EUROPEAN TELECOMMUNICATION STANDARD

ETS 300 355

June 1994

Source: EBU/ETSI JTC

Reference: DE/JTC-DMAC

UDC: 621.397.13

Key words: D-MAC, MAC/packet family, broadcasting

European Broadcasting Union — Union Européenne de Radio-Télévision

EBU UER

Television Systems;
Specification of the D-MAC/Packet system

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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 the EBU SPB 491 specification. In view of the urgency for presenting the technical content and for historic reasons, this ETS is presented in its original form as received from the EBU. Therefore, this ETS has not undergone the normal ETSI editing or quality control procedures relating to its 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 *.

The European Telecommunication Standard embodied in the present document is the result of studies carried out.

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Introduction

1. The members of the MAC/packet family

The MAC/packet family of systems consists of the following three members: C-MAC/packet, D-MAC/packet, D2-MAC/packet. These systems incorporate the following common features:

- time-division multiplexing
- MAC picture coding, with the capacity for extended aspect ratio and enhanced picture quality
- packet multiplexing for sound and data
- digital high and medium quality sound coding and error-protection method
- service identification
- conditional access with video, sound and data scrambling
- full-channel digital mode of operation.

The clock frequencies used in these systems have simple relationships with the sampling frequencies of the digital studio standard defined in CCIR Recommendation 601.

The specification of the three members of the MAC/packet family is given in EBU doc. Tech. 3258, 2nd issue. The present document is only concerned with the D-MAC/packet system.

2. Expected bearers

The D-MAC/packet system is suitable for satellite broadcasting with frequency modulation and for cable distribution with vestigial-sideband amplitude modulation, with full transparency.

3. Summary description of the system

The particular features of the D-MAC/packet system are:

- a baseband time-division multiplex in which analogue picture signals are combined with duobinary encoded digital sound, synchronization and data signals

- the capacity of the sound/data multiplex is about 3 Mbit/s in the case of normal television transmissions and nearly 20 Mbit/s in the case of full-channel sound and data mode of operation; these figures are respectively equivalent to eight and up to 52 high-quality sound channels with 15 kHz bandwidth, with near-instantaneous 14/10-bit companding (protected by one parity bit per sample). The spare data capacity can be used for other services.

It can be used with frequency modulation in a satellite channel, or with AM/VSB in a channel of at least 10.5 MHz bandwidth.

4. General organization of the system

The MAC/packet signal is the common bearer for numerous services. Each service is based on the use, possibly combined, of service components which can be classified in three categories:

- picture (analogue broadcasting)
- sound (digitally broadcast)
- data, a general term covering components which do not belong to the two preceding categories.

The following considerations introduce other important points:

- the multiplicity of channels available to the user imposes the need for channel identification to facilitate receiver tuning

- technology allows the design of a system broadcasting analogue and digital components in the same RF channel, and with the possibility to modify the respective parts of each component. For this purpose, a time-division multiplexing technique is used, referred to in this Introduction as TDM multiplex for transmission (time-division multiplex)

- the end of the traditional one-to-one relationship between service and RF channel imposes the need for the system to provide the user with tools to ease the service access

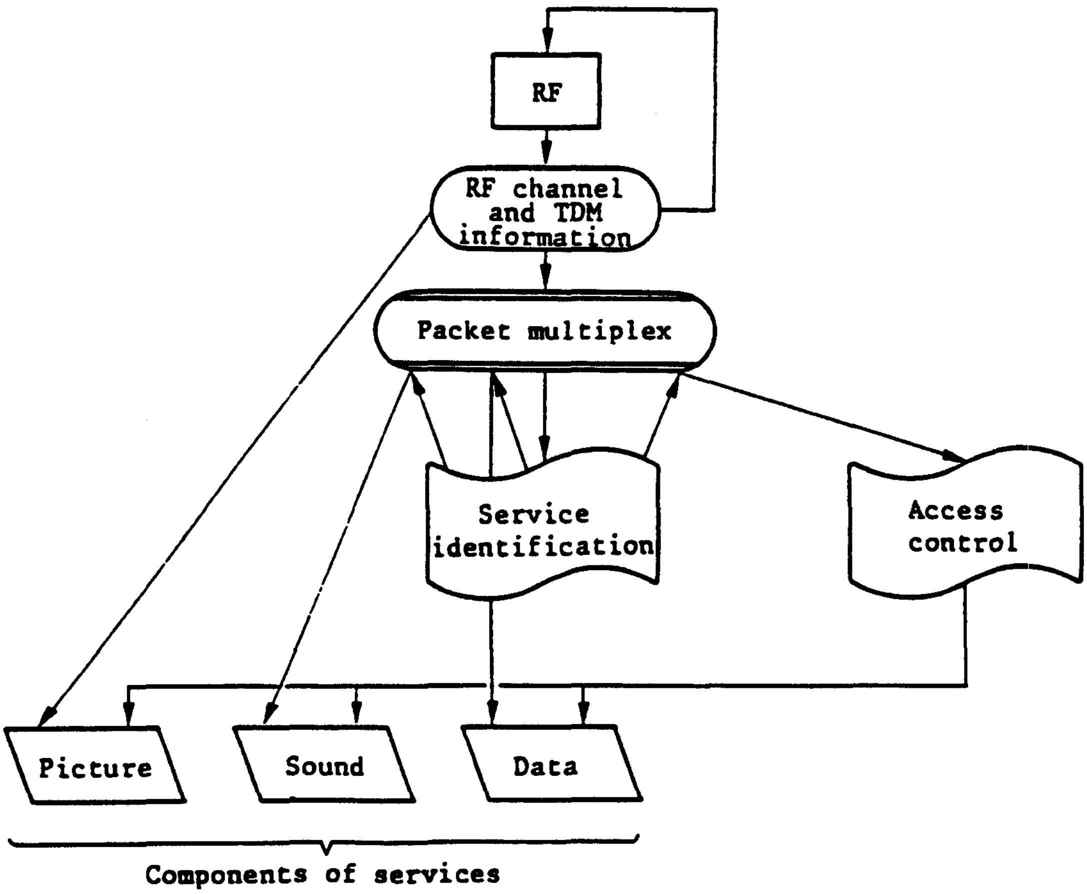
- to diversify operating modes, the system includes the tools for implementing access control to the services.

The list above leads to the conclusion that the system needs a skeleton, which connects the components of different services and the various mechanisms easing the utilisation of these components. This rôle of a skeleton is played by the service identification system inside the data broadcasting system, based on a packet multiplexing technique together with RF channel and TDM configuration data outside the packet multiplex.

Through these two mechanisms, the following important functions are realised:

- to facilitate access to the RF channel
- to define the configuration of the TDM
- to facilitate the access to services
- to control the access to services
- to broadcast the sound services
- to broadcast the data services.

The following diagram illustrates these:



The processing of the RF channel and TDM configuration data allows the receiver to separate the analogue part of the TDM, that is to say the picture, from its digital part, that is to say the packet multiplexes.

The service identification information itself is available in a fixed address data channel of the packet multiplexes. The processing of this information gives access to the channels carrying the different digital components of the various services, and to the data channel managing the access control to these components.

6. Methods of conveying satellite signals in cabled distribution networks

In the case of collective antenna installations where constraints from spectrum occupation or use of high frequencies do not exist, the distribution at the "first intermediate frequency" (950-1750 MHz)*, without remodulation, would provide a solution fully complying with the requirements. If the availability of a limited number of channels is acceptable to the user, signals received from the satellite could be distributed in the middle band (110-174 MHz) and in the extended super band (230-470 MHz) simply by means of frequency translation from the first intermediate frequency. These cases are therefore not discussed in more detail.

In the case of large cabled distribution networks, demodulation of the FM D-MAC/packet signal followed by remodulation in AM/VSB provides a simple method of conveying the satellite signals in cabled distribution networks with full transparency.

Transcoding from D-MAC/packet to D2-MAC/packet involves the loss of half of the sound/data capacity. In addition, the bandwidth of the luminance component must be limited which would inevitably involve a certain reduction in the picture quality.

In order to avoid the need for several variants of receivers, it would be desirable for all receivers in the future to be designed to be equally suitable for operation with all members of the MAC/packet family. It is also desirable that television receivers produced from now onwards should be equipped with the standard CENELEC "peritelevision" connector.

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* See EBU Technical Statement D46-1985

7. Presentation of the specification

The specification of the D-MAC/packet system is presented in eight parts:

Annexes to these Parts contain supplementary information which is nevertheless part of the specification or helpful to the understanding of the specification. Additional supplementary information are contained in Appendices of EBU doc. Tech. 3258, 2nd issue, particularly for the design of receivers, but they were elaborated before the present specification and were not updated.

8. Advantages of the system

Several CCIR Reports give additional details regarding advantages of this system and the experimental results obtained. In particular, Report 634 establishes the conformity of the protection ratios required by this system with the prescriptions of the Final Acts of the WARC-BS 77 and RARC-SAT 83. Nevertheless, the principal advantages of this system are recalled here; they are:

- to adopt the general principle of time-division multiplex for substantially improving the quality of the signals and in particular for eliminating the problems of intermodulation and cross-colour

- to introduce digital techniques for the sound

- to use a sound/data multiplex (associated with the service-identification system) making available the capacity required and at the same time the maximum of flexibility

- to permit the subsequent compatible introduction of further services or further improvements to the quality.

PART 1: SPECIFICATION OF THE TIME-DIVISION MULTIPLEX

Contents

1. Subject of Part 1

This Part of the document contains the specification of the multiplexing system for the D picture/sound/data system. D-MAC/packet is suited for use in satellite broadcasting. D-MAC/packet is also suited to use on a transmission medium which guarantees a baseband equal to or greater than 8.5 MHz. In particular, the D system can be used with AM/VSB modulation on cabled distribution networks with a channel spacing not less than 10.5 MHz.

2. Basic hypotheses

2.1 Vision standard

The present specification is compatible with the MAC vision standard, as described in Part 2. However, through the TDM control information (see Section 5.4) provision is made for future enhancements.

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

2.2 Capacity for the sound/data multiplex accompanying in a normal MAC vision signal

For the D system, the available capacity is equivalent to eight high-quality sound channels together with the video signal (see Part 3).

2.3 Sound/data capacity with full-channel digital mode of operation

For the D system, the available capacity is nearly 20 Mbit/s (equivalent to up to 52 high-quality sound channels - see Part 3).

3. Transmission multiplex structure

In the case where a normal MAC vision signal is transmitted for each line, the baseband multiplex structure for the D-MAC/packet signal is shown in Fig. 1a.

In the case of full-channel digital mode of operation, the time-multiplex structure for each line is shown in Fig. 1b.

A uniform sample numbering system is used. Data and vision sample timing uses a reference timing of equispaced sample instants given by a 20.25 MHz clock. Sample numbering runs from 1 to 1296 inclusive to describe the instants on a television line of the MAC/packet baseband signal. Associated with each sample instant is a clock period of 1/20.25 μs which is numbered so that it contains the sample instant of the same number.

Vision samples, synchronisation and data bits are placed at sample instants.

It should be noted that in Fig. 1a, the boundaries between regions lie on sample instants, and that each region includes the sample instant on its right-hand boundary. For example, the luminance region has boundaries at 589 and 1286, and therefore contains sample 590 to 1286. The run-in bit for the D system, which lies on sample 1296 therefore is included in region k, not a.

3.1 Instantaneous bit-rate

For D-MAC/packet, the instantaneous bit-rate of the data burst shall be 20.25 Mbit/s $\pm$ 2.5 parts in 10 million.

3.2 Data burst formats

3.2.1 Data burst format for normal television transmissions (see Fig. 2a)

For the D system, for standard television transmissions defined in Part 2, each data burst shall contain a total of 206 bits*. One bit shall be allocated as a run-in bit for differential demodulation. Six bits will be allocated as a line synchronization word. The last bit of the burst will not be allocated. The remaining 198 bits are used for sound and data.

The sound/data multiplex system occupies 623 data bursts per video frame, leaving one data burst free for insertion of a clamp marker** and one data burst free for the insertion of a frame synchronization word.

3.2.2 Data burst format for full-channel digital mode of operation (see Fig. 2b)

For full-channel digital mode of operation, the basic television multiplex structure described in Section 3.2.1 is used, except that the analogue vision signal is replaced by digital signals.

In principle, the digital TDM components of the D system are divided in two subframes, one of them being intended to be handed over to a D2 system. These components are identified by TDMCID codes 01-0E (see Section 5.4).

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* The whole of line 625 is available for transmission of special data (see Section 5).

** Provisionally, a clock run-in is transmitted in the spare bits on line 624, just preceding the clamp marker word (Figs. 2a and 2b).

However, certain operational requirements for the D system, such as high-speed data services, may demand a multiplex structure which is not compatible with the above general principle, i.e. the TDM components are not divided in two subframes. In this case, a D2 subset of the D system may not exist. These TDM components are identified by TDMCID codes 42-4F (see Section 5.4).

The digital signals occupy up to 623 lines of each frame, leaving line 624 free for insertion of a clamp marker* and reference signals, and line 625 free for insertion of a frame synchronization word and the special data burst described in Section 5.

3.3 Synchronization

Synchronization can be achieved by two independent methods. Line synchronization words provide both line and frame information. Frame synchronization words 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 Part 6 (see also the definition of FCNT and CAFCNT in Sections 5.4 and 5.3).

3.3.1 Line synchronization

Each data burst contains one of two 6-bit line synchronization words (LSWs).

The LSWs <including the run-in bit (X) for the D system>, are defined in their transmission order as:

$$W_1 = (X)\ 001011$$

$$W_2 = \bar{W}_1 = (X)\ 110100$$

The LSW is transmitted in its true (W_1) or inverted form (W_2) according to the pattern shown in Table 1.

The pattern of line sync words at the frame boundaries also provides the video frame synchronization. This is shown in Table 1.

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* Provisionally, a clock run-in is transmitted in the spare bits on line 624, just preceding the clamp marker word (Figs. 2a and 2b).

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Table 1

Frame number	Line number	Sync word	Frame number	Line number	Sync word
Even	620	W_2	Odd	620	W_1
	621	W_1		621	W_2
	622	W_2		622	W_1
	623	W_2		623	W_1
	624	W_1		624	W_2
	625	W_1		625	W_2
Frame Boundary					
Odd	1	W_2	Even	1	W_1
	2	W_1		2	W_2
	3	W_2		3	W_1
	4	W_1		4	W_2
	5	W_2		5	W_1
	etc.	etc.		etc.	etc.

The convention used for numbering the lines and fields is described in Section 3.2 of Part 2.

3.3.2 Frame synchronization

The data burst in line 625 shall contain a frame synchronising 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 synchronising sequence contains a clock run-in period of 32 bits followed by a 64 bit frame synchronising word.

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

3.3.3 Colour sequence identification

See Section 3.1 of Part 2.

3.3.4 Sound/data synchronization

The sound/data multiplex system transmits the sound and data in packets.

For any mode of operation, the start of the first packet of a digital TDM component is defined as the first bit of the first line of the TDM component. The positions of the subsequent data packets are found by counting (see Part 3).

3.4 Reference signals

3.4.1 Line 624

The video part of line 624 contains analogue and digital reference signals which may serve to permit self-adjusting equalisation for both digital and picture signals, which may be necessary on certain severely-disturbed transmission channels*.

The first part of line 624 contains grey, white and black reference signals which are specified as follows (see Fig. 3):

Grey level : from sample 210 to sample 372
White level: from sample 372 to sample 534
Black level: from sample 534 to sample 696.

The second part of line 624 contains the complex wobulation test signal defined in EBU doc. Tech. 3256 and in CCIR Report 1096-1.

3.4.2 Clamping

A clamp period is provided on lines 1-624, and is defined as following the data burst in the case of normal television transmissions (see Section 3.3 of Part 2) and as following the first data burst in the case of full-channel digital mode of operation. A clamp marker is provided in line 624 (see Figs. 2a and 2b). The clamp marker is a 32-bit word which is fixed in relationship to the end of the burst, independent of the burst duration. The 32-bit word, in hexadecimal notation, is EAF3927F. E is read out first, with the MSB first.

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* These reference signals are introduced in the satellite signal to avoid the need to insert them at all cable-heads.

3.4.3 Test signals

In addition to line 624, two lines of the frame (312 and 623) are reserved on a permanent basis within the field-blanking interval for the emission of test signals in the case that a vision signal is present. In the absence of vision as may occur in full frame data transmission, these lines may be omitted.

Three additional lines per frame may be used as an option for field-blanking interval test signals.

The additional line numbers which are used are signalled within the time- division multiplex control of line 625.

In the case of full-channel digital mode of operation, these lines should be omitted.

The test signals to be used are specified in the EBU document Tech. 3256 and in CCIR Report 1096-1.

3.5 Signalling of RF channel, TDM configuration and service-identification information

The 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 Section 5).

3.6 Bit interleaving*

Interleaving will be applied to a block length of 751 bits (i.e. a packet, see Part 3) in order to minimise the effect of multiple bit errors. The bits of each packet will be transmitted in the following order:

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

The interleaving is not applied to the special data burst on line 625, to the data burst in line 624, and to the teletext data in the field-blanking interval (see Sections 3.2 and 3.3 of Part 4A).

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* When scrambling is used, the digital sound and data signals are subjected first to scrambling for the conditional-access system and then to bit interleaving and transmision scrambling (for spectrum shaping purposes). The conditional-access scrambling process is described in Section 6 of Part 2 and in Section 3 of Part 6.

3.7 Data scrambling for spectrum shaping purposes*

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

- the first 7 bits of each data burst (run-in bit and line sync word) for the D system
- the special data burst in line 625 and the data on line 624.

The scrambling sequence is generated by a pseudo-random binary sequence (PRBS) generator as shown in Fig. 4. The PRBS generator operates continuously at a clock rate of 20.25 MHz for the D system. During periods when scrambling is not required, the scrambler continues to generate the PRBS sequence, but its action is inhibited by means of a gating signal.

The PRBS generator is initialized every 625 lines such that the first bit of the sequence is added to bit 8 of line 1 for the D system.

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

4. Data coding and multiplexing with the analogue TDM components

4.1 Digital component

The digital component is coded in duobinary form. The application of the basic coding method is illustrated in Fig. 5. With this coding system, the signal has three characteristic levels where the extreme levels represent logic "1" and the intermediate level represents logic "0".

4.2 Relationship between the bits of data and the sampling structure

It should be noted that bit y corresponds to sample number y-1.

4.3 Data precoding

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

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

where k is the sample number, " $\overline{\quad}$ " signifies the complementary function and "$\oplus$" signifies logic "exclusive or".

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* When scrambling is used, the digital sound and data signals are subjected first to scrambling for the conditional-access system and then to bit interleaving and transmision scrambling (for spectrum shaping purposes). The conditional-access scrambling process is described in Section 6 of Part 2 and in Section 3 of Part 6.

The first transmitted bit (run-in bit) of each line corresponds to A_{1296} (see Fig. 6).

During the periods where no digital component is transmitted, the data sequence $A_k = 0$; the corresponding B_k sequence consists of alternately 1 and 0.

Associated with the sequence B_k, with values 0 and 1, is the sequence C_k which takes values of -1 and +1.

$$C_k = 2\ B_k - 1$$

4.4 Duobinary coding

The signal coded in duobinary form is written as:

$$D(t) = \sum_k C_k\ h\ (t - kT)$$

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) is given by:

$$|f| \leqslant 1/2T: \qquad H(f) = (\frac{MT}{2}) \cos \pi\ f\ T$$

$$|f| \geqslant 1/2T: \qquad H(f) = 0$$

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

For all values of k, duobinary coded samples have the following values:

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

where + means arithmetic addition of the signal levels.

The duobinary burst amplitude M corresponds to 80% of that of the picture signal, disregarding overshoots (see Fig. 1a).

4.5 Multiplexing of analogue and digital TDM components

The multiplex is obtained by adding the analogue component to the duobinary data signal which is clamped to zero level during the period in which the digital component is not transmitted. This is illustrated for the case of normal TV transmission in Figs. 6, 7a and 7b.

5. Specification of data transmitted in line 625

In the following Sections, 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 CRC purposes.

5.1 Overall structure of line 625

The data in line 625 are organized according to Fig. 8 for the D-MAC/packet system. The allocated bits are not interleaved in the way described in Section 3.6, nor scrambled for spectrum shaping. There is one frame synchronization data (FSD) part, one unified date and time (UDT) part, one static data frame (SDF) data block and five other data blocks that are used for what is called the repeated data frames (RDF).

For the D-MAC/packet system, the line ends with 101 unallocated bits and the UDT, SDF and RDF bits are interleaved by alternation with unallocated bits; in each case the first bit of UDT, SDF and RDF is an information bit and the unallocated bits are filled for the present (until redefined) with pseudo-random data consisting of the output of the PRBS generator shown in Fig. 4.

Frame synchronization data (FSD)

For the D-MAC/packet system, the first bit is the demodulator run-in. The following 102 bits contain a fixed pattern which is complemented on alternate frames. The demodulator run-in bit (DRI) and line synchronization word (LSW - 6 bits) are common to all data bursts.

Then follows a 32-bit clock run-in (CRI) and 64-bit frame synchronization word (FSW) (see Fig. 10). 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.

Data blocks

The data block of the static data frame contains 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 contains 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 is generated by the polynomial:

$$(x^7 + x^3 + 1)\ (x^7 + x^3 + x^2 + x + 1)$$
$$= x^{14} + x^9 + x^8 + x^6 + x^5 + x^4 + x^2 + x + 1$$

The message to be sent is 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 are such that the polynomials of

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

and

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

are multiples, modulo 2, of the polynomial:

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

5.2 Unified date and time (UDT)

The following five bits contain data which change from frame to frame and which, over a 25-frame sequence, contains a statement of the unified date and time according to CCIR Recommendations 457 and 460, 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, is specified relative to UTC in multiples of 1/2 hour with an inclusive range +15 to -12 hours.

The first bit of the sequence is an element of a chain code such that five consecutive correct bits are sufficient to define the position within the 25-bit sequence. Fig. 9 indicates schematically the logic required to synchronize to, and reproduce, this chain code.

The boundary between the 25-frame sequence represents a second marker, and all the information in each sequence relates 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'.

5.3 Static data frame (SDF)

The static data frame is organized according to Fig. 10.

It 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 gives satellite position, channel number and polarisation, country of origin, responsible administration, ..., in accordance with ETSI Technical Report (ETR) XXX (JTC work programme reference DTR/JTC-001).

(SDFSCR) Services configuration reference (currently specified only for sound services transmitted in the subframes defined by TDMCID code 01 and 02, see Section 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 Section 2.1 of Part 5). However, if bit 2 of byte 1 of CI 1 of the interpretation block data (see Section 9.3 of Part 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		
packet address	: 10 bits		see Section 2.1 of Part 5
digital component location	: 2 bits		
audio configuration	: 3 bits		
scrambling	: 1 bit		see Table 3 of Part 3
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 must be allocated.

The first bit of SDFSCR is 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 are set to 0.

(MVSCG) Multiplex and video scrambling control group gives information on the physical signal organization within the satellite channel. The eight bits are allocated as follows:

Bit 1, Bit 2, Bit 3, Bit 4: This Time division multiplex configuration (TDMC)* sub-group is defined as follows:

Bit 1 = b_v = video configuration; if b_v = 1, the vision signal is compatible with decoders intended for the time compressed components system defined in Part 2 of this specification with the normal compression ratios of 3:1 for chrominance and 3:2 for luminance. If b_v = 0, the vision signal is not compatible or is absent (e.g. in the case of full-channel digital mode of operation).

Bit 2 = b_m = sound/data multiplex format; if b_m = 1, the sound/data multiplex is compatible with decoders intended for the normal burst multiplex, defined in this specification. Two 99-bit packet multiplexes will be present as shown in Fig. 2. If b_m = 0, the sound/data multiplex is not compatible.

* Section 3.3 of Part 2 states that it is important that all receivers decode and use the TDMCTL information defined in Section 5.4.

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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 Section 5.4.

Bit 3 = b_T = sound/data multiplex transcoding recommendation; if this bit is "1", then the subframe characterised by TDMCID = 01 is the only one that is recommended by the broadcaster to be handed on from D-MAC/packet system into a D2-MAC/packet system; if this bit is "0", then either of the subframes with TDMCID codes 01 and 02 may be handed on at the choice of the cable operators. For full-channel digital mode of operation, this bit is only relevant to the first data burst.

This bit has no technical function in the user's decoder. It must not be changed when transcoding from the D-MAC/packet system to the D2-MAC/packet system.

Bit 4 = b_A = aspect ratio indication; if b_A = 1 the ratio image width to image height within the source picture area is 4:3. The aspect ratio is 16:9 if b_A = 0. For full-channel digital mode of operation, this bit has no function in the user's decoder.

Bit 5 Reserved bit. Set to 1.

Bit 6
Bit 7
Bit 8
This Vision scrambling and access mode (VSAM) sub-group indicates whether the MAC video signal is scrambled by one of two techniques (see Section 6 of Part 2), and whether access is controlled or free. Five combinations are at present allocated:

bit 6	bit 7	bit 8	
0	0	0	free access, double-cut component rotation scrambling
0	0	1	controlled access, double-cut component rotation scrambling
0	1	0	free access, single-cut line rotation scrambling
0	1	1	controlled access, single-cut line rotation scrambling
1	0	0	free access, unscrambled

For full-channel digital mode of operation, these bits have no function in the user's decoder.

A change to the transmitted MVSCG (bits 4 to 8 only) must be made exactly 16 frames before the resultant modification takes effect. The change to the configuration 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.

(CAFCNT) Conditional access frame count: the twenty most significant bits of a 28-bit frame count (see Section 3 of Part 6). This information changes regularly every 256 frames. The eight least significant bits are sent in the Repeated Data Frame (see FCNT in Section 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 indicates* that Golay coded packets continaing 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 may 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.

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 consists of five successive identical 94-bit data blocks, as shown in Fig. 8. The coding structure of each RDF block is shown in Fig. 11.

---

* For rapid SI acquisition (important especially for access controlled services), the SIFT bit allows the receiver to positively identify the correct coding format within a few frames. The receiver/decoder may then be configured to acquire packet '0' based on the appropriate PT values (see Section 3.2 of Part 5).

(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) is 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) is used in conjunction with the conditional-access system specified in Part 6. See also (CAFCNT) in Section 5.3.

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

(UDF) Up-date Flag: if non-zero, this bit indicates 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 describes the present structure.

(TDMCID) TDM Component Identification: an 8-bit field which carries 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 : Unused (unallocated) capacity.

01-0E: For D-MAC/packet system, these codes are allocated to areas reserved for those data bursts that are organized as two related subframes; the odd TDMCID codes refer to the first subframe in each data burst, the even codes to the second subframe.

The data burst immediately following the line synchronization word is identified by TDMCID codes 01 and 02 for the two related subframes for D-MAC/packet system.

The data bursts organized as two related subframes which follow the clamping interval are labelled by TDMCID code pairs, i.e. 03 and 04, ..., 0D and 0E.

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 Section 3.3 of Part 2. (Black level reference is not allocated a code because, as specified in Section 4.5 of Part 2, it is always transmitted every field following the end of the first colour-difference signal. For normal video conditions as defined in Part 2, 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 (see also Section 2.2 of Part 5).

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).

30-3F : Reserved for field-blanking interval insertion test signals (see Section 3.4.3).

40-41 : Reserved values.

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

(TDMS) Time Division Multiplex Structure: defines 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 must 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 is coded as binary 0, clock period 1 is coded as binary 0; higher numbers are coded correspondingly. All 1's in FLN1, FLN2, etc. represent invalid codes and are used to signal undefined subframes. Thus, a TDMS field defining only one subframe has all 1's in FLN2 and LLN2.

(LINKS) Linked structure: one-bit switch used to link the group of TDMS field(s) needed to fully define one TDM component. This bit changes on each repetition of the linked TDMS field(s).

---

* A subframe is any rectangular shaped area within the television frame. It should be noted that for scrambled video, the boundaries of the subframe refers to the format before scrambling and after correct descrambling (see Part 2, Section 3.3).

TDMCTL data for different TDM components can be sent in any order in successive television frames. Linked structures must be described in increasing order of FLN1. TDMS fields having the same value of FLN1 must be transmitted in increasing order of FCP. The maximum number of different TDM components in one satellite channel must never exceed 128*.

Any change of the TDM configuration is synchronized by the frame counter. New TDMS data, which is flagged by the UDF bit, is 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 starts 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 is 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.

It is recommended that new TDMCTL data be sent shortly before any change of configuration in order to minimise the acquisition delay for those receivers which are turned on during this process.

5.5 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 is 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 is 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). Panning vectors may take values from -43 to +44 inclusive.

Each panning vector applies to one frame. The vectors are transmitted in groups of 7 for consecutive frames within the TDMS coding space of the RDF, and are 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 are unallocated and set to 1. Their values are ignored except for calculation of the error control group.

---

* This number corresponds to a maximum acquisition time of about 5 seconds for a particular TDM component.

The first vector after TDMCID = '81 applies to the second frame after the frame containing the panning vectors. The second vector applies 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.

It is intended that in normal operation, when panning is in effect, every seventh frame will contain panning vector information in line 625. It is also permitted, for events such as programme item changes, that a new set of panning vectors may occasionally be sent in an earlier frame than the seventh frame. In this case, the new set of vectors has priority and overwrites the old set. If panning vectors fail to be available for any reason, the decoder should initially maintain the last panning position sent. 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 wide to standard aspect ratio. When a transmission changes from standard to wide aspect ratio, the initial panning position should be central, and should remain so in default of panning vectors.

5.6 Unallocated bits

For the D-MAC/packet system, the unallocated bits interleaved with UDT, SDF and RDF bits as well as those after the repeated data frame (from bit position 1194 onwards) are filled for the present with pseudo-random data consisting of the output of the PRBS generator used for scrambling for spectrum shaping purposes (see Section 3.7 and Fig. 4). However, these bits may be used in the future for data yet to be defined.

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

Part 1

Table 2

Detail 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: $\boxed{0\ \ 0\ \ \pm\ \ 2^3}$
	0	4	$\boxed{2^2\ \ 2^1\ \ 2^0\ \ 2^{-1}}$
	1	x	Reserved for future use
	1	x	
	1	x	
	1	x	
	1	x	
	0	x	
	0	x	
	1	x	

Binary weights

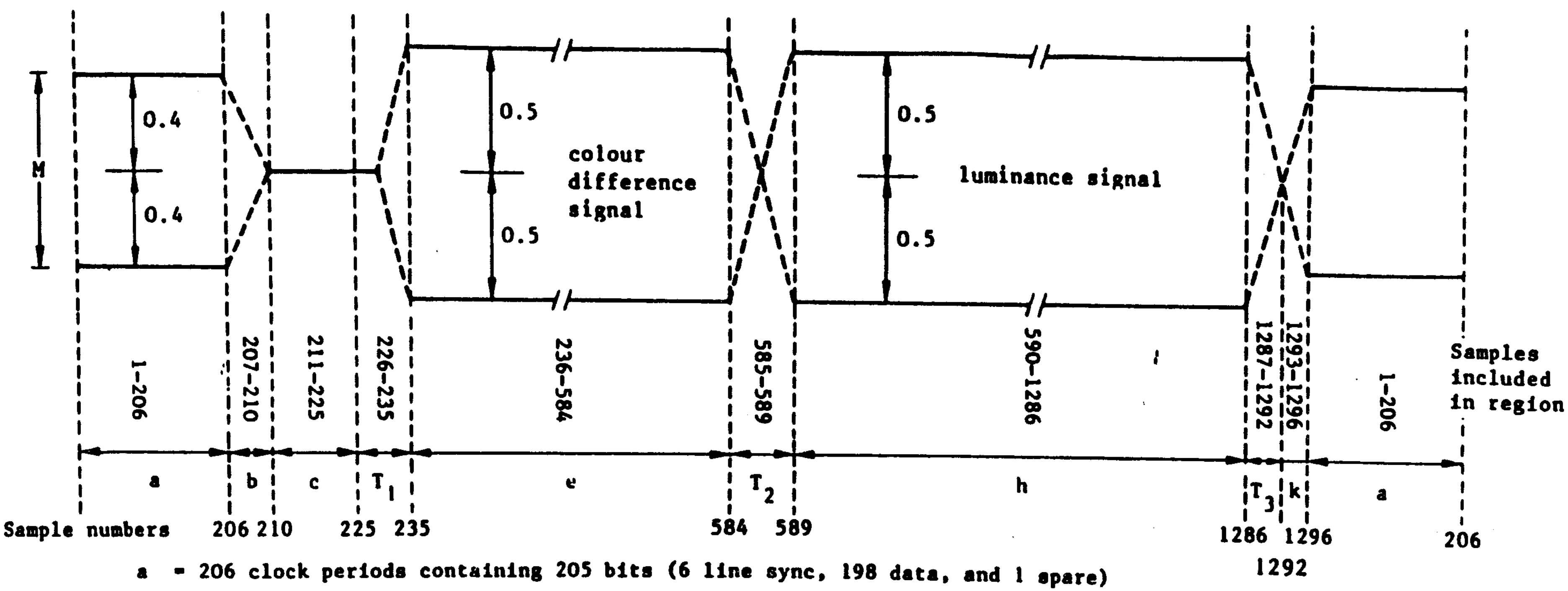
LOCAL OFFSET (hours)

Sign (0 = positive, Eastern hemisphere)

The example corresponds to 1983 April 19 09.31.17 French time

MJD Modified Julian Date

UTC Coordinated Universal Time



a = 206 clock periods containing 205 bits (6 line sync, 198 data, and 1 spare)

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 = 4 clock periods for transition into data, including one run-in bit

<u>Fig. 1a:</u> D-MAC/packet system - Approximate baseband signal waveform for normal unscrambled picture transmissions (not to scale)
Clock frequency: 20.25 MHz (see <u>Fig. 2</u> of <u>Part 2</u> for details)

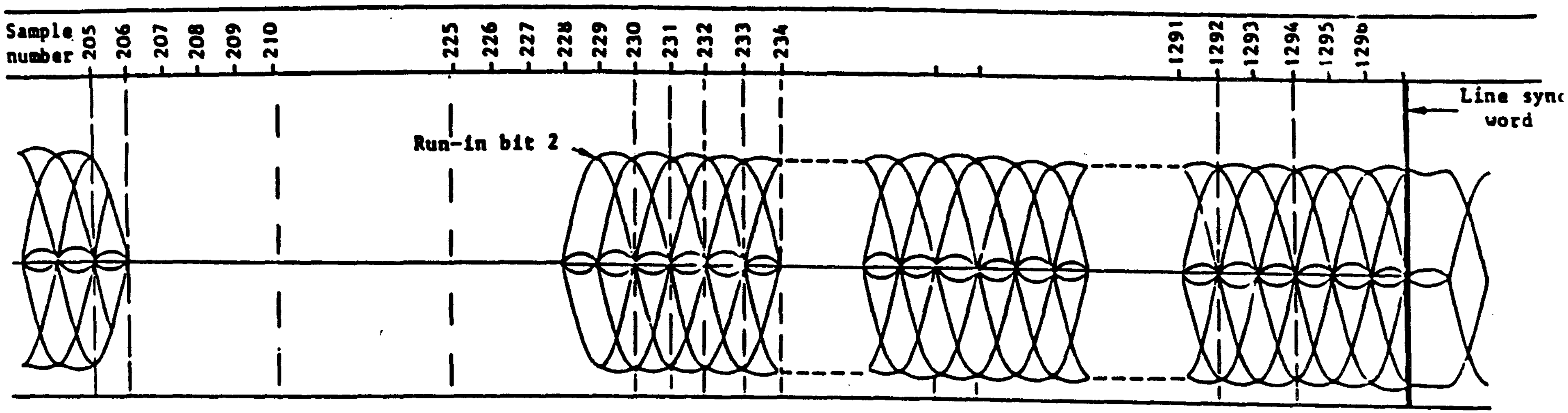


Fig. 1b: Relationship between the bits of data and the sampling structure in the case of full-channel digital mode of operation

Note to Fig. 1b

This Figure assumes that the first data burst is of normal length. If the first data burst is reduced in length, the clamping period and the start of the second data burst are moved forward accordingly. The first data burst must end not later than shown in the Figure.

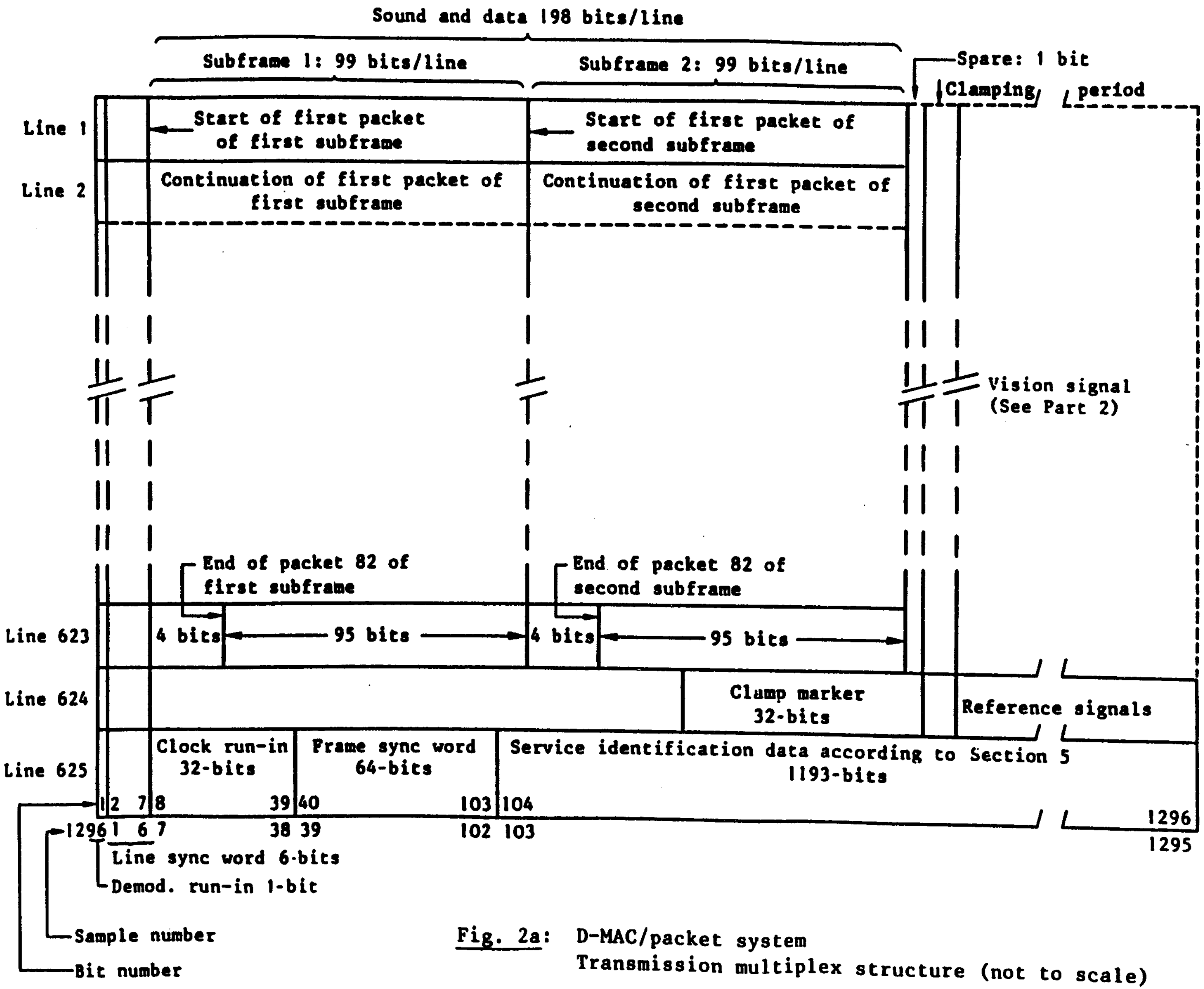
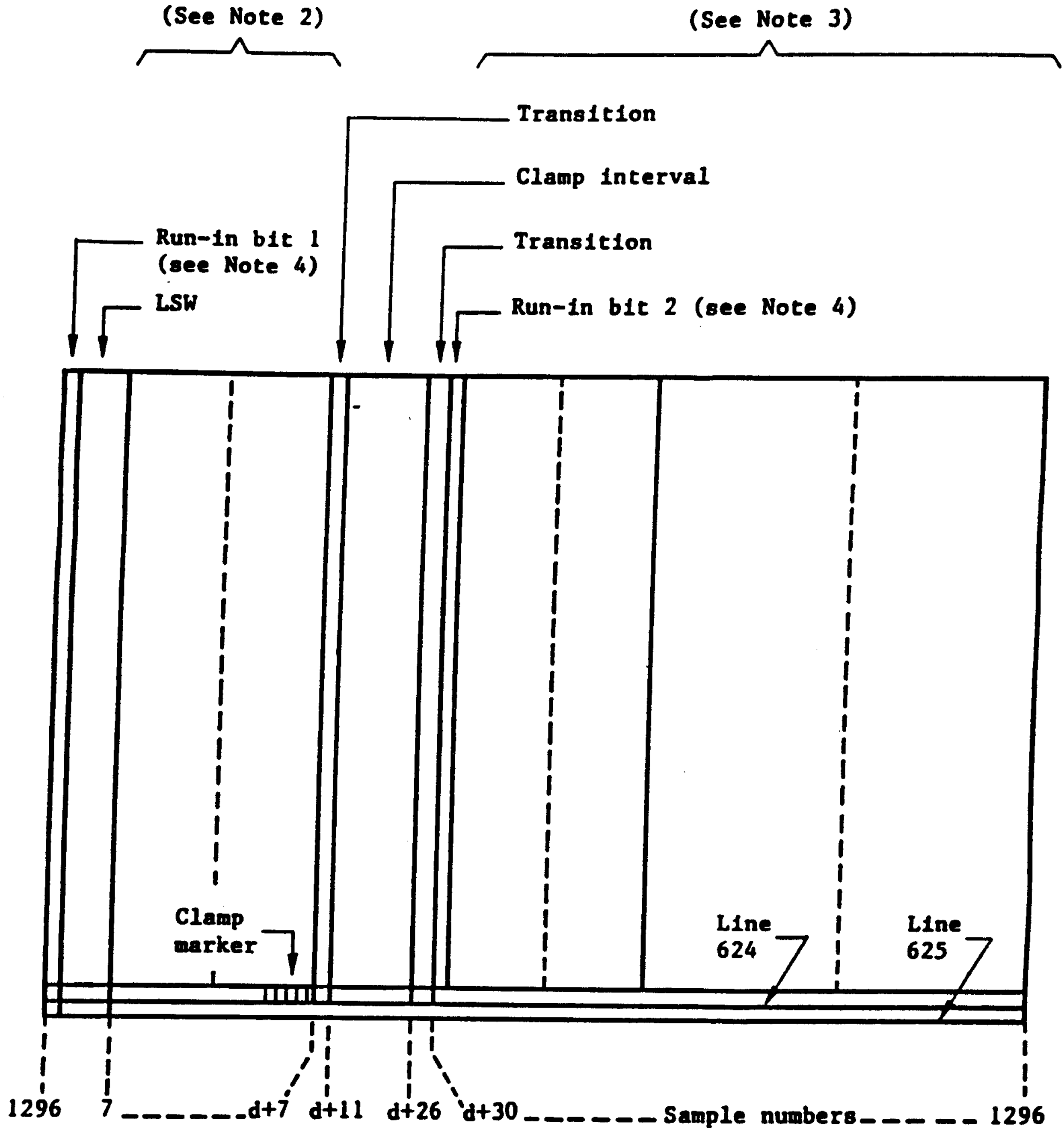


Fig. 2a: D-MAC/packet system
Transmission multiplex structure (not to scale)



d: duration in bits of the first data burst

Fig. 2b: Example of multiplex structure for D full-channel digital mode of operation (not to scale)

Notes to Fig. 2b

Note 1

The capacity for full-channel digital mode of operation is divided in TDM components. Each of the TDM components may occupy lines 1 to 623 inclusive.

Note 2

This part of the frame consists of TDM components with TDMCID codes 01 and 02. Its total duration is 198 bits (+ one spare bit) or less. It may contain a single TDM component.

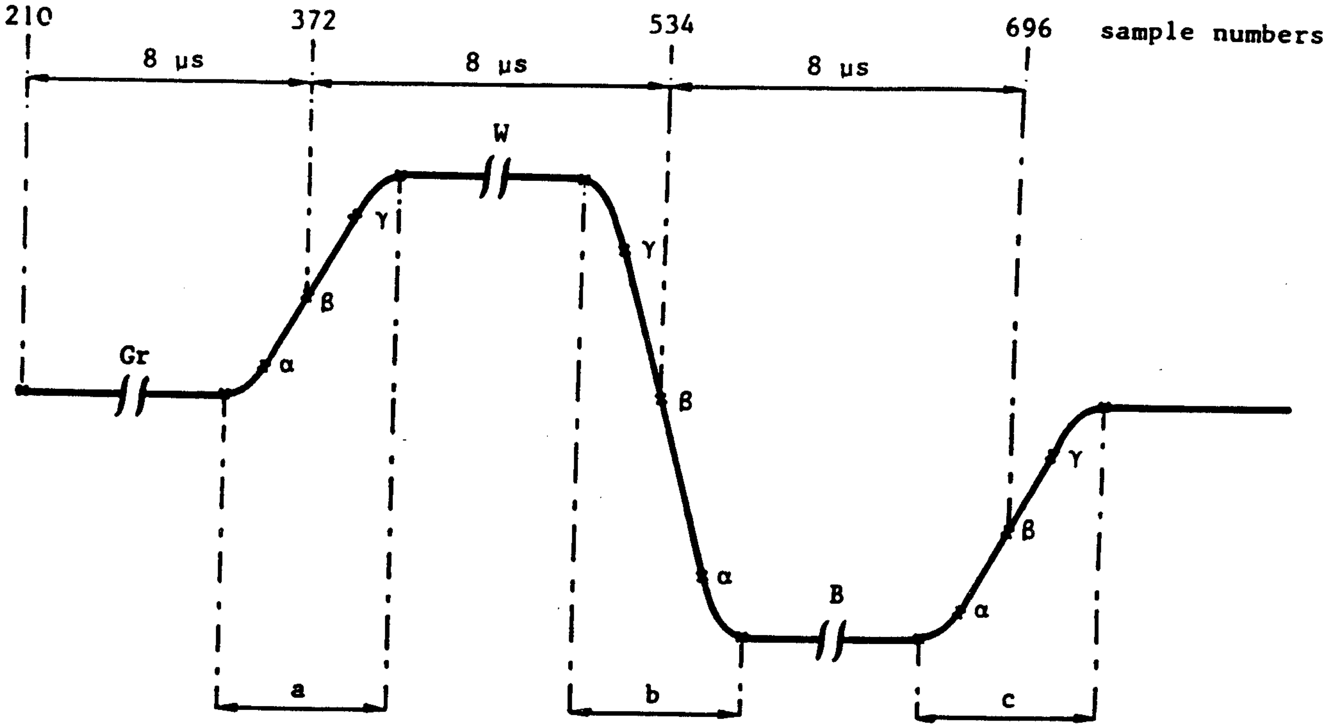
Note 3

This part of the frame consists of TDM components divided in two subframes or not. The TDMCID codes are 03 to 0E and 40 to 4F respectively. The duration of the TDM component is signalled in line 625.

Note 4

Run-in bits 1 and 2 are used in the C-MAC/packet system. Run-in bit 1 can be the last useful bit of the last digital TDM component as signalled in line 625.

Part I



Transitions

a: α = 7/8 (Gr) + 1/8 (W)
β = 1/2 (Gr) + 1/2 (W)
γ = 1/8 (Gr) + 7/8 (W)

b: α = 7/8 (B) + 1/8 (W)
β = 1/2 (B) + 1/2 (W)
γ = 1/8 (B) + 7/8 (W)

c: α = 7/8 (B) + 1/8 (Gr)
β = 1/2 (B) + 1/2 (Gr)
γ = 1/8 (B) + 7/8 (Gr)

Fig. 3: Waveforms of reference signals in line 624

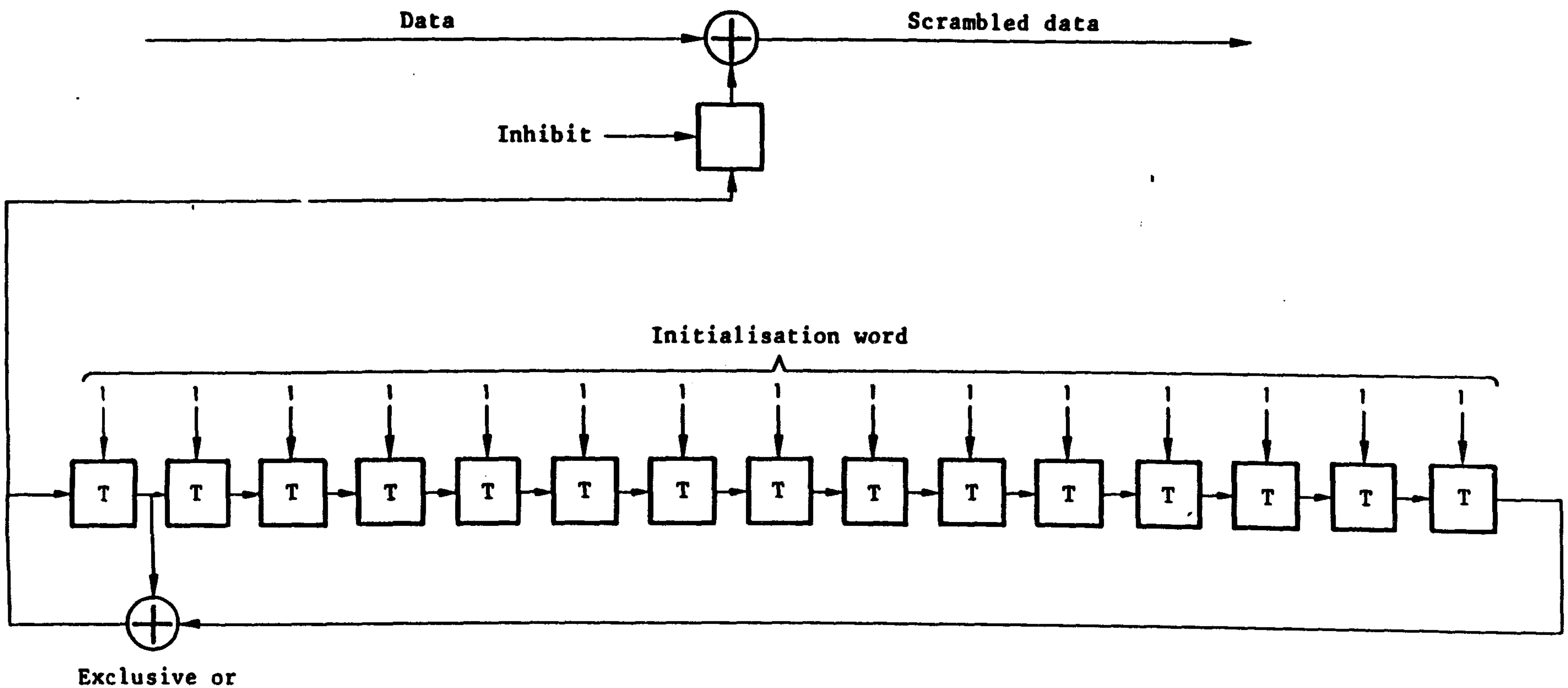
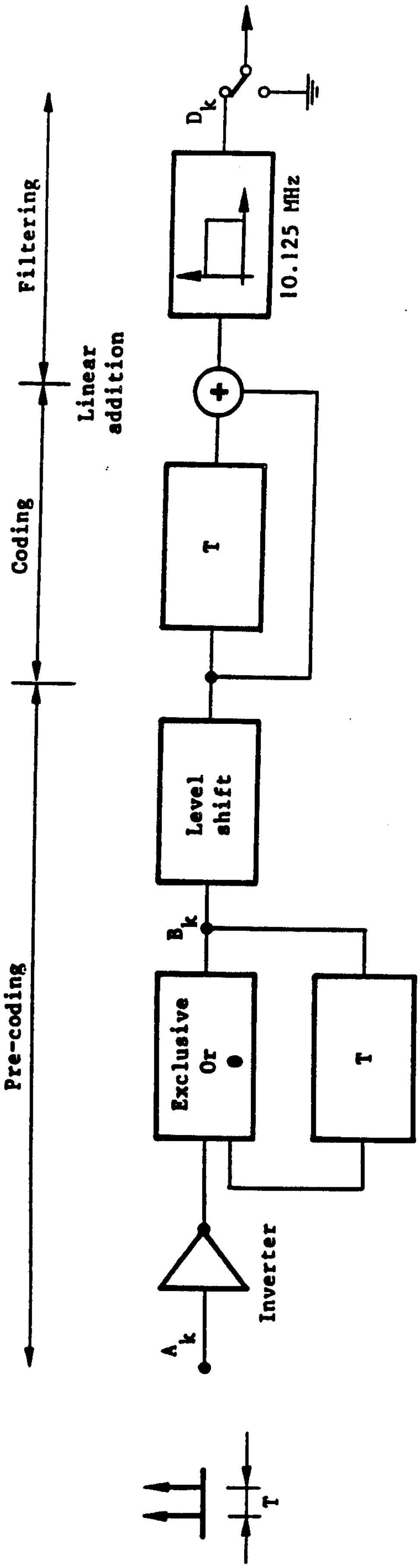


Fig. 4: Pseudo-random generator for scrambling and descrambling (for spectrum shaping purposes)



T = 49.4 ns (T is the bit interval)

Fig. 5: Coding method of a duobinary signal at 20.25 Mbit/s for D-MAC/packet system

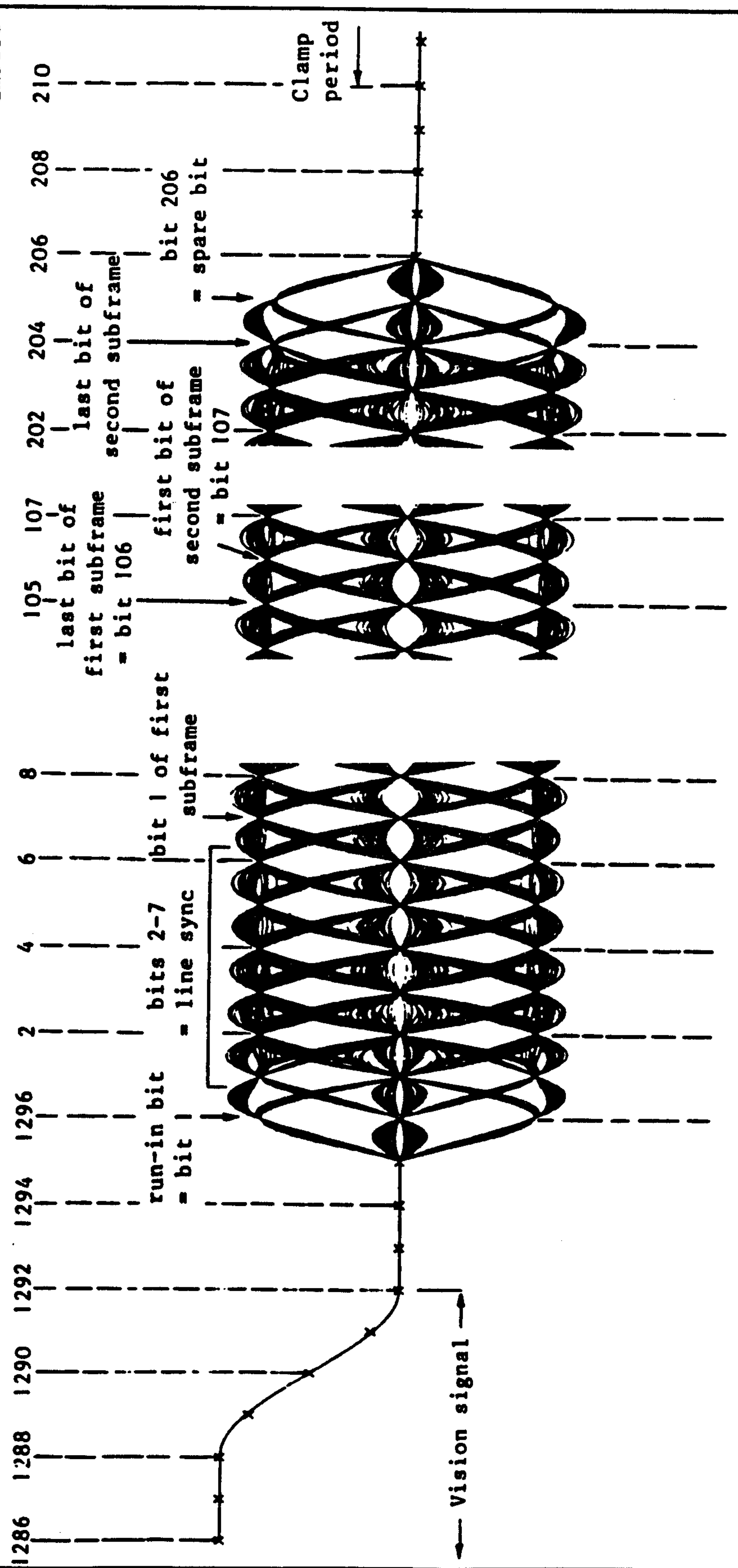


Fig. 6: Data period and transition for D-MAC/packet system

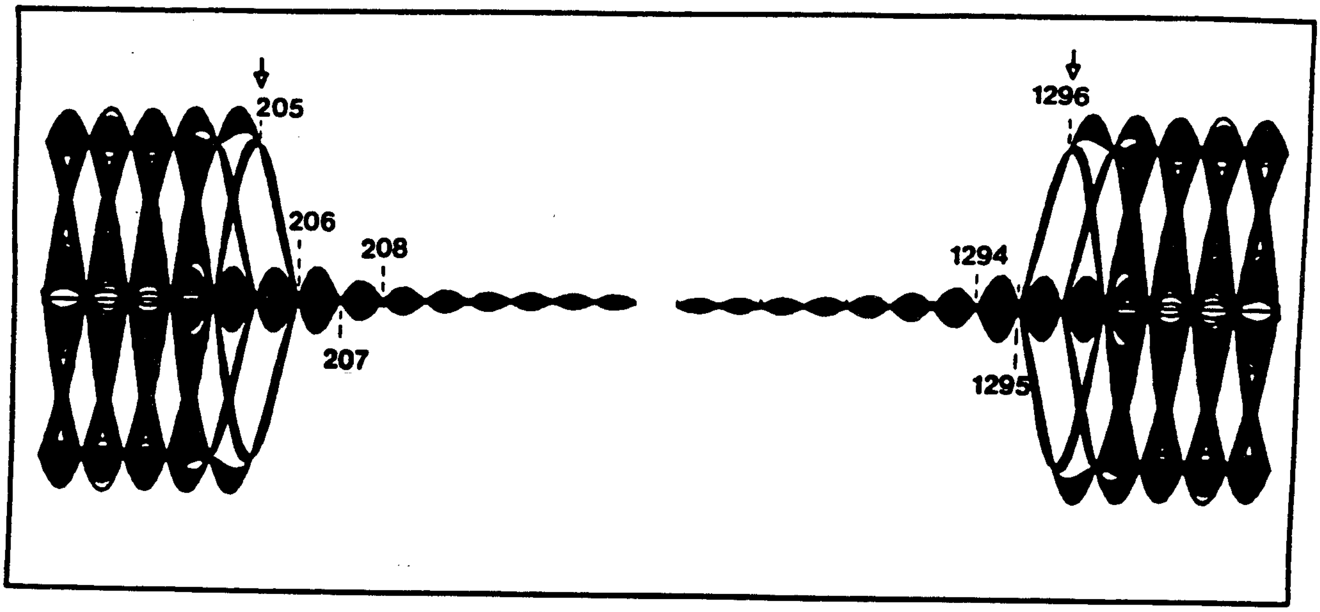


Fig. 7a

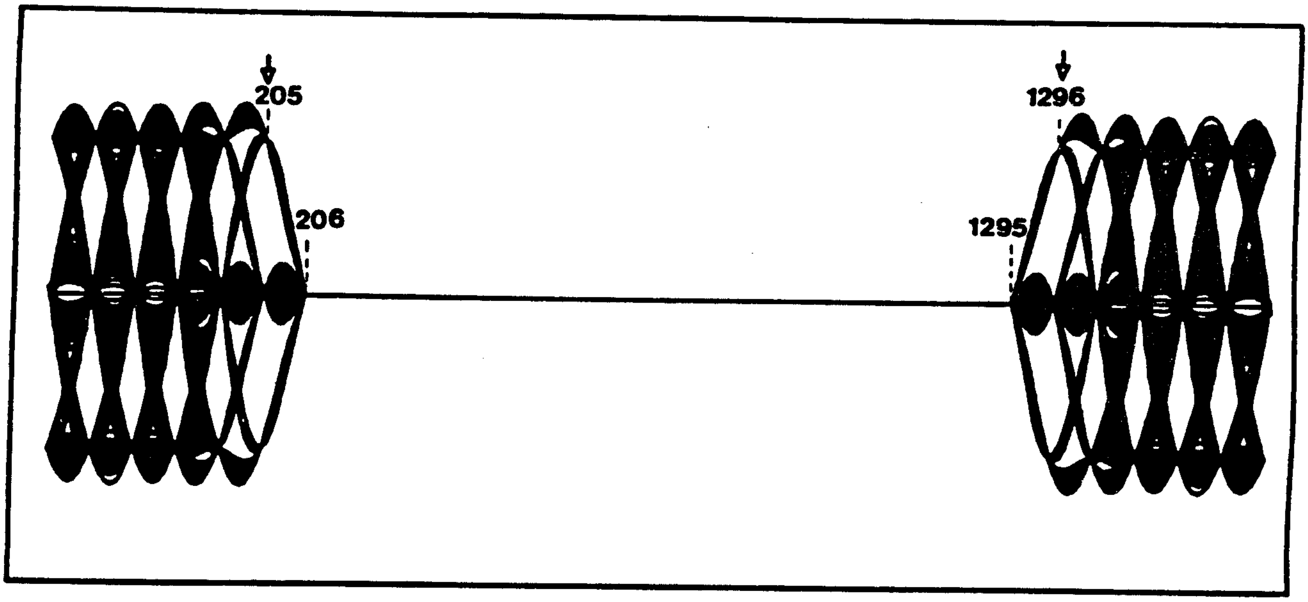


Fig. 7b

Data burst transitions obtained by interpolation with a half bit-rate rectangular filter in the D-MAC/packet system

7a: before forcing to zero clamping and vision periods

7b: after forcing to zero clamping and vision periods

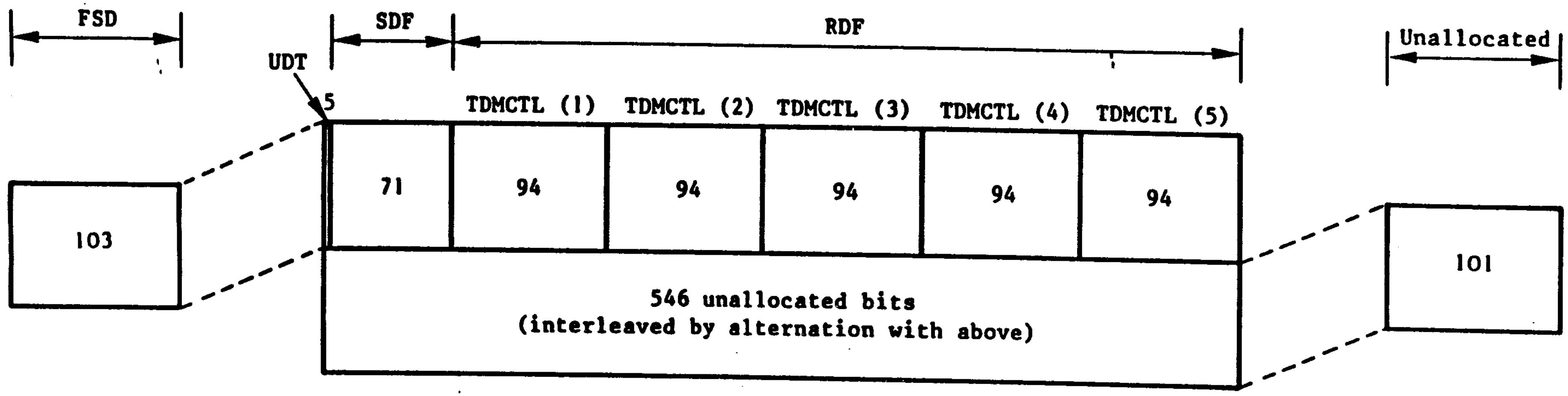


Fig. 8: Overall structure of line 625 for the D-MAC/packet system

The unallocated bits, some of which are interleaved by alternation with coding bits, are filled for the present with pseudo-random data (see Section 5.1).

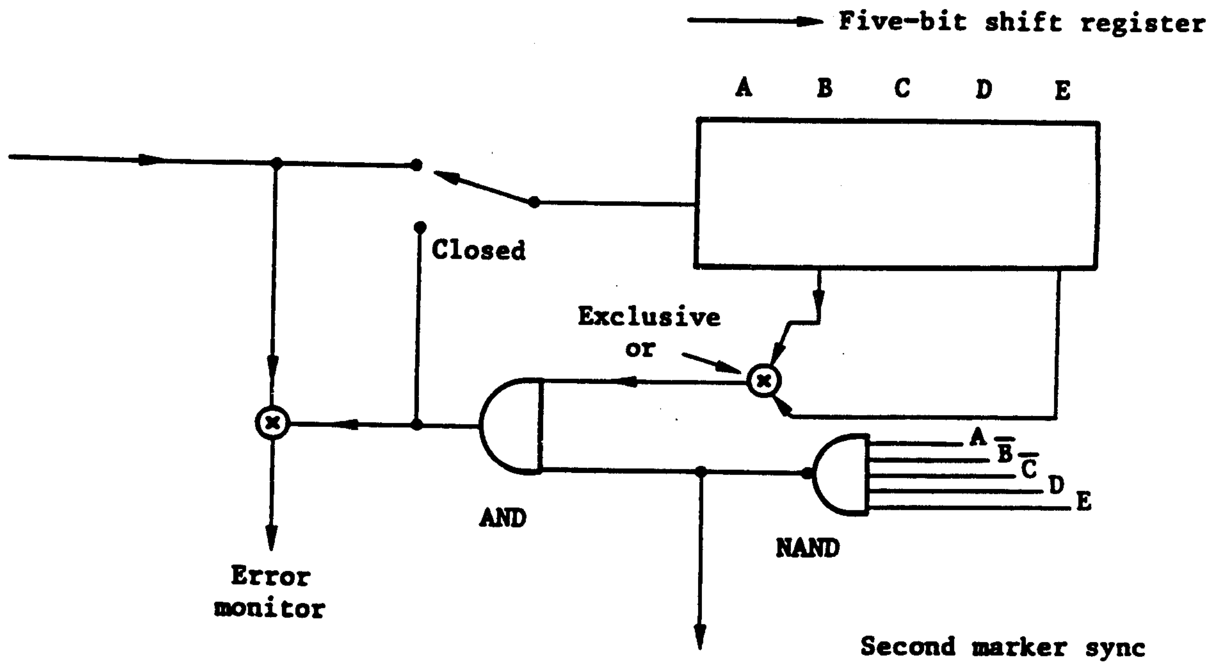
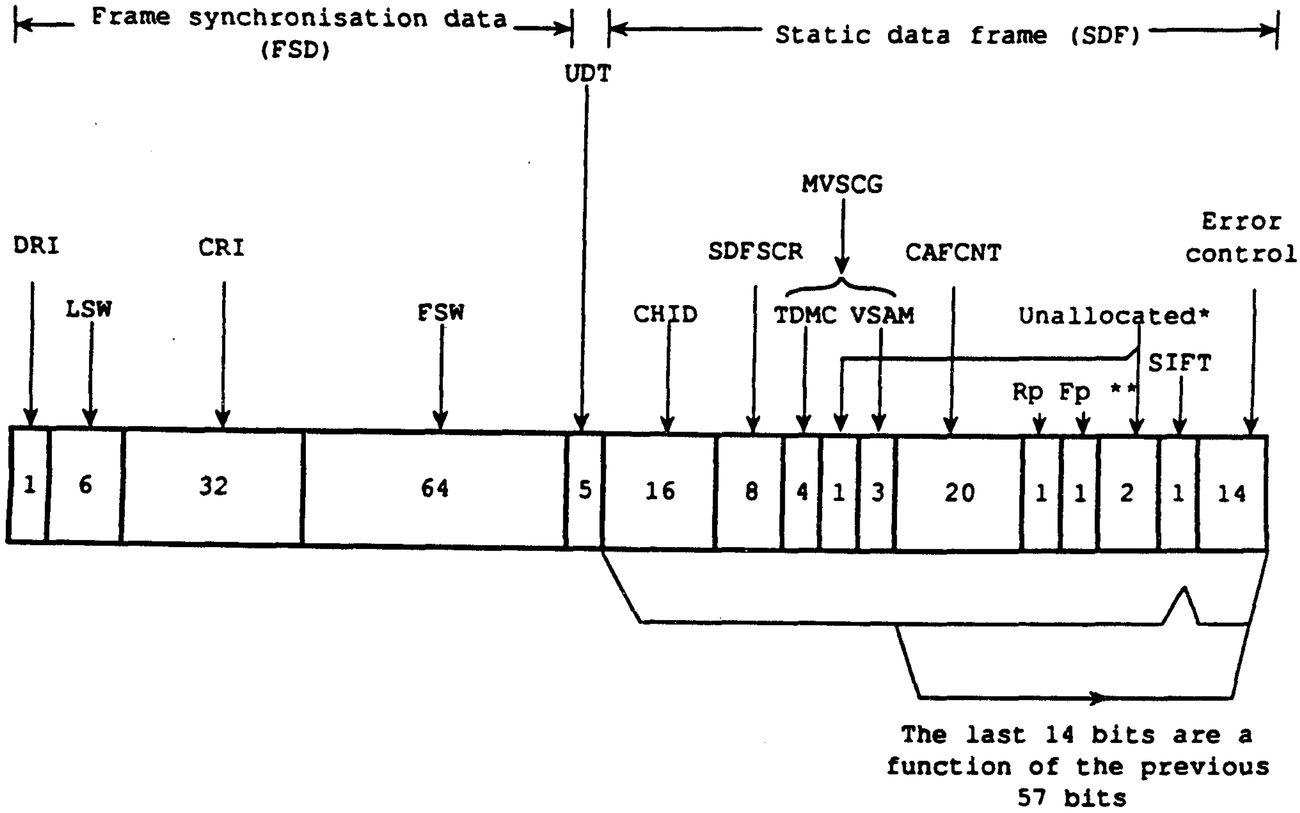


Fig. 9: Schematic diagram of sequence generator for the UDT chain code

Part 1



* These unallocated bits are set to '1'

** Rp, Fb: bits allocated in the conditional-access systems

Fig. 10: Frame synchronisation data, universal date and time part and static data frame block (line 625)

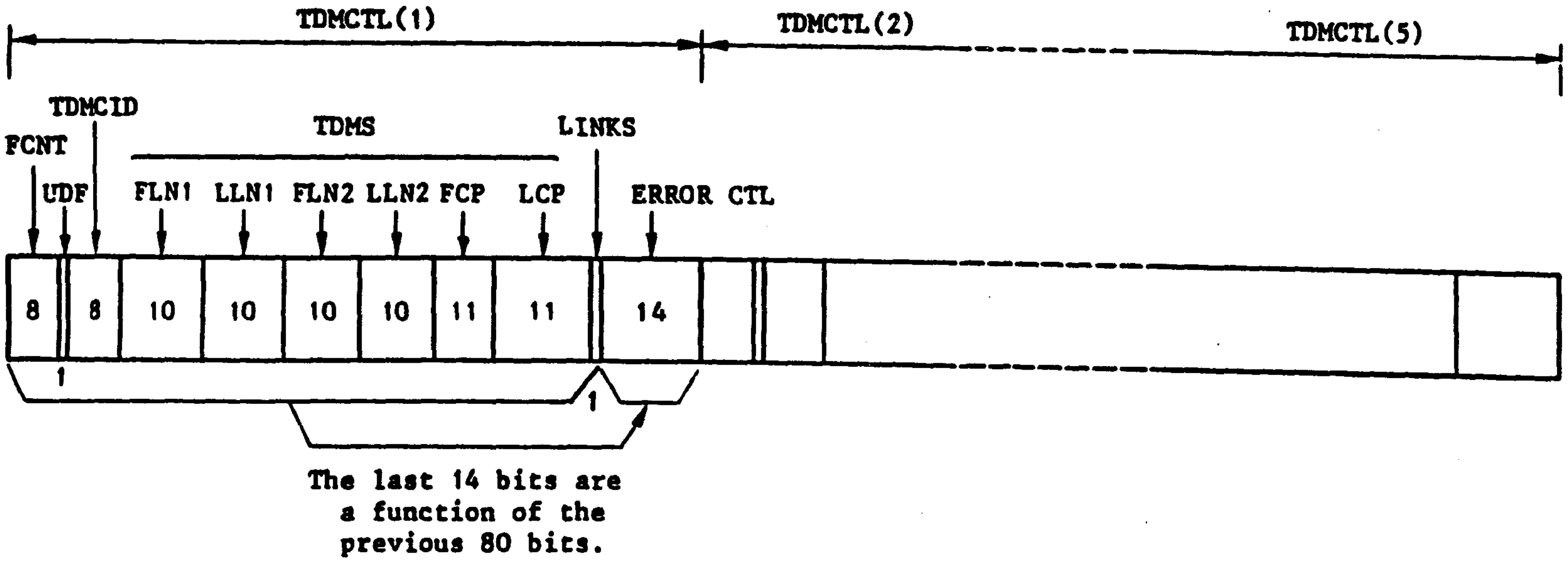


Fig. 11: Coding structure of the repeated data frame

CODING EXAMPLE FOR THE SPECIAL DATA IN LINE 625 FOR THE D-MAC/PACKET SYSTEM

In order to illustrate the specifications given in Section 5 of Part 1, the following example shows how information is sent in the static data frame, how the configuration of the TDM multiplex is described and how a modification of the configuration is signalled.

In the example, the initial TDM configuration comprises the following components:

- two sound/data subframes as of Fig. 2a of Part 1, identified by TDMCID codes 01 and 02 and designated as MPX 01 and MPX 02. Either one of them may be transcoded to D2 format at the choice of the cable operator
- a MAC vision signal under controlled access, scrambled according to the double-cut component rotation method and identified by TDMCID codes 10 and 11
- a fixed-format teletext signal on lines 1-11 and 313-323 inclusive, identified by TDMCID code 20
- a variable-format teletext signal on lines 12-22 and 324-334 inclusive, identified by TDMCID code 21
- insertion test signals on lines 312 and 623, identified by TDMCID code 30.

The initial configuration is modified by deletion of the MAC vision signal, the two teletext signals and the insertion test signals, and by addition of a new sound/data subframe, identified by TDMCID code 40 and designated as MPX 40. Note that no further details of this component are specified for the present, but it is used here to illustrate the mechanism for TDM modification and the potential flexibility of the D-MAC/packet system. The initial and modified TDM configurations are shown in Fig. 1.

Fig. 2 gives the coding example. The legend of codes in this Figure is the following:

a	CHID code value
b	SDFSCR code value corresponding to the sound services carried by the two sound/data subframes in the initial TDM configuration
c	SDFSCR code value corresponding to the sound services carried by the sound/data subframe identified by TDMCID code 01 in the modified TDM configuration
d	initial CAFCNT value

d+1 incremented CAFCNT value

E_{abd}, etc. bit combinations for error control of the static data frame

E_{247}, etc. bit combinations for error control of the repeated data frame (static over the fivefold RDF)

" indicates a code value that remains unchanged from the previous transmission having the same TDMCID code.

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 Section 5.4 of Part 1). In Fig. 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 Section 5.4 of Part 1.

Any unused bits in the static data frame are set to "1". This does not apply for the interleaved unallocated bits described in Section 5.1 of Part 1 and which are not shown in Fig. 2.

General comments on Fig. 2

The two initial sound/data bursts consist of only one subframe each, and consequently FLN2 and LLN2 are coded as 1023 (all "1"s). They occupy clock periods 7-105 and 106-204 in lines 1-623 inclusive (coded as 006-104, 105-203 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 236-584 in both fields (coded as 235-583), and the luminance comprises clock periods 590-1286 in both fields (coded as 589-1285). Refer also to Fig. 3 of Part 2.

The clock periods given for the teletext components are 230-1293 (coded as 229-1292). See Sections 2.1 and 2.2 of Part 4A. These signals occupy 11 lines each in field 1 and in field 2, as indicated in Fig. 1.

The new data subframe that replaces the MAC vision signal, the two teletext signals and the insertion test signals, turns the transmission format into a full-field data mode. This component is coded to use all free capacity outside the two initial data subframes and lines 624 and 625.

Notes to Fig. 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 TDM 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 80-160 ms (low BER). Maximum acquisition time for all (7) components is 360 ms (processing time is not included).

5. CAFCNT is incremented.

6. Start of new TDM structure definition. Old configuration is still transmitted. MPX 01 and MPX 02 remain unchanged and can be acquired.

7. TDM components flagged for deletion cannot be acquired 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 three times prior to the change of structure (corresponding to 960 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. The reconfiguration involves modification of both sound and vision (i.e. the television sound is deleted), which is reflected in the change of SDFSCR and MVSCG codes.

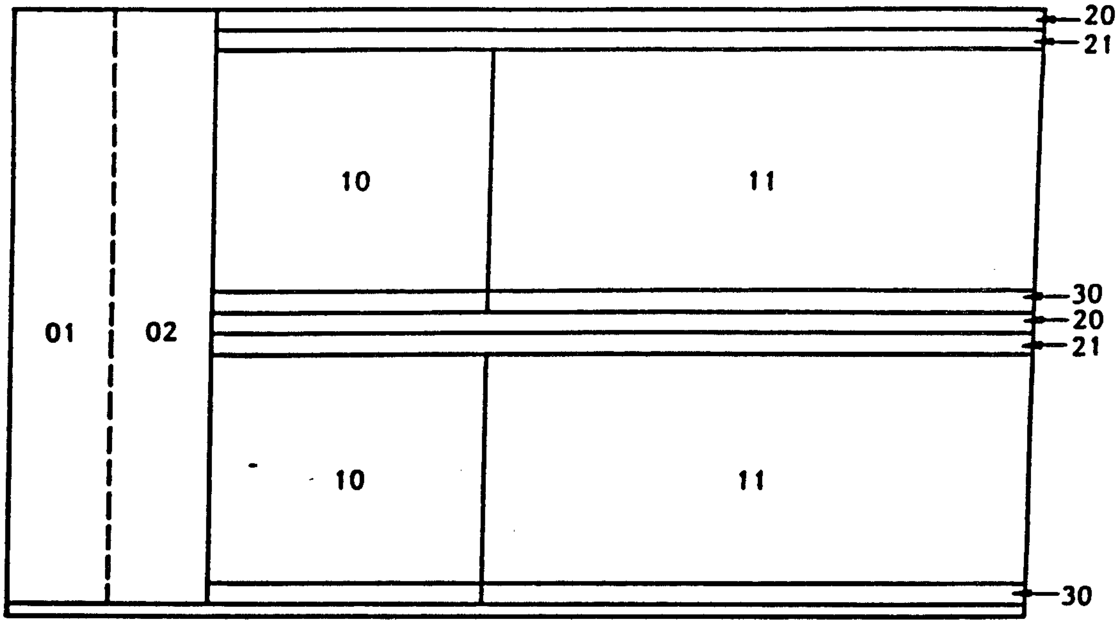
12. New configuration is transmitted (2 x 99-bit and 1067-bit bursts), starting from line 1 of this television frame.

13. Turns off decoders that may have failed to recognize the deletion earlier. These TDMS data will only be transmitted temporarily.

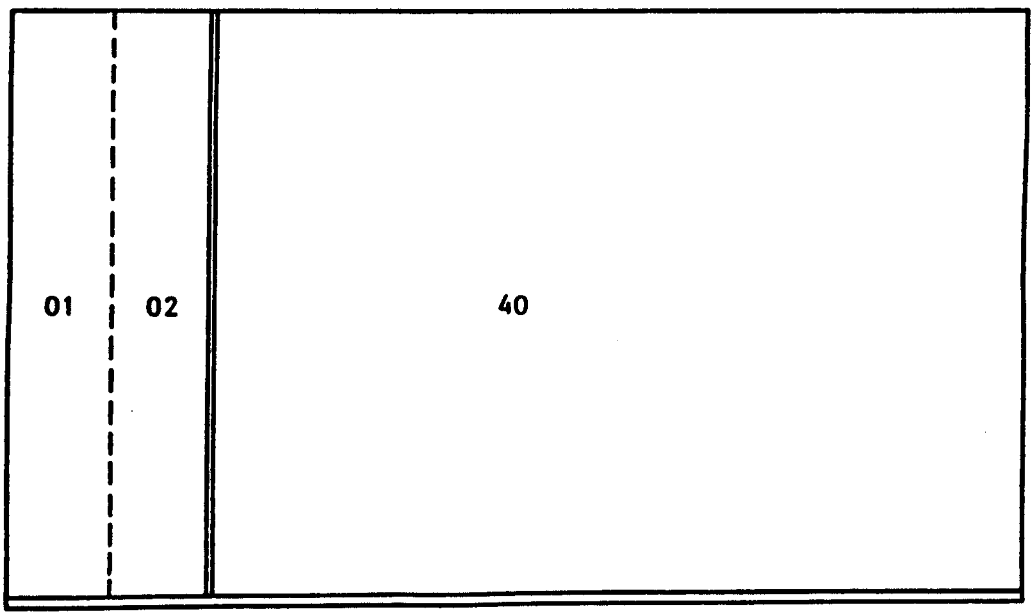
14. The VSAM code remains unchanged but has no meaning since no vision signal is transmitted.

Note 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 data burst(s).

Notes 9 and 13 indicate practical rules aimed at minimising the delay and maximising the security of TDM reconfigurations.



a) Initial TDM configuration



b) Modified TDM configuration

Fig. 1: The initial and modified TDM configurations with the corresponding TDMCID codes

Static Data Frame — Repeated Data Frame

114 — Clock 254 — period 256 — TDMCTL(1) — 442 — 444 TDMCTL codes are repeated 5 times

TDMS — LINKS

CHID	SDFSCR	MVSCG (TDMC)	CAFCNT (VSAM)	CAFCNT	UNALLOCATED	ERROR CONTROL	FCNT	UDF	TDMCID	FLN1	LLN1	FLN2	LLN2	FCP	LCP	LINKS	ERROR CONTROL	TDM COMPONENT DESCRIPTION	NOTES
16	8	8	20		5	14	8	1	8	10	10	10	10	11	11	1	14		
a	b	1101	001	d	ALL BITS SET TO "1"	E_{abd}	246	0	01	000	622	1023	1023	006	104	1	E_{246}	MPX 01 data burst,(99 bits)	(1)
.	.	.	.	.		.	247	0	02	900	622	1023	1023	105	203	1	E_{247}	MPX 02 " ,(9* bits)	
.	.	.	.	.		.	248	0	10	022	309	334	621	235	583	0	E_{248}	COIFF Colour DIFFerence signal	
.	.	.	.	.		.	249	0	11	022	309	334	621	589	1285	0	E_{249}	LUM LUMinance signal	(3)
.	.	.	.	.		.	250	0	01	.	.	.	.	.	.	0	E_{250}	MPX 01	(2)
.	.	.	.	.		.	251	0	02	.	.	.	.	.	.	0	E_{251}	MPX 02	
.	.	.	.	.		.	252	0	20	000	010	312	322	229	1292	0	E_{252}	FF Fixed Format teletext	
.	.	.	.	.		.	253	0	21	011	021	323	333	229	1292	0	E_{253}	VF Variable Format teletext	
.	.	.	.	.		.	254	0	30	311	311	622	622	235	1285	0	E_{254}	ITS	
.	.	.	.	d		E_{abd}	255	0	01	.	.	.	.	.	.	1	E_{255}	MPX 01	(4)
.	.	.	.	d+1		E_{abd+1}	000	0	02	.	.	.	.	.	.	1	E_{000}	MPX 02	(1),(5)
.	.	.	.	.		.	001	0	10	.	.	.	.	.	.	1	E_{001}	COIFF	
:	:	:	:	:		:	:	:	:	:	:	:	:	:	:	:	:	STATIC CONDITIONS	
.	.	.	.	.		.	106	0	11	.	.	.	.	.	.	0	E_{106}	LUM	
.	.	.	.	.		.	107	0	01	.	.	.	.	.	.	0	E_{107}	MPX 01	
.	.	.	.	.		.	108	0	01	.	.	.	.	.	.	1	E_{108}	MPX 01	(6)
.	.	.	.	.		.	109	0	02	1023	1023	1023	1023	2047	2047	0	E_{109}	MPX 02	
.	.	.	.	.		.	110	1	03	000	622	1023	1023	106	1285	0	E_{110}	MPX 40 new data burst, (1067 bits)	(8)
.	.	.	.	.		.	111	1	10	1023	1023	1023	1023	2047	2047	1	E_{111}	COIFF is flagged for deletion	
.	.	.	.	.		.	112	1	11	1023	1023	1023	1023	2047	2047	1	E_{112}	LUM "	(7)
.	.	.	.	.		.	113	1	20	1023	1023	1023	1023	2047	2047	0	E_{113}	FF "	
.	.	.	.	.		.	114	1	21	1023	1023	1023	1023	2047	2047	0	E_{114}	VF "	
.	.	.	.	.		.	115	1	30	1023	1023	1023	1023	2047	2047	1	E_{115}	ITS "	
.	.	.	.	.		.	116	0	01	.	.	.	.	.	.	0	E_{116}	MPX 01	
.	.	.	.	.		.	117	1	02	.	.	.	.	.	.	1	E_{117}	MPX 02 "	
:	:	:	:	:		:	:	:	:	:	:	:	:	:	:	:	:	TRANSIENT CONDITIONS	(9)
.	.	.	.	.		.	126	1	11	.	.	.	.	.	.	1	E_{126}	LUM	
.	.	.	.	.		.	127	1	20	.	.	.	.	.	.	0	E_{127}	FF	
.	.	.	.	.		.	128	1	21	.	.	.	.	.	.	0	E_{128}	VF	
.	b	1101	.	.		E_{abd+1}	129	0	01	.	.	.	.	.	.	0	E_{129}	MPX 01	(10)
.	c	0101	.	.		E_{acd+1}	130	0	02	.	.	.	.	.	.	1	E_{130}	MPX 02	(11)
.	.	.	.	.		.	131	0	40	.	.	.	.	.	.	1	E_{131}	MPX 40	(11),(12)
.	.	.	.	.		.	132	0	10	.	.	.	.	.	.	0	E_{132}	COIFF is deleted	
.	.	.	.	.		.	133	0	01	.	.	.	.	.	.	1	E_{133}	MPX 01	
.	.	.	.	.		.	134	0	02	.	.	.	.	.	.	0	E_{134}	MPX 02	
.	.	.	.	.		.	135	0	40	.	.	.	.	.	.	0	E_{135}	MPX 40	
a	c	0111	001	d+1		E_{acd+1}	136	0	11	.	.	.	.	.	.	0	E_{136}	LUM "	
:	:	:	:	:		:	:	:	:	:	:	:	:	:	:	:	:	STATIC CONDITIONS	(13),(14)

Fig. 2: Line 625 coding example

PART 2: SPECIFICATION OF THE VISION SIGNAL

Contents

Part 2

Part 2

1. Subject of Part 2

This Part gives the specification of the Multiplexed Analogue Components (MAC) vision signal which is used in association with the D type modulation system using the time-division multiplex as described in Part 1. Both non-scrambled and scrambled signals are specified.

2. General video characteristics

2.1 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.

2.2 Number of lines per picture

The number of lines per picture shall be 625.

2.3 Interlace

The interlace ratio shall be 2:1.

2.4 Aspect ratio

The normal ratio of the image width to image height within the picture area shall be 4:3. Provision is made by means of the TDM control, MVSCG (see Section 5 of Part 1), and VCONF (see Part 5) to allow transmission of an aspect ratio of 16:9 (see Section 7).

2.5 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.

2.6 Baseband format

The luminance component (Section 4.2) and the two colour-difference components (Section 4.3) are compressed in time so that each transmitted line can contain:

- the compressed luminance signal
- one of the two compressed colour-difference signals (which alternate from line to line)
- and a multiplex of digital sound and data.

This format gives complete separation of luminance, colour-difference and digital sound and data in time division multiplex as shown in Fig. 1. The non-scrambled waveform is shown in Fig. 2. The scrambled waveforms are shown in Fig. 6.

In the vision decoder the luminance and colour difference signals from each line must be expanded and read out for display.

The time compression and expansion are achieved by sampling the analogue signal, storing the samples, and reading them out at the appropriate time at a higher or lower clock frequency. All timing information in the transmitted signal is related to a notional 20.25 MHz sample rate for the compressed signal. This is 13.5 MHz multiplied by the luminance compression ratio (3:2). 13.5 MHz is the luminance sampling-frequency recommended by the CCIR for digital video studio signals, and therefore the 20.25 MHz clock frequency representation is a convenient reference system for describing the video signal in compressed form.

2.7 Sampling clock frequency

This is defined as:

$$f_{ck} = 20.25 \text{ MHz} \pm 2.5 \text{ parts in } 10^7$$

This frequency is exactly the same as that given in Section 3.1 of Part 1.

2.8 Relationship between clock frequency and line frequency

$$f_h = \frac{2}{3} \times \frac{1}{864} \times f_{ck}$$

(f_{ck} is defined in Section 2.7)

2.9 Field frequency

$$f_{field} = \frac{2}{625} \times f_h$$

3. Synchronization and TDM control

Details of line synchronization, field synchronization and overall TDM control are given in Part 1.

3.1 Colour sequence identification

The colour sequence identification is derived directly by a line count from the frame sync. The odd lines of the frame carry the U component of the colour sequence information and the even lines carry the V component.

3.2 Line numbering

Fig. 3 shows the line numbering system for the MAC/packet systems. Some waveform details are included for clarity. The MAC/packet systems have a two-frame sequence and the sequence is defined by the pattern of line synchronization words at either side of the frame boundary and also by sequence of frame synchronization words. All lines contain a data burst, and the data burst on line 625 is extended to cover the whole line. The lines are numbered consecutively in each field. The colour-difference signals alternate with consecutive line numbers and so appear as UU, VV, etc. on the displayed picture.

3.3 TDM control

The TDM control is described in Section 5.4 of Part 1.

Data for control of the time division multiplex is sent in the special data burst in line 625. This control data provides the means of obtaining flexibility for future developments. It is important that all receivers decode and use this information.

The TDM control information (TDMCTL) allows the definition of separate rectangular subframes for each component of the time-division multiplex within the frame of 625 lines by 1296 clock periods. Components for which subframes are defined include the sound/data burst(s), luminance, colour difference, field blanking teletext, etc. For scrambled video, the TDMCTL information refers to the format before scrambling and after correct descrambling.

Each subframe is defined by sending the following information:

- first clock period of the subframe
- last clock period of the subframe
- first line of the subframe
- last line of the subframe.

The transition 'b' and the clamp period 'c' (see Fig. 2) remain fixed with respect to the end of the data burst.

4. The picture signal

4.1 General specification

The colour picture signal shall comprise a luminance (brightness) component and a pair of colour-difference components, time compressed and transmitted sequentially in a time division multiplex. The luminance component is transmitted on each line with the colour-difference components being transmitted on alternate lines (Fig. 1). The scrambled picture signal is specified in Section 6.

4.2 Luminance component

An increase in the incident light intensity shall correspond to an increase in the amplitude of the video signal. For a colour signal, the relationship between the spectral composition of the incident light and the luminance amplitude shall be as defined in Section 5.2.

4.2.1 Luminance amplitude range

The maximum amplitude of the luminance signal shall be 1 V p-p (from black level to white level). Black level shall be at -0.5 V with reference to the clamping level.

4.2.2 Uncompressed amplitude/frequency characteristic

The uncompressed amplitude/frequency characteristic of the luminance signal shall comply with the digital component studio standard (CCIR Rec. 601).

4.2.3 Luminance compression

The luminance compression ratio shall be 3:2. Consequently the compressed amplitude frequency characteristic of the luminance signal shall be the one defined in Section 4.2.2 increased in the ratio 3:2.

4.3 Colour-difference components

The two colour-difference signals are defined in Section 5.3. These are scaled according to Section 5.4 to form the uncompressed colour-difference signal voltages $E'U_m$ and $E'V_m$.

Part 2

4.3.1 Colour-difference amplitude range*

The maximum amplitude of the colour-difference signals shall be 1.0 V p-p, (corresponding to 77% electrical saturation, or 98% displayed saturation, at 100% amplitude).

4.3.2 Uncompressed amplitude/frequency characteristic

The uncompressed amplitude/frequency characteristic of the colour-difference signal shall comply with the digital component studio standard (CCIR Rec. 601). It is assumed that a slow roll-off filter of the type envisaged in Annex 1 of CCIR Report 629-2 for picture monitors and composite encoders, will be provided in the receiver.

4.3.3 Colour-difference compression

The colour-difference signal compression ratio shall be 3:1. Consequently, the compressed amplitude/frequency characteristic of the colour-difference signal shall be the one defined in Section 4.3.2 increased in the ratio 3:1.

4.3.4 Colour-difference line sequence

The colour-difference signal voltages are 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 is spatially coincident with a luminance signal but is transmitted on the line prior to that carrying coincident luminance. $E'U_m$ is sent on odd lines of the picture. This arrangement is chosen to minimise the amount of memory required in receivers to implement a 1,2,1 post-filter used to interpolate the colour-difference information. Without field stores, the maximum vertical resolution theoretically obtainable without aliasing is equivalent to a horizontal resolution of 1.85 MHz. The practically obtainable resolution depends on the pre- and post-filters used to reduce alias effects to a level which is not disturbing. The post-filter design should be such that the luminance and chrominance are in registration on the display.

4.4 Transcoding from PAL or SECAM to MAC

Where a MAC video signal is derived directly from a PAL or SECAM signal, the odd frame of the MAC 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.

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* It should be noted that the results given in CCIR Report 634 used MAC with a higher colour-difference amplitude of 1.3 V p-p (corresponding to 100% saturation) but with reduced bandwidth (2 MHz). The compressed colour-difference and luminance channels specified here have identical characteristics. The interference performance of the colour-difference signal is therefore that of the luminance signal, which remains unchanged.

4.5 Field-blanking interval

Video signals shall be transmitted corresponding to picture lines 24 to 310 and 336 to 622 inclusive. There is colour-difference information on lines 23 and 335 as transmitted; on these lines the luminance is set to black to provide a black level reference. Line 624 is reserved for reference signals (see Section 3.4 of Part 1). Line 625 is completely reserved for frame synchronization and service indentification data (see Section 5 of Part 1). All remaining lines may be used for other purposes, such as field synchronization, teletext, test lines, additional picture information or additional sound/data. For some of these purposes the receiver displays will require field blanking.

4.6 Signal timing

Sections of the waveform are provided for clamping of the video signal (see Figs. 1 and 2 of Part 1). The clamping period is period 'c' of Fig. 2 and this period occurs on every line with the exception of line 625. The position of this period, together with other picture timing information, is specified by signals in the data burst as described in Part 1.

5. Colour signal equations

5.1 Primary colour chromaticities

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

5.2 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:

$$E'_Y = 0.299\ E'_R + 0.587\ E'_G + 0.114\ E'_B$$

5.3 Unscaled colour-difference signal voltage

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

$$E'_R - E'_Y = 0.701\ E'_R - 0.587\ E'_G - 0.114\ E'_B$$

$$E'_B - E'_Y = -0.299\ E'_R - 0.587\ E'_G + 0.886\ E'_B$$

These equations are derived from Section 5.2.

5.4 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 apply only for frequencies within the colour-difference bandwidth (see Section 4.2.2) and are subject to the limits in Section 4.3.1.

6. Picture scrambling

6.1 Introduction

The general principles of conditional access are described in Part 6. This Part gives only a description of the scrambled picture waveforms used for conditional access.

6.2 Description of the picture signal scrambling process

The non-scrambled video waveform has been described in Fig. 2. 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 and (b) single-cut in colour-difference line rotation.

The vision scrambling may be carried over to cable networks, and in this case the MAC/packet system would have to be transcoded for distribution as AM/VSB. The two scrambling principles described above differ in security and sensitivity to transmission impairments that might arise in AM/VSB cable systems used to distribute the signal. 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 VSB distortion. The scrambling scheme used is chosen by the service operator on the basis of the security and transmission requirements. The signalling which indicates which scheme is being used is described in Section 5.3 of Part 1 and in Section 2.3 of Part 5.

It is thought that either of the schemes would be sufficiently resistant to distortions arising in transmission and direct reception from satellites. However, clamping and low frequency distortions arising in AM/VSB cable systems may cause visible impairments in the descrambled signal.

Vision scrambling will only be applied to lines containing vision signals. These are defined in signalling given in the RDF of line 625, see Section 5.4 of Part 1.

6.2.1 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; to restore the signal to its intelligible form the cut points must be identified and the segments retransposed within the receiver. The cut points are varied in a pseudo-random way from line to line.

Figs. 4a and 4b illustrate 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 is chosen within the defined range, using numbers derived from a pseudo-random sequence. Independent cut-points are 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 is 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 is shown in Figs. 5a and 5b. For each line, the numbers of the two cut points are obtained from the 16-bit binary word generated by the PRBS system described in Section 3.3 of Part 6. The eight most significant bits represent the colour-difference cut-point number and the eight least significant bits represent the luminance cut-point number.

The scrambling process introduces extra transitions into the waveform. Edge effects at these transitions could impair the descrambled picture. To reduce the influence of edge effects, overlap samples are included on both sides of each cut and the transitions are 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 Fig. 6a, which describes the scrambled waveform in detail. Two possible shapings of the transitions are shown in Table 1. The shaping should ensure that the spectrum of the transitions is kept well within the signal bandwidth to minimise ringing.

To conceal landmarks in the waveform that could help piracy, the transitions between the segments of each component are replaced by smooth cross-fades as shown in Fig. 6a and Table 1.

6.2.2 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. To restore the signal to its intelligible form, the colour-difference segments must be identified and the first segment moved back to its correct position. The cut point is varied in a pseudo-random way from line to line.

Fig. 4c 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 is 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 is shown in Fig. 5a. For each line the number of the cut points is given by the eight most significant bits of the 16-bit binary word generated by the PRBS system described in Section 3.3 of Part 6.

The scrambling process introduces extra transitions into the waveform. Edge effects at these transitions could impair the descrambled picture. To reduce the influence of edge effects, overlap samples are included on both sides of the cut. The extra transitions are shaped by a raised-cosine function as shown in Table 2. The overlap samples are shown as P-1, P, P+1 and P+2 in Fig. 6b, which describes the scrambled waveform in detail. Two possible shapings of the transitions are shown in Table 2. The shaping should ensure that the spectrum of the transitions is kept well within the signal bandwidth to minimise ringing.

To conceal landmarks in the waveform that could help piracy, the transitions between the components are replaced by a smooth crossfade as shown in Fig. 6b and Table 2.

7. Extended aspect ratio pictures

The MAC waveform can be used to transmit pictures with an aspect ratio of 16:9. Fig. 7 describes this waveform:

- data burst is left unchanged (105 bits per burst)
- 349 samples (related to a notional 20.25 MHz sample rate) are used for the colour-difference component
- 697 samples (related to a notional 20.25 MHz sample rate) are used for the luminance component.

Fig. 7 also defines the compatible 4:3 part of the picture:

- 262 samples for colour-difference component
- 523 samples for luminance component.

This figure defines the central position of the 4:3 compatible part of the picture when panning vector has a value equal to 0. However, when the panning vectors are processed (see Section 5.5 of Part 1), the visible part of the picture varies accordingly.

Part 2

The expansion ratios at the decoder are 3:2 for the luminance and 3:1 for the colour-difference component when the whole 16:9 picture is to be displayed.

If only the compatible 4:3 part s to be processed, then the expansion ratios will be 2:1 for the lumi. ce and 4:1 for the colour-difference component.

Part 2

Table 1

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 level at 0.5 V.

Part 2

Table 2

Transition definitions 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 level at 0.5 V.

Part 2

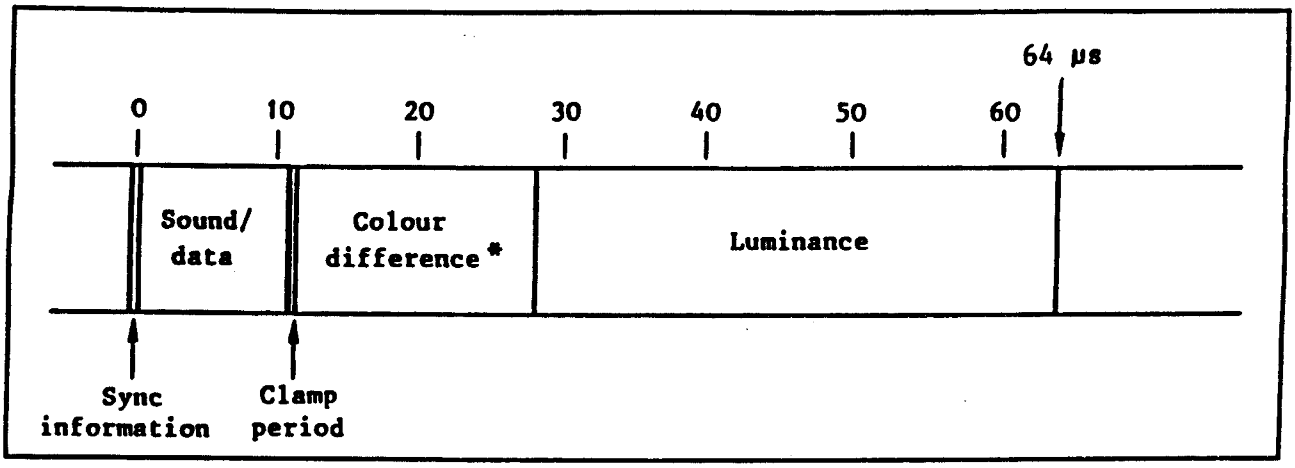
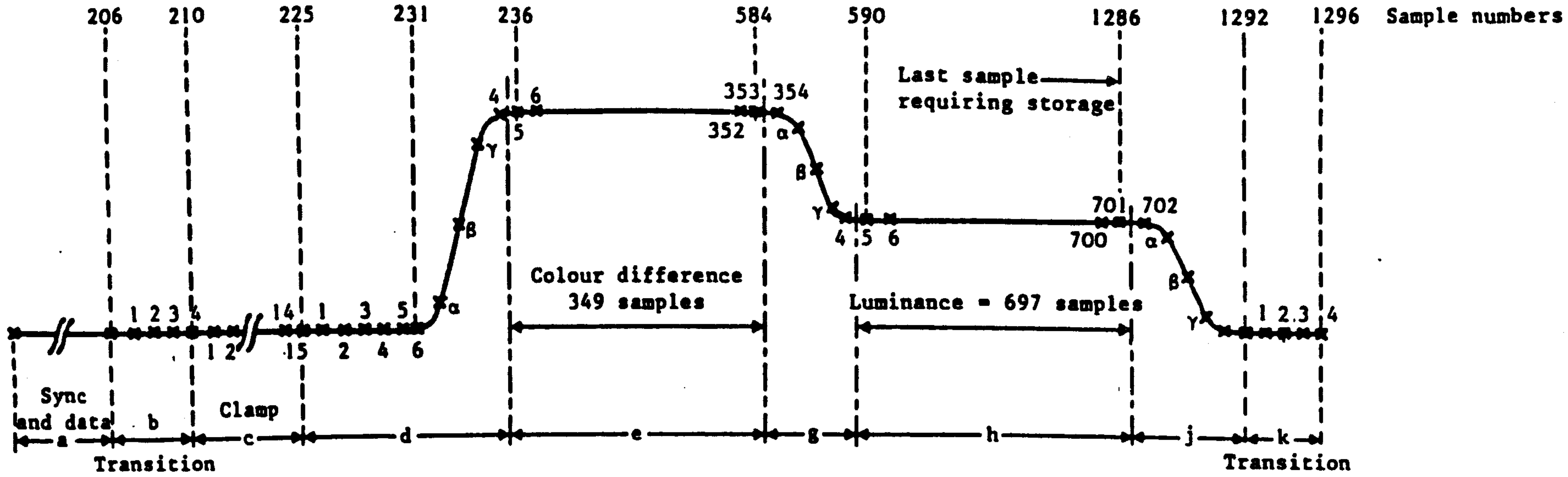


Fig. 1: This figure illustrates the time-division multiplex between the component parts of the signal (more details are given in Fig. 1 of Part 1)

* U and V colour difference signals are transmitted on alternate lines.



Notes

(1) (n) - value of sample n

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

(3) Gr - grey level (0.5 V)
(black level = 0.0 V; maximum luminance level = 1 V;
colour difference signal range = 0.5 V ± 0.5 V)

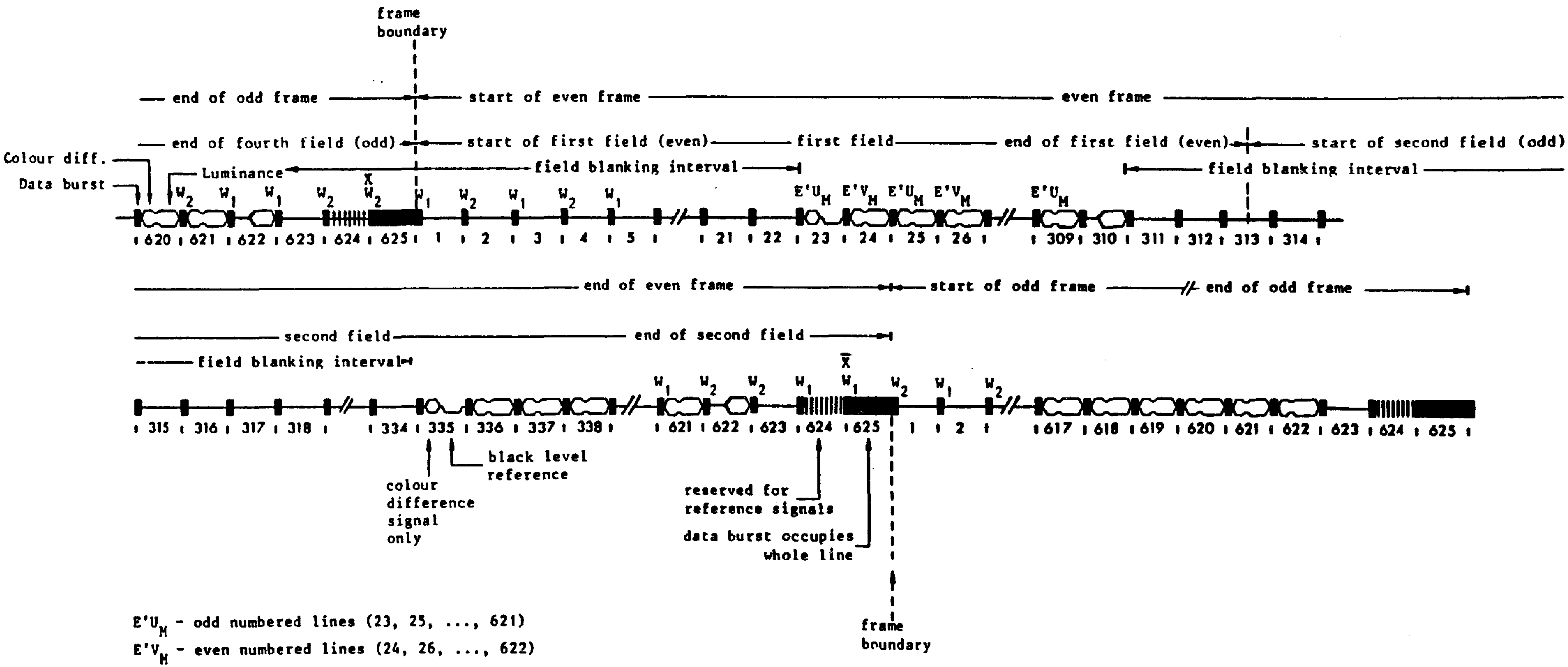
(4) for definition of a - k, see also Fig. 1a to Part 1

(5) Because the MAC standard allows 697 active samples per line, the first 12 and the last 11 active samples out of the 720 per line in the 4:2:2 standard are omitted. For the colour-difference signals, MAC allows 349 active samples and 6 active samples of the digital 4:2:2 standard are omitted at the start and 5 at the end of the line.

Gr = grey level
Bl = black level

Transitions	Method A (continuing samples)	Method B (held samples)
d:	α = 1/8 (1) + 7/8 Gr.	α = 1/8 (4) + 7/8 Gr.
	β = 1/2 (2) + 1/2 Gr.	β = 1/2 (4) + 1/2 Gr.
	γ = 7/8 (3) + 1/8 Gr.	γ = 7/8 (4) + 1/8 Gr.
g:	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) + 1/8 (4)
j:	α = 7/8 (703) + 1/8 Gr.	α = 7/8 (702) + 1/8 Gr.
	β = 1/2 (704) + 1/2 Gr.	β = 1/2 (702) + 1/2 Gr.
	γ = 1/8 (705) + 7/8 Gr.	γ = 1/8 (702) + 7/8 Gr.

Fig. 2: Representation of the waveform (unscrambled) of the transmitter modulating signal (before pre-emphasis)



$E'U_M$ - odd numbered lines (23, 25, ..., 621)

$E'V_M$ - even numbered lines (24, 26, ..., 622)

W_1 and W_2 are the line synchronization words

(phase transitions 2 to 8 in the data burst)

X - frame synchronising sequence - true form

$\bar{X}$ - frame synchronising sequence - inverted form

Fig. 3: Diagrammatic representation of the transmitted signal showing two frame sequence and line numbering system (not to scale)

Part 2

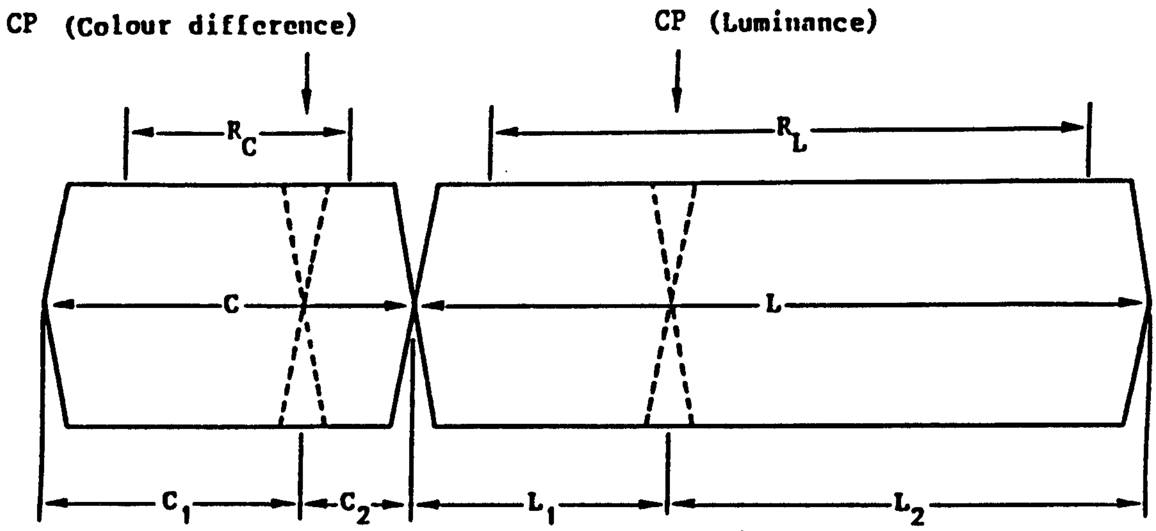


Fig. 4a: Simplified diagram of non-scrambled waveform showing possible cut points in the case of double-cut component rotation

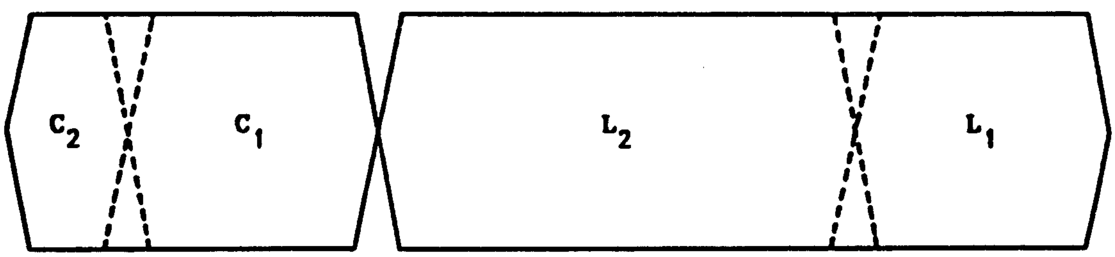


Fig. 4b: Simplified diagram of waveform scrambled by double-cut component rotation

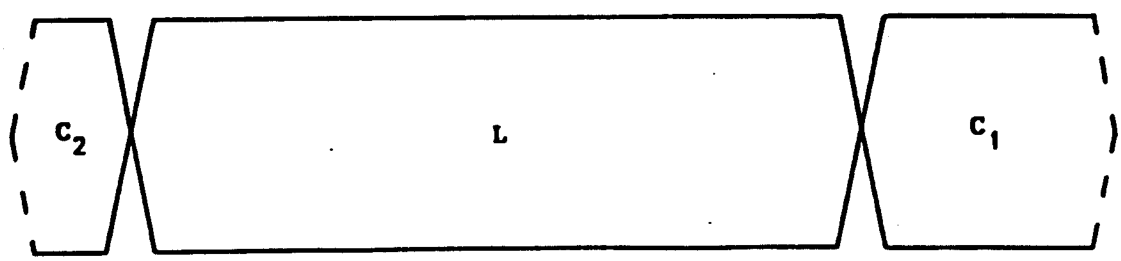


Fig. 4c: Simplified diagram of waveform scrambled by single-cut-in-colour-difference line rotation

Part 2

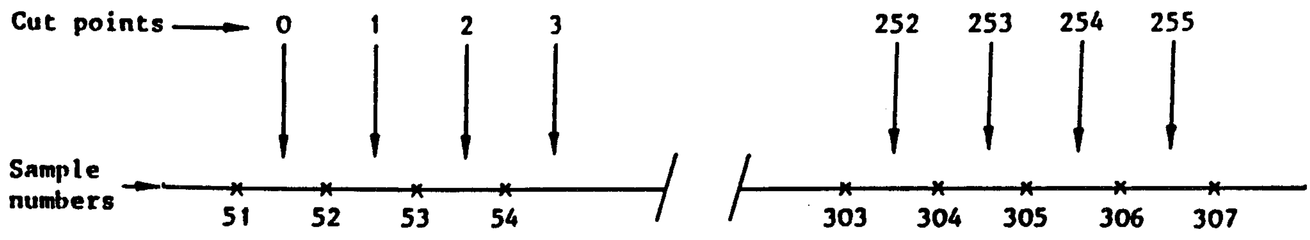


Fig. 5a: Permissible cut-point positions for colour-difference signal (double-cut component rotation and single-cut line rotation)

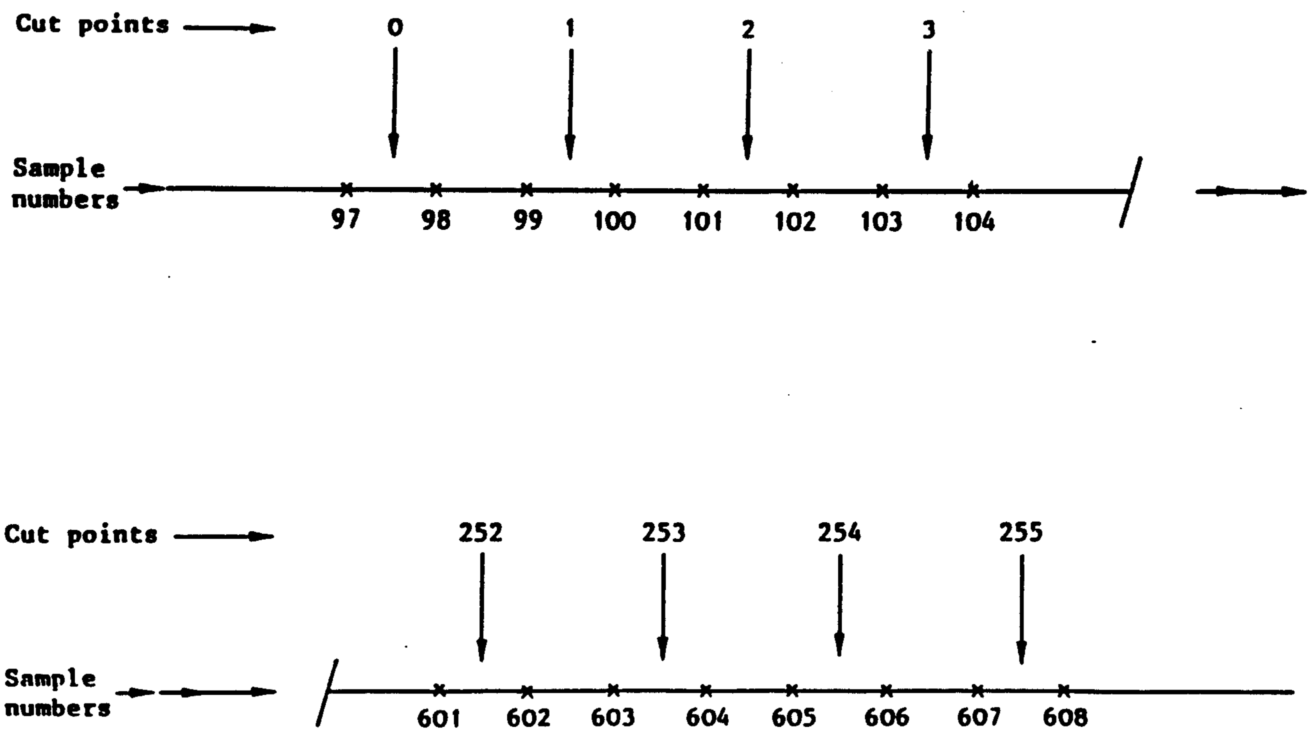


Fig. 5b: Permissible cut-point positions for luminance signal (double-cut component rotation)

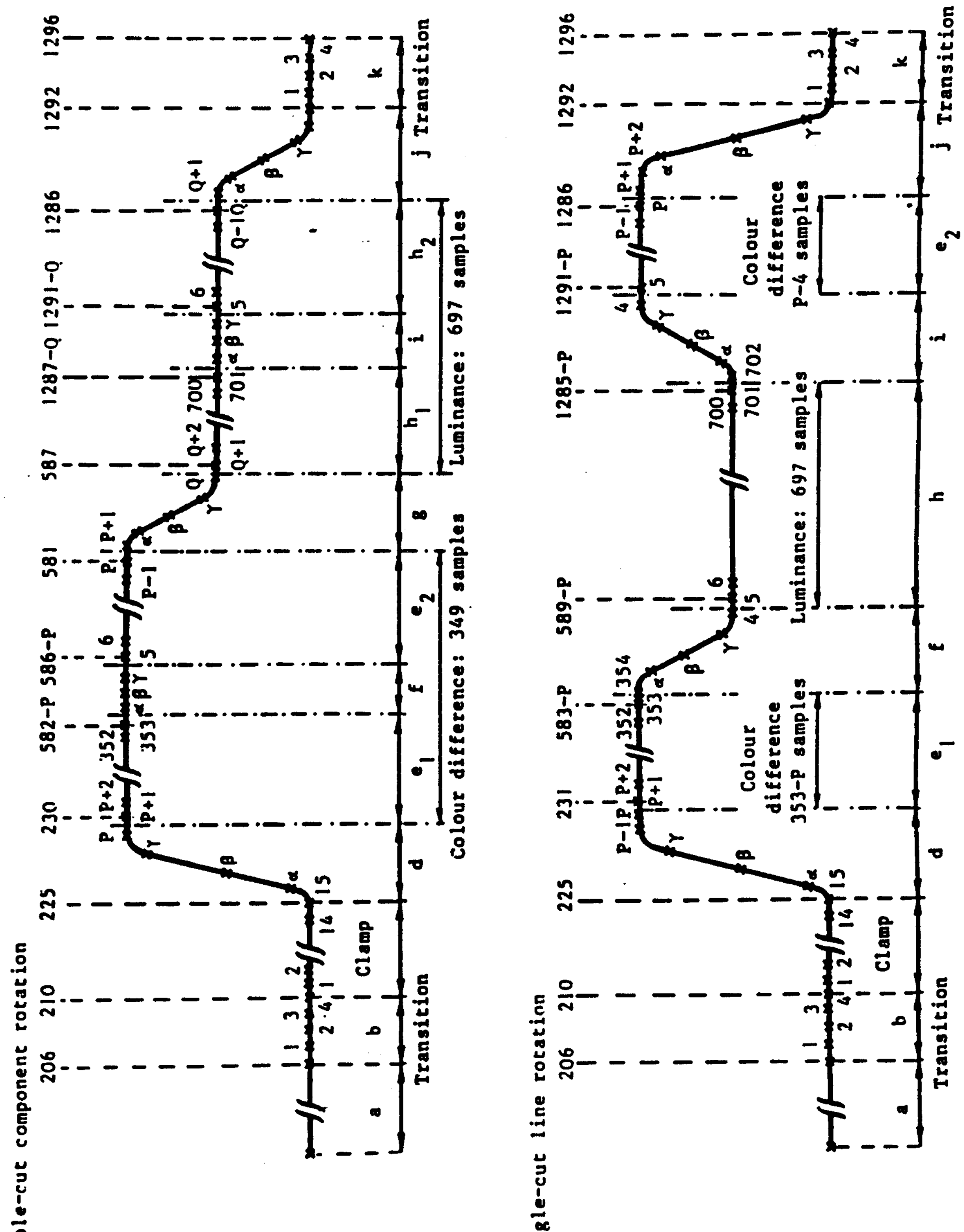
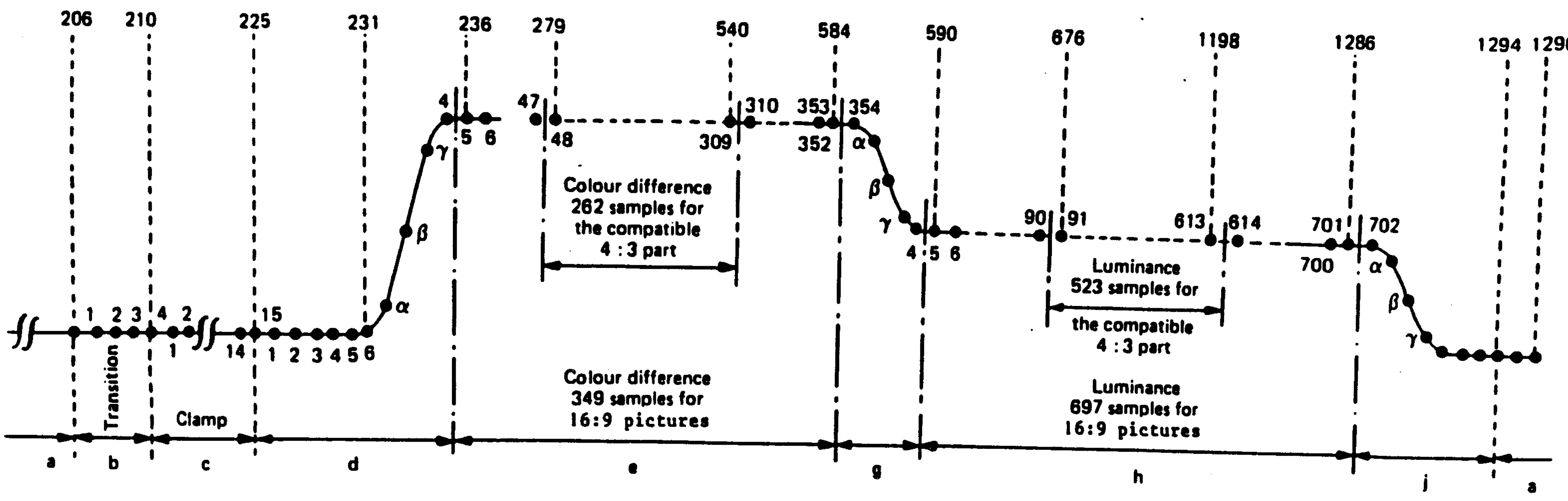


Fig. 6: Representation of the scrambled MAC video waveform



Notes

(1) (n) – value of sample n

(2) colour-difference and luminance samples correspond at the display level according to the relationship $C_{(n)} = L_{(2n-5)}$

(3) Gr – grey level (0.5 V)
(black level = 0.0 V ; maximum luminance level = 1 V ; colour difference signal range = 0.5 V ± 0.5 V)

(4) for definition of a - k, see also Fig. 1a to Part 1

Gr = grey level

Bl = black level

Transitions	Method A (continuing samples)	Method B (held samples)
d :	α = 1/8 (1) + 7/8 Gr β = 1/2 (2) + 1/2 Gr γ = 7/8 (3) + 1/8 Gr	α = 1/8 (4) + 7/8 Gr β = 1/2 (4) + 1/2 Gr γ = 7/8 (4) + 1/8 Gr
g :	α = 7/8 (355) + 1/8 (1) β = 1/2 (356) + 1/2 (2) γ = 1/8 (357) + 7/8 (3)	α = 7/8 (354) + 1/8 (4) β = 1/2 (354) + 1/2 (4) γ = 1/8 (354) + 7/8 (4)
j :	α = 7/8 (703) + 1/8 Gr β = 1/2 (704) + 1/2 Gr γ = 1/8 (705) + 7/8 Gr	α = 7/8 (702) + 1/8 Gr β = 1/2 (702) + 1/2 Gr γ = 1/8 (702) + 7/8 Gr

Fig. 7: Representation of the MAC video waveform for 16:9 aspect ratio, after descrambling if appropriate.

Annex to Part 2

VIDEO BANDWIDTH FOR WIDE-ASPECT RATIO PICTURE

1. Video bandwidth

The normal uncompressed amplitude/frequency characteristic of the picture signal is given in Part 2. In the case of an extended aspect ratio transmission, this leads to a reduction of horizontal resolution for the 4:3 sub-picture.

Where it is required to retain the resolution afforded by the digital component studio standard (CCIR Rec. 601) for the 4:3 sub-picture, the bandwidth of the picture signal can be increased* by a factor of 4/3. (In analogue terms, this would result in a luminance bandwidth, before compression, of approximately 7.6 MHz for the 16:9 picture.)

2. Receiver compatibility

Where receivers are designed to take advantage of an increase in bandwidth of an extended aspect ratio picture signal, these should include additional suitable filtering to limit the video bandwidth to the basic value of 8.4 MHz (before decompression). When receiving transmissions with increased bandwidth, the luminance bandwidth would revert to approximately 11.4 MHz (before decompression).

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* If such a picture signal of wider bandwidth is used for BSS, it will be necessary to confirm that WARC 77 conditions are met, particularly with regard to adjacent channel interference.

PART 3: SPECIFICATION OF THE SOUND/DATA MULTIPLEX AND THE SOUND CODING METHODS

Contents

Part 3

1. Subject of Part 3

This Part contains the specifications of the packet multiplexing system used for the audio and data signals, together with the specifications (conforming to CCIR Report 953) of the sound coding methods. The specifications concerning signal coding for data services within the total capacity of the packet multiplex are a matter for further study. However, such information can be given later, while maintaining full compatibility with the present specification.

In particular, Part 3 of this document presents the specifications for a multiplex system that provides flexible use of the available transmission capacity in the two subframes specified in Part 1. In order to convert the various digital sound and data signals into single, continuous bit streams that can be transmitted in these subframes, a form of multiplexing known as packet multiplexing is adopted (see CCIR Report 954). In doing so, two modes of operation can be defined, depending on whether packets conveying particular services are transmitted in either or both subframes. In the first mode, where any service is transmitted in one subframe only, one configuration can provide for each subframe, for example, 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 of each subframe.

In the second mode, where some components may be transmitted in both subframes, the total capacity can be used as one entity. It is, however, recommended that services be organized according to the first mode, whenever possible, in order to allow transcoding of at least one subframe into the D2-MAC/packet format.

2. Packet structure

Each packet has a constant length and consists of a string of 751 bits (see Fig. 1). The packet is divided into two parts: header (described in Section 2.1), data area (described in Section 2.2).

2.1 Header

In all cases, the header is formed of 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 is thus:

Address	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 LSB is a — The address MSB is k

The continuity index LSB is l — The continuity index MSB is m

2.1.1 Address field

A unique address code is allocated to each broadcast service. 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 1024 different simultaneous services.

The packet address "0" is permanently allocated to the service identification system (see Part 5).

The packet address "1023" is permanently allocated to dummy packets which are inserted to fill the multiplex.

2.1.2 Continuity index

The length of this index is two bits. It assures the link between successive packets of a same service and can be used to detect possible packet loss (see Section 7.4 and Fig. 6 of Part 3, Section 3.2 of Part 5, Section 5 of Part 6).

2.1.3 Protection suffix

This field has 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 generator polynomial is equal to:

$$x^{11} + x^{10} + x^6 + x^5 + x^4 + x^2 + 1$$

2.2. Useful data

The total length of the useful data area is 91 bytes.

Part 3

3. Insertion of packets into the transmission multiplex for standard television transmissions

3.1 Bits per subframe

The transmission multiplex structure described in Part 1 of this Report allows:

- 99 useful bits per line in each subframe
- 623 lines per subframe (two remaining special data bursts are for modulation synchronization, signalling and other purposes).

Consequently, 61 677 bits are available per subframe, corresponding for the D system, to a total of 123 354 bits for the two subframes.

3.2 Packets per subframe

Each subframe contains 82 packets of length 751 bits as shown in Fig. 2 of Part 1. Hence each frame contains a total of 164 packets for the D system.

3.3 Modes of operation

3.3.1 Independent subframe

All packets conveying a particular digital component are multiplexed into only one of the two subframes.

In this mode, the same packet address can be assigned to different services in the two subframes. Either subframe can be transcoded into the D2 format, and all its components can be recovered subsequently (see Part 5).

3.3.2 Related subframes

Packets conveying a particular digital component are multiplexed either into only one of the two subframes, or into both. In the latter case, two types of components have to be identified:

- the packets of the digital component are inserted in both subframes. The service component can be recovered when considering only one of the subframes
- the packets of the digital component are inserted in both subframes. The service component cannot be recovered when considering only one of the subframes. In this mode, packets conveying the same component must never occupy the same relative position in the two subframes and must be in the correct time sequence in the overall multiplex.

The same packet address can be assigned to different services entirely separated into the two subframes. A packet address assigned to a service that is transmitted in both subframes must not be assigned to any other service. Either subframe can be transcoded into the D2 format.

3.4 Buffer storage

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

3.5 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 Sections 3.6 and 3.7 of Part 1, the sound coding blocks (as well as the useful data of packets carrying data services) 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 Part 6 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 Section 9 below

c) for the D system, the data burst is arranged into two subframes (see Parts 1, 3 and 5). The data scrambling is so arranged that a given service component would be scrambled with the same scrambling sequence whichever subframe it occupied. In effect, each subframe is scrambled independently but it is as if the two subframes were time coincident from the point of view of applying a scrambling sequence for a given service component.

4. Sound coding methods

The present specification considers three different types of sound signals:

1. high quality stereophonic sound
2. high quality monophonic sound
3. monophonic sound with a reduced bandwidth.

Two coding methods are recommended. Their main characteristics are presented in Table 1. Fig. 2 details the actual codes for linear and companded signals.

Part 3

Table 1

Bandwidth	High quality 40-15000 Hz	Medium quality* 40-7000 Hz
Type	stereophonic or monophonic	monophonic
Sampling frequency	32 kHz**	16 kHz**
Accuracy	10^{-6}	10^{-6}
Conversion	14 bits/sample	14 bits/sample
Overload level	12 dBm	12 dBm
Pre-emphasis	J.17-CCITT (-6.5 dB at 800 Hz)	J.17-CCITT (-6.5 dB at 800 Hz)

1. Linear coding	Linear code:	14 bits per sample 2's complement configuration
2. Near-instantaneous coding	Law:	near-instantaneous law with blocks of 32 successive samples
	Coding:	10 bits per sample 2's complement configuration scale factor on five ranges
Fig. 2 describes the structure of both linear and companded samples		
In stereophonic mode, the right and left channels are sampled simultaneously and coded separately		

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* It should be noted that 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.

** See Section 9.5 for the synchronism between sampling frequencies of channels intended for mixing.

5. Protection of sound signals against errors

For the D system, two specific protection levels have been specified to offer, in the two subframes, a total of respectively either eight companded or six linear sound channels with a basic protection but enough for a national coverage area, or a lower capacity (six companded or four 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.

5.1 First level protection

5.1.1 Companded mode

For companded 10-bit coding, one parity bit per sample is 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 service or to 32 consecutive samples of each channel of a stereophonic service are modified to signal the scale factor and control information (see Section 6.2).

Table 2 presents the scale factor codes on 3 bits corresponding to the different companding and protection ranges.

a) the coding ranges are related to the characteristics of the near-instantaneous law (5 ranges)

b) the protection ranges are 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 2

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

5.1.2 Linear mode

For linear 14-bit coding, one parity bit per sample is 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 service or to 32 consecutive samples of each channel of a stereophonic service are modified to signal the scale factor and control information (see Section 6.1). The recovery of 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.

5.2 Second level protection

5.2.1 Companded mode

For companded 10-bit coding, the extended Hamming (11,6) code is applied to each sample. Subsequently, one parity bit for 32 consecutive samples of a monophonic service or the 32 consecutive samples of each channel of a stereophonic service is modified to signal the scale factor and control information (see Section 6.4 and Fig. 4d).

- Fig. 3 presents coding and decoding tables.
- Table 2 (see Section 5.1.1) presents the 3-bit scale factor codes corresponding to the different companding and protection ranges.
- The same procedure as described in Section 5.1.1 can be applied to perform the protection of the samples.

5.2.2 Linear mode

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

Subsequently, one parity bit corresponding to 18 consecutive samples for a monophonic service or to 18 consecutive samples for each channel of a stereophonic service is modified to signal part of the scale factor information. The remaining part of the scale factor information is transmitted using spare bits available. (Note that the control information is sent using also five spare bits available in the coding block.) (See Section 6.3 and Fig. 4c.)

Fig. 3 presents coding and decoding tables.

The same procedure as described in Section 5.1.2. can be applied to perform protection of the samples.

6. Structure of the sound coding blocks

Four coding block structures have to be considered.

6.1 Linear mode and first level protection

One coding block contains 64 samples of 15 bits:

- Either, 64 successive samples in monophonic mode. The scale factor FE1 is related to the first 32 samples. The scale factor FE2 is 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 is related to the 32-A channel samples. The scale factor FE2 is related to the 32-B channel samples.

Fig. 4-a describes the structure of the coding block.

The three-bit information of the scale factor code R_2, R_1, R_0 (see Table 2 in Section 5.1.1) is 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 Sections 6.1.1 and 6.1.2.

6.1.1 Stereophonic mode

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

6.1.2 Monophonic mode

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

6.1.3 Control information

Control information is used to assure high-speed switching for a process which is identified in the interpretation block of the sound service (e.g. music/speech control, see Section 9 and Table 3). This binary control information bit (CIB) is related to each 32-sample group and is transmitted by the use of the original parity bit P_i of samples which becomes P'_i by the relations:

$$P'_i = P_i \oplus CIB_1 \quad \text{for } i = 55, 56, 57, 58, 59$$
$$P'_i = P_i \oplus CIB_2 \quad \text{for } i = 60, 61, 62, 63, 64$$

Scale factor information and control information are extracted by majority decision logic. Subsequently, the original parity is restored.

6.2 Companded mode and first level protection

One coding block contains 64 samples of 11 bits:

- Either 64 successive samples in monophonic mode. The scale factor FE1 is related to the first 32 samples. The scale factor FE2 is 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 is related to the 32-A channel samples. The scale factor FE2 is related to the 32-B channel samples.

- At the beginning of the block, 16 unallocated bits are free for future needs.

Fig. 4-b describes the structure of the coding block.

The scale factor is sent by signalling in parity as follows:

The three-bit information of the binary range code R_2, R_1, R_0 (see Table 2 in Section 5.1.1) is 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 Sections 6.2.1 and 6.2.2.

6.2.1 Stereophonic mode

Two scale factor values are transmitted:

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

and

$P'_i = P_i \bullet R_{2A}$ for i = 1, 7, 13, 19, 25, 31, 37, 43, 49
$P'_i = P_i \bullet R_{1A}$ for i = 3, 9, 15, 21, 27, 33, 39, 45, 51
$P'_i = P_i \bullet R_{0A}$ for i = 5, 11, 17, 23, 29, 35, 41, 47, 53

$P'_i = P_i \bullet R_{2B}$ for i = 2, 8, 14, 20, 26, 32, 38, 44, 50
$P'_i = P_i \bullet R_{1B}$ for i = 4, 10, 16, 22, 28, 34, 40, 46, 52
$P'_i = P_i \bullet R_{0B}$ for i = 6, 12, 18, 24, 30, 36, 42, 48, 54

6.2.2 Monophonic mode

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

$P'_i = P_i \bullet R_{2n}$ for i = 1, 4, 7, 10, 13, 16, 19, 22, 25
$P'_i = P_i \bullet R_{1n}$ for i = 2, 5, 8, 11, 14, 17, 20, 23, 26
$P'_i = P_i \bullet R_{0n}$ for i = 3, 6, 9, 12, 15, 18, 21, 24, 27

$P'_i = P_i \bullet R_{2n+1}$ for i = 28, 31, 34, 37, 40, 43, 46, 49, 52
$P'_i = P_i \bullet R_{1n+1}$ for i = 29, 32, 35, 38, 41, 44, 47, 50, 53
$P'_i = P_i \bullet R_{0n+1}$ for i = 30, 33, 36, 39, 42, 45, 48, 51, 54

6.2.3 Control information

Control information is used to assure high speed switching for a process which is identified in the interpretation block of the sound service (see Section 9 and Table 3). This binary control information bit (CIB) is related to each 32-sample group and is transmitted by the use of the original parity bit P_i of samples which becomes P'_i by the relations:

$$P'_i = P_i \oplus CIB_1 \qquad \text{for } i = 55, 56, 57, 58, 59$$
$$P'_i = P_i \oplus CIB_2 \qquad \text{for } i = 60, 61, 62, 63, 64$$

Scale factor information and control information are extracted by majority decision logic. Subsequently, the original parity is restored.

6.3 Linear mode and second level protection

One coding block contains 36 samples of 19 bits:

- Either 36 successive samples in monophonic mode. The scale factor FE1 is related to the first 18 samples. The scale factor FE2 is 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 is related to the 18-A channel samples. The scale factor FE2 is related to the 18-B channel samples.

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

  8 unallocated bits, free for future needs

  10 control information bits (CI) used to send control information as described in Section 6.3.3

  18 bits (B_i) used to partly send scale factor information as described in Sections 6.3.1 and 6.3.2.

Fig. 4-c describes the structure of the coding block.

The scale factor value is 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.

6.3.1 Stereophonic mode*

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 $R_{0B} = B_{12} = B_{15} = B_{18}$

6.3.2 Monophonic mode*

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 $R_{2n} = B_1 = B_4 = B_7$

---

* See Figs. 3 and 4-c for definition of notations.

$P'^5_i = P^5_i \oplus R_{1n}$ 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 $R_{0n+1} = B_{12} = B_{15} = B_{18}$

6.3.3 Control information

Control information is used to assure high-speed switching for a process which is identified in the interpretation block of the sound service (see Section 9 and Table 3). This binary control information (CIB) is related to each 18-sample group and is transmitted in the 10 (CI_i) bits reserved for this purpose at the beginning of the block.

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

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

Scale factor information and control information are extracted by majority-decision logic. Subsequently, the original parity is restored.

6.4 Companded mode and second level protection

One coding block contains 64 samples of 15 bits:

- Either 64 successive samples in monophonic mode. The scale factor value FE1 is related to the first 32 samples. The scale factor value FE2 is 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 is related to the 32-A channel samples. The scale factor value FE2 is related to the 32-B channel samples.

- Scale factor value is transmitted in the parity bits as follows:

The three-bit information of the binary range code R_2, R_1, R_0 is 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'^5_1, ..., P'^5_{64}).

- Fig. 4-d describes the structure of the coding block.

6.4.1 Stereophonic mode

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

6.4.2 Monophonic mode

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

6.4.3 Control information

Control information is used to assure high speed switching for a process which is identified in the interpretation block of the sound service (see Section 9 and Table 3). This binary control information bit (CIB) is related to each 32-sample group and is transmitted by the use of the original parity bit P^{5}_{i} 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

Scale factor information and control information are extracted by majority decision logic. Subsequently, the original parity is restored.

7. Insertion of the coding block into packet structure

7.1 General

The length of the sound coding blocks depends 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 is 91 bytes (i.e. 728 bits).

The present specification describes only the structure of the data area concerned with sound services. Data area structures concerned with other possible services will be covered by a supplement to the present specification, except for the service-identification system which is specified in Part 5.

For sound transmission, the first byte of the data block is always used to indicate the type of packet (i.e. sound packet or control packet). It is designated the Packet Type (PT) byte. The remaining 90 bytes convey either control or sound information.

Two types of coding blocks have to be distinguished:

- sound coding block (BC) as described in Section 6 above
- interpretation block (BI) as described in Section 9 below.

The coding blocks follow immediately the Packet Type byte (PT) which characterizes their nature.

The PT byte is required to inform 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 is conveyed by occasional "interpretation" packets. These packets have the same address as the packets containing the sound samples and scale-factor information but are identified by having different PT bytes at the start of the useful data block. The useful data block is designated as an "interpretation block" (BI) when it contains data for setting-up the sound decoder; interpretation blocks 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 is designated as a "sound coding block" (BC), and the PT byte has either the BC1 configuration or the BC2 configuration.

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

The configurations BI1 and BI2 characterize the interpretation blocks, the structure of which is described in Section 9. These blocks are 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 services, 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 service may then have to operate simultaneously.)

The four configurations for the PT byte are defined as follows:

	MSB ↓ LSB ↓	
BC1:	11000111	"C7"
BC2:	11111000	"F8"
BI1:	00000000	"00"
BI2:	00111111	"3F"

with least significant bit transmitted first.

7.2 Insertion of a 90-byte coding block

The insertion of a 90-byte coding block into the packet structure as described in Section 6 is realised by placing the start of each coding block immediately after the PT byte. This relation is presented in Fig. 5.

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* For example, this may be required in the case of sound channels intended for automatic mixing (see Section 9.5).

7.3 Insertion of a 120-byte coding block

The insertion of a 120-byte coding block into the packet structure as described in Section 6 is realised by inserting three successive coding blocks into four successive packets in the manner shown in Fig. 6. They must be understood as successive packets which contain the same sound service address. They can be interleaved in the global bit stream with packets related to other services.

7.4 Continuity index (see Section 2.1)

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

The continuity index of any BI packet (PT = BI1 or BI2) (see Section 9) is decremented from that of the previous packet of the same address, regardless of its packet type.

8. 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 are:

- 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 is realised in the scale factor (see Table 2 in Section 5.1.1).

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

9. Specification of sound interpretation data

9.1 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 Section 7.1:

- sound coding blocks (BC) and
- interpretation blocks (BI) (see Section 7.1 for the normal repetition rate).

In each case, the address code will be that allocated for the particular sound component. As specified in Section 7.4, the continuity index of any BI packet is decremented from that of the previous packet of the same address, regardless of its packet type.

The sound interpretation data are inserted in interpretation blocks BI in accordance with the coding procedure specified in Section 9.2 below. 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 service.

b) They prepare the decoder for forthcoming changes in the characteristics of the selected sound service.

They are grouped in the "commands" defined in Section 9.3 below, for which the indicators have the value "CI = 1" and "CI = 2".

9.2 General structure for the coding of sound interpretation data

a) Commands

The data are assembled into commands, each of which is introduced by one of 16 command indicators (CI) as a Hamming-coded byte*. This is followed by a Hamming-coded byte (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 indicates length 15 and the next pair of Hamming-coded bytes are also LI bytes indicating the length.

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* The Hamming code used for all Hamming-protected bytes defined in this Section is described in Table 4.

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

The command indicator CI = 0 introduces 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 Section 9.4 below.

The command indicators CI = 1 and CI = 2 introduce a fivefold repetition of the sequence whose significance is detailed in Section 9.3 below and which contains the data needed for commanding the configuration of the decoder.

The sound interpretation data is thus coded in a byte-oriented way. For each byte, the least significant bits are transmitted first.

b) Data groups

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

- Structure of the interpretation blocks

The sequence of interpretation blocks within one digital sound channel provides a transmission capacity which is 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 is done using the fourth possible value (BI2) of the packet type (PT) byte (see Section 7.1). Both values BI1 and BI2 identify an interpretation block; BI1 indicates that the beginning of the block coincides with the beginning of a data group; BI2 indicates 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.)

- The data group transport

The positioning of the data group in the interpretation block packets is detailed in Fig. 7. Although this general coding scheme allows a maximum of 22692 bytes in the data group, the data group is restricted to a maximum of 264 bytes (i.e. three packets). In the cases of eight companded sound channels with first-level error protection or six linear sound channels with first-level error protection in the D-MAC/packet system, the length of the data group should 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 Section 7.4.

Commands within a data group are 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 are sent first (see in particular an application of this sort in Section 9.4).

The protection against errors is achieved by the combination of:

- the continuity check CC, itself Hamming protected

- Hamming protection of the structuring bytes LI, CI, S1, S2, F1, F2.

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

The data needed to control the automatic configuring of the sound decoder are 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 refers to the existing state and CI = 2 refers to the future state. The organization of these parameter fields is in accordance with the following description. They are formed from two bytes labelled Byte No. 1 and Byte No. 2 in the order in which they are transmitted; bit 1 of each byte is transmitted first.

<u>Byte No. 1</u>

<u>Bit 1</u>: Parity check

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

<u>Bit 2</u>: SDFSCR flag

This bit is set to 1 to indicate that details of the selected sound channel should <u>not</u> be stored in the receiver in association with the current SDFSCR code (see Section 5.3 of <u>Part 1</u>). In particular, this bit will be set to 1 in the case of a sound component in the controlled-access mode.

<u>Bit 3</u>: News flash indication

The value 1 of this bit in the main (or "original") sound, or optionally also in a commentary (or "additional") sound, indicates that a digital news flash sound is currently being broadcast (the corresponding channel address is given in channel 0). The value 0 indicates 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 will, 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 is achieved by BC1-BC2 alternation (see Section 7.1).

Bit 4 ensures 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 corresponds to the configuration BC1
bit 4 = 0 corresponds 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 Sections 9.5 and 9.6). Bit 5 signals this characteristic.

bit 5 = 0 : continuous
bit 5 = 1 : intermittent.

Bit 6
Bit 7 } Function of the command information

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 Sections 9.5 and 9.6):

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 will be set to 0.

Bit 8: State

This bit is used when bit 5 = 1. It indicates 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 Section 9.5. It is set to 0 when bit 5 = 0.

Byte No. 2

Bit 1
Bit 2 } Audio configuration
Bit 3

Eight options are possible:

bit 1 = 0 bit 2 = 0 bit 3 = 0	bandwidth 40 Hz - 15 kHz, monophonic
bit 1 = 0 bit 2 = 0 bit 3 = 1	bandwidth 40 Hz - 15 kHz, stereophonic
bit 1 = 0 bit 2 = 1 bit 3 = 0	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 indicates if this sound should be reproduced alone or if it is intended to be mixed in the receiver with the main sound. This bit is 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 indicates 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 Sections 9.5 and 9.6.

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 indicates whether the sound channel is subject to scrambling (see Part 6).

bit 5 = 0 no scrambling
bit 5 = 1 scrambling

Bit 6: Conditional access

This bit indicates whether the sound channel is subject to free-access or to controlled-access (see Part 6).

bit 6 = 0 free-access mode
bit 6 = 1 controlled-access mode

Bit 7: Coding law

Two coding laws are specified. This bit indicates 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 indicates which level is in use:

bit 8 = 0 first level
bit 8 = 1 second level.

Table 3 lists all these functions and the codes.

These two bytes are 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.

9.4 Procedure for updating the parameters of a sound configuration

In a steady state, the command with CI = 0 is either not transmitted or transmitted without 1 and 2 in its parameter field, and the command with CI = 1 is 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 is adopted:

- During the two seconds preceding the transition, the new configuration will be signalled at least 3 times per second. The structure of the corresponding data group will 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 will 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 is effected in accordance with the procedure described in Section 7.1.

- After the change, the content of the command with CI = 1 describes the new configuration and the process comes back to the steady state described at the beginning of this Section 9.4.

9.5 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 satellite 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 1 in Part 5). 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 Section 2.1 of Part 5).

b) Digital TV additional sound signals which are specified by the command and parameter identifiers '90 and 'A5 (see Table 1 in Part 5), 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 Section 2.3 in Part 5) 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 must be mixed as follows:

output x = α main sound + β commentary sound

α and β values can vary as described below.

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 3):

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) should 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, it is recommended that the user should be able to select the exact value. It must be noted 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 is the responsibility of the broadcaster and is 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 must 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 has 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 Section 6). 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 3).

The operation is illustrated by Fig. 8.

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

The same procedure as described in Section 9.5 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 is specified in the command and parameter identifiers under the heading "Digital radio sound" (A8: see Table 1 of Part 5).

b) The additional digital sounds are specified by the identifier: "Digital radio additional sound" (A9: see Table 1 of Part 5).

Part 3

Table 3

Function and code for each bit of the bytes defined in Section 9.3

Byte 1	Function	Code
b_1	Parity check	to give modulo-2 sum of all bits in bytes 1 and 2 equal to 0
b_2	SDFSCR flag	1 don't 0 store
b_3	News flash indication	1 yes 0 no
b_4	Identification of sound coding blocks	1 BC1 0 BC2
b_5	Timing characteristic	0 continuous 1 intermittent
b_6 b_7	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
b_8	State	0 signal present 1 interrupted

Table 3 (continued)

Byte 2	Function	Code	
b_1 b_2 b_3	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
b_4	Automatic mixing	0	mixing not intended
		1	mixing intended
b_5	Scrambling	0	no
		1	yes
b_6	Controlled access	0	no
		1	yes
b_7	Coding law	0	linear
		1	companded
b_8	Level of error protection	0	first level
		1	second level

Part 3

Table 4

Hamming code used for the Hamming protected bytes

(Defined in Section 9)

ENCODING

Protection bits: b1, b3, b5, b7

Hexadecimal number	Decimal number	b1	b2	b3	b4	b5	b6	b7	b8
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

Information bits: b2, b4, b6, b8

First binary element transmitted: b1

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

DECODING

$\oplus$ = EXCLUSIVE - OR
$\overline{b2}$ = inverse of b2

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

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

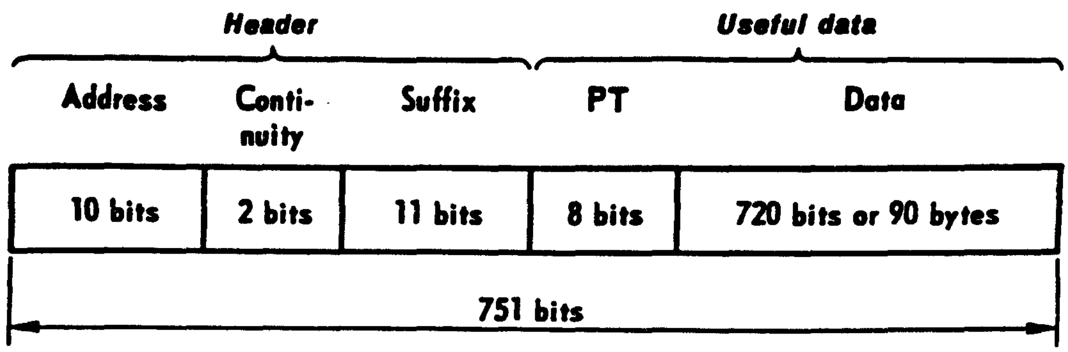


Fig. 1: Packet structure

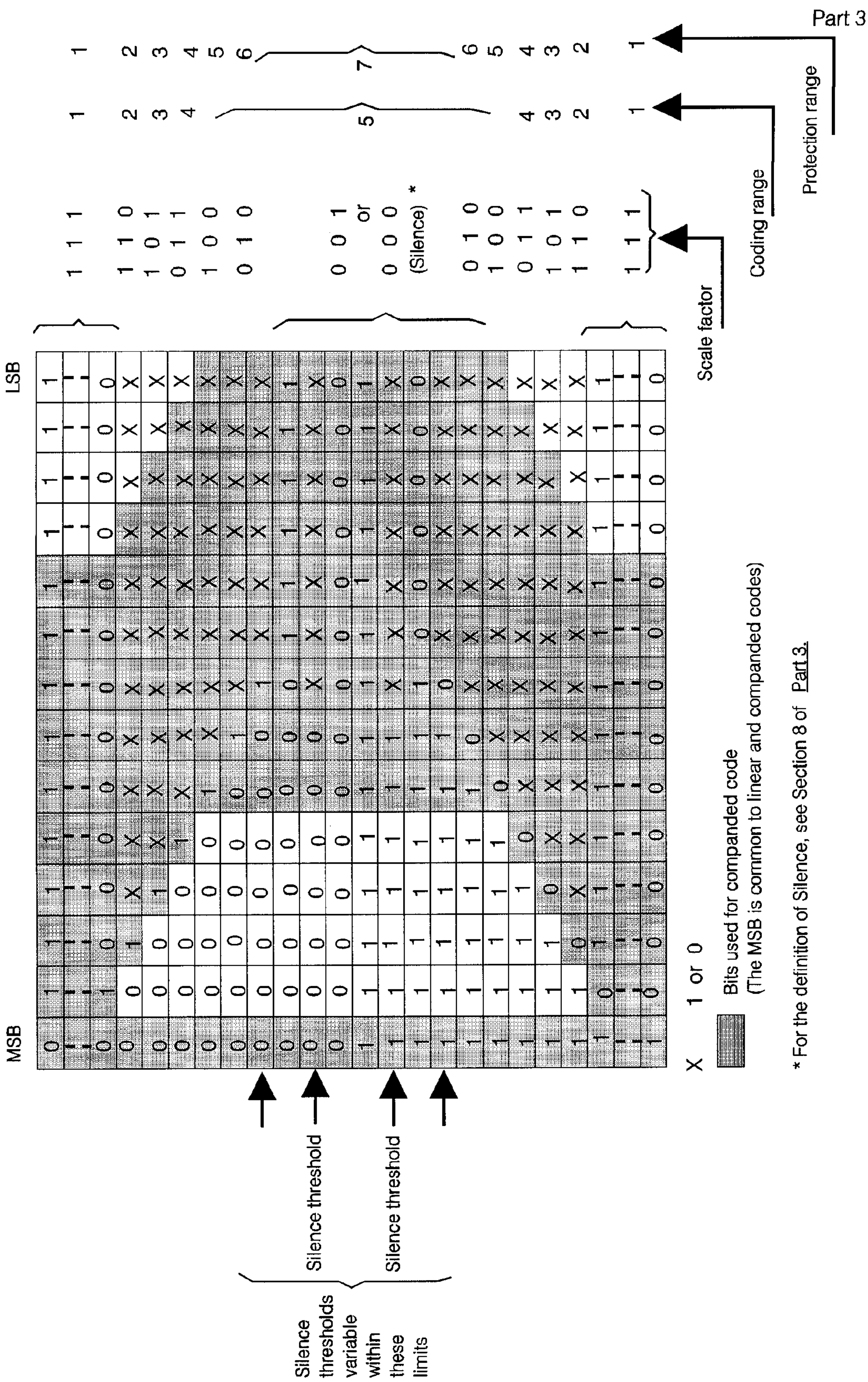


Figure 2: Coding for linear and companded signals (2's complement)

Part 3

Fig. 3: Extended Hamming code (16,11) and (11,6) for companded mode

Relation between code table, sample bits and parity bits

	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 Fig. 4-c)	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 Fig. 4-d)	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

Coding relations

$$P_i^1 = b_1 \oplus b_4 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_9 \oplus b_{11}$$

$$P_i^2 = b_1 \oplus b_2 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{10}$$

$$P_i^3 = b_1 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_{10}$$

$$P_i^4 = b_2 \oplus b_3 \oplus b_5 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{11}$$

$$P_i^5 = b_3 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_8 \oplus b_{10} \oplus b_{11}$$

$$P'^5_i = P_i^5 \oplus R_j$$

Decoding relations

$$S_4 = P_i^1 \oplus b_1 \oplus b_4 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_9 \oplus b_{11}$$

$$S_3 = P_i^2 \oplus b_1 \oplus b_2 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{10}$$

$$S_2 = P_i^3 \oplus b_1 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_7 \oplus b_8 \oplus b_{10}$$

$$S_1 = P_i^4 \oplus b_2 \oplus b_3 \oplus b_5 \oplus b_6 \oplus b_7 \oplus b_9 \oplus b_{11}$$

$$S_0 = \hat{P}_i^5 \oplus b_3 \oplus b_4 \oplus b_6 \oplus b_7 \oplus b_8 \oplus b_{10} \oplus b_{11}$$

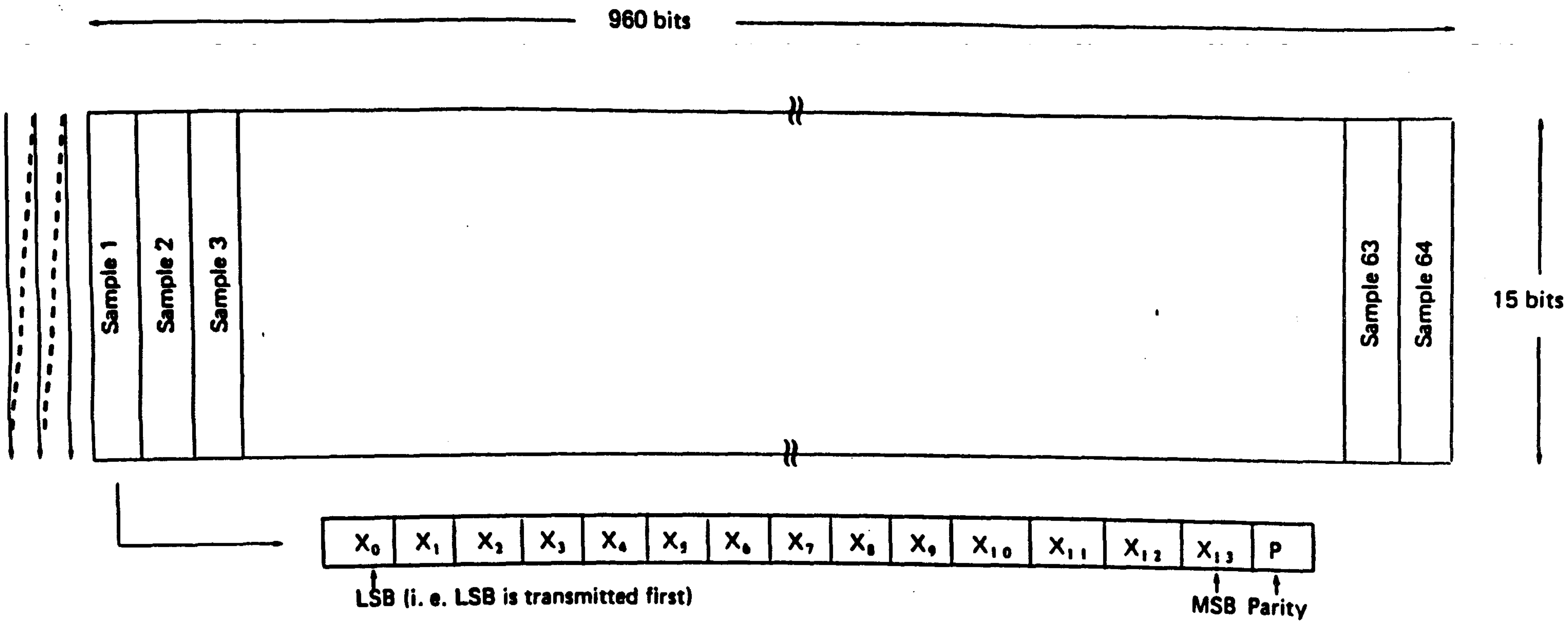
where $\hat{P}'^5_i = P'^5_i \oplus R_j$

Part 3

$$S_P = \sum_{k=0}^{k=4} S_k = \sum_{k=1}^{k=11} b_k \oplus \sum_{n=1}^{n=5} P_i^n$$

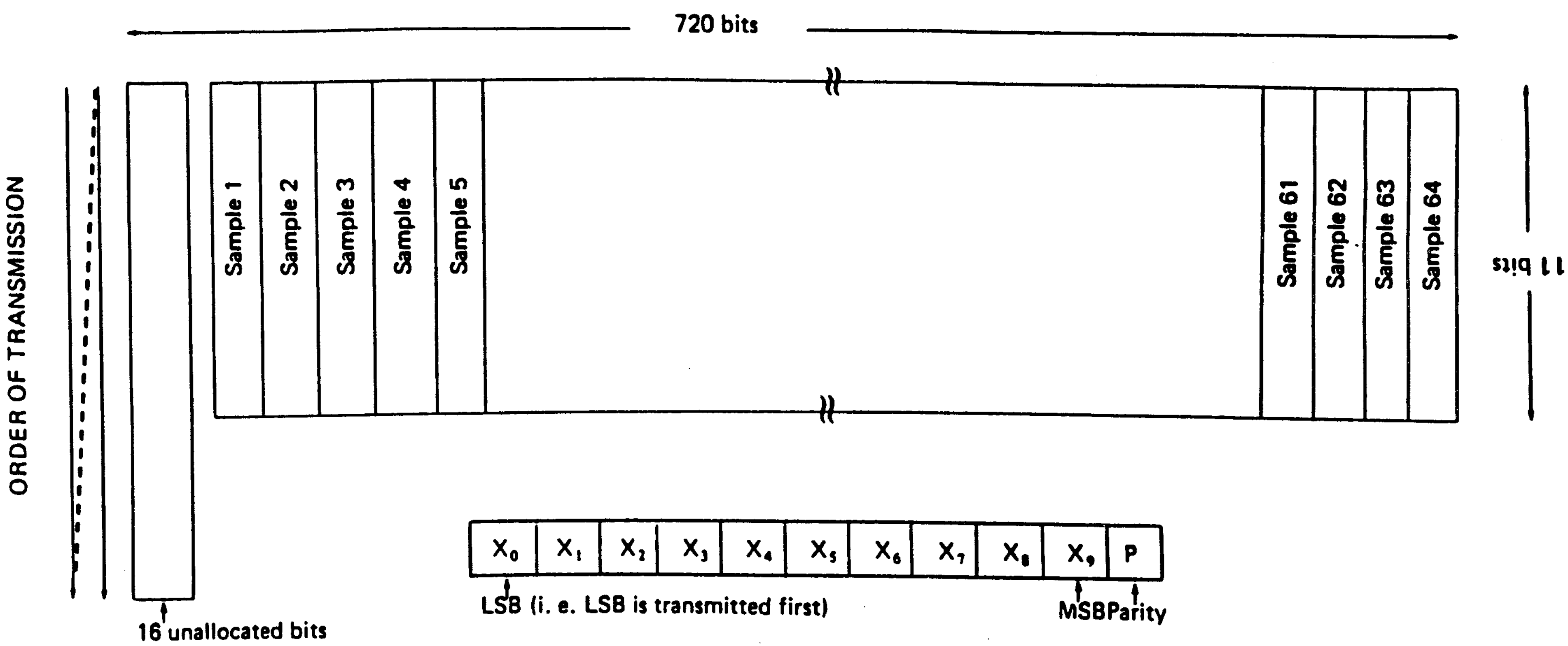
Correction table

S_4	S_3	S_2	S_1	S_0	S_P	Interpretation	Sample treatement: Linear	Sample treatement: 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	



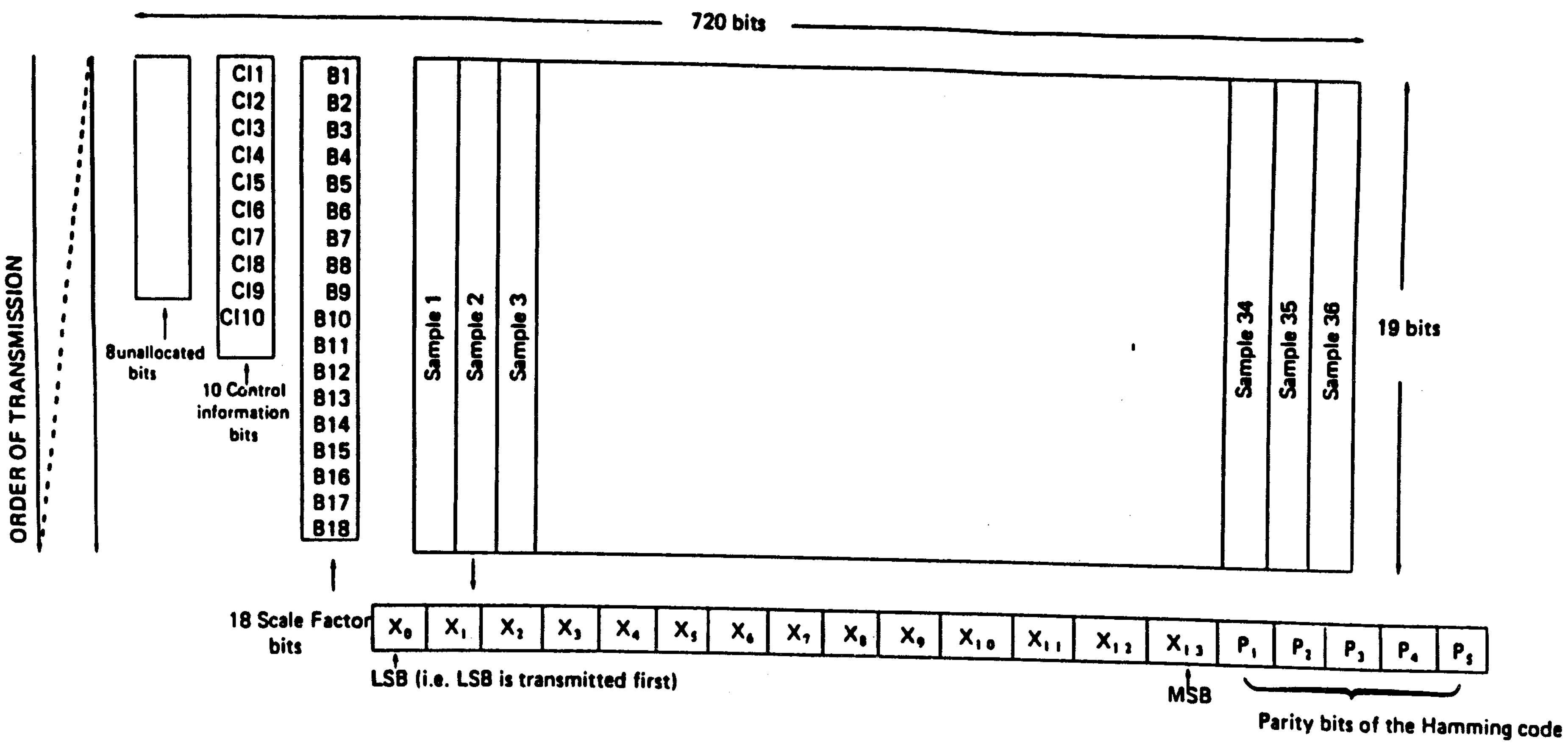
	SAMPLES																		
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ Stereophonic Linear sound	A_1	B_1	A_2	B_2	....	A_{15}	B_{15}	A_{16}	B_{16}	A_{17}	B_{17}	A_{18}	B_{18}	A_{19}	...	A_{31}	B_{31}	A_{32}	B_{32}
	Block n																		
HQ/MQ monophonic Linear Sound	A_1	A_2	A_3	A_4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

Fig. 4a: Arrangement of 120-byte LINEAR sound coding blocks in relation with the FIRST-LEVEL PROTECTION



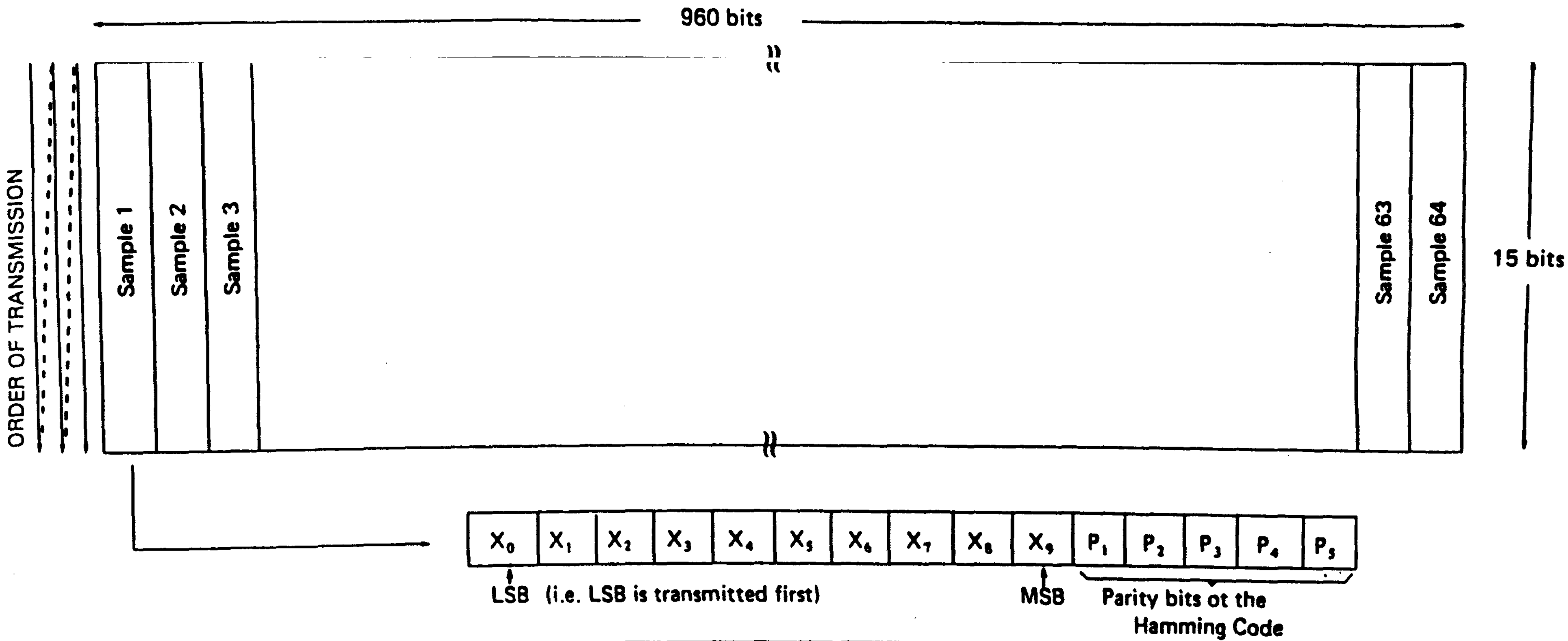
	SAMPLES																		
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ stereophonic companded sound	A_1	B_1	A_2	B_2	....	A_{15}	B_{15}	A_{16}	B_{16}	A_{17}	B_{17}	A_{18}	B_{18}	A_{19}	....	A_{31}	B_{31}	A_{32}	B_{32}
	Block n																		
HQ/MQ Monophonic companded sound	A_1	A_2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

Fig. 4b: 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						

Fig. 4c: 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									

Fig. 4d: Arrangement of 120-byte COMPANDED sound coding blocks in relation with the SECOND-LEVEL PROTECTION

Part 3

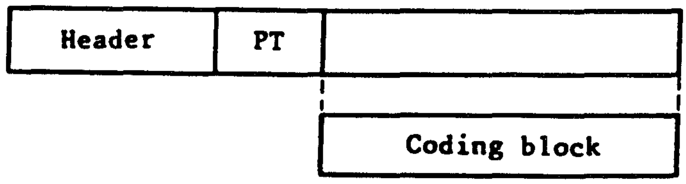
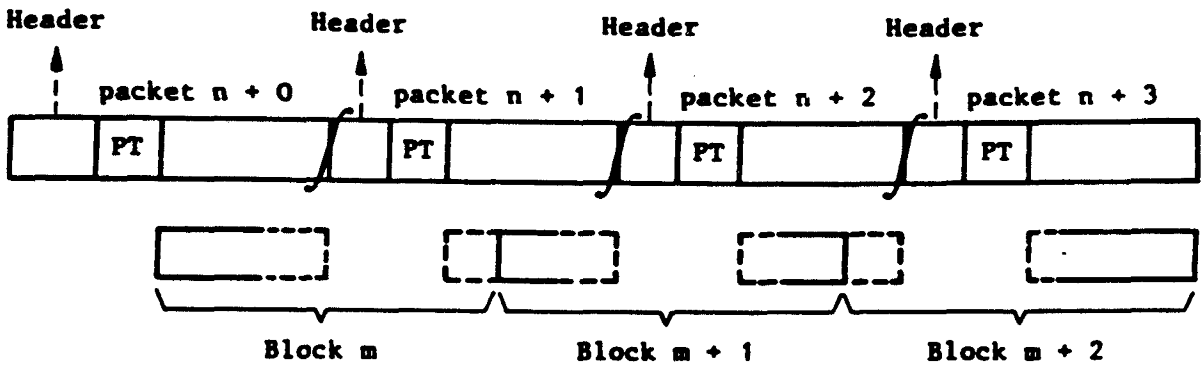


Fig. 5: Insertion of a 90 byte coding block

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.

Fig. 6: Insertion of a 120 byte coding block

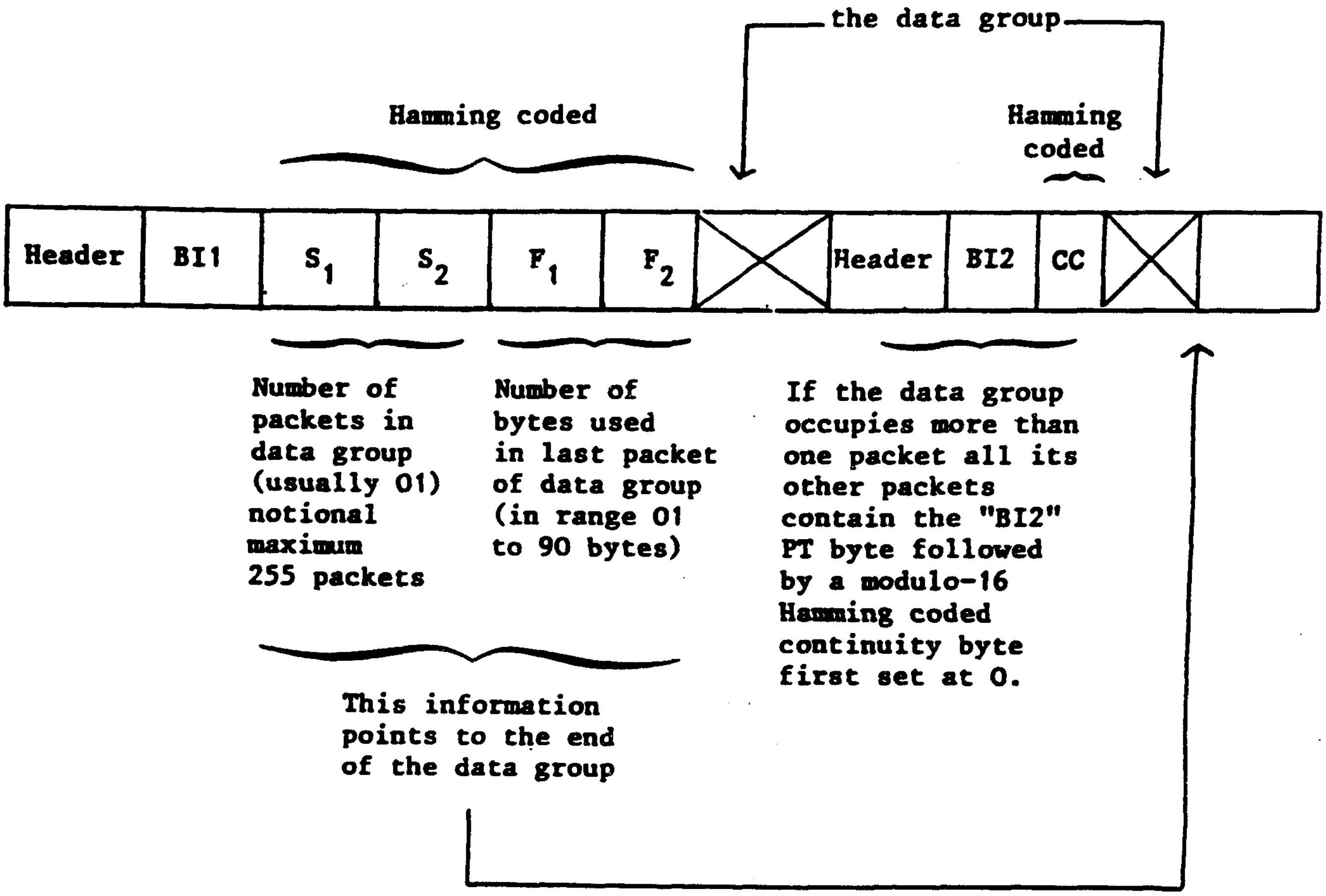


Fig. 7: Positioning of the data group in the interpretation packets. (The values of BI1 and BI2 are defined in Section 7.1)

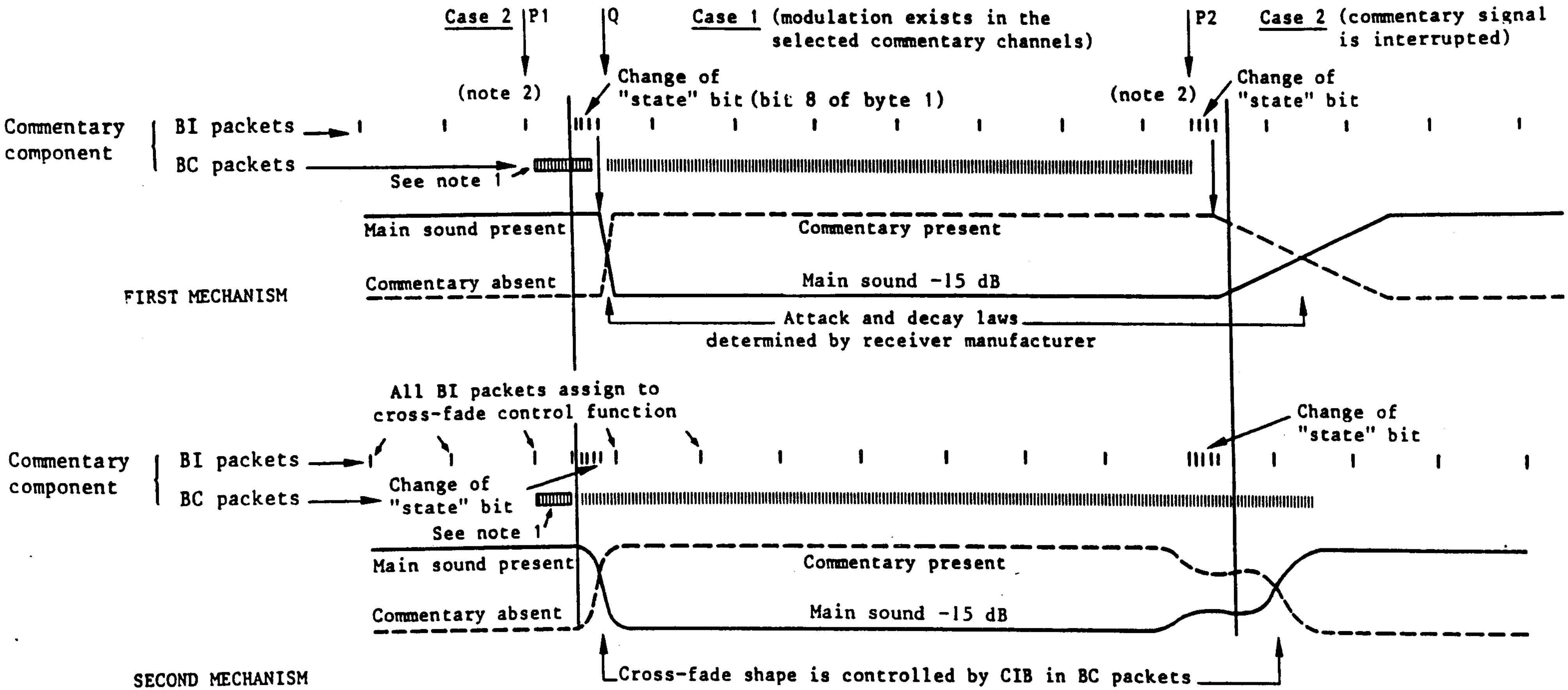


Fig. 8: Automatic mixing of main sound and commentary (BC and BI packets of main sound continue normally throughout this operation)

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

Annex 1 to Part 3

CODING EXAMPLES FOR THE INTERPRETATION BLOCKS FOR A SOUND CHANNEL IN VARIOUS CONFIGURATIONS

1. General

In order to illustrate the specifications set out in Section 9 of Part 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.

2. 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	BI_1 = "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	10010000	00001110
9		Byte 2	00010001	01001010
10	Repetition 2	Byte 1	10010000	00001110
11		Byte 2	00010001	01001010
12	Repetition 3	Byte 1	10010000	00001110
13		Byte 2	00010001	01001010
14	Repetition 4	Byte 1	10010000	00001110
15		Byte 2	00010001	01001010
16	Repetition 5	Byte 1	10010000	00001110
17		Byte 2	00010001	01001010

3. 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	BI$_1$ = "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 COMMAND	CI	0 Hamming code: "15"
7		1ST COMMAND	LI	1 Hamming code: "02"
8		1ST COMMAND	Command No. changed	2 Hamming code: "49"
9	2ND COMMAND		CI	1 Hamming code: "02"
10	2ND COMMAND		LI	A Hamming code: "8C"
11	2ND COMMAND	Repetition 1	Byte 1	10010000
12	2ND COMMAND	Repetition 1	Byte 2	00010001
13	2ND COMMAND	Repetition 2	Byte 1	10010000
14	2ND COMMAND	Repetition 2	Byte 2	00010001
15	2ND COMMAND	Repetition 3	Byte 1	10010000
16	2ND COMMAND	Repetition 3	Byte 2	00010001
17	2ND COMMAND	Repetition 4	Byte 1	10010000
18	2ND COMMAND	Repetition 4	Byte 2	00010001
19	2ND COMMAND	Repetition 5	Byte 1	10010000
20	2ND COMMAND	Repetition 5	Byte 2	00010001
21	3rd COMMAND		CI	2 Hamming code: "49"
22	3rd COMMAND		LI	A Hamming code: "8C"
23	3rd COMMAND	Repetition 1	Byte 1	11001110
24	3rd COMMAND	Repetition 1	Byte 2	01001010
25	3rd COMMAND	Repetition 2	Byte 1	11001110
26	3rd COMMAND	Repetition 2	Byte 2	01001010
27	3rd COMMAND	Repetition 3	Byte 1	11001110
28	3rd COMMAND	Repetition 3	Byte 2	01001010
29	3rd COMMAND	Repetition 4	Byte 1	11001110
30	3rd COMMAND	Repetition 4	Byte 2	01001010
31	3rd COMMAND	Repetition 5	Byte 1	11001110
32	3rd COMMAND	Repetition 5	Byte 2	01001010

Annex 2 to Part 3

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.

Annex 2 to Part 3

TABLE

Possible combinations of sound mixing informations and expected behaviour of the receiver

Additional sound available*	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	No	-	Additional sound with 0 dB attenuation
	Yes	Equal	Additional sound with 3 dB attenuation + main sound with 3 dB attenuation
(Intermittent and present)	Yes	Attenuated	Additional sound with 0 dB attenuation + main sound with 15 dB attenuation***
Intermittent and interrupted	No		Silence**
	Yes	-	Main sound with 0 dB attenuation

* The additional sound is considered available when its description in the Service Identification channel matches the language selection criteria.

** This mode of operation is not recommended in the transmitted channel.

*** Practical tests have indicated that an attenuation of 15 dB is a maximum.

<u>Note</u> : 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.

PART 4: SPECIFICATION OF THE SYSTEM FOR DATA SERVICES

Subject of Part 4

This Part of the document contains the specifications for the transmission of data.

Part 4A specifies the transmission of teletext systems A & B given in Parts I and II respectively of the draft CCIR descriptive booklet. 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.

Part 4B 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.

Part 4C specifies the protocol of general purpose data services carried via the sound/data multiplex.

PART 4A: TELETEXT TRANSMITTED IN THE FIELD-BLANKING INTERVAL

Contents

Part 4A

1. Subject of Part 4A

Part 4A specifies the transmission of teletext systems A & B given in Parts I and II respectively of the draft CCIR descriptive booklet. 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.

The teletext data are transmitted in the order defined by the CCIR descriptive booklets for systems A and B (see Section 5), whereas the service identification data concerning these signals are transmitted in the order defined in Section 1 of Part 5.

2. Data transmission in the field-blanking interval

2.1 Data coding

The data corresponding to the teletext components are transmitted in lines of the field-blanking interval at a bit-rate of twice 10.125 Mbit/s. Signal generation starts from a 20.25 MHz sampling clock. Even samples carry bits from a teletext data line of a teletext service or subtitle component of the television service. Odd samples may carry bits from a teletext data line of another teletext service or subtitle component of the television service. Other than in these cases, the odd samples are filled with pseudo-random data consisting of the output of the PRBS generator used for scrambling for spectrum shaping purposes.

For the D-MAC/packet system, these data are time-multiplexed with the data described in Part 1 (burst of 206 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 (see Sections 4.2.3 and 4.2.4 of Part 1) as a whole and in a single operation.

2.2 Relationship between the bits of data and the sampling structure (Fig. 1)

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 corresponds to sample 230, and the last bit to sample 1292. For the D-MAC/packet system, the first bit of the teletext data line carried by odd sample numbers (if present) corresponds to sample 231 and the last bit to sample 1293. The remaining samples, i.e. 1294, 1295 and 1296, are unallocated.

2.3 Clamping transitions

In the lines of the field-blanking interval allocated to teletext transmission, the clamping period is obtained by forcing the duobinary signal to zero from sample 206 to 228 inclusive. In the other lines, the duobinary signal is forced to zero from sample 206 to 1295 inclusive prior to addition to the MAC signal as already described in Part 1.

3. Transmission of teletext components

3.1 Allocated lines

The lines allocated to CCIR systems A and B teletext are defined in line 625 according to TDMCTL procedure described in Section 5.4 of Part 1. Any free lines of the intervals from line 1 to 22 inclusive and from line 311 to 334 inclusive may be used.

3.2 Transmission of CCIR system A teletext

CCIR system A teletext is transmitted as data lines of 320 bits as specified in the document given in <1>. These data lines are inserted in the first part of the 532 bits defined in Section 2.2, without scrambling or interleaving. The remaining 212 bits are unused and filled with pseudo-random data consisting of the output of the PRBS generator used for scrambling for spectrum shaping purposes.

3.3 Transmission of CCIR system B teletext

CCIR system B teletext is transmitted as data lines of 360 bits as specified in the document given in <2>. These data lines are inserted in the first part of the 532 bits defined in Section 2.2, without scrambling or interleaving. The remaining 172 bits are unused and filled with pseudo-random data consisting of the output of the PRBS generator used for scrambling for spectrum shaping purposes.

4. Service-identification data

Access coordinates of teletext components transmitted in the field-blanking interval are given by the parameter DCINF of the service-identification channel (see Section 2.3 of Part 5) 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 is as follows:

a) CCIR system A teletext:

b_{16} = bit D defined in <1> (information on bit-rate)
b_{15} = 0
b_{14}, b_{13}: two most significant bits of data channel address
(A_1, A_2, A_3 of <1>)
b_{12} = 0
b_{11} = 0 if teletext component was inserted on even numbered samples
= 1 if teletext component was inserted on odd numbered samples
b_1 to b_{10}: ten least significant bits of data channel address
(A_1, A_2, A_3 of <1>)

b) CCIR system B teletext:

b_{16} = 0
b_{13} to b_{15}: magazine number
b_{12} = 0
b_{11} = 0 if teletext component was inserted on even numbered samples
= 1 if teletext component was inserted on odd numbered samples
b_9, b_{10}: irrelevant
b_1 to b_8: page number.

The complementary access coordinates are composed of 2 bytes coded as a 16-bit sequence as follows:

a) CCIR system A teletext: page number on the 12 LSB
b) CCIR system B teletext: page sub-code on the 13 LSB.

The transmission of the complementary access coordinates is mandatory when CCIR system A teletext is used for subtitling.

5. References

<1> Draft CCIR descriptive booklet - Part I, Teletext system A

<2> Draft CCIR descriptive booklet - Part II, Teletext system B

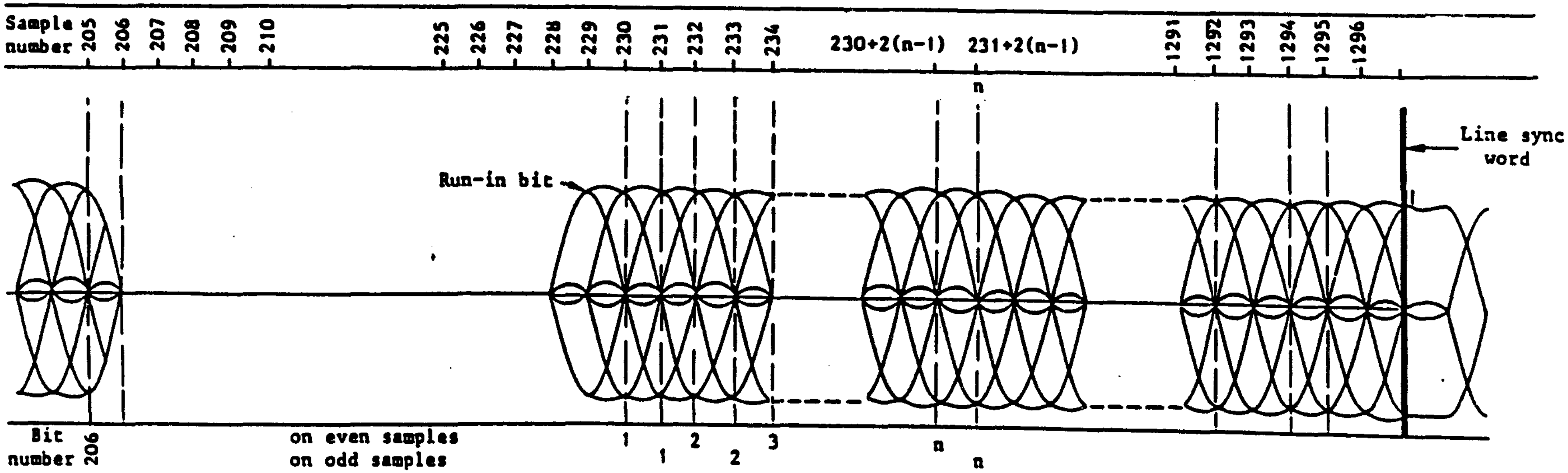


Fig. 1: Relationship between the bits of data and the sampling structure

PART 4B: TELETEXT SERVICES WITHIN THE PACKET MULTIPLEX

Contents

1. Subject of Part 4B

Part 4B gives the specification of the transport mechanisms using the sound/data multiplex of the D-MAC/packet system, 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, is described in the service identification channel, which is formed by packets in the sound/data multiplex with the packet address "0" (see Part 5).

2. Teletext transmission structure

2.1 Teletext data block description

CCIR teletext system A or B is transmitted as teletext data blocks of 360 bits (45 bytes). These are drawn from the data bits of the system A or B teletext lines, as specified in References <1> and <2>, 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 is divided into a 6-byte prefix, an information field of 36 useful bytes plus 1 filling byte, and a 2-byte CRC suffix (see Fig. 1a). The 6-byte prefix contains in sequence:

- one control byte (CB)
- three address bytes (P1, P2, P3)
- one continuity index (CI)
- one format indicator (PS).

In the case of CCIR system B, the 45-byte teletext data block is divided into a 3-byte prefix, an information field of 40 bytes and a 2-byte CRC suffix (see Fig. 1b). The 3-byte prefix carries in sequence:

- one control byte (CB)
- the 16-bit magazine and teletext data packet address group.

2.2 Data block prefix description

All prefix bytes are (8,4) Hamming encoded.

The coding of the message bits b2 b4 b6 b8 of the control byte (CB) is 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 is 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 is as specified in Ref. <1>; in the case of CCIR system B, the coding of the message bits of the 16-bit magazine and teletext data packet address group is as specified in Ref. <2>.

2.3 Data block suffix description

The data block suffix is 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 are such that the polynomial:

$$m_{295}\, x^{311} + m_{294}\, x^{310} + \ldots + m_0\, x^{16} + r_{15}\, x^{15} + \ldots + r_1\, x + r_0$$

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 are such that the polynomial:

$$m_{319}\, x^{335} + m_{318}\, x^{334} + \ldots + m_0\, x^{16} + r_{15}\, x^{15} + \ldots + r_1\, x + r_0$$

is a multiple (modulo 2) of the polynomial:

$$x^{16} + x^{12} + x^5 + 1$$

2.4 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 is 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 is 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.

3. Insertion of teletext data blocks into the MAC/packet multiplex

The structure of the MAC/packet is shown in Fig 2a in the case of first-level protection and in Fig. 2b in the case of second-level protection. The MAC/packet commences with a 23-bit packet header followed by a PT-byte which is used to signal, on a packet basis, the access mode. Scrambling as specified in Part 6 (Section 3) and Part 3 (Section 3.5) may be applied to the content of the packet. The PT-byte values are assigned as follows:

PT-byte	access mode
'C7	free access, scrambled
'F8	controlled access, scrambled
'00	unscrambled.

The fourth PT-byte value ('3F) remains unallocated.

4. Service identification data

Access coordinates of the service or service component transmitted in the packet multiplex are given by the parameter DCINF of the service identification channel (see Section 2.3 of Part 5) 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 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 Section 2.3 of Part 5.

In the case of system A teletext only, the first two bytes of the complementary access coordinates 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; this transmission is mandatory.

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 is optionally 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) are 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) are 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.

The transmission of the complete complementary access coordinates is mandatory when CCIR teletext systems A or B are used for subtitling.

5. References

<1> CCIR Descriptive Booklet - Part I, Teletext system A

<2> CCIR Descriptive Booklet - Part II, Teletext system B

Part 4B

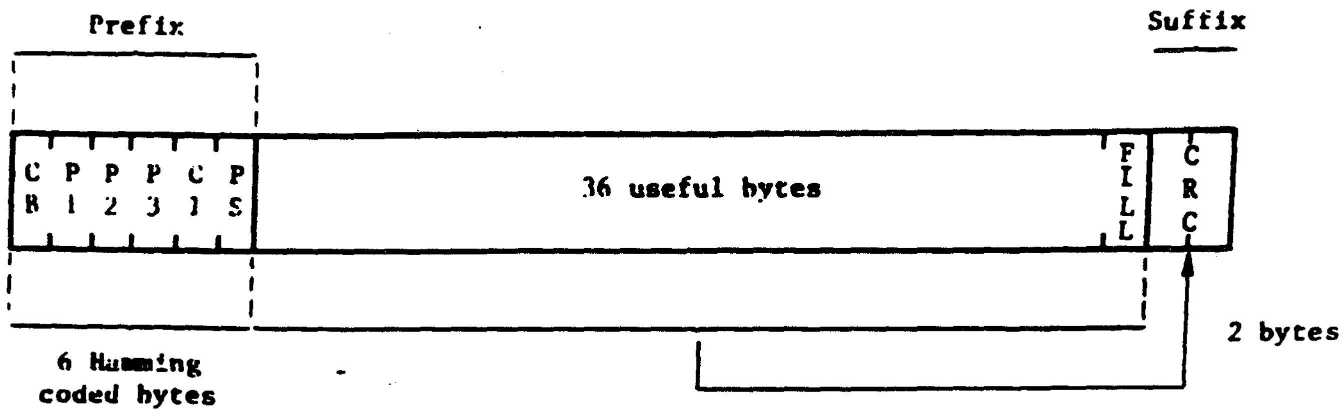


Fig. 1a: CCIR system A teletext

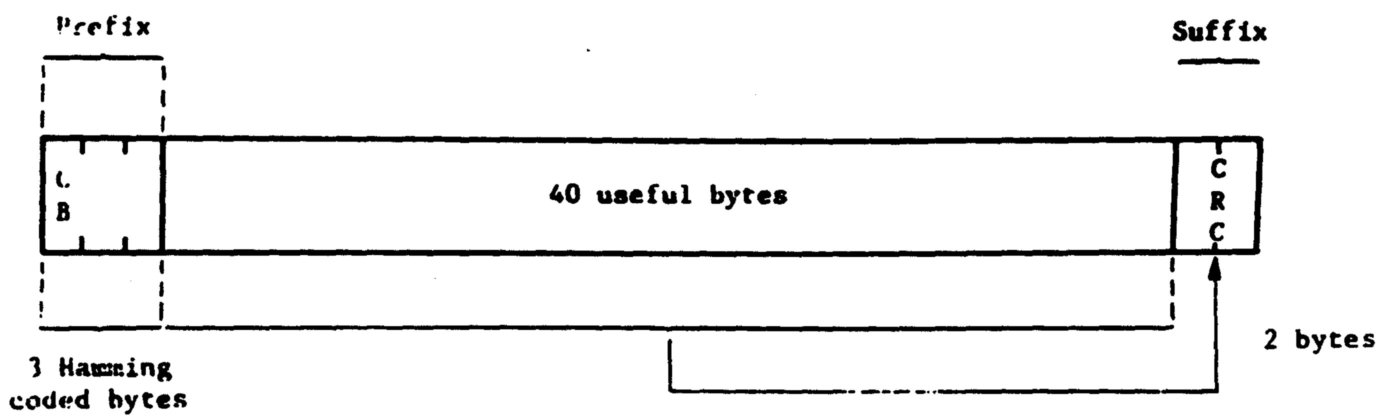


Fig. 1b: CCIR system B teletext

Fig. 1: Construction of a 45-byte teletext data block from a teletext data packet with CRC suffix

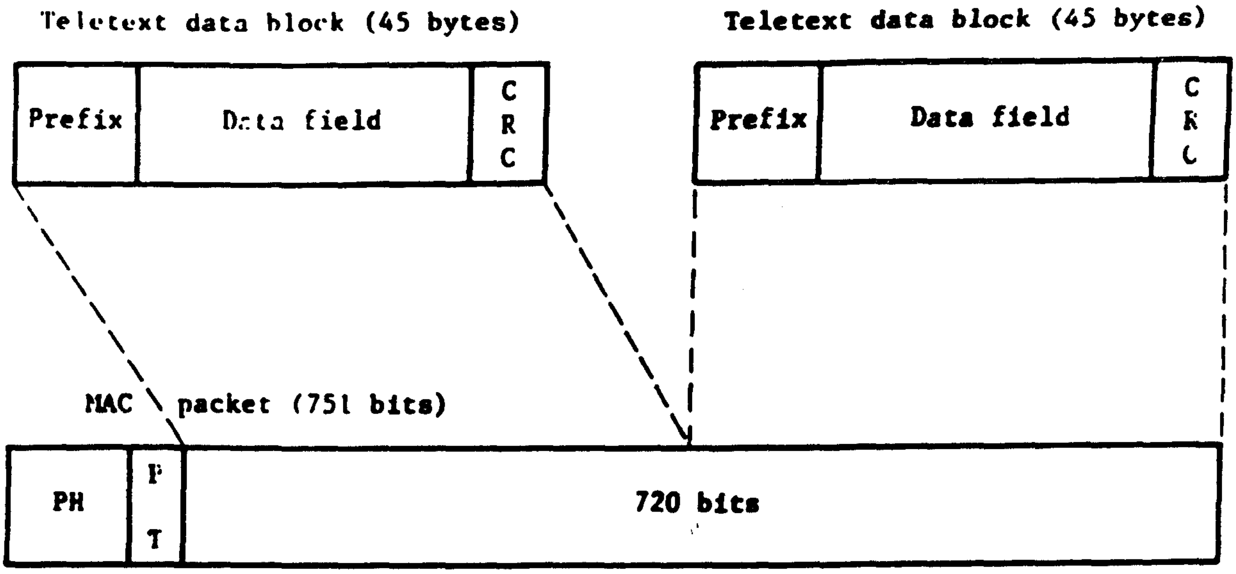


Fig. 2a: First-level protection

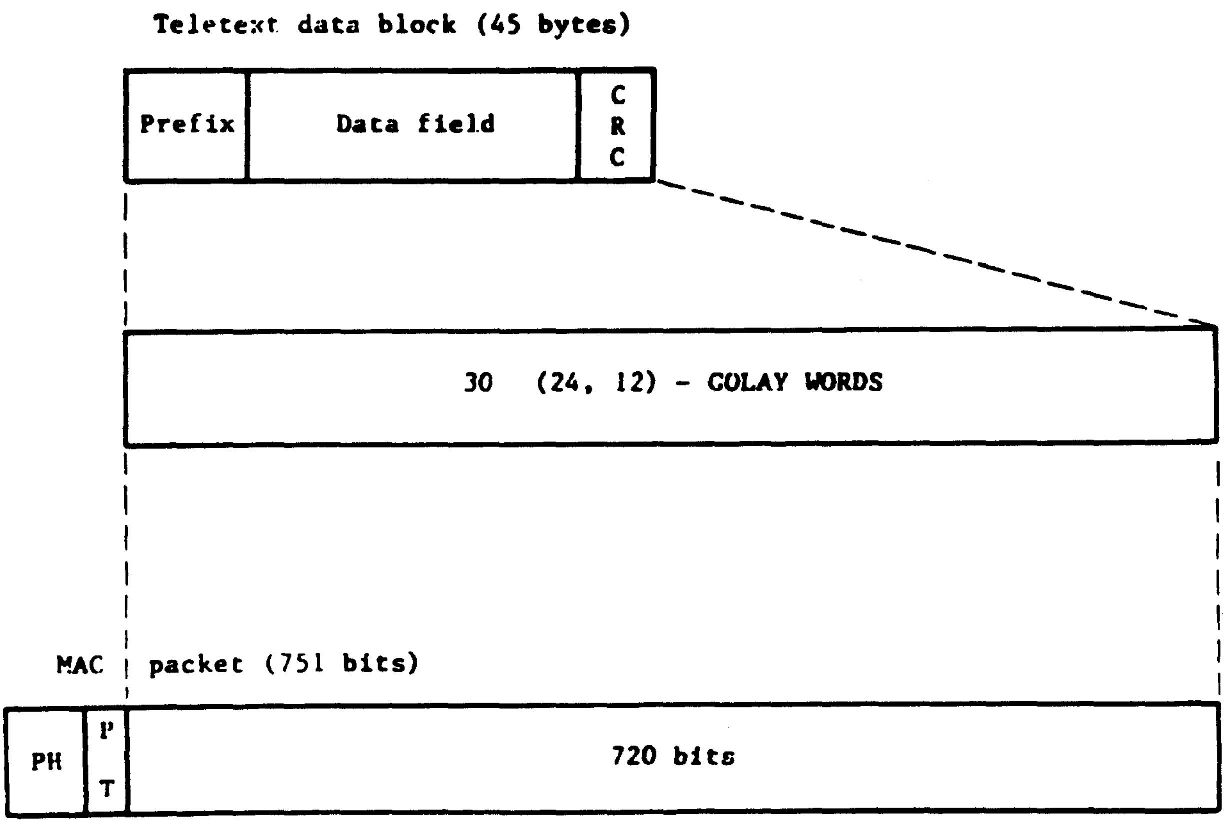


Fig. 2b: Second-level protection

Fig. 2: The insertion of teletext data blocks in a MAC packet

PART 4C: GENERAL PURPOSE DATA WITHIN THE PACKET MULTIPLEX

Contents

Part 4C

1. Subject of Part 4C

Part 4C specifies the protocol of general purpose data (GPD) services carried via the sound/data multiplex of the D-MAC/packet system. 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 multiplex by having a different format. The one-way distribution of general data may be carried either by teletext components or by GPD components.

2. Description of GPD service protocol

2.1 General

In terms of the OSI model, a satellite 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 should be external to the MAC/packet specification.

GPD services may be carried as one or more MAC/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) - User-data, defined as data interpreted at layer IV or above, is carried in data messages. Data messages (see Fig. 1a) 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 Part 6, Section 5.2. The performance improvement using Golay coding is illustrated in the Annex to Part 4C .

2.2 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.

2.3 Identification of GPD services

The SI description of a GPD service component includes 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 is 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.

2.4 Data message structure (see Fig. 1)

A data message consists of:

- an optional variable length data message header
- a segment of user-data
- an optional segment protection CRC.

The data message header consists of:

- a format descriptor (FD) byte
- a header extension, of variable length according to FD coding.

2.4.1 Data message header

The FD byte is Hamming (8,4) coded according to Part 3, Table 4. The four information bits are 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, consists 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.

The address extension field, if present, consists of an address extension length indicator byte coded as Hamming (8,4) according to Part 3, Table 4 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, define different data channels.

The segment counter field, if present, consists 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 increments modulo-65536 for successive data messages.

The segment length indicator, if present, consists 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.

2.4.2 Segment description

The data segment, immediately following the message header, consists of a sequence of bytes of user-data, each transmitted LSB first.

2.4.3 Data segment CRC description

The segment CRC, if present, immediately follows the data segment. It is computed from all bytes of the data segment in transmission order. The message header is not included in the computation. The generator polynomial and computation method are as specified in Part 5, Section 3.2.

3. Assembly of messages into packets

When a message header is not used, packets will 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.

3.1 Non-linked component

Each data message, with Golay coding if used, must fit within the 90 bytes following the PT byte of a single packet. The packet header CI value increments 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 must fill the packet with no bytes left unused. Thus the last byte of the packet must 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 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 must be contained within a maximum of 90 bytes. If necessary, arbitrary padding bytes are 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 must be contained within a maximum of 45 bytes. If necessary, arbitrary padding bytes are added after the message to complete 45 bytes. The block of 45 bytes is then coded into thirty Golay (24,12) words as specified in Part 6, Section 5.2. The resultant 90 byte block is then loaded into the 90 byte space following the PT byte of a single packet.

3.2 Linked component

Each data message, with Golay coding if used, must fit within the 360 bytes following the PT bytes of four linked packets, which are linked by packet header CI. CI = 0 for the synchronising packet (i.e. the first packet in the linked group), and increments for successive packets in the linked group.

If the message header FD byte bit b2 = 0, the data message must 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 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 is 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 is 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 is then coded into thirty Golay (24,12) words as specified in Part 6, Section 5.2. The resultant 90 byte blocks are then loaded into the 90 byte spaces following the PT bytes of the linked packets.

3.3 Use of PT byte

GPD service components packets are 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*

PT = 'C7 Reserved.

4. Multiplexing rules

Packets of one linked group of a GPD service component must 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 must not be interleaved with the packets of any other linked group having the same packet address but different address extension field.

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* Control or interpretation packets may be found useful to provide extra service management within each GPD component.

5. 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 Sections 1 to 4. A GPD service is defined as containing 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 MAC 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 is carried under a separate packet address, the value of which may be from 1 to 1022 (decimal) inclusive. The value of packet address used for GPD SI in any particular case is included in the information conveyed by the DATX parameter in the Network Command in DG0 of packet 0 SI, as described in Part 5, Section 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 are included within the GPD service description, and do not appear in packet 0 SI.

A GPD service must 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 must remain constant for as long as the associated GPD service remains in existence in the commercial or other sense, and must be uniquely associated with one, and only one, GPD service transmitted on a given MAC network. In this context, "Network" has the same meaning as defined in Part 5, Section 1 (footnote), ie., the ensemble of signals broadcast in the satellite channel at a given time. 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 - (2^{56} - 1) which is 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 is 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.., has 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.

5.1 General description of GPD service identification

All GPD SI information is 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 are 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 must continue to be described in GPD SI messages, and the service is considered active. When a GPD service becomes inactive, the GPD SI description should 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 must 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 are identified by Type of List (TL), and the following values are defined:

- Type '0: Network Summary

This is an optional list, consisting 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 is a mandatory list, and 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 message index (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 zer 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 must be described in the same message having MI = PX, and within this message, no two service descriptions may have the same value of SX.

- Type '2: NSI cross-reference list

This is an optional list, but any active GPD services which have an NSI must 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 is an optional list. 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) must 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 is signalled in DATX.

- Type '4: Name server list

This is an optional list. All active GPD services which do not appear in either the NSI or UI cross-reference lists must appear in the name server list. Where this list consists of more than one message, the messages must be transmitted cyclically with message index values which increment modulo-N, where N is the number of messages which comprise the complete list. The value of N is 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 has 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 consists of the SX value plus a plain text service name.

All other values of TL are unallocated and reserved.

5.2 Description of GPD SI list transport protocol

The following details, illustrated in Fig. 2a, are common to all types of GPD SI list.

A GPD SI list may consist of one or more messages. Each message is 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 Part 5, Section 3.2.

The message header is 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 Section 5.1, with values '05 to 'FF reserved. All messages in a given GPD SI list must 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 Section 5.3 below.

- 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 is 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, subject to confirmation, in connection with full channel digital mode of operation. Initial normal mode transmissions should set the byte value 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 Part 5, Section 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 message index (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.

5.3 Use of Message Index (MI) in each type of list

5.3.1 Network Summary list

If present, this consists of one message, for which MI is set to 0.

5.3.2 Service Description list

In each message of the list, MI must 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, must be described in the message identified by MI = PX. At any time, there must 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.

5.3.3 NSI cross-reference list

In each message of the list, MI must have the value of a pointer computed from NSI. All services having values of NSI yielding the same pointer value, and no other services, must appear in the message identified by MI equal to this pointer. At any time, there must be only one message in the NSI cross-reference list, transmitted repetitively, having a given value of MI. The NSI message pointer is 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.

5.3.4 UI cross-reference list

In each message of the list, MI must have the value of a pointer computed from UI. All services having values of UI yielding the same pointer value, and no other services, must appear in the message identified by MI equal to this pointer. At any time, there must be only one message in the NSI cross-reference list, transmitted repetitively, having a given value of MI. The UI message pointer is 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.

5.3.5 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 should be set to zero. When there are N messages in this list, they should be transmitted with MI incrementing modulo-N, i.e., in a repeating cycle from 0 to (N-1).

5.4 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 are signalled in the DATX parameter (see Part 5, Section 2.1). The same algorithm is 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.

5.5 Assembling messages into packets

All GPD SI messages are 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 is divided, from Byte 1 of its header, into blocks of 45 bytes, adding arbitary padding bytes if necessary after the CRC to complete the last (or only) block. Each block is coded into 90 bytes each consisting of 30 Golay (24,12) words as described in Part 6, Section 5.3.2.

When a message, so coded, consists of only one 90-byte block, it is 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 is 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 is set to zero for the first (or synchronising) packet, and increments 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 should be transmitted in empty form.

5.6 Rules of multiplexing GPD SI messages

5.6.1 Transmission rates

If present, the Network Summary message must 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 must 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.

5.6.2 Subframe location

There is no requirement that a GPD message is transmitted in the same subframe as the GPD service components which it references. The choice of subframe(s) for GPD services and the corresponding service identification information is at the discretion of the network management, according to requirements.

When any list currently being transmitted has a subframe location designation of binary 00, the packets of one GPD SI message must not be interleaved in transmission with the packets of any other GPD SI message, when considering both subframes together. When all messages currently being transmitted have subframe location designations of binary 01, 10, or 11, the packets of one GPD SI message must not be interleaved in transmission with the packets of any other GPD SI message, when considering either subframes individually.

5.7 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.

5.7.1 Network Summary message (see Fig. 2b)

The network summary message consists 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 will be the Parameter Identifier (PI) of whichever parameter comes first. The parameters should be ordered in increasing value of PI.

A change in the coding of any parameter in the Network Summary message must be accompanied by an increment in the value of the message CSN byte, and must occur at nominally the same time as the associated coding change in the packet 0 channel.

5.7.2 Service Description list (see Fig. 2c)

The contents of each message in this list must consist of one or more GPD service descriptions. Each service description is 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

An optional parameter 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

A mandatory parameter when conditional access scrambling is in use: data field as ACCM.

DGPD PI = '2

A mandatory parameter to describe each GPD component: data field as follows:

Bytes 1 and 2: Bytes 1 and 2 are coded as a 16-bit field. The 12 LSBs are coded as for DCINF, and the 4 MSBs are 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 : this byte is optional. If provided it is 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 kbits/sec
'7 - maximum throughput 7680 bits/sec.

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 : an optional language byte, coded as for packet 0 sound DCINF. If this byte is provided when byte 3 is not required, byte 3 must 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

A mandatory parameter 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

A mandatory parameter if video carried in the sub-frames identified by TDMCID '10 and '11 is part of a GPD service: the data field consists of one mandatory byte coded as for VCONF.

Other values of PI are unallocated and reserved. Values '9 to 'F are reserved, subject to confirmation, for use in connection with full channel digital mode of operation.

The association between conditional access parameters (DACMM and DACCM) and service component parameters (DGPD, DSOUND, DTEXT, DVIDEO) is effected by parameter ordering. There is no requirement to order parameters in ascending order of parameter indentifier. Component parameters are associated with the closest preceding conditional access parameter, if present. The parameters for all unscrambled components must appear before any DACCM parameters.

GPD service descriptions, coded as above, are assembled into service description messages in ascending order of SX value. All service descriptions in a given message must 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 may have the same value of SX.

5.7.3 NSI cross-reference list (see Fig. 2d)

An NSI cross-reference list message consists 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, are assembled into messages in ascending order of NSI value. Each NSI cross-reference message must 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.

5.7.4 UI cross-reference list (see Fig. 2e)

A UI cross-reference list message consists of binary information in fixed-length entries, coded as follows. The length of the entry depends 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, are assembled into messages in ascending order of UI value. Each UI cross-reference message must 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.

5.7.5 Name server list (see Fig. 2f)

Entries in the name server list are 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 are 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 above in Section 5.3.5.

The first byte of each name service message gives the value of PX for all entries in the message. The rest of the message consists 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 Sections 3.1 to 3.2 of Annex 1 to Part 5.

5.8 Coding changes

Whenever any content of a GPD SI message is changed, the Coding Sequence Number (CSN) in the message header must 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 should 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 should stop before the first SI message containing the change, and should 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 should be made at a sufficient time interval before the new service or component starts in order to allow all decoders to acquire it.

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)

Fig 1: 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	

Fig. 2: 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

Fig. 2: Format of GPD SI messages (continued)

Annex to Part 4C

PERFORMANCE OF GPD PROTOCOL

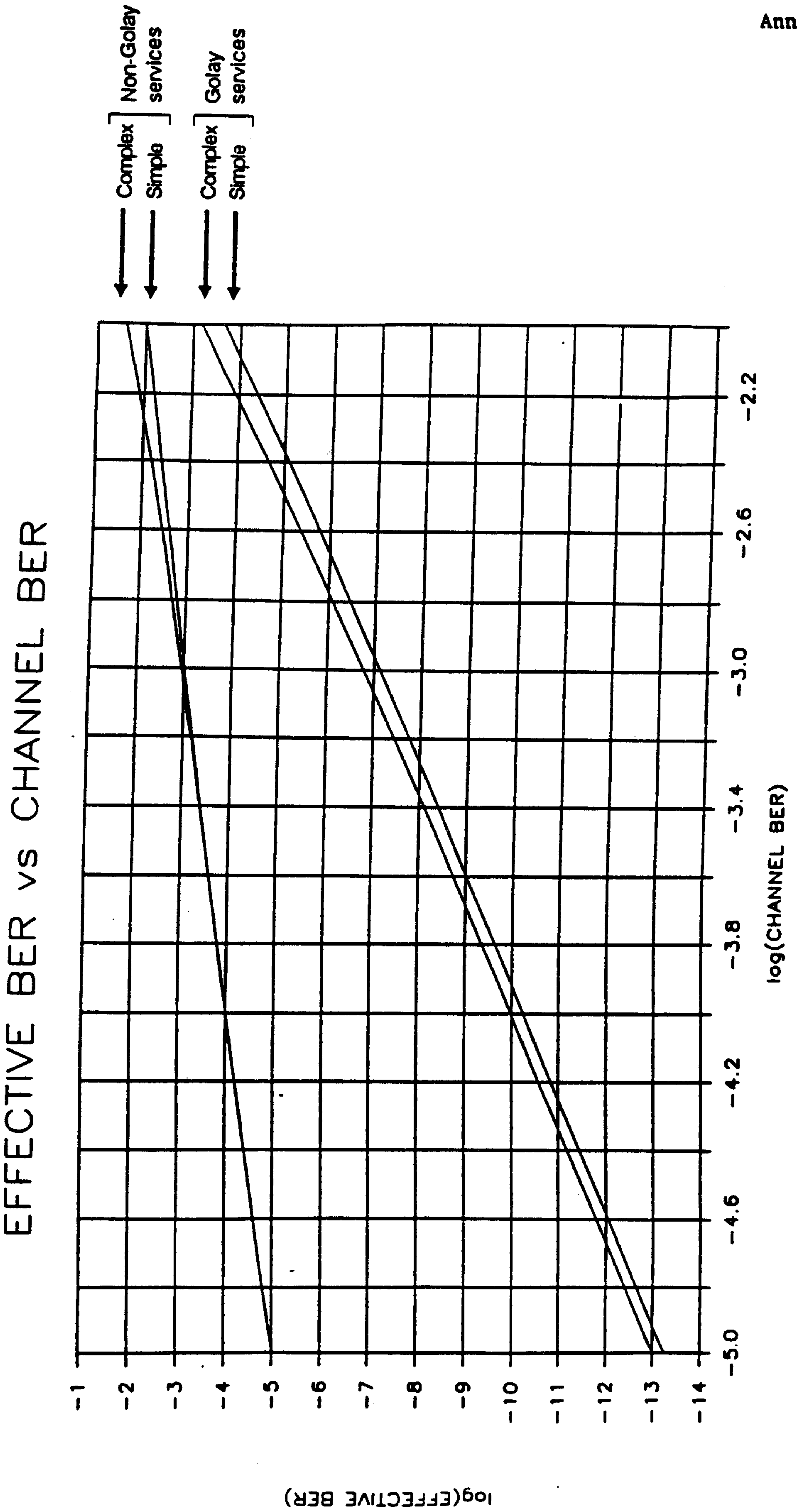
IN THE PRESENCE OF CHANNEL BIT ERRORS

The Figure 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 Part 4C, but should be implemented at OSI layer IV (transport).

Annex to Part 4C



Simple reference message : I 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.

PART 5: SPECIFICATION OF THE SERVICE IDENTIFICATION CHANNEL

Contents

1. Subject of Part 5

Part 5 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 Section 2.1.1 of Part 3). For normal television transmissions, the information broadcast in this channel gives the user access to the various television, sound, and data services, that may co-exist in a channel carrying a signal of the MAC/packet family. In the case of full-channel digital mode of operation, each digital TDM component carries its own SI channel. It provides the information needed by the user to access the sound services and the data services present in that TDM component.

A service may comprise several components; for example, a television service might comprise a vision component, one or 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 Section 2. For transmission, the parameters may be grouped into parameter groups and are coded into commands (see Section 3.1). The commands are assembled into data groups each of which consists of one or more whole packets (see Section 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. Section 4 gives rules for multiplexing the packets carrying the data groups in the whole sound/data multiplex.

Annex 1 to this Part 5 specified coding tables for certain information transmitted by the service identification system. Annex 2 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 Sections, the least significant bits are 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.

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* The word "network" is used here to refer to the ensemble of signals broadcast in the satellite channel at a given time.

2. Parameters

The following parameters are used to describe the network and the services available.

2.1 Network identification and description in terms of its services

(NWO) Network origin. This information consists of three bytes followed by a variable-length text. It is a mandatory parameter. 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 is 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
  0011: Linear polarisation Y*, tilting from the vertical plane.

The text gives the name of the country of origin coded according to the standard given in Section 3 of Annex 1.

(NWNAME) Name of the network as it is known by the public, requiring a variable length of text, coded according to the standard given in Section 3 of Annex 1. This is a mandatory parameter.

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* The X and Y polarisations 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.

Decoders should not react to any bit combinations not defined above.

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*. This is an optional parameter with parameter identifier '16. This information consists 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 is 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 D-MAC and D2-MAC channels.

- 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:

0000 : linear pre-emphasis only (see Section 2.3.1 of Part 7)

0001 : both linear and E7 non-linear pre-emphasis (see Section 2.3.2 of Part 7)

other values: both linear and other non-linear pre-emphasis

The least-significant four bits are unallocated.

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* This parameter is mandatory when the transmit parameters do not correspond to the default values of:

a) centre frequency specified by the WARC channel number (see parameter NWO)
b) deviation equal to 13.5 MHz/V for both the vision and sound/data signals
c) energy dispersal equal to 600 kHz peak-to-peak
d) linear pre-emphasis.

- optional ninth byte: 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) This is a mandatory parameter which 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 Section 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 has a variable length:

- 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 Section 5.3 of Part 6) 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 have been allocated the following codes (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 is 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 Section 4 of Part 4A).

The two remaining bits (bits 11 and 12) are defined at present only in subframes characterized by TDMCID 01-0E, where they are used for digital component location in the following way:

bit 11 = 1, bit 12 = 0: The packets of the digital component are inserted in the subframe characterized by an odd TDMCID code in the range 01-0E (see Section 5.4 of Part 1).

bit 11 = 0, bit 12 = 1: The packets of the digital component are inserted in the subframe characterized by an even TDMCID code in the range 01-0E (see Section 5.4 of Part 1).

bit 11 = 1, bit 12 = 1: The packets of the digital component are inserted in both subframes of a D-MAC type sound/data multiplex. The service component can be recovered when considering only the D2 subframe.

bit 11 = 0, bit 12 = 0: The packets of the digital component are inserted in both subframes of a D-MAC type sound/data multiplex. The service component cannot be recovered when considering only the D2 subframe.

The first eight entries in the list are eight ordered default choices for that type of service.

The parameters NWO, NWNAME, LISTX, COMD (for network commentary) and TIMD (for local time description) together with UPDAT (see Section 2.4) if applicable are always sent in data group type '0' of packet address '0'.

(DATX) A parameter with PI = '19 which must 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 Part 4C, Section 5, for details of GPD SI description and acquisition.

DATX consists 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, subject to confirmation, in connection with full channel digital mode of operation.

Byte 3	Set to zero, reserved for future use, subject to confirmation, in connection with full channel digital mode of operation.

Byte 4

bit 8	Presence (bit = 1) or absence (bit = 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 = 1) or absence (bit = 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 = 1) or absence (bit = 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 = 1) or absence (bit = 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 must be transmitted in rotation with MI incrementing modulo-n.

All unallocated bits are reserved for future use, and should be set to 0 in initial transmissions.

(TIME) Local time, of medium precision, is an optional parameter intended for display and related to a relevant time-zone. The information is coded, in accordance with the standard given in Section 3 of Annex 1, as text of limited length, for example in the form: day of week/year/month number/day of month/hour/minute. The length is limited to 32 characters. Changes to the value of TIME are simply broadcast; they are not announced through the UPDAT mechanism. However, UPDAT is used in association with TIME if the TIME parameter is being added to or deleted from a data group.

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, composed of two items of information (coded in the given order):

- 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 Section 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 Section 3 of Annex 1. The length is limited to 32 characters.

SREF is a mandatory parameter for all non-TV service descriptions which are introduced with a LISTX parameter. It is recommended to be used also for the TV service description.

(PREF) Programme item reference is an optional parameter which when used is composed of three items of information (coded in the given order):

- 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 Section 4 of Annex 1. Value '0' indicates that no programme-type information is provided. As only one code can be transmitted for each programme, the broadcasting organization must decide 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 consists of a variable-length text, coded according to the standard given in Section 3 of Annex 1. The length is limited to 32 characters.

(NXPREF) Next programme item reference is an optional parameter which when used provides information about the next programme item to be transmitted. It is coded in the same manner as PREF.

(VPS) Video Programme System* is an optional parameter, 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).

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* This parameter can be used as an option. 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. The addition of this signal in line 16 is permitted in order to provide compatibility with video tape recorders for which D-MAC/packet signals are converted into a composite TV signal. This line, when used, is signalled within the time division multiplex control of line 625.

2.3 Service components description

Depending on the service type and the complexity of the programme, one or several components may be present. 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 gives the information concerning entitlement management messages. It is a mandatory parameter 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 applies to one type of conditional-access system. When used within another type of service description it applies to one CA system, itself applicable to the service components following it in the command or parameter group. The parameter field is 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
- 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 gives the information concerning entitlement checking messages (ECMs) described in Section 5.2 of Part 6. It is a mandatory parameter for all service components which are subject to conditional-access control. The ACCM content is associated with one or more service components. The coding is 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 is derived from the 20 bit CAFCNT parameter carried in line 625. If bit 15 is set to 0, the control word is 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 Section 2.2, 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 applies 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 applies 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 : this mandatory parameter allows the correct "interpretation" of the video signal. It is coded as one byte according to the following rules:

the three least significant bits duplicate the VSAM sub-group data of MVSCG (see Section 5.3 of Part 1), 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			
00100xxx	AR = 4:3	Cy = 3:2	Cu = 3:1
00110xxx	AR 16:9	Cy = 3:2	Cu = 3:1
00010xxx**	Not to be used		
00011xxx**	Not to be used		

* See Section 7 of Part 2.

** This combination may be reassigned at a later date.

(ASCONF) Analogue sound configuration: This parameter allows the correct "interpretation" of an analogue sound signal. It could be introduced for terrestrial applications, when an analogue sound component is present.

(DCINF) Digital component information: This a mandatory parameter for any service with one or more digital components. A separate DCINF parameter is supplied for each digital sound, data and teletext component. DCINF consists of three items (the last being optional), coded in the given order:

- 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 Section 5 of Annex 1.

- 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). As in LISTX the two bits adjacent to the packet address bits (bits 11 and 12) give an indication on the subframe related location of this digital component. The four most significant bits are coded according to the type of component. For teletext within the packet multiplex, these bits of the data field are coded as specified in Section 4 of Part 4B. For teletext in the field- blanking interval, this data field is coded as described in Section 4 of Part 4A.

- 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 Section 4 of Part 4A for components in the field-blanking interval and to Section 4 of Part 4B for components in the packet multiplex.

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* See however Section 4 of Part 4A and Section 4 of Part 4B for teletext components.

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 is identical to the corresponding BI data specified in Part 3 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 must be set to zero. (Bit 8 in the BI data specifies whether the sound signal is present or interrupted.)

All other bits will 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.

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, signalling the changing of the content of a previous network or service description. The coding of those messages will be given in Section 3.1. These messages are mandatory.

(ULIST) The update list parameter (PI = '01) is an optional parameter 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 are specified in terms of the corresponding data bytes of the associated UPDAT parameter:

UPDAT	ULIST
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 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 must 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, information on an existing direct or indirect commentary. This is an optional parameter. 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 Section 3 of Annex 1. The field coding of COMD is 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 give the value of the data group type which carries this commentary.

(TIMD) Description of local time with the same coding as that specified for COMD, in order to provide the access coordinates of the local time information (TIME) (see Section 2.1). This is an optional parameter.

(DCOM) Direct commentary, a variable-format clear text giving information on the network or on the services broadcast. This is an optional parameter. It is 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 Section 3 of Annex 1.

(LKPNT) Link pointer, 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 is 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.

2.5 Further parameters required for additional service identification (ASI)

(MID) Multiplex identity. A parameter used in the ASI service to identify a subframe other than the one which carries the ASI. It is coded as one byte representing either the lower numerical value of the pair of TDMCID codes (see Part 1, Section 5.4) which characterize a dual subframe, or the TDMCID code of an unpaired subframe. Services and service components described by parameters following a MID parameter are carried in the subframe identified by MID (if applicable), unless a different TDMCID is explicitely indicated with an ACMEM, ACCEM, or EDCINF parameter.

(EDCINF) Extended digital component information: this consists of four items (the last being optional), coded in the given order:

- 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).

  The two bits adjacent to the packet address bits (bits 11 and 12) indicate the location of the digital component within a dual subframe in the same manner as bits 11 and 12 in LISTX, but where the dual subframe is that characterized by the pair of TDMCID values defined below. If the digital component is carried in an unpaired subframe (characterized by TDMCID values '40 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, either representing the lower numerical value of the pair of TDMCID codes (see Part 1, Section 5.4) characterizing the dual subframe which carries the digital component, or representing the TDMCID code of the unpaired subframe which carries the digital component.

- Complementary access coordinates: definition as for DCINF.

(ACCEM) Access checking-related extended messages: the parameter field is as defined for ACCM except that the first 16-bit field is extended to 24 bits. The additional 8 bits (MSBs) provide the multiplex location as defined in EDCINF.

(ACMEM) Access management-related extended messages: the parameter field is as defined for ACMM, except that the first 16-bit field is extended to 24 bits. The additional 8 bits (MSBs) provide the multiplex location as defined in EDCINF.

3. Coding structure

Most of the parameters listed in Section 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 is also transmitted in the uncoded form.

3.1 Commands and parameters

Such a coding structure is currently used for the control procedure for the teletex service (CCITT Rec. S.62), and is now under study at the ISO and the CCITT as a general coding method for session protocol for teletex, telecopy and videotex. The coding principles are illustrated in Fig. 1.

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 is 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 are 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 indicates 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 is 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*
- digital replacement teletext
- digital replacement sound

for the radio sound services: - digital radio sound
- digital radio additional sound
- digital news flash sound*
- 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 Part 6).

Table 1 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 receiver's 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 2.

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* A given service cannot have more than one component of this type.

Updating messages introduced by the UPDAT parameter identifier are 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 indicates 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 are 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.

3.2 Data channel

Packet address '0' is 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 contains 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 contains 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 is Hamming coded (see Tables 2 and 3).

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 indicated by the packet type byte (PT) in the corresponding packet header (see Tables 2 and 3).

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 applies 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 is 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 refers 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 is applied to the lengths of data groups. In the subframes characterized by TDMCID values '01 and '02 and when in line 625, bit 2 (b_m) 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 Section 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 2) or Golay encoding of the data.

The value of the continuity index I is 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 is reset to an initial value, i.e. 00.

The packet suffix has a length of 2 bytes and is a CRC generated by the polynomial:

$$x^{16} + x^{12} + x^{5} + 1$$

The message to be sent is 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 is 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 2).

When the identical information is repeated several times in consecutive data groups of the same type, the data group repetition byte is set and decremented at each transmission until it is zero at the last. When the repetition continues indefinitely, the repetition byte is 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 is incremented by one (modulo 16), each time a data group of the same type (TG) is transmitted.

When used, (24,12) Golay encoding is 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 is 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 2 and Table 3). The data group suffix is 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 is permanently 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 2 and Table 3, which are arranged in accordance with the layer architecture. All the bytes of the data group header are Hamming protected according to the Hamming code table given in Table 4 of Part 3.

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.

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 must 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.

For the D-MAC/packet system, in the case of the related subframe configuration as described in Part 1 (TDMCID codes 01 to 0E), the foregoing considerations apply within each subframe. In addition, the following rules apply:

- data group '0' shall exist in each subframe in identical form

- data groups other than '0' shall exist in a particular subframe such that any digital component which occupies capacity in that subframe is described

- where a service has components which occupy capacity in both subframes, the data groups associated with this service must exist in each subframe.

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.

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 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 must always 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 must always 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 system 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 1

Coding of commands and parameters identifiers (CI, GPI, PI)

PARAMETER NAME and (MNEMONIC)	PI \ CI	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
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 1 (continued)

PARAMETER NAME and (MNEMONIC)	PI \ CI	Continuation Command '00	Network Command Priority Medium '10	Network Command Priority Low '11	TV Command Priority Medium '90	TV Command Priority Low '91	Radio sound Command Priority Medium 'A0	Radio sound Command Priority Low 'A1	Teletext Command Priority Medium 'B0	Teletext Command Priority Low 'B1	Over-air Addressing Comm'd Priority Medium 'C0	Over-air Addressing Comm'd Priority Low 'C1	Additional SI Command Priority Medium 'D0	Additional SI Command Priority Low 'D1	Extension Command '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 1 (continued)

PARAMETER NAME and (MNEMONIC) — PI / CI		Contin-uation Command	Network Command		TV Command		Radio sound Command		Teletext Command		Over-air Addressing Comm'd		Additional SI Command		Exten-sion Command
			Priority		Priority		Priority		Priority		Priority		Priority		
			Medium	Low	Medium	Low	Medium	Low	Medium	Low	Medium	Low	Medium	Low	
		'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				*										

Notes:
1. Only index values must be repeated here.
2. High priority limited length (32 byte) network commentary.
3. To be confirmed by further study.
4. In case of use of the Eurocrypt® conditional-access system, these codes are reserved for the conditional-access assistance teletext (see reference in Part 6, Section 6).

Part 5

Table 2

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	service-identification data 1	/ CI 2	LI	service-identification data 2	/ CI 3	LI	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 ":".

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* 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 3

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 1	LI	service-identification data 1	/	CI 2	LI	service-identification data 2	/	CI 3	LI	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)	DGS: data group suffix A total of 8 bytes, each in Hamming code (see Table 4 of Part 3) before Golay encoding.

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)*

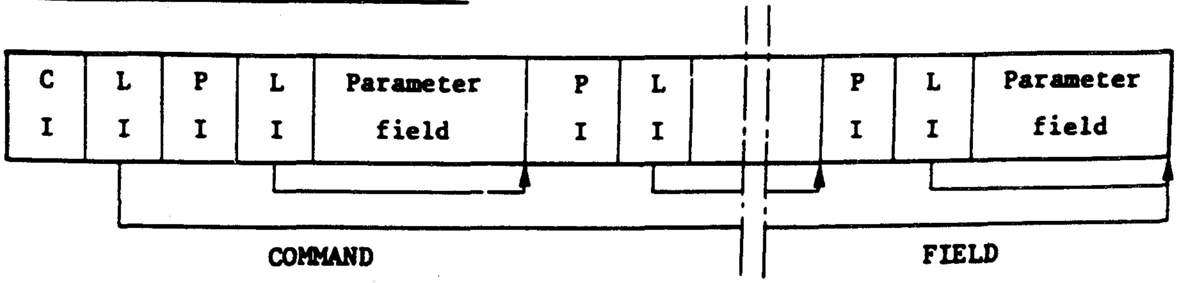
The extremities of a command are shown by "/".
The extremities of a sequence of command data (data group) are shown by "+".
The extremities of a data group (useful part of data blocks) are shown by ":".

---

* 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)

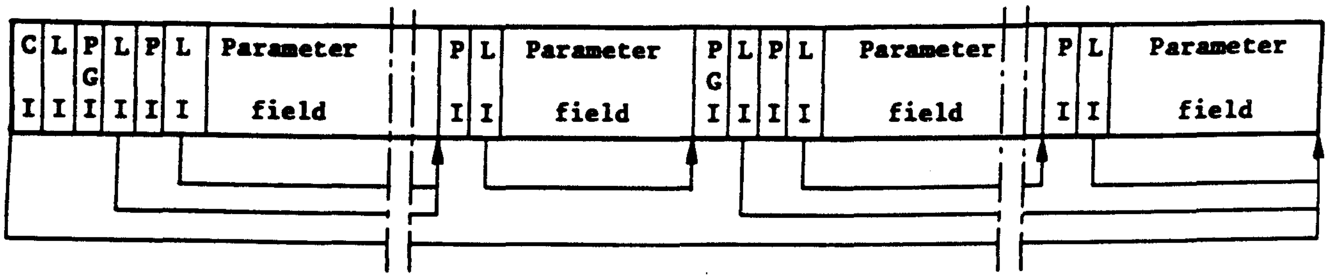
Part 5

a) Without PG (parameter group)



- within the command field, the PI codes are transmitted in an increasing order*.

b) With PG (parameter group)

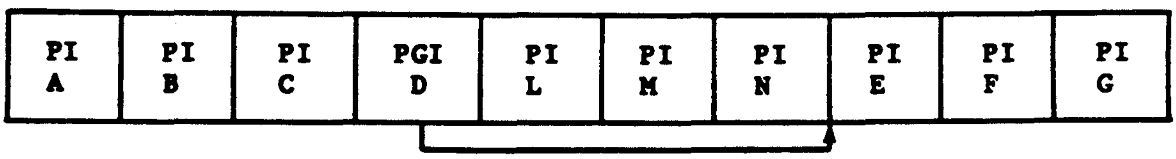


- within the command field, parameter groups can be mixed with isolated parameters

- 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 in general



Notes

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 PIs ascend from A to G and independently from L to N i.e.:

   $A \leqslant B \leqslant C \leqslant D \leqslant E \leqslant F \leqslant G$

   and $L \leqslant M \leqslant N$
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

---

* Value greater than or equal to the previous one.

Fig. 1: Coding structure of a command

Annex 1 to Part 5

CODE TABLES

1. Purpose of the present Annex

A number of special codes are defined in Part 5 of the specification, with reference to code tables or coding methods. The corresponding details are included in the present Annex.

2. CHID codes

The reference list of allocated codes is found in ETR XXX (JTC work programme reference DTR/JTC-001).

3. Coding of characters used for various text messages transmitted in the dedicated channel (see Section 2 of Part 5)

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 should be that given in Appendix 2 to EBU doc. Tech. 3232 (2nd edition, 1982). 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.

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* has been adopted directly for use in the service identification system. They are reproduced here as Figs. 1, 2 and 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.

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* EBU document Tech. 3244, Appendix 5.

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 the EBU Radio data system specification, the applicable coding table is determined by the non-spacing control characters as follows:

0/15, 0/15: selection of code table of Fig. 1
0/14, 0/14: selection of code table of Fig. 2
1/11, 6/14: selection of code table of Fig. 3.

In default of a control character, the code table shown in Fig. 1 here is applicable. Receivers must 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 the EBU Radio data system specification.

3.4 Display of clear text messages

The maximum length of any clear text message is 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.

4. Coding of programme type (see Section 2.2 of Part 5)

4.1 Introduction

In Section 2.2 of Part 5, 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.

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 the specification of the Radio data system 'RDS' for VHF/FM sound broadcasting (doc. Tech. 3244). The EBU Statistics Group has produced an elaborate multi-dimensional classification system known as 'ESCORT' (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".

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.

Code table for programme type

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 Section 2.2 of Part 5 and hexadecimal code 3F has been assigned to identify alarm/emergency messages.

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 " "

Code (Hexadecimal)	Programme type	ESCORT ref. number
	INTENDED AUDIENCE	
08	General Audience	2.0.0
	Special Groups:	
10	Ethnic & 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

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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 & 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 & financial affairs	1.3.0
44	Housing, environment & health affairs	1.4.0
45	Communication affairs	1.5.0
46	Educational and cultural affairs	1.6.0
47	International relations & defence affairs	1.7.0
48	Science & 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	
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-relig us philosophy of life	4.3.0
67	Other phi sophies 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 & equestrian sports	5.5.0
6E	Athletics	5.6.0
6F	Martial arts	5.7.0

70	<u>Leisure and hobbies</u>	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	<u>Light entertainment, folklore and human interest</u>	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.

5. Coding of languages (see Section 2.3 of <u>Part 5</u>)

5.1 Introduction

In Section 2.3 of <u>Part 5</u>, 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.

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 doc. Tech. 3232 (second edition, 1982), "Displayable character sets for broadcast teletext" 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.

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 <u>in the languages themselves</u>. 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 Section 2.3 of <u>Part 5</u>, code 00 indicates 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.

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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	)
17	Latin	37	Reserved for
18	Latvian	38	national assignment
19	Luxembourgian	39	)
1A	Lithuanian	3A	)
1B	Hungarian	3B	)
1C	Maltese	3C	)
1D	Dutch	3D	)
1E	Norwegian	3E	)
1F	Occitan	3F	)

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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

Annex 1 to Part 5

8-BIT CODING TABLES OF DISPLAYABLE CHARACTERS FOR THE SERVICE-IDENTIFICATION SYSTEM

The complete EBU Latin-based repertoire

Additional displayable characters for:

					Displayable characters from the code table of ISO Norm 646:						EBU common-core (7 languages)		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	Ǧ	±	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	[(1)	k	{ (1)	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	] (1)	m	} (1)	β	ǧ	↑	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	↓	§	Ŀ	ŀ	ð	

Fig. 1: Code table for 218 displayable characters forming the complete EBU Latin-based repertoire. The characters shown in positions marked (1) 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 doc. Tech. 3232 (2nd edition, 1982).

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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	.	0	2	"	2	B	R	b	r	é	ê	©	²	Б	ď	6	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	Ђ	Џ	η	Ξ
1	0	0	1	9	)	9	I	Y	i	y	ù	ü	Ľ	¿	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)	ß	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	

Fig. 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 (1) 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 doc. Tech. 3232 (2nd edition, 1982).

Annex 1 to Part 5

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	ـبـ	ظ	א	נ	Є	Ў	Π	π
0	0	0	1	1	!	1	A	Q	a	q	تـ	عـ	ב	ס	Я	Љ	α	Ω
0	0	1	0	2	"	2	B	R	b	r	ة	غـ	ג	ע	Б	d′	β	ρ
0	0	1	1	3	#	3	C	S	c	s	ثـ	فـ	ד	ף	Ч	Ш	ψ	σ
0	1	0	0	4	¤	4	D	T	d	t	جـ	قـ	ה	פ	Д	Ц	δ	τ
0	1	0	1	5	%	5	E	U	e	u	حـ	كـ	ו	ץ	Э	Ю	ε	ξ
0	1	1	0	6	&	6	F	V	f	v	خـ	لـ	ז	צ	Ф	Щ	φ	Θ
0	1	1	1	7	'	7	G	W	g	w	د	ـهـ	ח	ק	Ѓ	Њ	γ	Γ
1	0	0	0	8	(	8	H	X	h	x	ذ	نـ	ט	ר	Ъ	Џ	η	Ξ
1	0	0	1	9	)	9	I	Y	i	y	ر	هـ	י	ש	И	Й	ι	υ
1	0	1	0	10	*	:	J	Z	j	z	ز	و	ך	ת	Ж	З	Σ	ζ
1	0	1	1	11	+	;	K	[(1)	k	{ (1)	سـ	يـ	כ	°	Ќ	č	κ	ς
1	1	0	0	12	,	<	L	\	l	|	شـ	←	ל	¼	Л	š	λ	Λ
1	1	0	1	13	-	=	M	] (1)	m	} (1)	صـ	↑	ם	½	Ћ	ž	μ	Ψ
1	1	1	0	14	.	>	N	—	n	~	ضـ	→	מ	¾	Ђ	đ	ν	Δ
1	1	1	1	15	/	?	O	_	o	\	ط	↓	ן	§	Ы	ć	ω	

<u>Fig. 3</u>: 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 (1) 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 doc. Tech. 3232 (2nd edition, 1982)

Annex 2 to Part 5

CODING EXAMPLE FOR INFORMATION WITHIN THE DEDICATED CHANNEL OF THE SOUND/DATA MULTIPLEX

1. General

In order to illustrate the specifications given in Part 5, 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 Fig. 1.

It is assumed that a D2-MAC/packet signal can be obtained by transcoding of the D-MAC/packet signal.

The channel carries the following services (see also Fig. 2):

- in the first subframe of the D-MAC/packet signal (TDMCID 01), which is the subframe handed on to form the D2-MAC/packet signal: one television service with one original sound component (companded, first level protected, stereo), one additional sound component (half bandwidth, companded, first level protected, mono), and an entitlement checking message component
- in the second subframe of the D-MAC/packet signal (TDMCID 02), which is discarded in transcoding to the D2-MAC/packet signal: one radio service (linear, first level protected, stereo)
- in both subframes simultaneously (but capable of being recovered from only one subframe): one teletext service.

A subtitle component carried by fixed-format teletext in the field-blanking interval is described in the channel. In addition, a new radio service (linear, first level protected, stereo using capacity in both subframes of the D-MAC/packet signal and therefore not capable of being recovered from the D2-MAC/packet signal) is commencing. This service is described in the D2-MAC/packet signal but the digital component location code '00 in LISTX indicates that it cannot be recovered and therefore its service packets should be treated by the D2-MAC/packet decoder as dummy data.

Within the dedicated channel of the D-MAC/packet signal, data groups are placed in the two subframes such as to obey the rules of Section 4 of Part 5 without unnecessary duplication (see also Figs. 3a and 3b).

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 Section 3 of Annex 1 to Part 5. The notations MPX 01 and MPX 02 are used to describe the subframes corresponding with TDMCID 01 and TDMCID 02 respectively. Thus only the subframe denoted by MPX 01 is available in the D2-MAC/packet signal. (It must be noted that the dedicated channel in MPX 01 carries some information concerning services which existed only in MPX 02.)

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. Section 3 illustrates the coding of commands in their entirety.

Annex 2 to Part 5

2.1 Network identification and description in terms of services

2.1.1 Medium priority (M)

Data group 0 in MPX 01 (and MPX 02 in the D-MAC/packet system).

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 01 alone, packet address 341 (0011, 01, 01 0101 0101)	'01 '02 '3555
LIST X	'18	- for radio services - list of indices: i) index value 1, data group 4, in MPX 02 alone, packet address 343 (0100, 10, 01 0101 0111) ii) index value 2, data group 4, in MPX 01 and MPX 02, packet address 344 (0100, 00, 01 0101 1000)	'02 '01 '4957 '02 '4158
LIST X	'18	- for teletext service - list of indices: index value 1, data group 6, in MPX 01 and MPX 02, 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 01 and MPX 02, 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 01 and MPX 02, packet address 0 (1001, 11, 00 0000 0000)	'0F '9C00

2.1.2 Low priority (L)

Data group 9 in MPX 01 (and MPX 02 in the D-MAC/packet system).

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	 'OF "NOUVEAU : EUROJAZZ COMMENCE BIENTOT"

2.2 TV service description

2.2.1 Medium priority (M)

Data group 3 in MPX 01 alone.

Parameter name	PI	Details	Parameter field coding
SREF	'40	- service reference - index value 2 - name of service in clear text	 '02 "TF1"
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 01 alone, packet address 347 (0000, 01, 01 0101 1011) - page number 123 (0000 0001 0010 0011)	 'OF '055B '0123
ACCM	'88	- access related message - entitlement checking messages exist, CAFCNT not used for key generation, in MPX 01 alone, packet address 346 (10, 00, 01, 01 0101 1010)	 '855A
VCONF	'90	- video configuration - controlled access, double-cut component rotation scrambling, AR = 4:3, Cy = 3:2, Cu = 3:1	 '24

Parameter name	PI	Details	Parameter field coding
DCINF	'A4	- (television) original sound component - language French - in MPX 01 alone, packet address 341 (0000, 01, 01 0101 0101)	 '0F '0555
DCINF	'A5	- (television) additional sound component - language English - in MPX 01 alone, packet address 342 (0000, 01, 01 0101 0110)	 '09 '0556
DCINF	'F8	- (television) subtitles carried by fixed-format teletext in the field-blanking interval - language German - magazine number 8, carried by even samples, page number 88 (0, 000, 0000, 1000 1000)	 '08 '0088

2.3 Radio sound service description

2.3.1 Medium priority (M)

Data group 4 in MPX 01 (and MPX 02 in the D-MAC/packet system).

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 01 and MPX 02, packet address 344 (0000, 00*, 01 0101 1000)	 '00 '0158

* Indicates that this component cannot be recovered in the D2-MAC/packet system.

Annex 2 to Part 5

2.3.2 Medium priority (M)

Data group 4 in MPX 02 (and therefore applicable to D-MAC/packet system only).

Parameter name	PI	Details	Parameter field coding
SREF	'40	- service reference - index value 1 - service name in clear text	 '01 "FRANCE RADIO 1"
PREF	'48	- programme reference - type: serious music - number (from scheduled broadcast start time) (1983 May 29, 23:00 French time) - name (not given)	 '51 '83 '21 '00 -
COMD	'61	- pointer to commentary (in the dedicated channel in this example) - language French - in data group 9, MPX 02 alone, packet address 0 (1001, 10, 00 0000 0000)	 'OF '9800
DCINF	'A8	- (radio) original sound component - language French - in MPX 02 alone, packet address 343 (0000, 10, 01 0101 0111)	 'OF '0957

2.3.3 Low priority (L)

Data group 9 in MPX 02 (and therefore applicable to D-MAC/packet system only).

Parameter name	PI	Details	Parameter field coding
SREF	'40	- service reference - index value 1	 '01
DCOM	'60	- direct commentary - language French - clear text	 'OF "DEBUSSY : LA MER"

2.4 Teletext service description

2.4.1 Medium priority (M)

Data group 6 in MPX 01 (and MPX 02 in the D-MAC/packet system).

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 01 and MPX 02, packet address 345, page 1, row 1 (0000, 11*, 01 0101 1001) (0000, 0000, 0000, 0001) (0000, 0001)	 'OF 'OD59 '0001 '01
DCINF	'BO	- cyclic teletext component - language French - in MPX 01 and MPX 02, packet address 345 (0000, 11*, 01, 0101 1001) - teletext data channel 0	 'OF 'OD59 '0000

* Indicates that this component can be recovered from MPX 01 alone in the D2-MAC/packet system.

3. Coding structure of commands and parameters

3.1 Network identification and description in terms of services

3.1.1 Medium priority (M)

Data group 0 in MPX 01 (and MPX 02 in the D-MAC/packet system), total length 53 bytes.

CI	LI	PI	LI	Parameter field
'10	'33			
		'00	'01	'18
		'10	'09	'17 '81 '91 "FRANCE"
		'14	'04	"TDF1"
		'18	'04	'01 '02 '3555
		'18	'07	'02 '01 '4957
				'02 '4158
		'18	'04	'03 '01 '6D59
		'61	'03	'OF '9C00
		'62	'03	'OF '9C00

3.1.2 Low priority (L)

Data group 9 in MPX 01 (and MPX 02 in the D-MAC/packet system), total length 63 bytes.

CI	LI	PI	LI	Parameter field
'11	'3D			
		'20	'14	"LUNDI 83 05 30 00:59"
		'60	'25	'OF "NOUVEAU : EUROJAZZ COMMENCE BIENTOT!"

3.2 TV service description

3.2.1 Medium priority (M)

Data group 3 in MPX 01 alone, total length 51 bytes.

CI	LI	PI	LI	Parameter field
'90	'31			
		'40	'04	'02 "TF1"
		'48	'0C	'40 '83 '22 '15 "MINUIT 1"
		'61	'05	'0F '055B '0123
		'88	'02	'855A
		'90	'01	'24
		'A4	'03	'0F '0555
		'A5	'03	'09 '0556
		'F8	'03	'08 '0088

3.3 Radio service description

3.3.1 Medium priority (M)

Data group 4 in MPX 01, 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 '0158

3.3.2 Medium priority (in D-MAC/packet system only)

Data group 4 in MPX 02, total length 63 bytes.

CI	LI	PGI	LI	PI	LI	Parameter field
'A0	'3D					
		'80	'21			
				'40	'OF	'01 "FRANCE RADIO 1"
				'48	'04	'51 '83 '21 '00
				'61	'03	'OF '9800
				'A8	'03	'OF '0957
		'80	'18			
				'00	'00	
				'40	'09	'02 "EUROJAZZ"
				'48	'04	'54 '83 '23 '00
				'A8	'03	'00 '0158

3.3.3 Low priority (in D-MAC/packet system only)

Data group 9 in MPX 02 only, total length 24 bytes.

CI	LI	PI	LI	Parameter field
'A1	'16			
		'40	'01	'01
		'60	'11	'OF "DEBUSSY : LA MER"

Multiplex 02

Medium priority:	data group 0	(network)	1 packet
	data group 4	(radio)	1 packet
	data group 6	(teletext)	1 packet
		Medium priority total:	3 packets.
Low priority:	data group 9	(network & radio)	2 packets
		Low priority total:	2 packets.

Assuming again 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 examples becomes:

MPX 01	$(4 \times 4) + 1 =$	17 packets/second
MPX 02	$(3 \times 4) + 2 =$	14 packets/second
		31 packets/second

3.4 Teletext service description

3.4.1 Medium priority (M)

Data group 6 in MPX 01 (and MPX 02 in the D-MAC/packet system), total length 34 bytes.

CI	LI	PI	LI	Parameter field
'BO	'20			
		'40	'0C	'01 "FRANCE SOIR"
		'48	'01	'40
		'61	'06	'OF 'OD59 '0001 '01
		'BO	'05	'OF 'OD59 '0000

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 Fig. 4. The SI dedicated channel capacity requirement is as detailed below:

Multiplex 01

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.

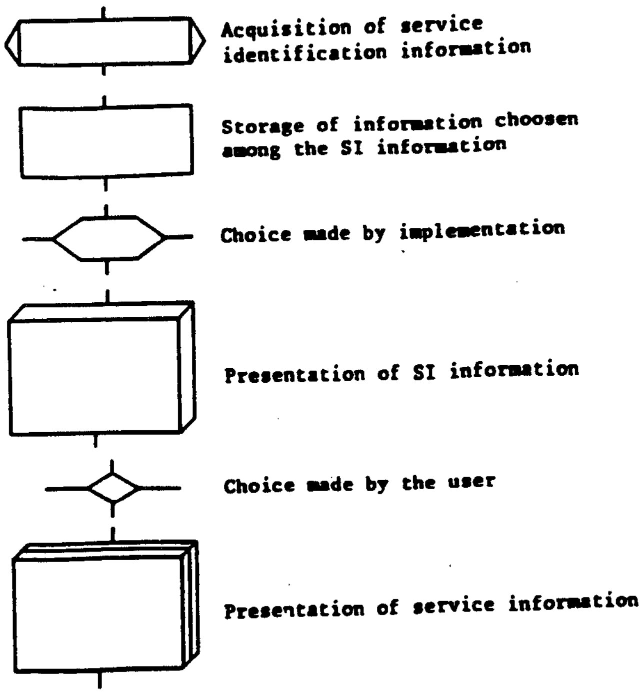
Assuming that the mean rate of repetition of medium priority information is four times per second and that of low priority information is one per second, the mean capacity utilised by the SI examples becomes:

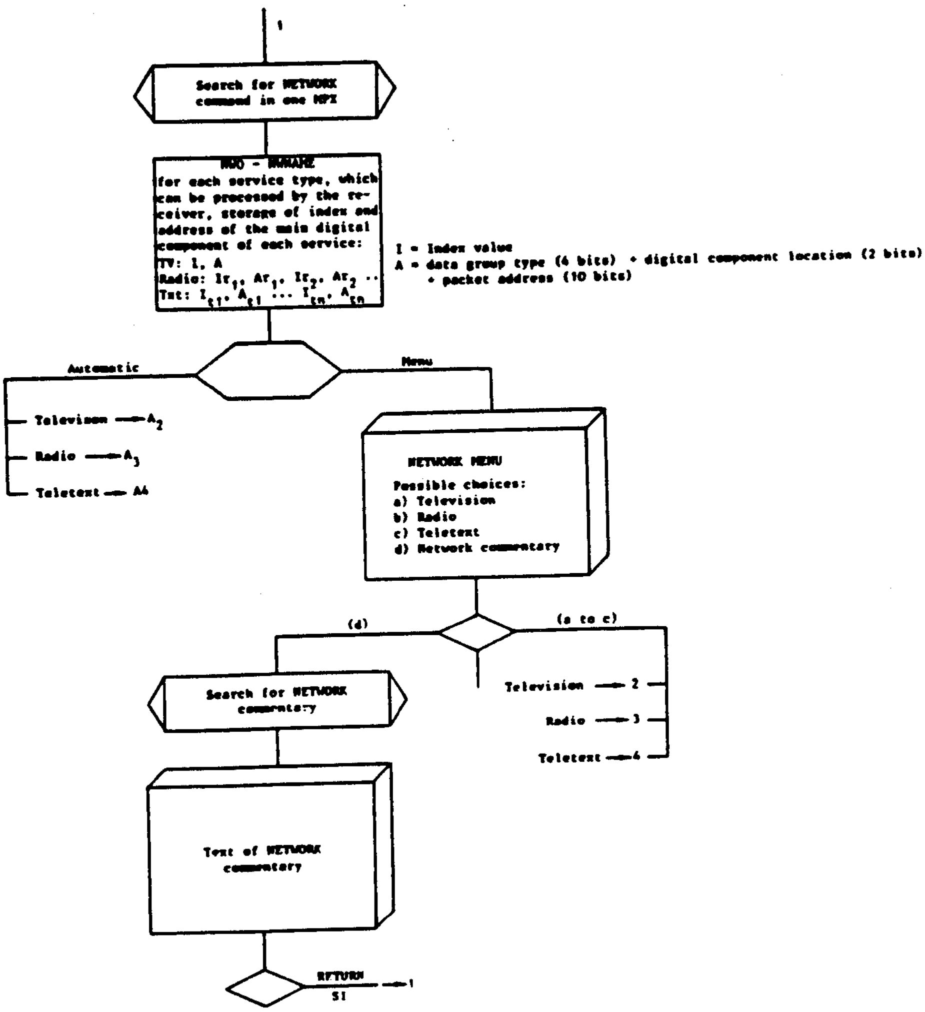
$$(4 \times 4) + 1 = \underline{17 \text{ packets/second}}$$

Annex 2 to Part 5

Fig. 1: Functional description of the access procedure

LEGEND





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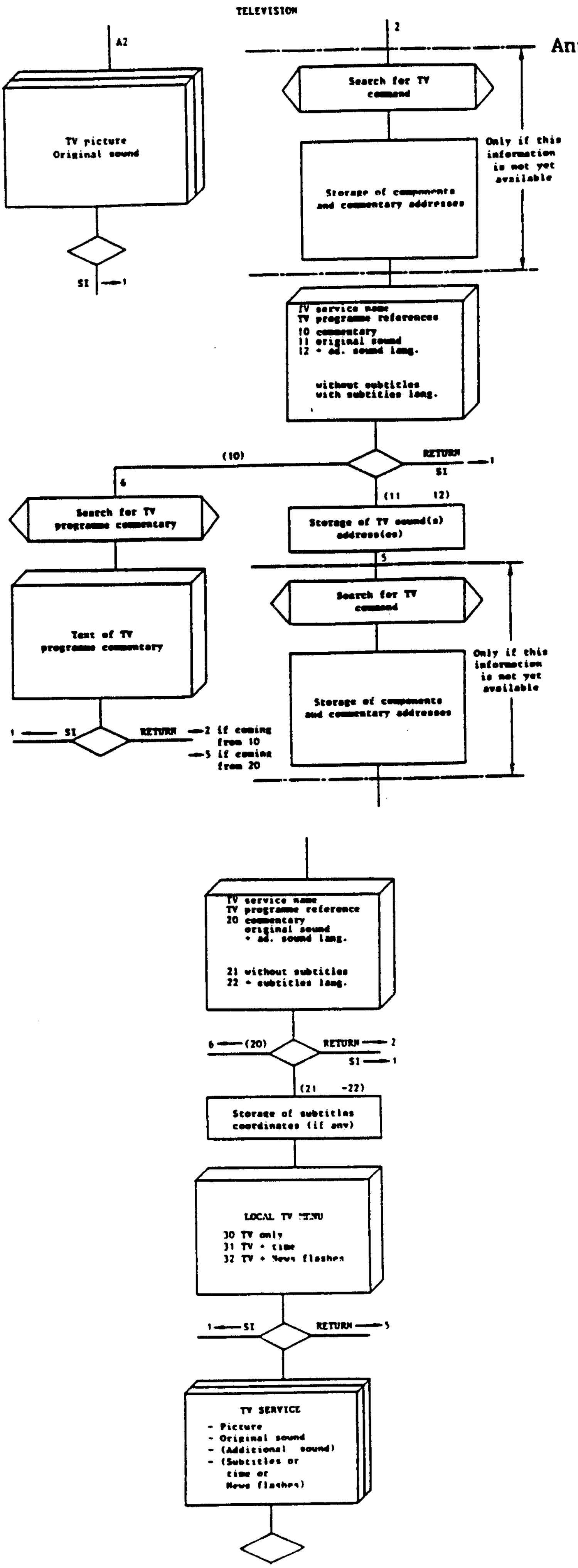

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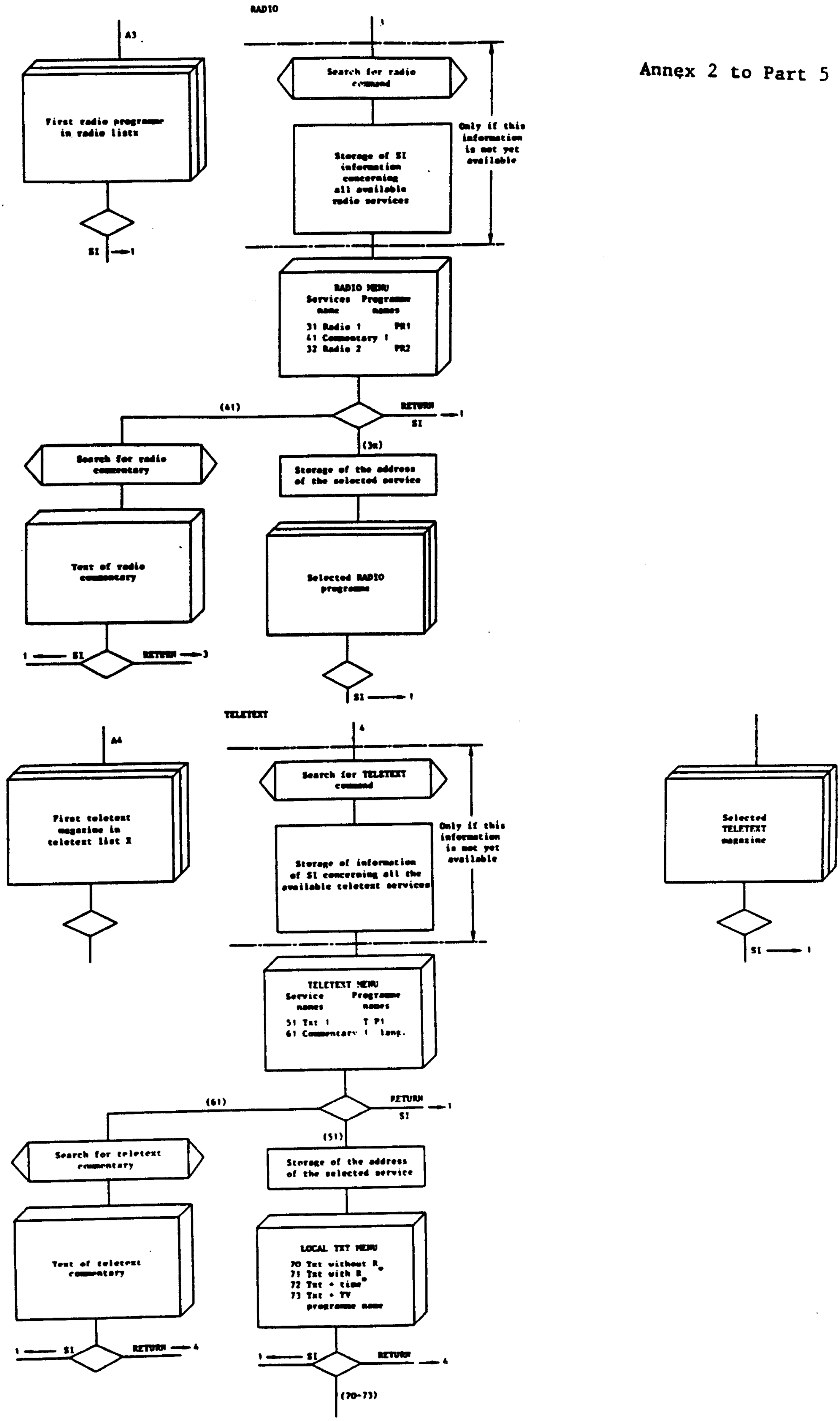
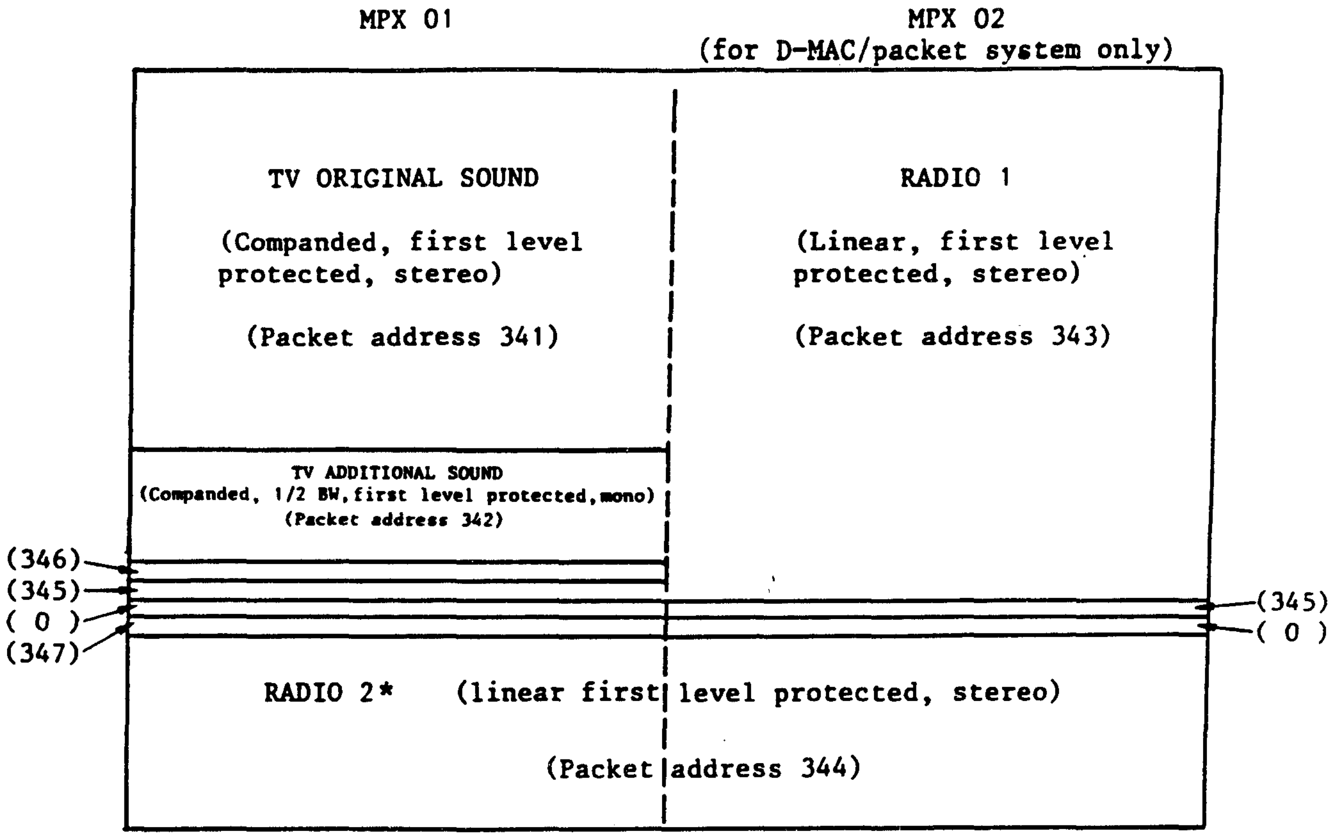


Annex 2 to Part 5



(346): ENTITLEMENT CHECKING MESSAGES /
(Packet address 346)

(345): TELETEXT /
(Packet address 345)

(0): SERVICE IDENTIFICATION / (345): TELETEXT
(Packet address 0) (Packet address 345)

(347): TV INDIRECT COMMENTARY / (0): SERVICE IDENTIFICATION
(Packet address 347) (Packet address 0)

Fig. 2: Capacity map for the SI coding example

* The Radio 2 service cannot be recovered by a D2-MAC/packet receiver, and its packets must be considered as dummy information.

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DATA GROUP 0

- Television service, D G 3
- Radio 1 service, D G 4
- Radio 2 service, D G 4
- Teletext service, D G 6
- Network commentary, D G 9

DATA GROUP 3

- TV main sound, PA 3 4 1
- TV addit. sound, PA 3 4 2
- TV ECM, PA 3 4 6
- TV commentary, PA 3 4 7
- TV subtitles,* -

DATA GROUP 4

- Radio 2,** PA 3 4 4

DATA GROUP 6

- Teletext, PA 3 4 5
- Teletext indirect commentary, PA 3 4 5

DATA GROUP 9

- Network commentary

Note:

DG x stands for data group x of the SI dedicated channel, PA xxx for packet address xxx of MPX 01.

Fig. 3a: Example data group map for the SI dedicated channel in MPX 01

* Through teletext in the field-blanking interval.
** Cannot be recovered by D2-MAC/packet receivers.

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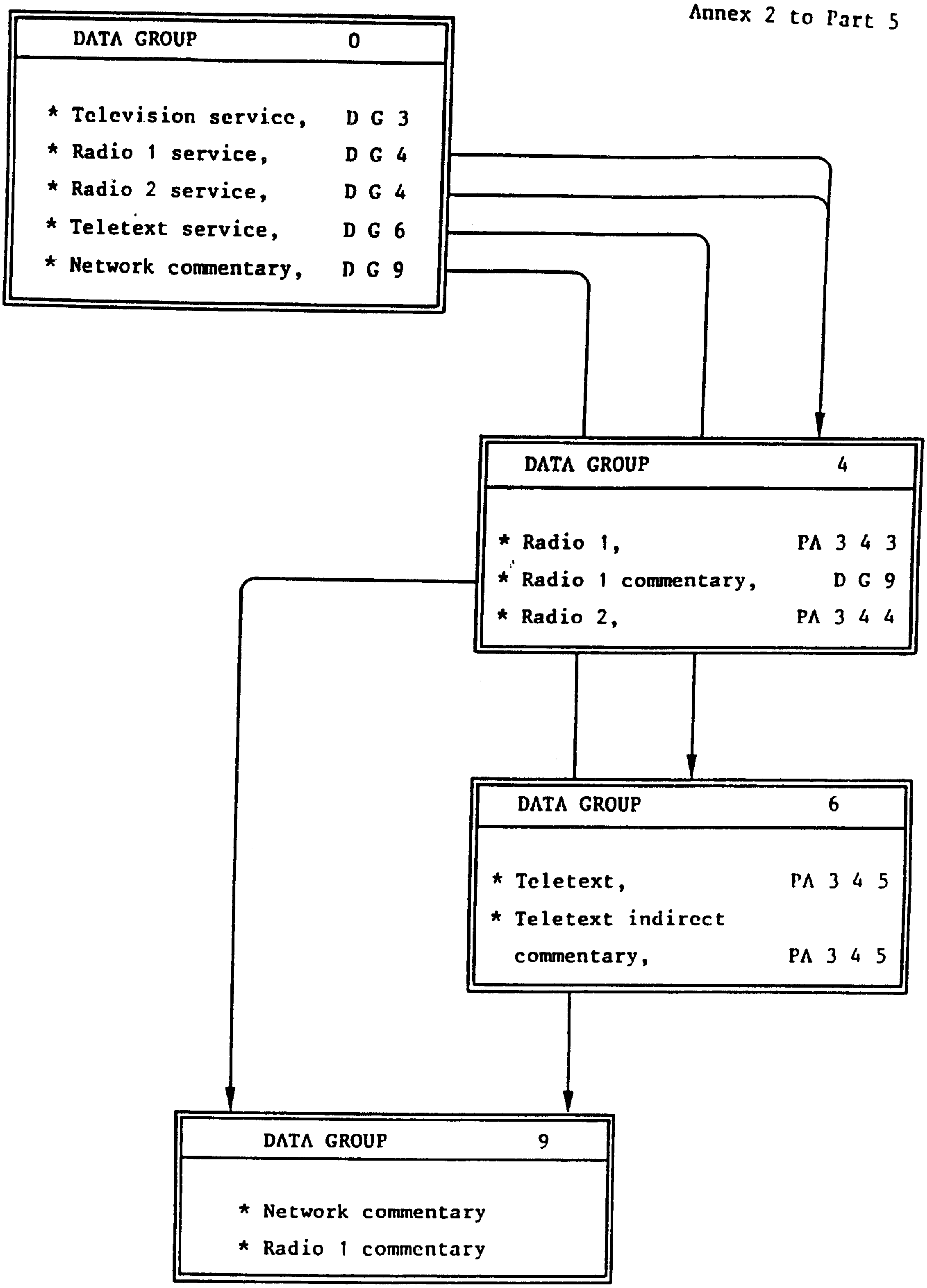


Note: DG x stands for data group x of the SI dedicated channel, PA xxx for packet address xxx of MPX 02.

Fig. 3b: Example data group map for the SI dedicated channel in MPX 02, for the D-MAC/packet system only

Annex 2 to Part 5

PT('F8)

PH	DGH	NETWORK MEDIUM PRIORITY	DGS	UNUSED (25 BYTES)	PS

DATA GROUP 0 IN MPX 01 (and MPX 02 in the D-MAC/packet system)

PT('F8)

PH	DGH	TV MEDIUM PRIORITY	DGS	UNUSED (27 BYTES)	PS

DATA GROUP 3 IN MPX 01

PT('F8)

PH	DGH	RADIO MEDIUM PRIORITY	DGS	UNUSED (52 BYTES)	PS

DATA GROUP 4 IN MPX 01

PT('F8)

PH	DGH	RADIO MEDIUM PRIORITY	DGS	UNUSED (15 BYTES)	PS

DATA GROUP 4 IN MPX 02 (in the D-MAC/packet system only)

PT('F8)

PH	DGH	TELETEXT MEDIUM PRIORITY	DGS	UNUSED (44 BYTES)	PS

DATA GROUP 6 IN MPX 01 (and MPX 02 in the D-MAC/packet system)

PT('F8)

PH	DGH	NETWORK LOW PRIORITY	DGS	UNUSED (15 BYTES)	PS

DATA GROUP 9 IN MPX 01

PT('F8)

PH	DGH	NETWORK LOW PRIORITY	RADIO LOW PRIORITY	PS

DATA GROUP 9 IN MPX 02 (in the D-MAC/packet system only)

PT('C7)

PT	RADIO LOW P. cont.	DGS	UNUSED (79 BYTES)	PS

DATA GROUP 9 IN MPX 02 (Continued)

Fig. 4: SI dedicated channel coding example - Packets and contents

PART 6: SPECIFICATION OF THE CONDITIONAL-ACCESS SYSTEM

Contents

Part 6

1. Subject of Part 6

This Part of the specification 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. Section 2 of this Part is 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 <u>scrambling system</u> and the <u>encryption system</u>. 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. Section 3 is a description of the scrambling system. Section 4 is a description of a conceptual internal interface across which key and control signals pass in the receiver. Section 5 is 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 Sections 2.2 and 2.3 of <u>Part 5</u> is given in EN 50094 (EuroCrypt) and prEN 50071 (VideoGuard).

As far as the terminology used in this Part is concerned, the reader should note the following equivalences:

ECM (entitlement checking message) = SMM (service management message)
EMM (entitlement managing message) = CMM (customer management message).

2. General description of the conditional-access system

2.1 The scrambling and encryption systems

A block diagram of the overall conditional-access system is given in Fig. 1. Certain functions (such as demultiplexing) are omitted for clarity, and Fig. 1 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 Fig. 1 is an example. The source component in Fig. 1 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 user's 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 Fig. 1). 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 cryptogramme. The cryptogramme, associated with programme data (P), is encrypted by the service key (SK) (see Fig. 1). The control word is 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 cryptogramme SK(P,CW) is sent in an entitlement checking message (ECM), which is 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 is 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 is 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 is 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) are encrypted with each customer's distribution key (DK) or unique key (UK). These cryptogrammes are sent in entitlement management messages (EMMs).

The time required to cycle around the audience base is 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 are sent in unique entitlement management messages, which are 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 Fig. 1.

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 Part 5) 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 customer's 'signature' on selected frames of the vision signal.

2.2 Example of functional operation

The MAC channel may contain one or more services (e.g. television, radio, or teletext services) with service components (e.g. TV = video + 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.

3. Sound, data and picture scrambling

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 is always used for the controlled-access mode, but is optional 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 Section 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 Section 6.2 of Part 2.

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 Section 3.5 of Part 3.

3.2 Description of the sound and data scrambling process

To scramble sound and data, a pseudo-random binary sequence (PRBS) is added modulo 2 to the useful data of the sound or data service packets.

Note that:

- the sound encoding blocks are 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) are not scrambled.

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 Fig. 2. This would be the necessary configuration for the basic television service (vision and associated sound components); in full-channel digital mode of operation only the sound/data part of the configuration is relevant. 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/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 Fig. 2 shows three PRBS generators. These are described in Figs. 3, 4 and 5.

The PRBS system requires an 8-bit frame count, and the location of this information within line 625, and its specification, are given in Section 5.4 of Part 1 (code FCNT). As shown in Fig. 2, the initialisation 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 Section.

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 CW_1 and CW_2 (Fig. 2) to generate new values of IW_1 and IW_2, 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 F_1.

The following diagram summarises the relationships between the frame number, FCNT, CAFCNT, and the derivation of the initialisation word (IW).

---

* For the D system: the first packet in each of the two related subframes received during frame F...: in full-channel digital mode of operation: the and one set for the sound/data packets. One additional set with an independent control word is required for each additional sound/data service with independent scrambling.

	......Frame 255.....	.....Frame 256......	.....Frame 257.....
FCNT &	255	000	001
CAFCNT	0	1	1
in line 625			
IW derivation	(CWx & FCNT 254)	(CWx & FCNT 255)	(CWx+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.

4. The conditional-access interface

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 (Fig. 6) allows that

1. different operating/controlling organizations may control services within the same channel

2. different revenue collection systems may be used

3. three identified methods for delivery of the entitlement management message may be used; over-air addressing, post and user action, telecommunications networks.

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) initialisation (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 Section 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.

4.3 Basic functions of the CA interface

4.3.1 General diagram

$\boxed{A} \longrightarrow$ CA interface $\longleftarrow \boxed{B}$

A: main receiver circuits
B: access-control system (possibly including a detachable sub-system).

4.3.2 Rôle of the CA interface

Through the interface bus, the receiver first initialises 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.

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 Fig. 6, so as to allow the use of any form of access-control system.

This document 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 document.

4.3.4 Data crossing the CA interface

Table 1 shows the data which can be handled by the interface. A specific conditional-access system may use a sub-set of the messages listed.

4.3.5 Structure of information exchanged across the interface

The main traffic (illustrated in Table 1) is devoted to entitlement checking. Indeed, the main rôle of the access-control system (ACS) is to check entitlements presented by the user. The access-control system is also able to manage entitlements and to provide information on their status. A fuller description of the message types is given in Section 5.

a) Entitlement checking

Rx ⟶ ACS	initialisation (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

Rx ⟶ ACS	initialisation (e.g. ACMM parameter)
ACS ⟶ Rx	addresses for the demultiplexer (1)
Rx ⟶ ACS	contents of selected packets (2)
ACS ⟶ Rx	display data, configuration data, external data (e.g. to modem) (3)
Rx ⟶ ACS	key-pad entry (3).

(1) may also occur after (2)
(2) occurs whenever appropriate packets arrive in multiplex
(3) depends on the content of the EMM packets

c) Control (e.g. entitlement status, man/machine dialogue)

Rx ⟶ ACS	initialisation
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.

Part 6

Table 1

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
Initialisation	Initialisation
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)
Initialisation	CA-byte, plus optional parameters

5. Conditional-access data

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 Fig. 6).

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 Section 2.3 of Part 5).

For selection of keys and entitlements in the access-control system, a set of parameters is required from the SI (see Section 2 of Part 5) 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 Section 2.1 of Part 5). Four different types of EMMs <EMM-U (Unique), EMM-S (Shared), EMM-C (Collective) and EMM-G (General), see Section 5.2> which may contain a customer address, are specified. These addresses together with the ACMM parameter in the SI (see Section 2.2 of Part 5) 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.

5.2 Message formats

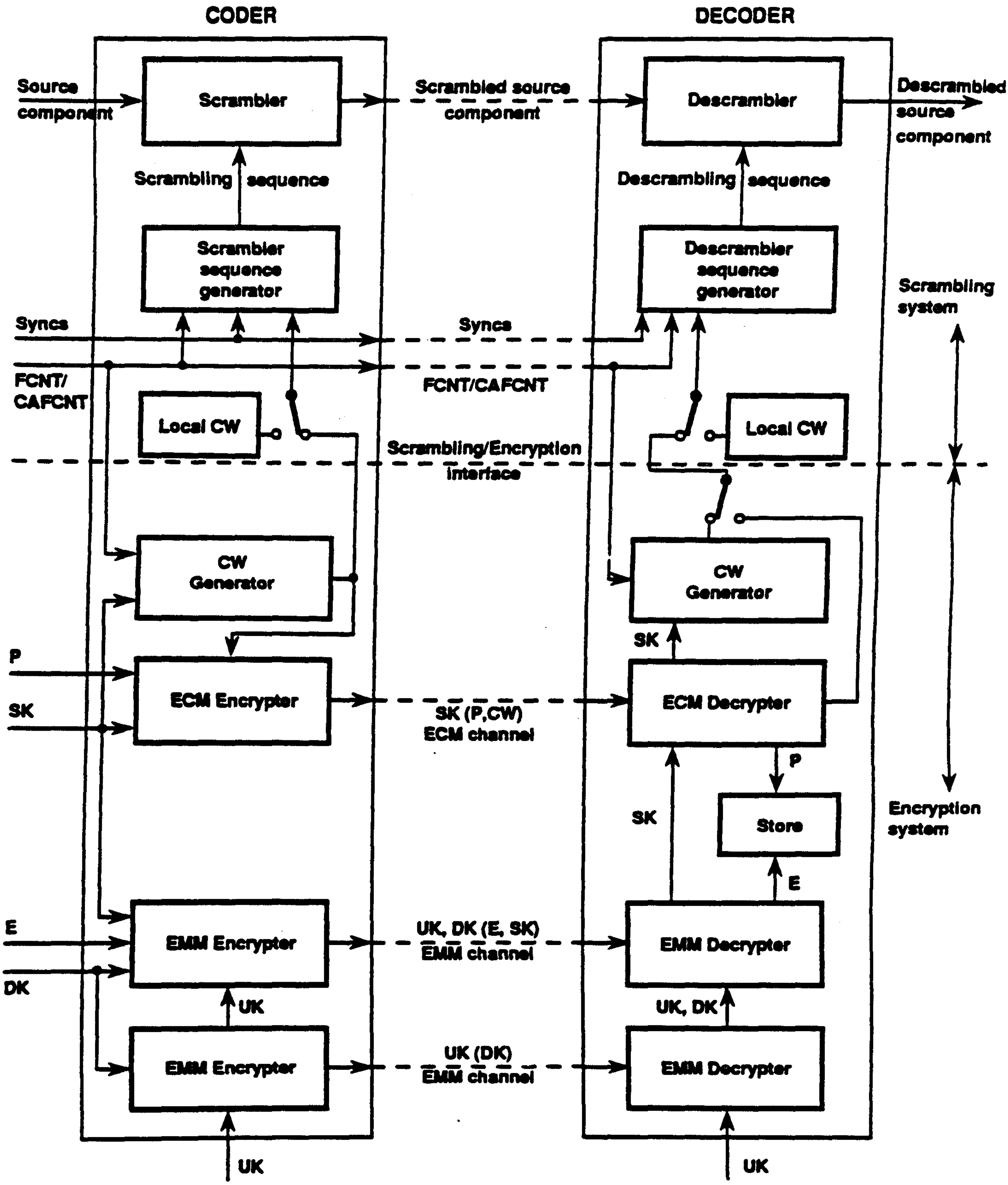
The five general packet formats which may be used for the entitlement management messages and the entitlement checking messages are shown in <u>Fig. 7</u>.

The first two fields, PH and PT, are common to all packets. 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 always has a zero continuity index (I=0), while subsequent packets have continuity index values in the range (<1> to <3>), the value of I being incremented by 1 in modulo 3 sequence each time a packet of the same type is transmitted. The PH field is protected by a (23,12) Golay code.

The type of message is indicated by the packet type (PT) parameter which is (8,2) Hamming protected. The values of the PT parameter defined for EMMs are as shown in <u>Fig. 7</u>.

After the PT parameter, all packets contain an error protected data field. Protection is 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 are formed by the (23,12) code specified by the packet headers, the 24th bit specifying an overall odd parity. The Golay protection is always 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

Fig. 1 : General structure of conditional-access system with an example of key hierarchy

Part 6

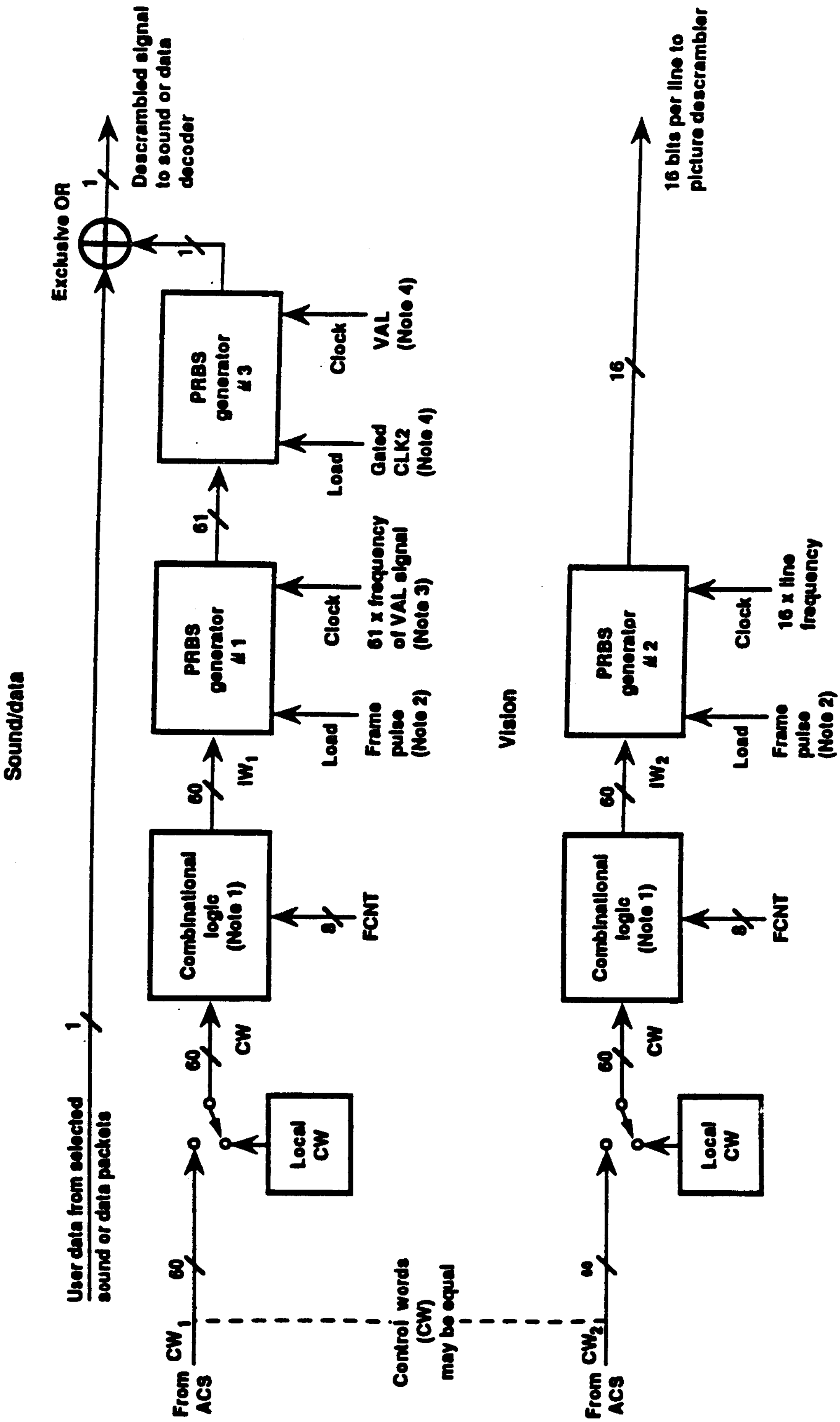


Fig. 2 : PRBS system for conditional-access system (ACS) (shown for vision and one sound/data channel)

Notes on Fig. 2

Note 1: The combinational logic functions, used to combine the two 60-bit control words (CW_1 and CW_2) with the 8-bit frame count (FCNT), to give the two 60-bit initialisation words (IW_1 and IW_2), are the same in both cases and are defined as follows:

The 8-bit frame count (FCNT) is written as F_0, F_1, ... F_7

The 60-bit control word (CW) is written as follows:

$$\underbrace{C_0C_1 \cdots C_7}_{CW0} \quad \underbrace{C_8C_9 \cdots C_{15}}_{CW1} \quad \underbrace{C_{48}C_{49} \cdots C_{55}}_{CW6} \quad \underbrace{C_{56}C_{57}C_{58}C_{59}}_{CW7}$$

Then the 60-bit initialisation word (IW) may be written as follows:

$$\underbrace{I_0I_1 \cdots I_7}_{IW0} \quad \underbrace{I_8I_9 \cdots I_{15}}_{IW1} \quad \underbrace{I_{48}I_{49} \cdots I_{55}}_{IW6} \quad \underbrace{I_{56}I_{57}I_{58}I_{59}}_{IW7}$$

IW is obtained from CW and FCNT as follows:

IW0 = CW0 O FCNT IW1 = CW1 O FCNT IW2 = CW2 O FCNT IW3 = CW3 O FCNT
IW4 = CW4 O FCNT IW5 = CW5 O FCNT IW6 = CW6 O FCNT IW7 = CW7 O FCNT

In addition, it should be noted that:

a) FCNT is the complement of FCNT (i.e. FCNT ⊕ FCNT = 11111111)

b) In the transformation from CW7 to IW7 only the least significant half byte of FCNT is used.

Note 2: The frame pulse occurs once per frame such that its leading edge (rising edge) marks the frame boundary.

Note 3: PRBS generator 1 is clocked 61 times between each VAL signal. The validation signal (VAL) is a clock signal which occurs at the packet rate.*

---

* For the D system: at the packet rate for the subframe being descrambled.

<u>Note 4</u>: The clock is 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.

<u>Note 5</u>: In the scrambled free-access mode, all the bits of the control word are set at logic '1'.

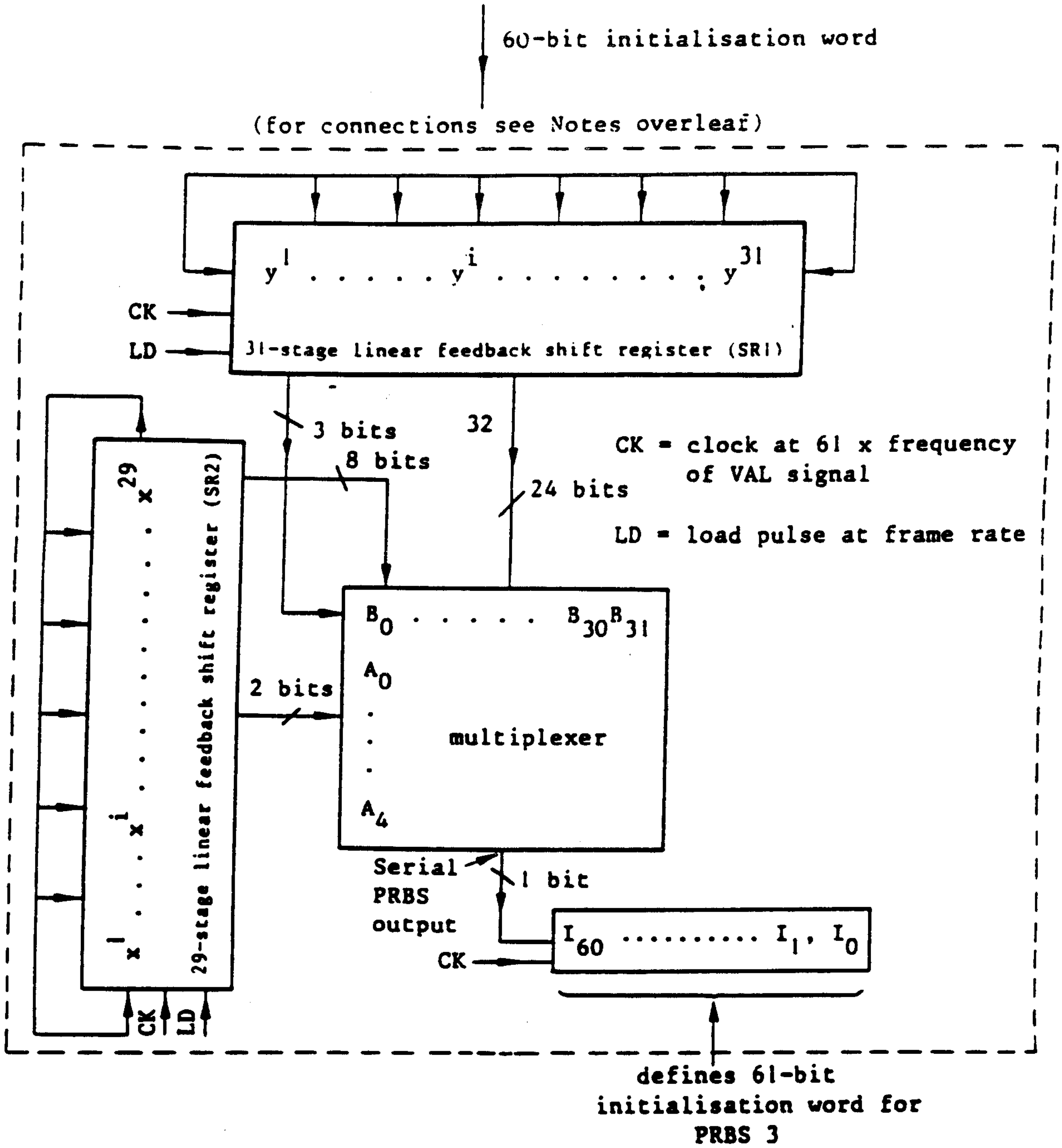


Fig. 3: PRBS generator 1 in Fig. 2

Notes on Fig. 3

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 initialisation word loading for SR1 and SR2 are defined as follows. The 60-bit initialisation 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 initialisation word for PRBS 3 at a rate of once per packet (i.e. VAL signal frequency) such that the least significant bit of the initialisation word (I_0) is output first.

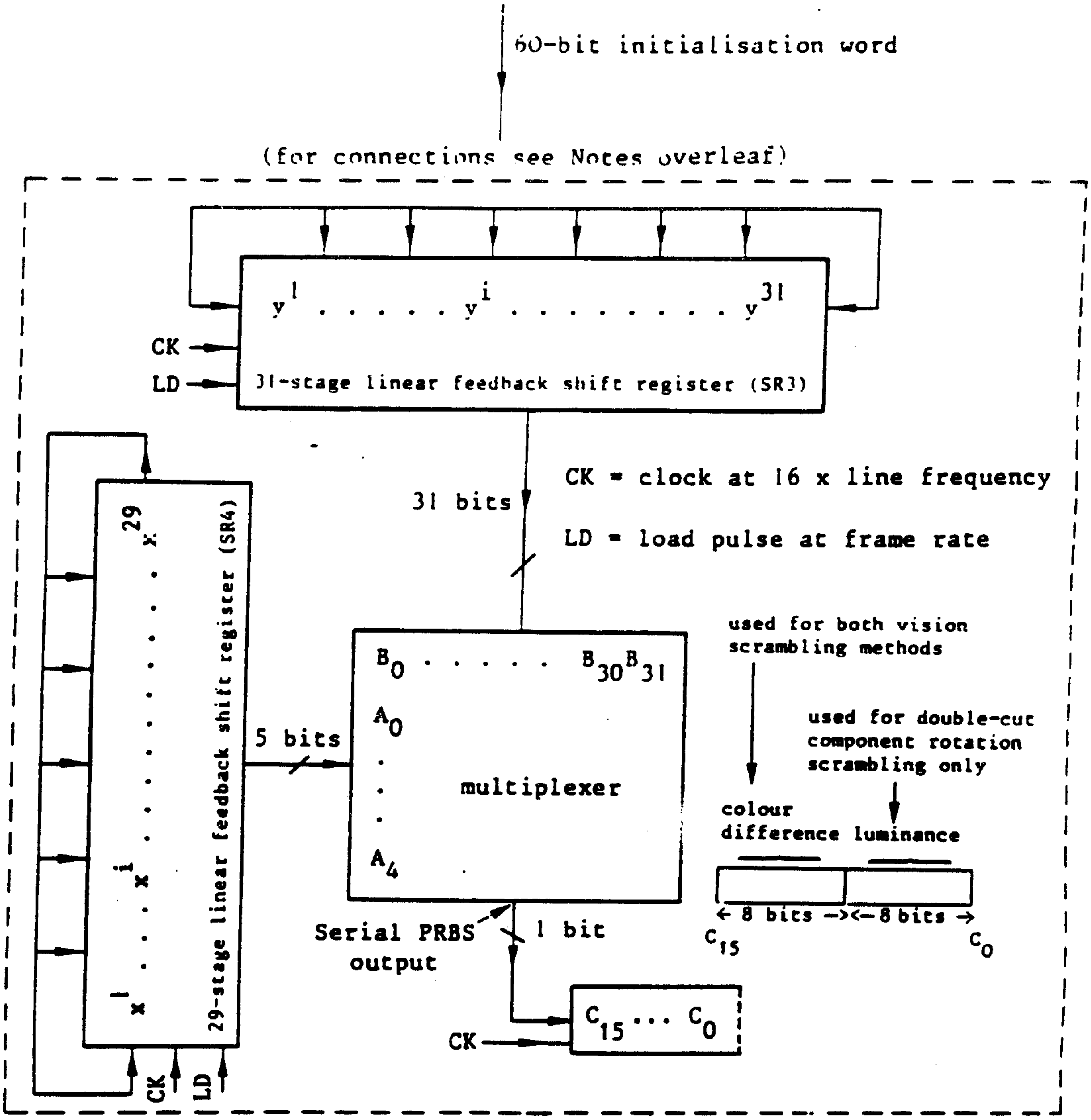


Fig. 4: PRBS generator 2 in Fig. 2

Notes on Fig. 4

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 initialisation word loading for SR3 and SR4 are defined as follows. The 60-bit initialisation 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 Fig. 5 b of Part 2) 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 Fig. 5 a of Part 2) 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.

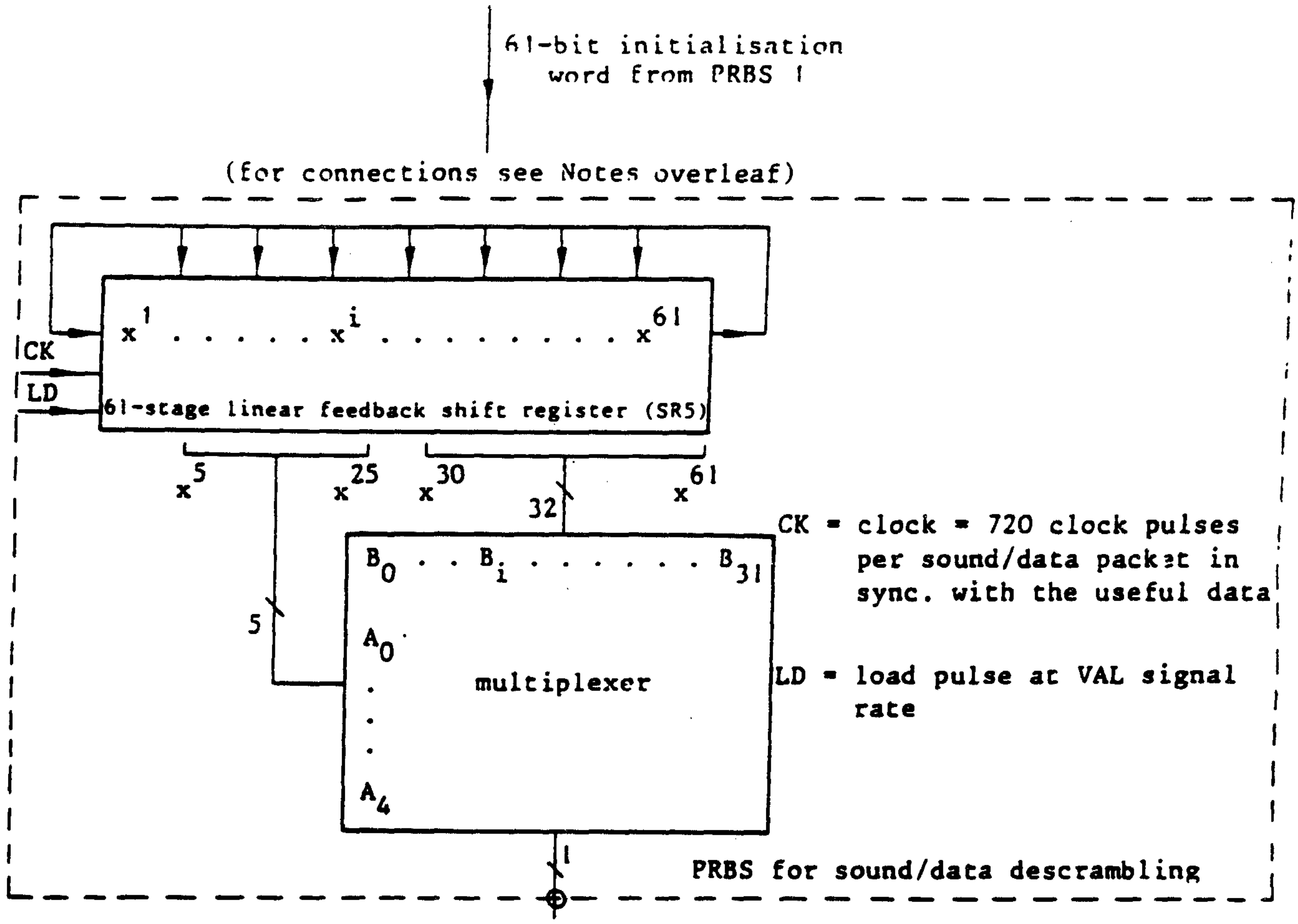


Fig. 5: PRBS generator 3 in Fig. 2

Notes on Fig. 5

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 initialisation word loading for SR5 is as follows.

The 61-bit initialisation 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 ... 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.

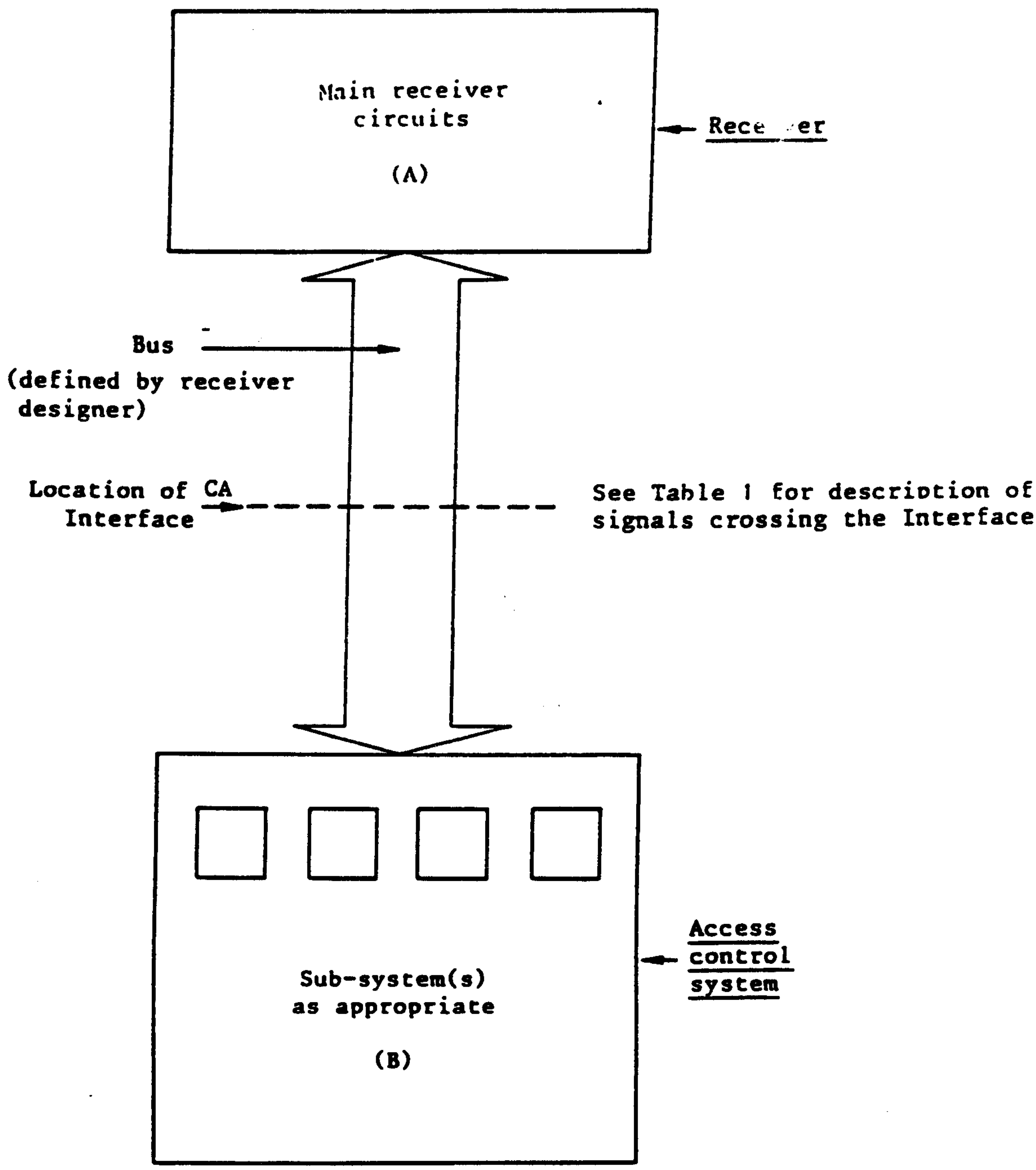
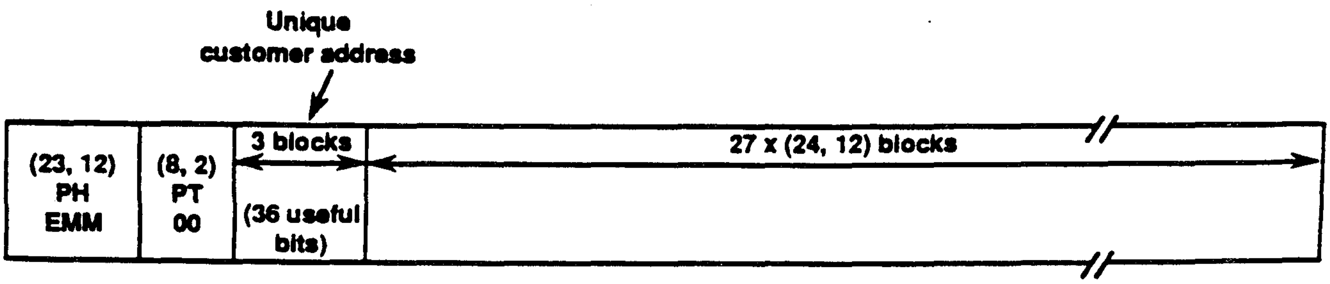
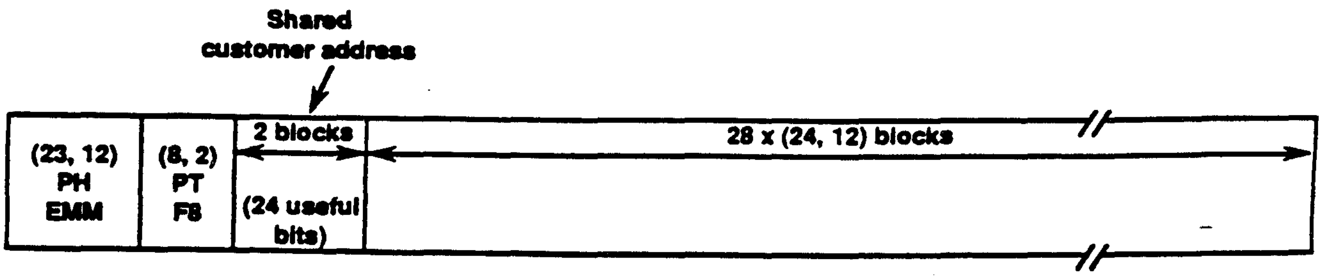


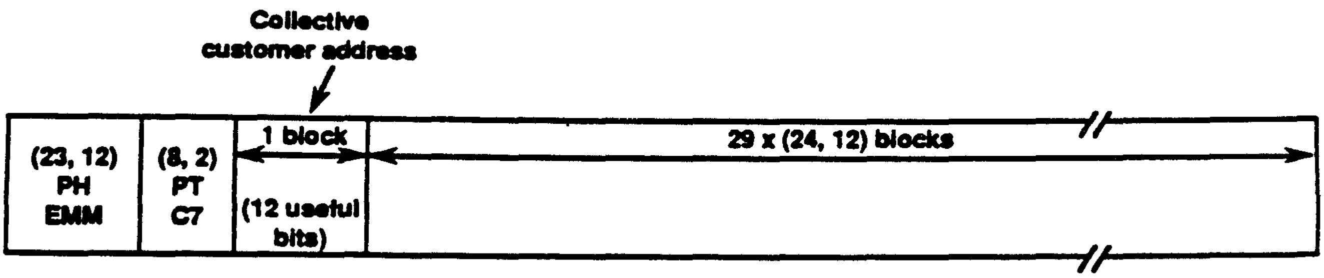
Fig. 6: Conditional-access interface



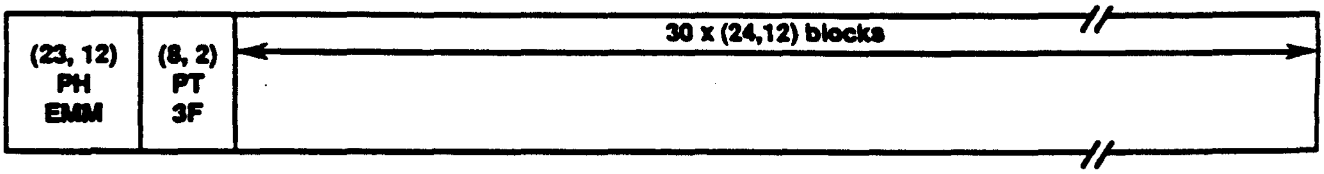
a) EMM-U: Packet for a unique customer



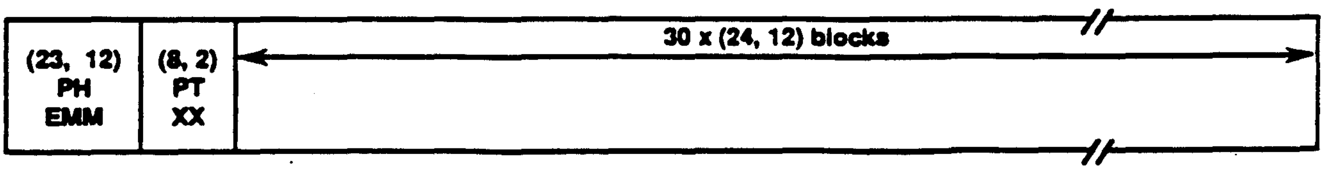
b) EMM-S: Packet for a small number customers



c) EMM-C: Packet for a large collection of customers



d) EMM-G: Packet for the customers audience in general



e) ECM: Packet

Fig. 7: EMM and ECM packet formats

PART 7: SPECIFICATION OF THE MODULATION PARAMETERS

Subject of Part 7

This Part contains the specification of the modulation parameters of the D-MAC/packet system for satellite broadcasting receivable by domestic satellite receiving equipment and for cabled distribution networks.

The first Part, Part 7A, is concerned with the modulation parameters suited to satellite channels in the broadcasting satellite service (BSS) and in the fixed-satellite service (FSS).

The second Part, Part 7B, is concerned with the modulation parameters suited to both wideband and narrowband cabled distribution networks which use AM/VSB.

PART 7A: MODULATION PARAMETERS FOR SATELLITE BROADCASTING

Contents

Part 7A

This Part contains the specification of the D-MAC/packet system for use in satellite broadcasting both in the broadcasting satellite service (BSS) and in the fixed-satellite service (FSS) frequency ranges.

Frequency modulation is used for the whole baseband signal. Pre-emphasis is applied to the whole baseband of the D signal*, energy dispersal is added, and the signal is used to frequency modulate a carrier.

Both data and vision are processed together in the modulating equipment.

In the full-channel digital mode of operation, the only analogue components are the clamping interval and the analogue reference signals in line 624.

For normal television transmissions, 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 radiofrequency regulations. In the following, they are referred to as:

- Option a: narrow bandwidth channels (< 30 MHz)
- Option b: wide bandwidth channels ($\geq$30 MHz).

Note: The figures given hereafter for the deviation sensitivity and the energy dispersal are extracted from the specification of the D2 system. They are subject to confirmation for the D system.

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 rate of 10^{-3} is achieved at a carrier-to-noise ratio of 8 dB, measured in a 27 MHz noise bandwidth.

2. Baseband channel

2.1 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.

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* However, the additional non-linear pre-emphasis, if used, is applied to the vision samples only (see Section 2.3.2).

** For full channel digital mode of operation, a third set of parameters (option c) could be used corresponding to optimal values for the data signal only.

2.2 Deviation sensitivity

The frequency deviation sensitivity* in MHz/V at the transition frequency of the pre-emphasis network shall be as follows**:

Channel bandwidth \ Frequency bands	BSS	FSS
Option a	13.5-16	13.5-16
Option b	18-22	22

2.3 Pre-emphasis

The pre-emphasis consists of two parts, a linear pre-emphasis and also, as an option, a non-linear pre-emphasis (E7)***. The pre-emphasis used is signalled via MODPRM, Part 5, Section 2.1.

2.3.1 Linear pre-emphasis

The baseband signal of the D2-MAC system will, before transmission, be pre-emphasised by means of a network with transfer characteristic (as shown in Fig. 1) defined by the expression:

$$H(f) = A \times \frac{1+j\ (f/f_1)}{1+j\ (f/f_2)}$$

The characteristics of the pre-emphasis network are given in the Table below and Fig. 1.

A	$= 1/\sqrt{2}$
f_1	= 0.84 MHz
f_2	= 1.5 MHz

______

* Also applicable to HD-MAC.

** The actual value of the deviation sensitivity is conveyed in the service identification (SI) channel, parameter MODPRM (see Part 5).

*** The characteristics of this network are such that the use of non-linear pre-emphasis without corresponding non-linear de-emphasis does not degrade the subjective picture quality. Compatibility problems arise when transmission is made without E7 pre-emphasis and the receiver is equipped with E7 de-emphasis.

2.3.2 Additional non-linear pre-emphasis (E7)

Additional non-linear pre-emphasis can be included in a compatible way for the video signal.

The use of the additional non-linear pre- and de-emphasis improves the picture quality for D-MAC and for the compatible reception of HD-MAC.

The network can be implemented either in analogue or digital form. The digital network is recommended, but the analogue network can be used in early services.

2.3.2.1 Digital non-linear pre-emphasis

The block diagram is shown in Fig. 2. The non-linear pre-emphasis is applied to vision samples only.

High-pass filter F1: clock rate : 20.25 MHz
phase response : linear

F1 is a 7-tap digital filter with the following coefficients:

$$C_o = 180/256$$
$$C_1 = C_{-1} = -58/256$$
$$C_2 = C_{-2} = -25/256$$
$$C_3 = C_{-3} = -7/256$$

These are scaled for an AC gain of nominally unity.

Delay element T : 3 clock periods

Non-linear function N_1 is defined by the relationship:

$$V_o = f\ (V_i) - V_i$$

where V_o = output from network
V_i = input to network

$f(V_i)$ is defined from the relationship $V = f\ (V_i)$

where

$$V_i = \frac{V}{C} + \frac{1}{B} \log_e \left(\frac{V + \sqrt{V^2 + (2AC)^2}}{2AC} \right)$$

with A = 0.011
B = 19.8
C = 1.5225

Non-linear function N_2 is defined by the relationship

$$V_o = V_i\left(\frac{1 - C}{C}\right) + \frac{1}{B}\log_e\left(\frac{V_i + \sqrt{V_i^2 + (2AC)^2}}{2AC}\right)$$

where $A = 0.011$

$B = 19.8$

$C = 1.5225$

V_o = output from network

V_i = input to network

The pre-emphasis network is complex to permit a simple implementation of the de-emphasis network.

Pre-emphasis is applied only to active vision samples, and the transitions between E7 mode and non-E7 mode are controlled transitions. The weighting network is included as shown in Fig. 2.

For active vision lines 23 to 310 and 335 to 622 inclusive, transition weights are as follows:

Transition weight	Sample numbers	
	Scrambled MAC (double cut)	Unscrambled MAC
0	1 to 225 inclusive	1 to 231 inclusive
1/8	226	232
1/2	227	233
7/8	228	234
1	229 to 1287 inclusive	235 to 1287 inclusive
7/8	1288	1288
1/2	1289	1289
1/8	1290	1290
0	1291 to 1296 inclusive	1291 to 1296 inclusive

2.3.2.2 Analogue non-linear pre-emphasis

The block diagram is shown in Figs. 3 and 4. The non-linear pre-emphasis is applied to vision samples only.

Low-pass filter F

The transfer function is given by:

$$\frac{1}{1 + j\, f/f_o}$$

where f = frequency
f_o = 2.0 MHz

Non-linear function N^{-1}

The output V_o of the non-linear function is related to its input V_i by the relationship:

$$V_i = \frac{V_o}{C} + \frac{1}{B}\log_e\left(\frac{V_o + \sqrt{V_o^2 + (2AC)^2}}{2AC}\right)$$

where A = 0.009
B = 19.8
C = 1.5642

The gain G must be sufficiently high that the error is small compared to the input signal for baseband frequencies up to 8.4 MHz. The output filter is 12 MHz (-3 dB) bandwidth low pass.

2.4 Energy dispersal

An energy dispersal signal shall be added* to the whole baseband signal of the D2-MAC system (see Fig. 5). The dispersal signal shall consist of a frame synchronous triangular waveform of frequency 25 Hz with a deviation after modulation n the radiofrequency channel as given in the following Table:

Frequency bands / Channel bandwidth	BSS	FSS
Option a	600 kHz p-p	1.2 MHz p-p
Option b	600 kHz p-p	2 MHz p-p** ***

* For full channel digital mode of operation (option C) energy dispersal is not always required.

** A lower value can be used if it complies with the Radio Regulations requirement not to cause interference (this depends on the satellite e.i.r.p. and the angle of elevation).

*** 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*.

2.5 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.

3. Digital component multiplexing

The digital components are multiplexed at baseband and the whole signal is modulated as described in 2.1 above.

4. 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.

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* The actual value of the energy dispersal is conveyed in the service identification (SI) channel, parameter MODPRM (see Part 5).

** The satellite transponder bandwidth can be greater than the transmission channel bandwidth.

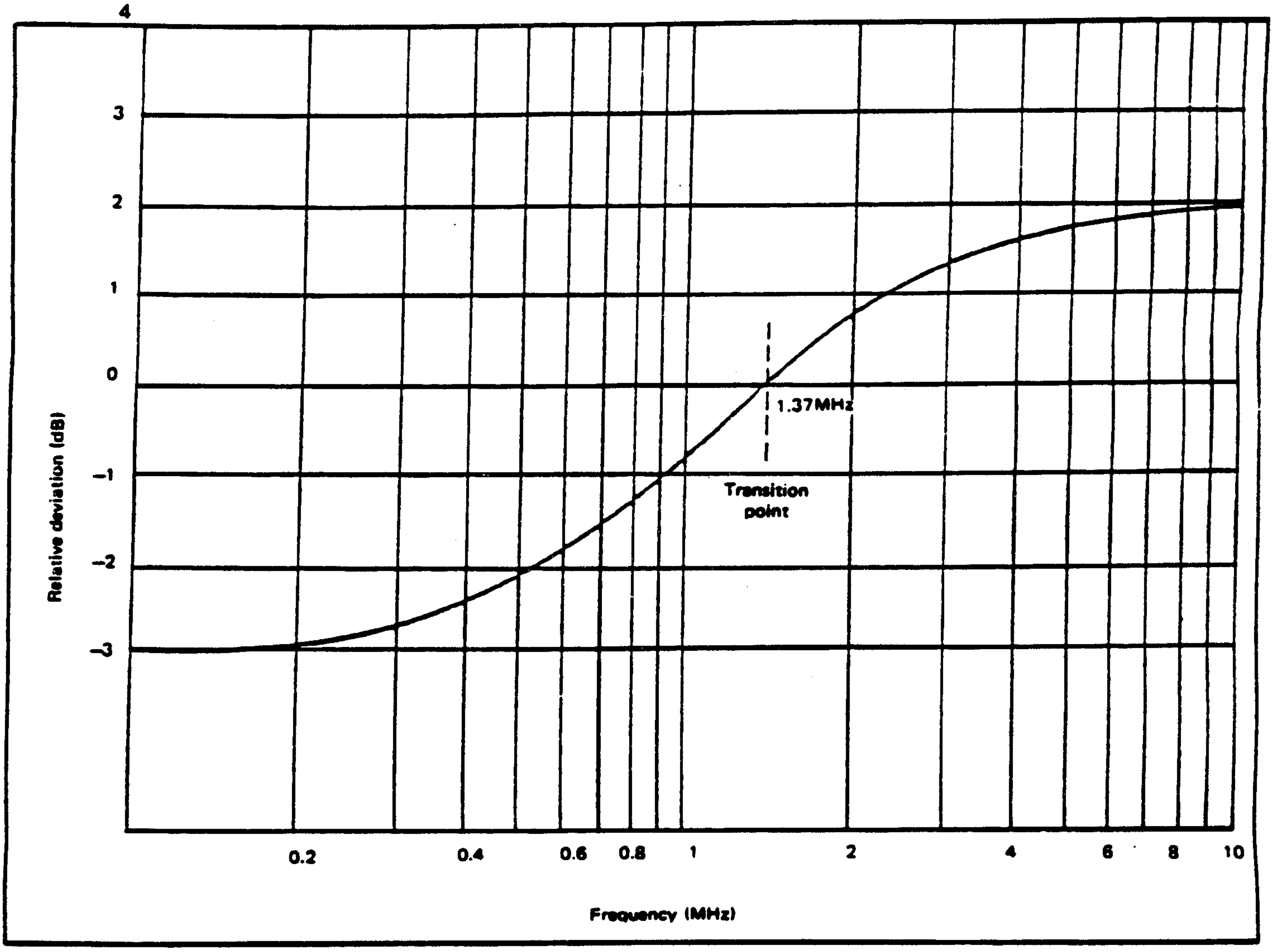
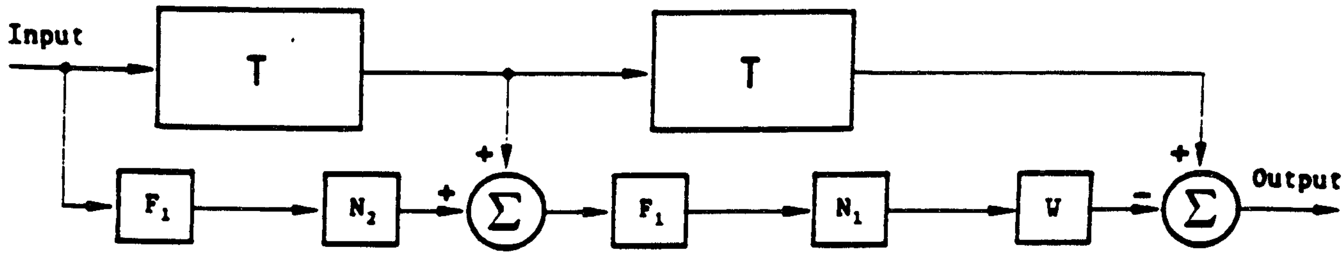


Fig. 1: MAC linear pre-emphasis network

Part 7A



Σ: adder
T: delay element

F_1: High pass filter
N_1, N_2: non-linear elements
W: weighting coefficient

Fig. 2: Digital pre-emphasis network

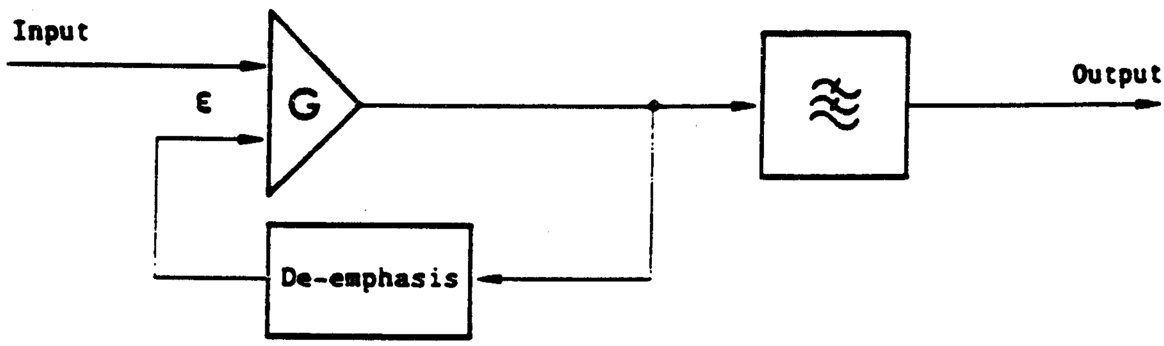


Fig. 3: Analogue pre-emphasis network

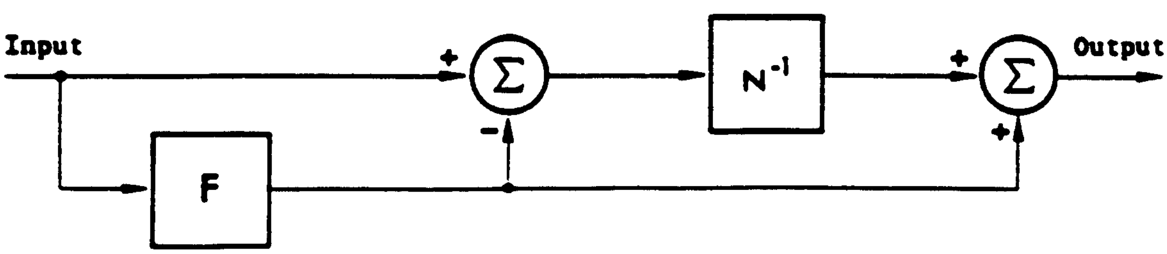
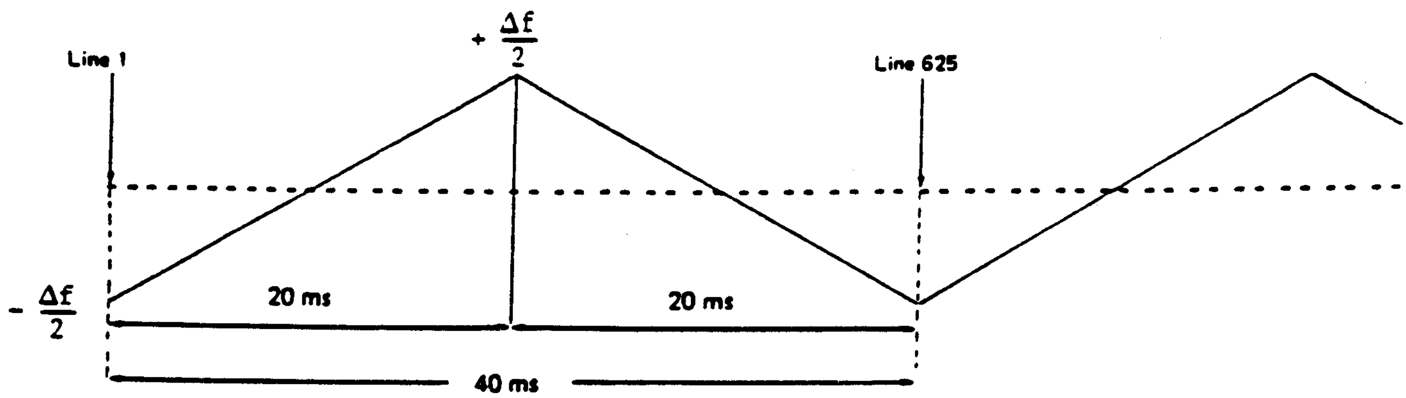


Fig. 4: The de-emphasis network of Fig. 3

Part 7A



Note: Δf corresponds to the peak-to-peak deviation of the energy dispersal

Fig. 5: Energy dispersal waveform added to the baseband signal of the D system

PART 7B: MODULATION PARAMETERS FOR CABLE DISTRIBUTION

Contents

Part 7B

1. Introduction

This Part contains the specification of the D-MAC/packet system for use in cabled distribution networks using AM/VSB.

Basically, the D-MAC/packet system for cable distribution has the same baseband signal as for the satellite D-MAC/packet system. A 12 MHz channel spacing maintains the full D-MAC quality.

Only the 12 MHz channel spacing allows the transmission of HD-MAC signals.

This Part conforms to the existing European Standard EN 50080 "RF characteristics of MAC AM-VSB cable receivers", which is applicable to 12 MHz channels in the hyperband.

2. 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 the UHF band is under study.

The use of a PAL or SECAM channel adjacent to a D-MAC channel shall be avoided without special measures.

3. Modulation method

VSB amplitude modulation shall be used for the composite multiplex signal. There shall be no energy dispersal or linear pre-emphasis.

4. 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).

5. 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 Fig. 1).

6. Transmitter filtering

6.1 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:

$$H(f) = \frac{1}{2}\left[1 + \sin\left(\frac{\pi}{2}\frac{f_o - f}{0.75}\right)\right]$$

where f_o is the carrier frequency, f is the current frequency, f and f_o are expressed in MHz, and $|f_o - f| \leqslant 0.75$ MHz (see both continuous and dashed lines in Fig. 2).

The important characteristic is the symmetry of the Nyquist slope around the carrier. Tolerances are given in Fig. 3.

6.2 Transmitter amplitude-frequency response in the upper sideband

For 12 MHz channel spacing, the upper sideband shall be transmitted without attenuation up to at least 9.75 MHz above the carrier frequency in order to obtain the maximum eye opening of the data signal. The nominal characteristic is shown in Fig. 2. The tolerances are given in Fig. 3, which includes also the requirements for HD-MAC.

6.3 Transmitter phase-frequency response

No pre-correction for the receiver group delay characteristic shall be applied; the transmitter filter shall be substantially phase linear.

Precise template of phase-frequency response is given in Fig. 4.

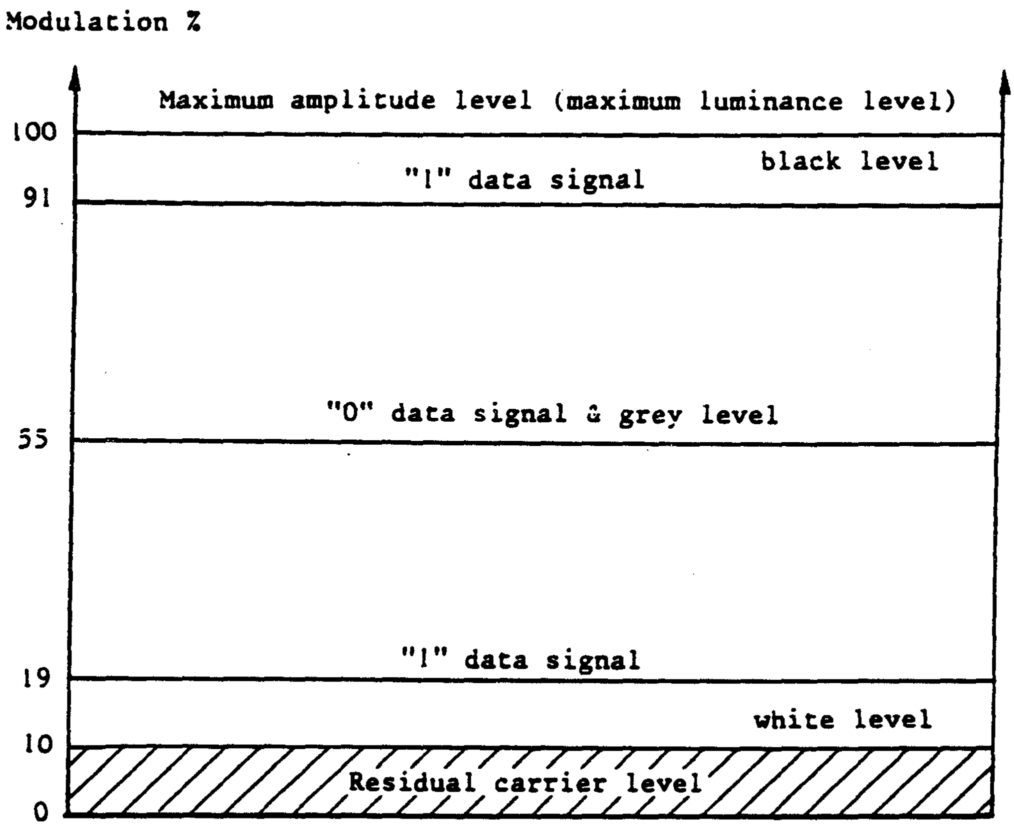


Fig. 1: Modulation levels (before transmission filtering)

Part 7B

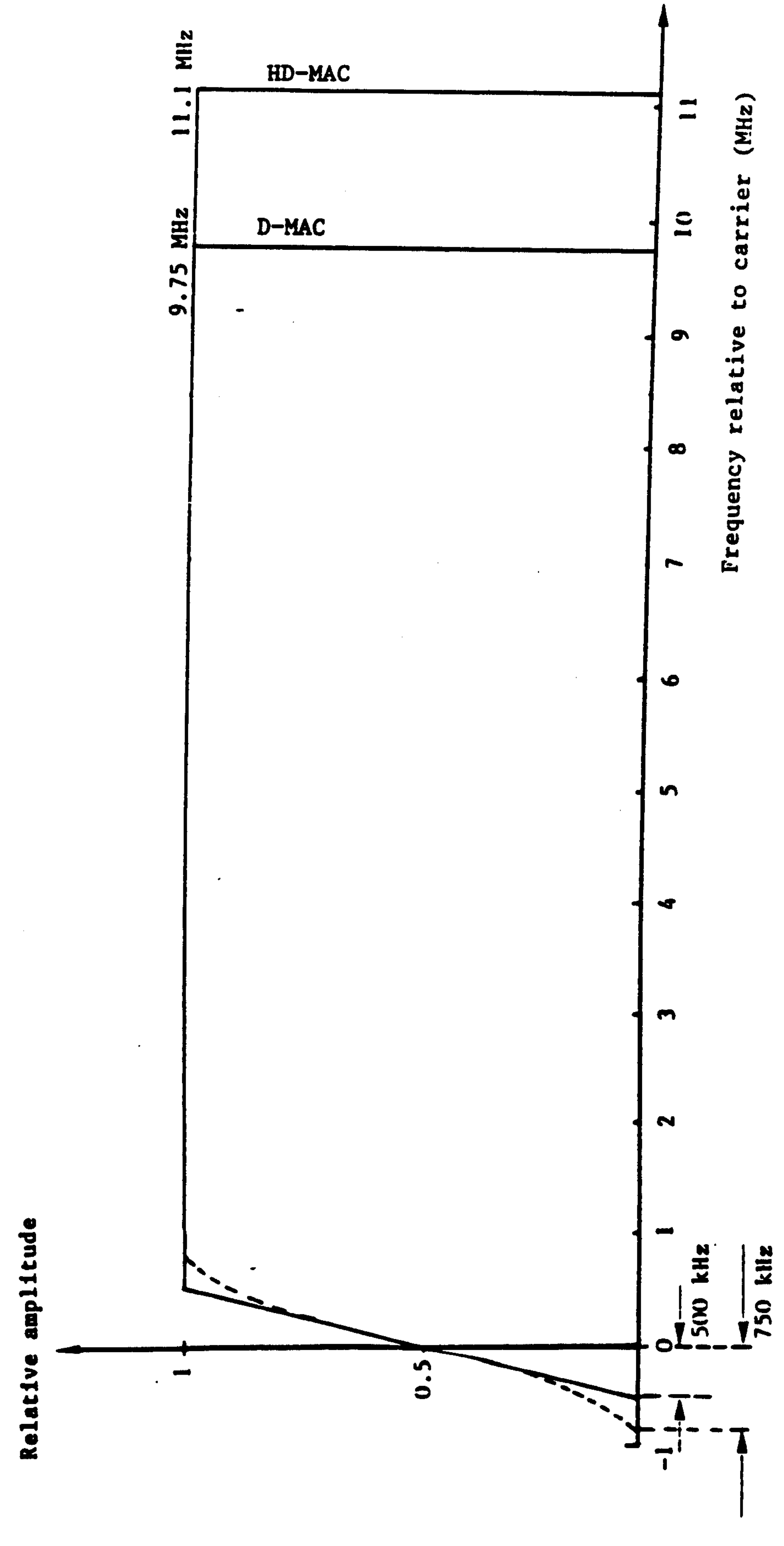


Fig. 2: Schematic amplitude-frequency characteristic of the transmitter

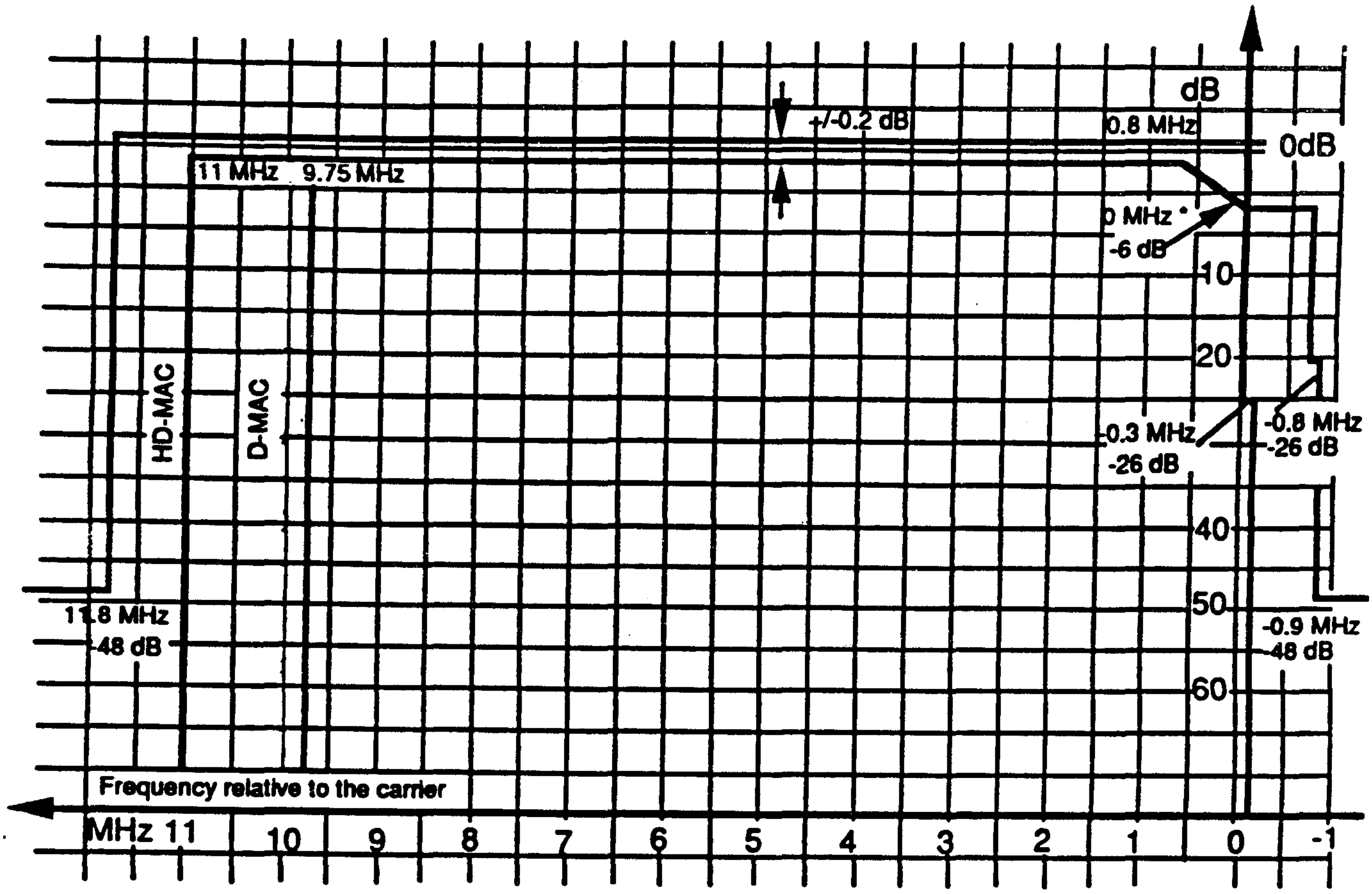
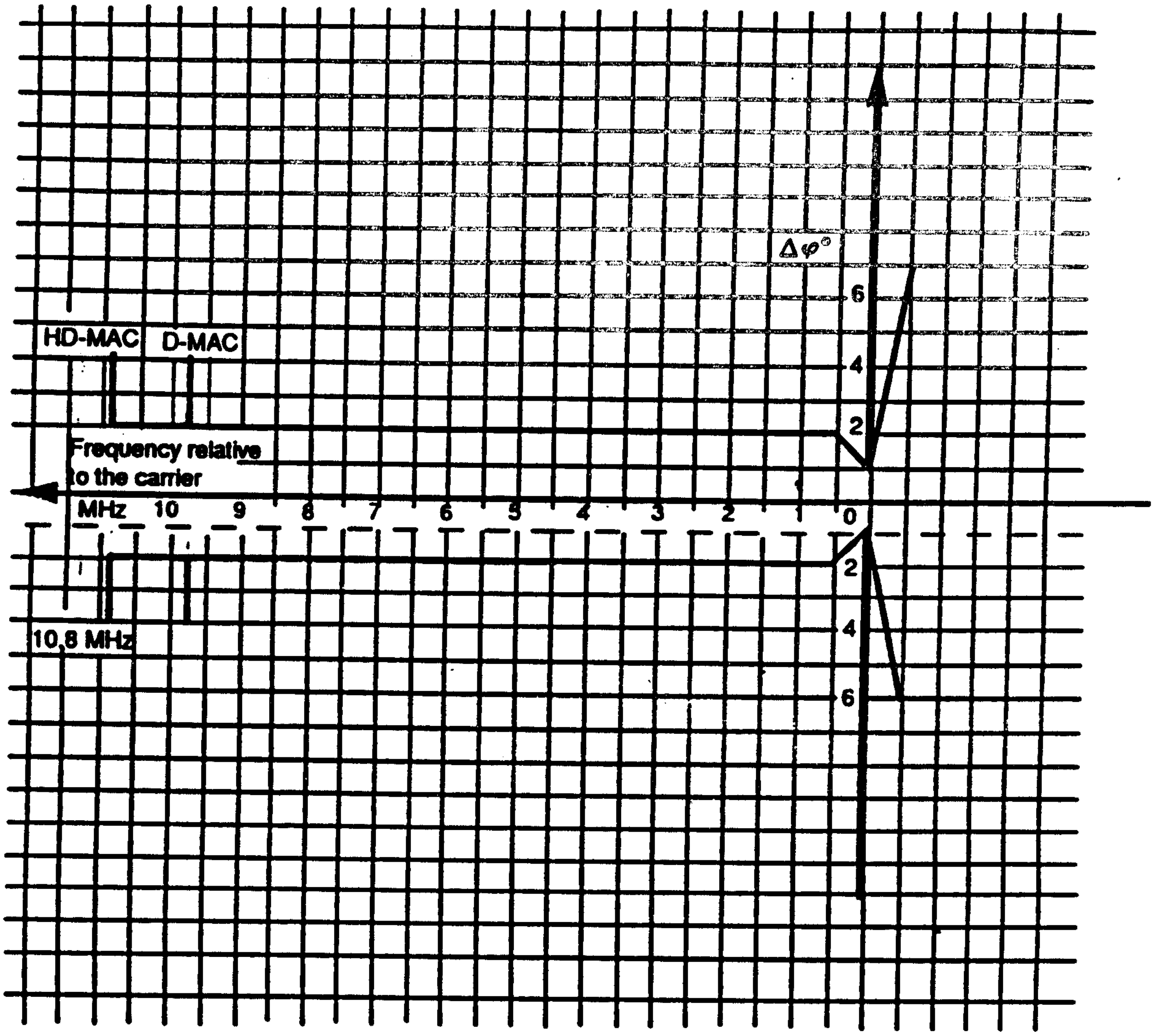


Fig. 3 : Amplitude / frequency template for the for the transmitter filtering

* Within the range of the Nyquist slope, the sum of the upper and lower sideband amplitudes shall also meet the given parameter tolerance of ± 0.2 dB.



Note: "0" reference is equal to 1/2 ($\Delta\phi$ max + $\Delta\phi$ min), where $\Delta\phi$ max and $\Delta\phi$ min are maximum and minimum values respectively, reached between:

0 and 9.75 MHz for D-MAC
0 and 10.8 MHz for HD-MAC

Fig. 4: Phase/frequency template for the transmitter filtering

History

Document history		
June 1994	First Edition	