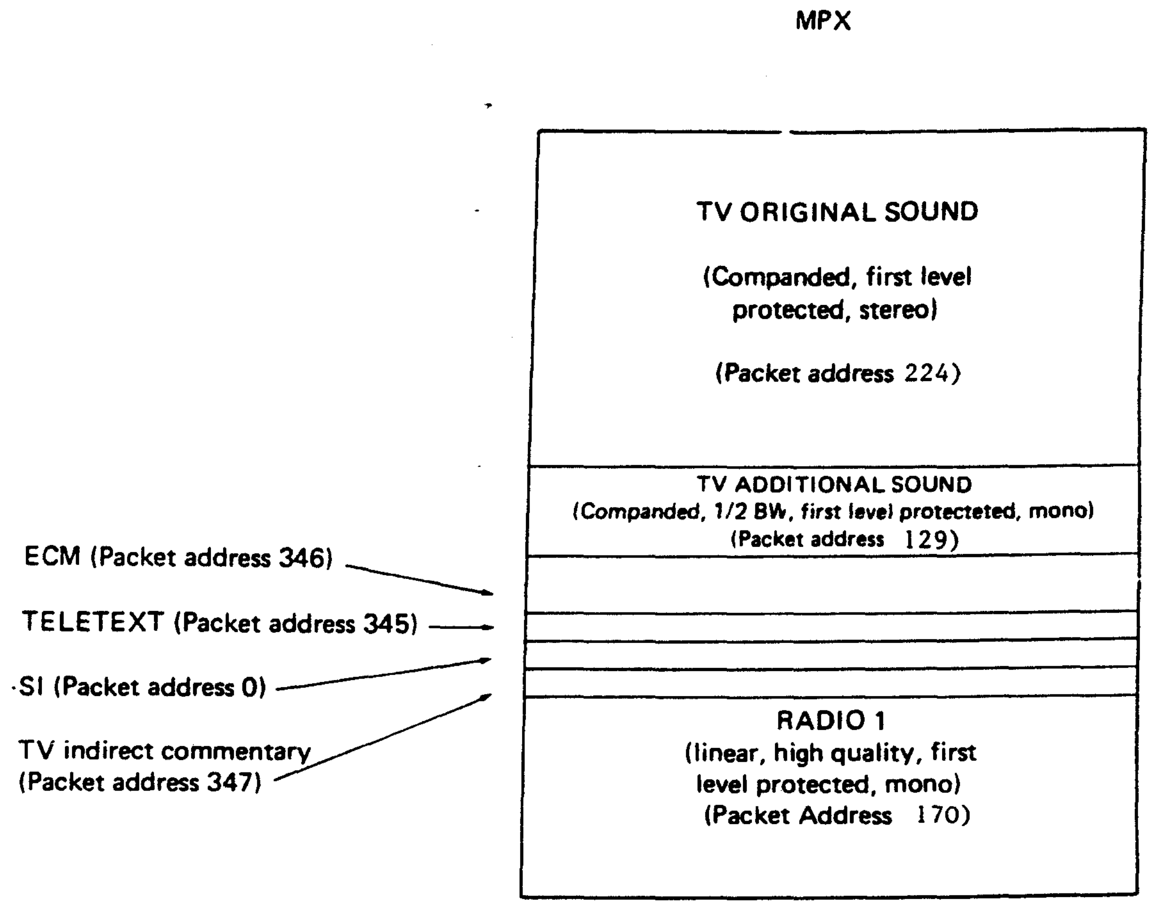
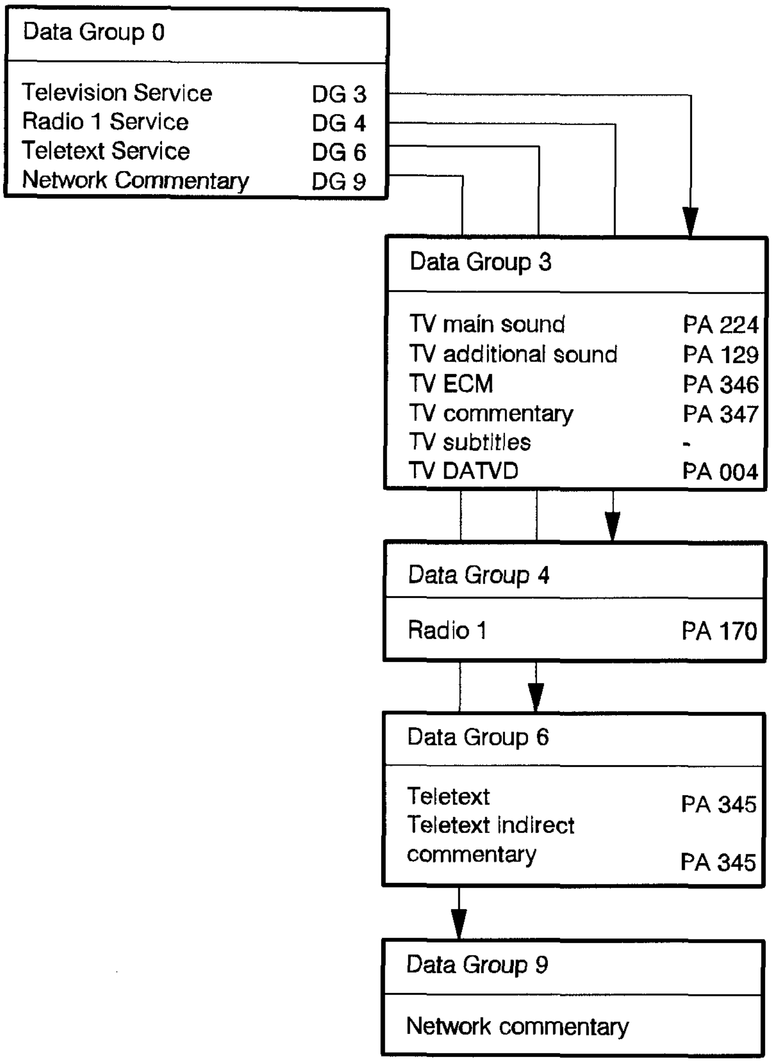


Figure G.2: Capacity map for the coding example



DG x stands for data group x of the SI dedicated channel.
PA xxx stands for packet address xxx of MPX (except for DATV, which is present in FBI1 and FBI2).

Figure G.3: Example data group map for the SI dedicated channel in MPX

PH	PT ('F8)	DGH	NETWORK CATEGORY M	DGS	UNUSED (28 bytes)	PS

DATA GROUP 0

PH	PT ('F8)	DGH	TV CATEGORY M	DGS	UNUSED (19 bytes)	PS

DATA GROUP 3

PH	PT ('F8)	DGH	RADIO CATEGORY M	DGS	UNUSED (52 bytes)	PS

DATA GROUP 4

PH	PT ('F8)	DGH	TELETEXT CATEGORY M	DGS	UNUSED (44 bytes)	PS

DATA GROUP 6

PH	PT ('F8)	DGH	NETWORK CATEGORY L	DGS	UNUSED (15 bytes)	PS

DATA GROUP 9

Figure G.4: SI dedicated channel coding example - packets and contents

Annex H (informative): D2-MAC and D2-HDMAC compatibility

H.1 Introduction

One of the principles of the D2-HDMAC system is the compatibility with the D2-MAC system. The following paragraphs describe some aspects of this compatibility.

H.2 D2-MAC signal and D2-HDMAC decoder

A D2-HDMAC decoder should be fully compatible with the D2-MAC system, as described in ETS 300 250 [1]. When it recognizes a D2-MAC signal (i.e. the bit 5 of MVSCG in line 625 is set to 1), it should be able to switch to the behaviour of a standard D2-MAC decoder:

- the bandwidth is limited to 8,4 MHz;
- no half-Nyquist filtering is required;
- the MAC part is processed in normal definition mode;
- the E7 deemphasis, if so signalled in the SI channel ,is applied instead of E7E;
- an orthogonal sampling grid as defined for D2-MAC is applied to the video information.

H.3 D2-HDMAC signal and D2-MAC decoder

A D2-MAC-only decoder ignores the high definition attribute given by the bit 5 of MVSCG in line 625. The DATV should be broadcast in a manner such that the overlapping mechanism does not disturb the teletext processing in D2-MAC decoders (see subclauses 4.3.2.2 and 4.5.5)

H.4 D2-HDMAC signal in a limited channel bandwidth

As described in subclause 4.1, the baseband bandwidth for normal operation of a D2-HDMAC signal is equal to or greater than 11,4 MHz. If the available bandwidth is less than this value (for instance when using 8 MHz channel spacing), the D2-HDMAC signal can in principle be transmitted after low pass filtering and be processed as a D2-MAC signal by a D2-MAC decoder.

However, DATV is rendered unrecoverable by the bandwidth limiting, such a signal cannot be processed correctly by a D2-HDMAC decoder even though this will consider the signal to be D2-HDMAC.

Therefore, inserting a D2-HDMAC signal in a 8 MHz channel should be avoided.

Annex J (informative): Specification of an interface for digital D2-HDMAC signals

J.1 Introduction

In the HDMAC multiplexer the signal is available in a digital format before being converted to analogue. When transmitting HDMAC signals via digital links it is advantageous to use the signal in its digital form for further signal processing. Then A/D and D/A conversions, which would introduce additional noise, can be avoided.

The interface defined in the following paragraphs can also be used for D2-MAC/packet signals.

In Clause J.2 the signal format is described. In Clause J.3 the particular characteristics of the interface are described.

J.2 Signal format

J.2.1 General description of the interface

The interface provides a unidirectional interconnection between a single source and a single destination. The data signals are in the form of binary information coded in 8-bit words (9-bit words for future applications are allowed). The data words are conveyed at 20,25 Mword/s. The structure of the multiplex is based on a 40 ms digital frame which contains 625 lines with 1 296 words each (see figure J.1).

The multiplex is composed of two main components:

- vision data;

- duobinary data burst which carries the sound and the DATV/data multiplex.

J.2.2 Vision data

The vision data transmitted in orthogonal mode on each line contain the compressed luminance signal samples and one of the compressed colour difference signals samples. Scrambling of the HDMAC vision is possible conforming to the MAC/packet specification.

The correspondence between video signal levels and 8-bit quantization levels is referred to the reference line 624:

- black level corresponding to decimal level 16;

- grey (clamp) level corresponding to decimal level 128;

- white level corresponding to decimal level 240.

The vision signals at this interface have been filtered by the E7E non-linear preemphasis, but the half-Nyquist filter has not been applied.

J.2.3 Duobinary data burst

The data signal is coded in duobinary form. With this coding system, the signal has three characteristic levels where the extreme levels (-1, +1) represent logic "1" and the intermediate level represent logic "0". The duobinary burst amplitude corresponds to 80% of that of the vision signal (reference line 624). The 8-bit quantized duobinary data burst decimal levels are defined as:

- -1 corresponding to 38;
- 0 corresponding to 128;
- +1 corresponding to 218.

For the sound/data data burst the values of the transition words conform to ETS 300 250 [1].

J.3 Particular characteristics

J.3.1 Electrical characteristics of the interface

J.3.1.1 General

The interface carries data in the form of 8 parallel data bits and a separate synchronous clock. For future applications pins for a 9^{th} bit (LSB) are reserved.

The interface employs nine line drivers and nine line receivers (for future applications a tenth pair may be reserved). Each line driver (source) has a balanced output and the corresponding line receiver (destination) a balanced input (see figure J.2).

Although the use of ECL technology is not specified, the line driver and receiver should be ECL-compatible, i.e. they permit the use of ECL for either driver or receiver.

J.3.1.2 Logic convention

The A terminal of the line driver is positive with respect to the B terminal for a binary "1" and negative for a binary "0".

J.3.1.3 Line driver characteristics (source)

- Output impedance: 110 Ω maximum.
- Common mode voltage: -1,29 V ± 15% (both terminals relative to ground).
- Signal amplitude: 0,8 to 2,0 V peak-to-peak, measured across a 110 Ω resistive load.
- Rise and fall times: less than 5 ns, measured between the 20% and 80% amplitude points, with a 110 Ω resistive load. The difference between rise and fall times should not exceed 2 ns.

J.3.1.4 Line receiver characteristics (destination)

- Input impedance: 110 Ω ± 10 Ω.
- Maximum input signal: 2,0 V peak-to-peak.
- Minimum input signal: 185 mV peak-to-peak.
- Differential delay: data should be correctly sensed when the clock-to-data differential delay is within the range of ± 11 ns.

J.3.2 Clock signal

J.3.2.1 General

The clock signal is a 20,25 MHz square wave where the 0-1 transition represents the data transfer time.

J.3.2.2 Clock-to-data timing relationship

The positive transition of the clock signal shall occur midway between data transitions as shown in figure J.3.

Clock period:	$T \approx 49{,}4$ ns.
Clock pulse width:	$t = 24{,}7 \pm 3$ ns.
Data timing - sending end:	$t_d = 24{,}7 \pm 3$ ns.

J.3.3 Mechanical details of the connector

The interface uses the 25 contacts type D subminiature connector specified in ISO Document 2110-1980. The contact assignment is described in figure J.4.

Connectors are locked together by a one-piece slide lock on the cable connectors and locking posts on the equipment connector. Cable connectors employ pin contacts and equipment connectors employ socket contacts.

Shielding of the interconnecting cable and its connectors should be employed.

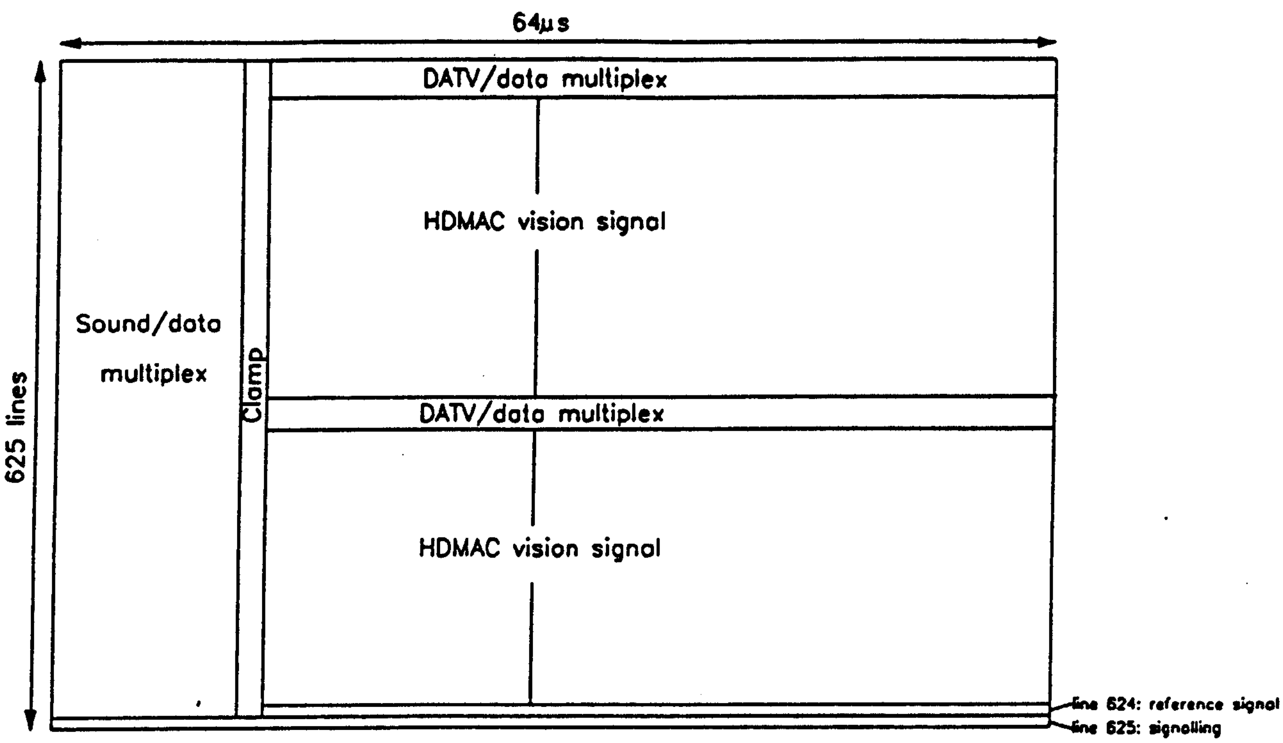


Figure J.1: Simplified D2-HDMAC/packet TDM structure

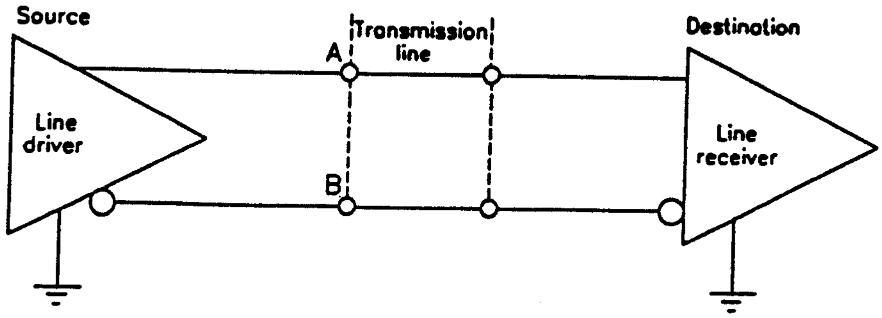


Figure J.2: Line driver and line receiver interconnection

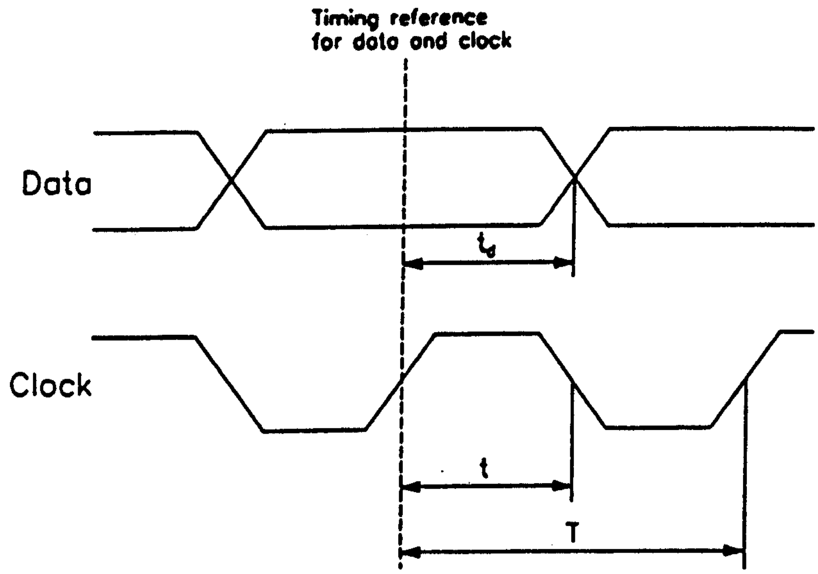


Figure J.3: Clock to data timing (at source)

Clock period:	$T \approx 49,4$ ns.
Clock pulse width:	$t = 24,7 \pm 3$ ns.
Data timing (sending end):	$t_d = 24,7 \pm 3$ ns.

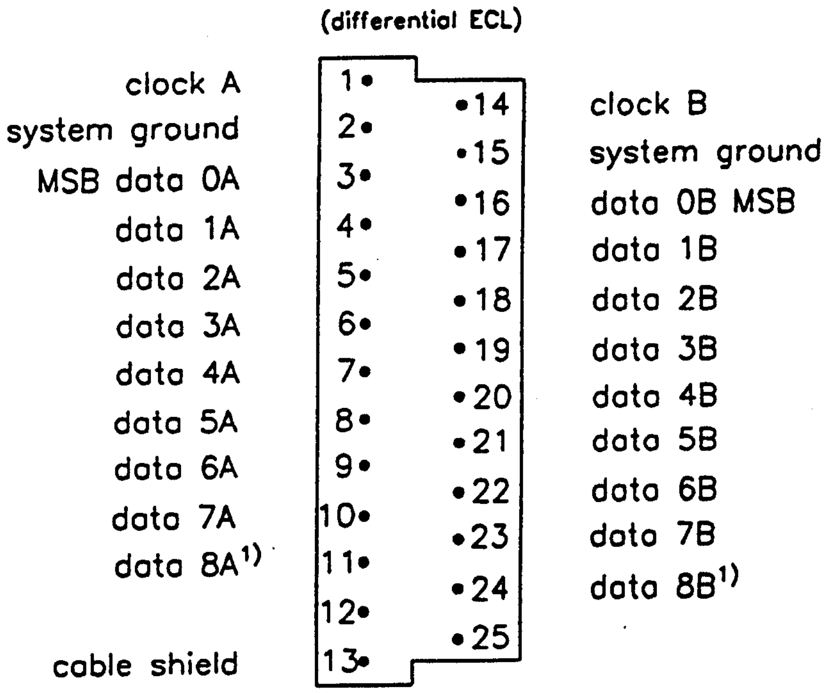


Figure J.4: HDMAC interfacing contact assignment (25 pole subminiature D-connector)

1): reserved for future applications

History

Document history	
June 1994	First Edition
March 1996	Converted into Adobe Acrobat Portable Document Format (PDF)